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 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 extend 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 Deskphones, 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 recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in this DevConnect Application Note included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with this Application Note, the interface between Avaya systems and Teo 7810 and 7810 TSG-6 Series IP Phones did not include use of any specific encryption features as requested by Teo.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- SIP registration of Teo 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.

- Teo IP Phones support local 3-way conference only. Two call appearances are required.

- Teo IP Phones support their own implementation of bridge line appearances (BLA) for SIP phones. Basic bridged scenarios were covered during the compliance test, including:

    - Answering a call on a BLA.
    - Bridging onto an active call on a BLA of another phone.
    - Placing a call from a bridged line appearance.
    - Transferring a call on a BLA.

- Ensure that the Initial IP-IP Direct Media option on the Communication Manager signaling group form is disabled so that all BLA 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: https://www.teotech.com/support

https://www.teotech.com/company/contact-us/

- Email: info@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.
- Avaya Aura® Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- Avaya Aura® System Manager used to configure Session Manager.
- Avaya Aura® Messaging serving as the voicemail system.
- Avaya 96x1 Series 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.

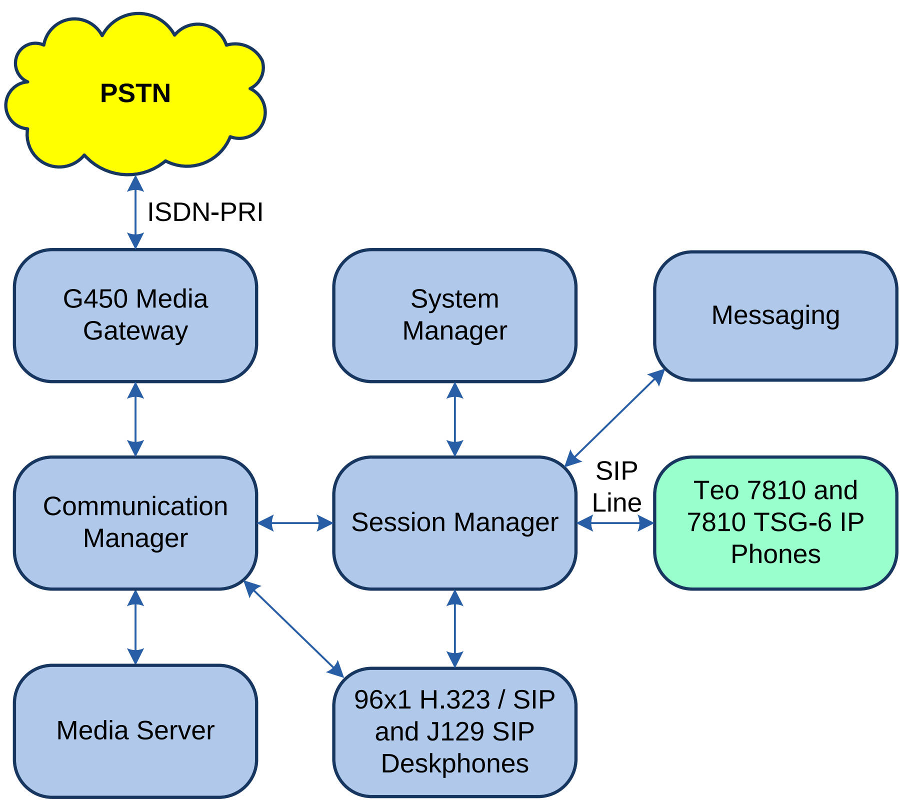


Figure 1: Avaya SIP Network with Teo 7810 and 7810 TSG-6 Series IP Phones

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	8.1.0.1.0-SP1 (R018x.01.0.890.0 with Patch 25393)
Avaya G450 Media Gateway	FW 40.25.0
Avaya Aura® Media Server	v.8.0.1.121
Avaya Aura® System Manager	8.1.0.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.0.0.079814
Avaya Aura® Session Manager	8.1.0.0.810007
Avaya Aura® Messaging	7.1.3.1.0-FP3SP1
Avaya 96x1 Series IP Deskphones	6.8003 (H.323) 7.1.5.0.11 (SIP)
Avaya J129 SIP Deskphones	4.0.2.0.8
Teo 7810 and 7810 TSG-6 Series IP Phones	05.04.28.02

5. Configure Avaya Aura® Communication Manager

This section provides the procedure for configuring Communication Manager. The procedure includes the following areas:

- Verify Communication Manager license
- Administer IP Node Names
- Administer IP Network Region and IP Codec Set
- Administer SIP Trunk Group to Session Manager
- Administer AAR Call Routing
- 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 System Manager in **Section 6.2**.

5.1. Verify Communication Manager License

Using the SAT, verify that the Off-PBX Telephones (OPS) option is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 1**, verify that the number of OPS stations allowed in the system is sufficient for the number of SIP endpoints that will be deployed.

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

     G3 Version: V18                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                          USED
                            Platform Maximum Ports:  48000     80
                                  Maximum Stations:  36000     19
                          Maximum XMOBILE Stations:  36000      0
                Maximum Off-PBX Telephones - EC500:  41000      0
                Maximum Off-PBX Telephones -   OPS:  41000      8
                Maximum Off-PBX Telephones - PBFMC:  41000      0
                Maximum Off-PBX Telephones - PVFMC:  41000      0
                Maximum Off-PBX Telephones - SCCAN:      0      0
                     Maximum Survivable Processors:  313        0

        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer IP Node Names

In the **IP Node Names** form, assign an IP address and host name for Communication Manager (*procr*) and Session Manager (*devcon-sm*). The host names will be used in other configuration screens of Communication Manager.

```
change node-names ip                                            Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
default             0.0.0.0
devcon-aes          10.64.102.119
devcon-ams          10.64.102.118
devcon-sm           10.64.102.117
procr               10.64.102.115
procr6              ::

( 6  of 6    administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.3. Administer IP Network Region and IP Codec Set

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *avaya.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G450 Media Gateway 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 2                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 2       NR Group: 2
Location: 1       Authoritative Domain: avaya.com
    Name:                        Stub Network Region: n
MEDIA PARAMETERS                 Intra-region IP-IP Direct Audio: yes
      Codec Set: 2               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 2** command. Note that IP codec set '2' was specified in IP Network Region '2' 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.

Note: In the IP Codec Set associated with the Teo IP phones, the Media Encryption section should only list *none*. If multiple encryption methods are advertised in the SIP Invite SDP, the call will fail.

```
change ip-codec-set 2                                          Page   1 of   2

                          IP MEDIA PARAMETERS
    Codec Set: 2

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

     Media Encryption                  Encrypted SRTCP: best-effort
 1: none
 2:
 3:
 4:
 5:
```

5.4. Administer SIP Trunk to Session Manager

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tls*.
- Set the **Enforce SIPS URI for SRTP** field to *n*.
- Specify Communication Manager (*procr*) and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TLS port value of *5061* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *avaya.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Disable **Initial IP-IP Direct Media**.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

```
add signaling-group 10                                       Page   1 of   2
                                SIGNALING GROUP

 Group Number: 10             Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? y  Peer Server: SM                       Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                  Far-end Node Name: devcon-sm
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1

Far-end Domain: avaya.com
                                             Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                  RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload            Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                     IP Audio Hairpinning? n
         Enable Layer 3 Test? y                    Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n            Alternate Route Timer(sec): 6
```

Configure the **Trunk Group** form as shown below. This trunk group is used for SIP calls to Teo IP phones, Avaya SIP Deskphones, and Messaging. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

```
add trunk-group 10                                                Page   1 of  22
                                TRUNK GROUP

Group Number: 10                   Group Type: sip           CDR Reports: y
  Group Name: To devcon-sm               COR: 1        TN: 1        TAC: 1010
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 10
                                                     Number of Members: 10
```

5.5. Administer AAR Call Routing

SIP calls to Session Manager are routed over a SIP trunk via AAR call routing. Configure the AAR analysis form and enter add an entry that routes digits beginning with "78" to route pattern 10 as shown below.

```
change aar analysis 78                                               Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 1

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
    78                    5    5     10        lev0         n
```

Configure a preference in **Route Pattern** 10 to route calls over SIP trunk group 10 as shown below.

```
change route-pattern 10                                              Page   1 of   3
                    Pattern Number: 10     Pattern Name: To devcon-sm
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                           DCS/ IXC
    No          Mrk Lmt List Del  Digits                             QSIG
                             Dgts                                    Intw
 1: 10   0                                                            n   user
 2:                                                                   n   user
 3:                                                                   n   user
 4:                                                                   n   user
 5:                                                                   n   user
 6:                                                                   n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               unk-unk   none
 2: y y y y y n  n            rest                                         none
```

5.6. Enable Bridged Calls (Optional)

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.

Note: Bridged Call Alerting and a Bridge Appearance are configured for each SIP user via System Manager in **Section 6.3.5**.

```
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

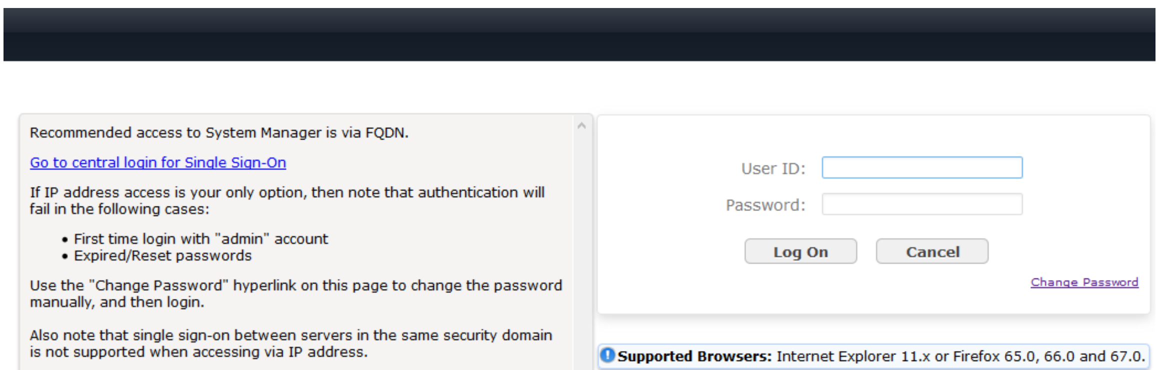
This section provides the procedure for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Set Network Transport Protocol for Teo 7810 and 7810 TSG-6 Series IP Phones
- 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.

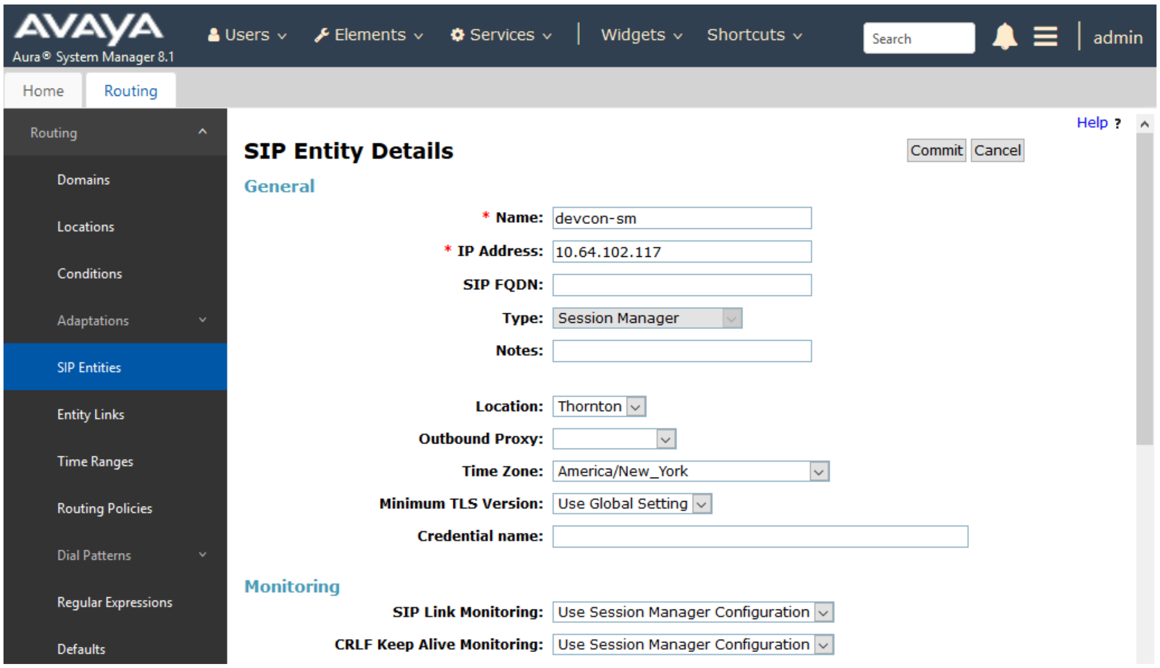
6.1. Launch System Manager

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.

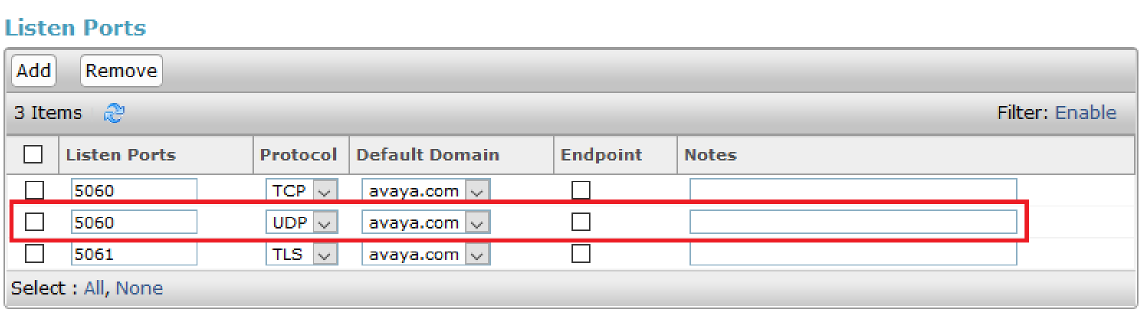


6.2. Set Network Transport Protocol for Teo 7810 and 7810 TSG-6 Series IP Phones

From the System Manager **Home** screen, select **Elements → Routing → SIP Entities** and edit the SIP Entity for Session Manager shown below.

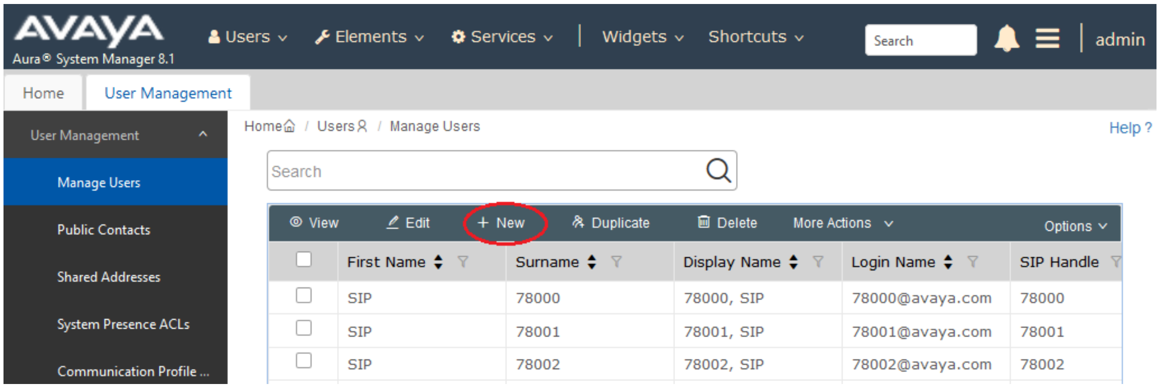


Scroll down to the **Listen Ports** section and verify that the transport network protocol used by Teo IP phones is specified in the list below. For the compliance test, the solution used UDP network transport.



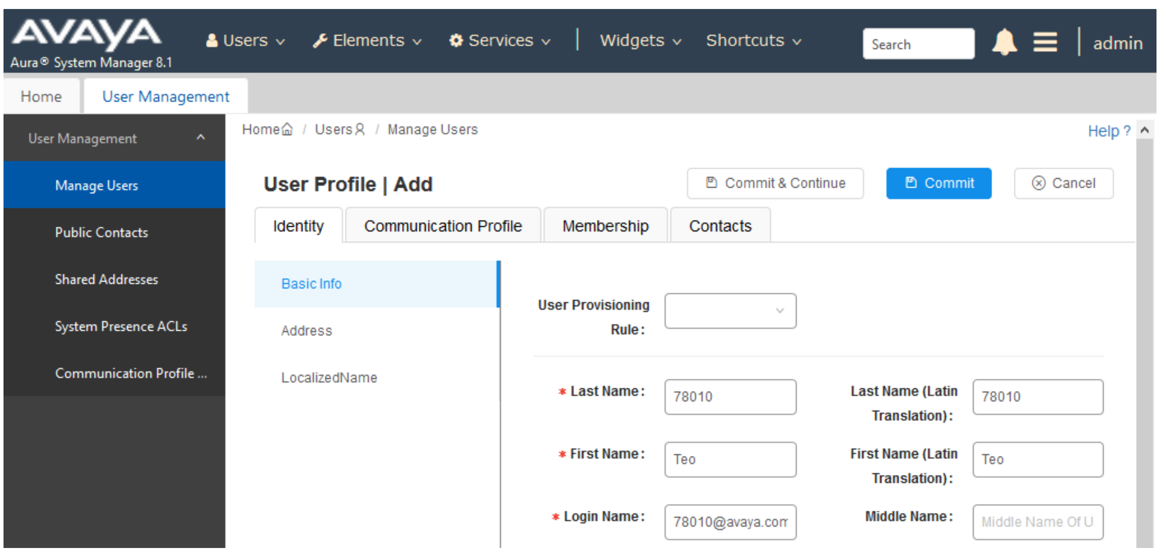
6.3. Administer SIP User

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.



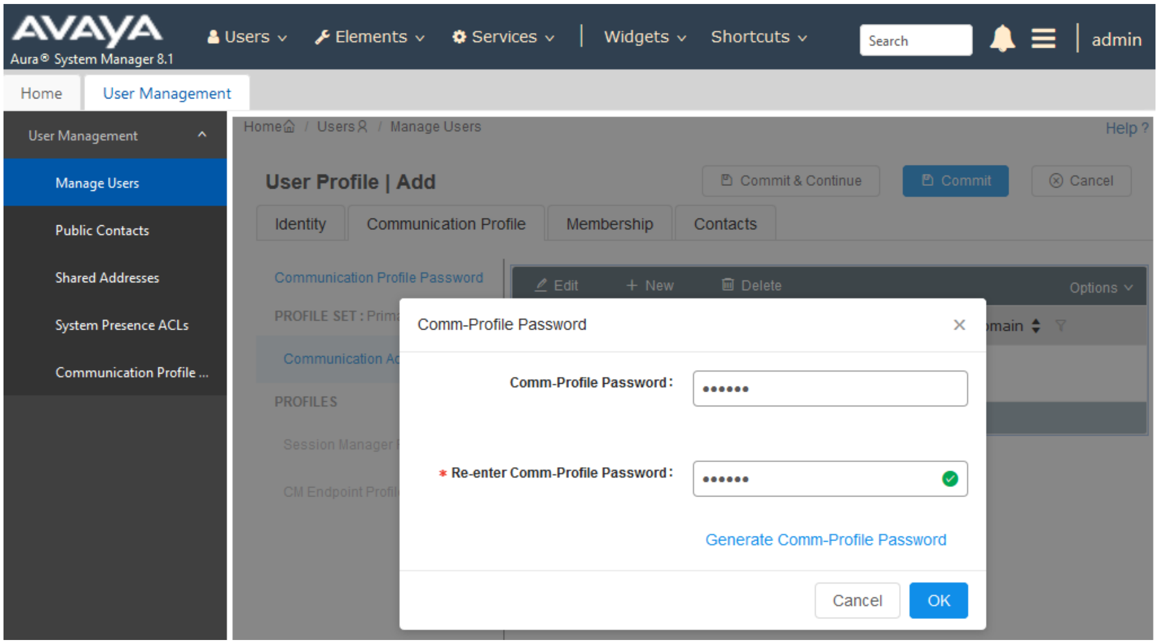
6.3.1. Identity

The **New User Profile** screen is displayed. Enter desired **Last Name** and **First Name**. For **Login Name**, enter "*<ext>@<domain>*", where "*<ext>*" is the desired Teo IP phone SIP extension and "*<domain>*" is the applicable SIP domain name from **Section 5.2**. Retain the default values in the remaining fields.



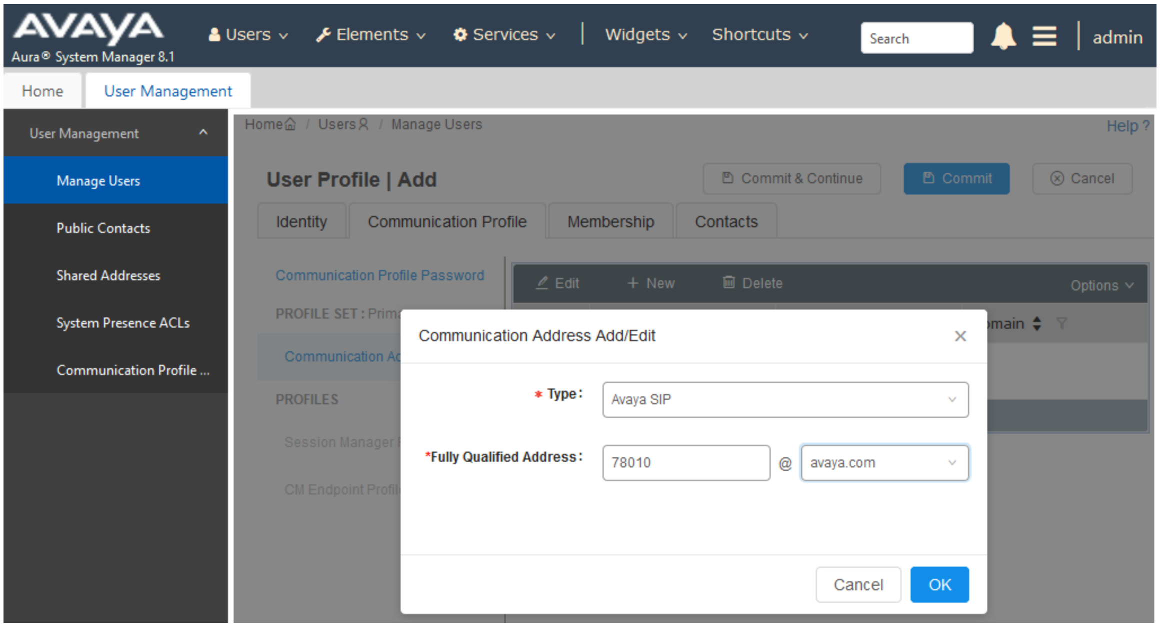
6.3.2. Communication Profile

Select the **Communication Profile** tab. Next, click on **Communication Profile Password**. For **Comm-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration. Click **OK**.



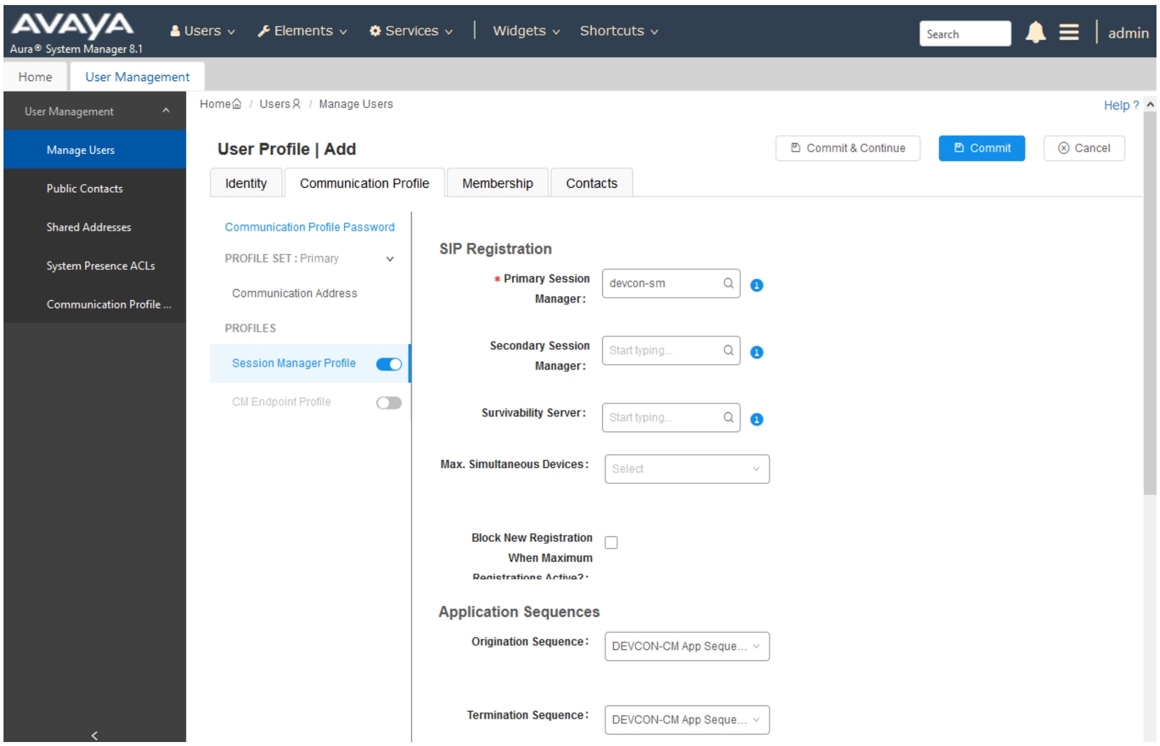
6.3.3. Communication Address

Click on **Communication Address** and then click **New** to add a new entry. The **Communication Address Add/Edit** dialog box is displayed as shown below. For **Type**, retain *Avaya SIP*. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.3.1**. Click **OK**.

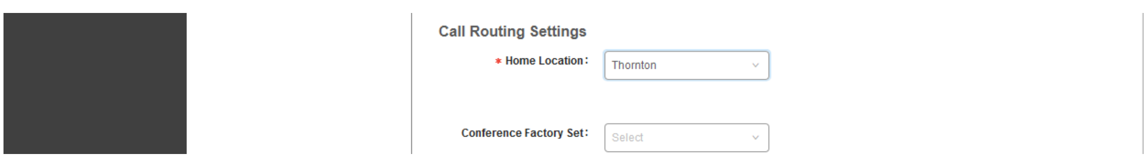


6.3.4. Session Manager Profile

Click on toggle button by **Session Manager Profile**. For **Primary Session Manager**, **Origination Application Sequence**, **Termination Application Sequence**, and **Home Location**, select the values corresponding to the applicable Session Manager and Communication Manager. Retain the default values in the remaining fields.

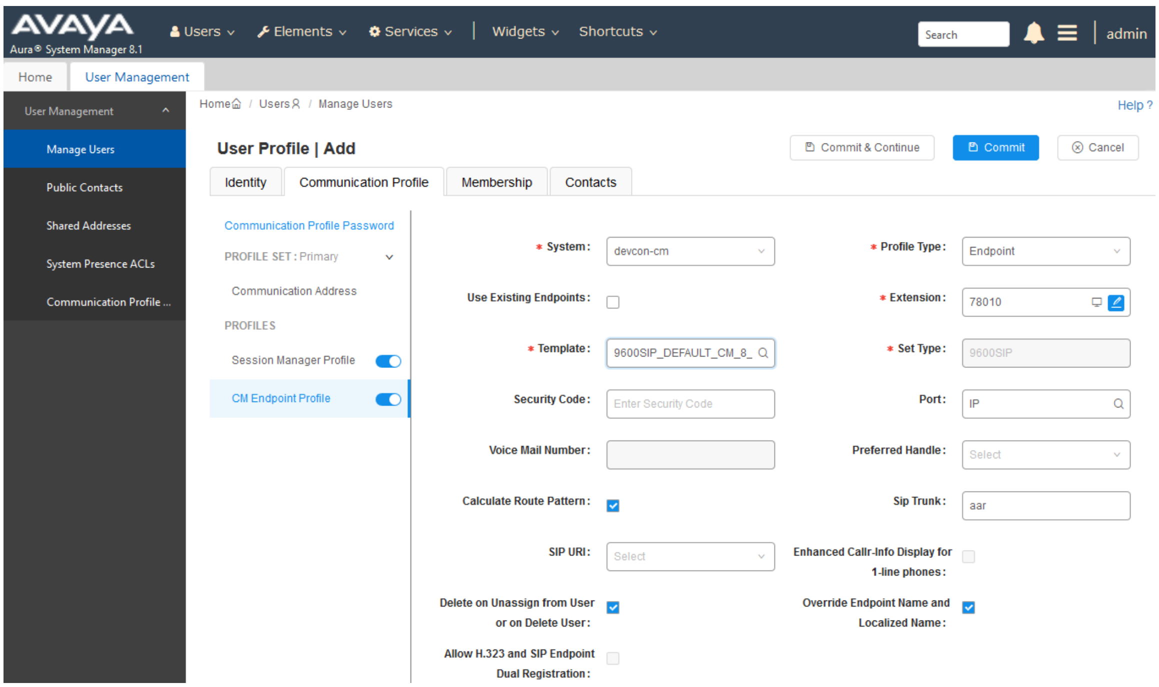


Scroll down to the **Call Routing Settings** section to configure the **Home Location**.

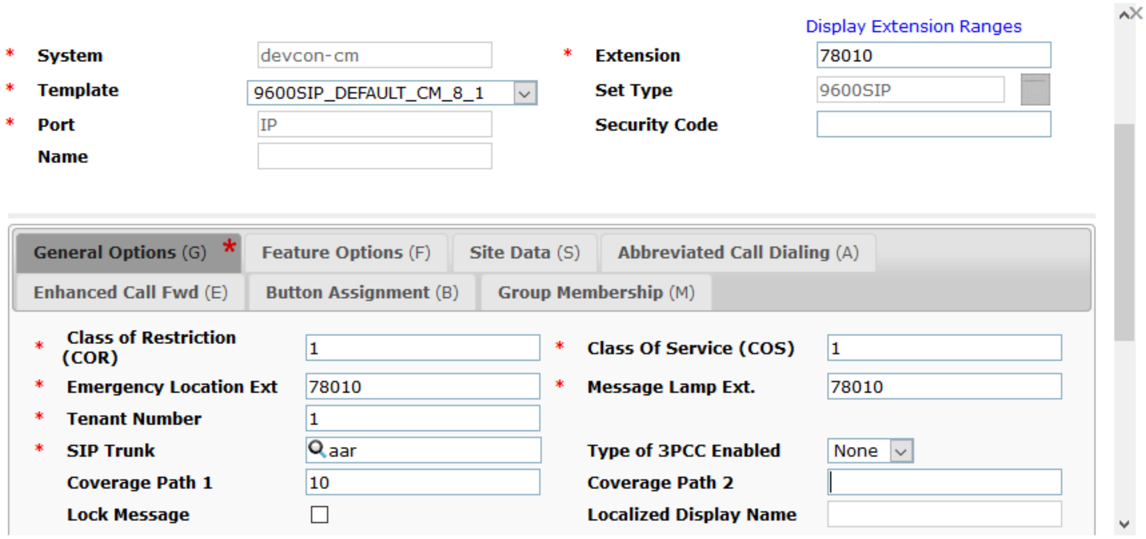


6.3.5. CM Endpoint Profile

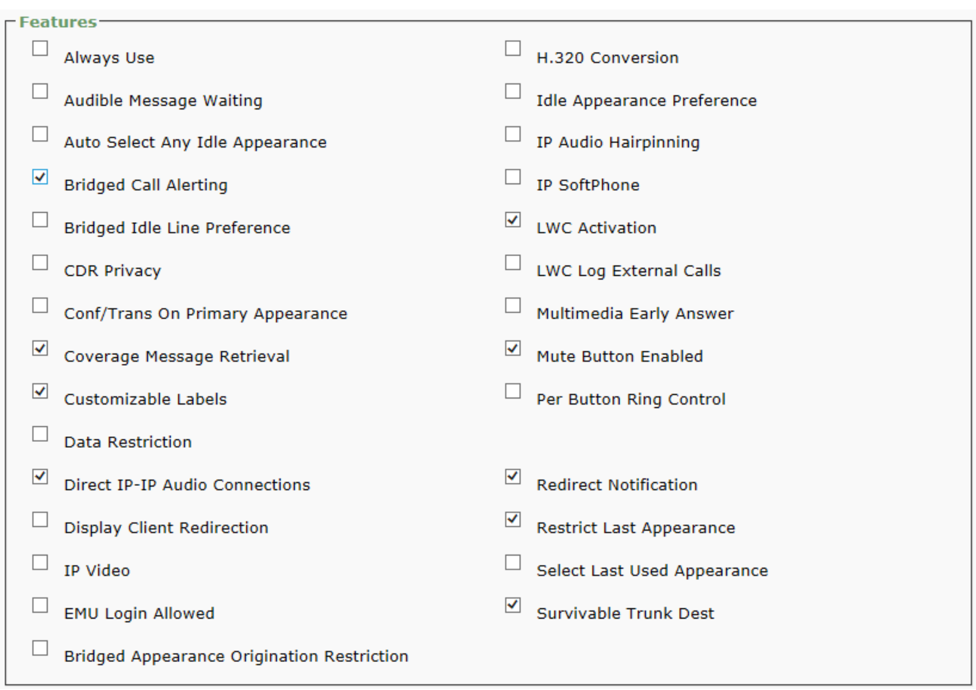
Click on the toggle button by **CM Endpoint Profile**. For **System**, select the value corresponding to the applicable Communication Manager. For **Extension**, enter the SIP user extension from **Section 6.3.1**. For **Template**, select *9600SIP_DEFAULT_CM_8_1*. For **Port**, click and select *IP*. Retain the default values in the remaining fields. Click on the Endpoint Editor (i.e, Edit icon in **Extension** field) to configure the **Coverage Path** and enabled bridged calls.



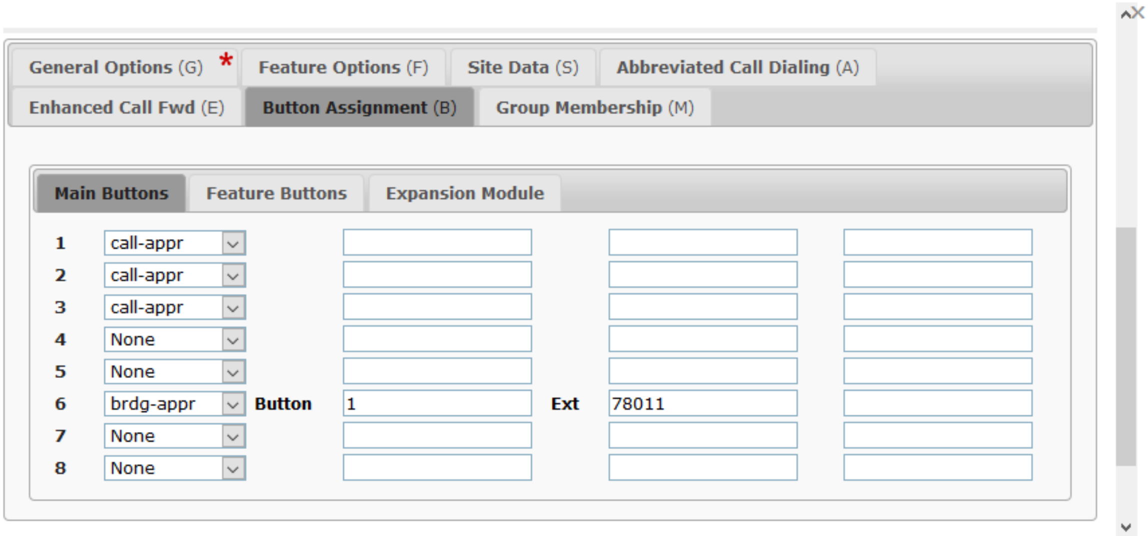
Navigate to the **General Options** tab and set the **Coverage Path 1** field to the voicemail coverage path. This provides voicemail coverage for the SIP user. In this example, coverage path 10 was used.



Navigate to the **Feature Options** tab and scroll down to the **Features** section and enable **Bridged Call Alerting**.



In the **Button Assignment** 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** (not shown) when complete, followed by **Commit** on the previous web page.

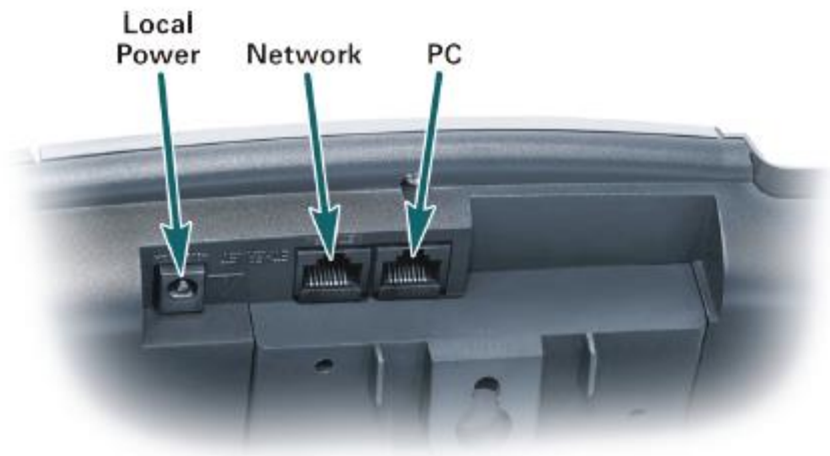


7. Configure Teo 7810 and 7810 TSG-6 Series IP Phones

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

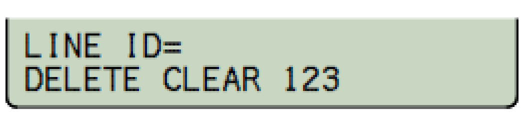
Connect power to the Teo telephone using the **Local Power** option, or in models where Power over Ethernet (PoE) is supported, connecting a CAT 5 or better cable to the **Network** jack of the telephone. Both the Teo 7810 and the 7810PoE-TSGB models used in the compliance test support PoE. During the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



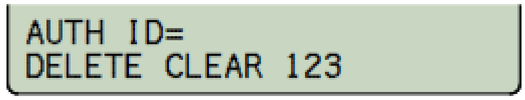
7.2. SIP Configuration

The configuration steps in this section represent the basic steps required to register 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].**

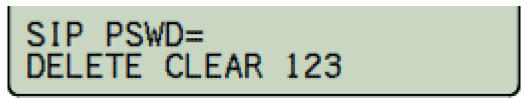
Once power is applied and the telephone initializes, the display shows different progress messages and it will then prompt for the **LINE ID**. This is the extension number assigned to the user, previously in **Section 6.3.1**. Extension **78010** was used in the example and it was entered at the prompt below. Press the **OK** on the phone when the entry is completed.



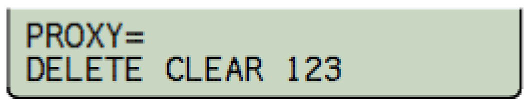
The telephone prompts next for the Authentication ID. Enter the same extension number that was assigned to the **LINE ID**. Press the **OK** key on the phone when the entry is completed.



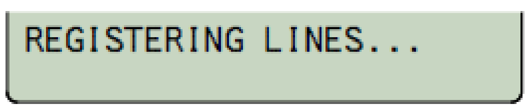
Next, the phone will prompt for the SIP Authentication Password. This is the **Communication Profile Password** assigned to the user previously in **Section 6.3.2**. Press the **OK** on the phone when the entry is completed.



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.



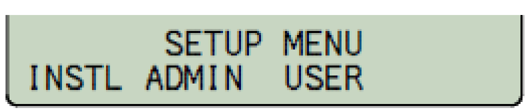
At this point, the telephone will register with Session Manager. Upon successful registration, all affected line key indicators will turn solid green momentarily, and then will go out.
The following status messages will be displayed.

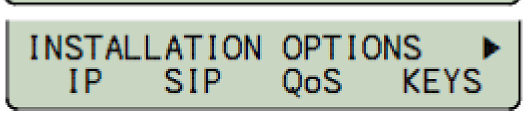


The idle display will then appear indicating that the phone is ready for use (not shown).

To ensure that MWI is enabled, press SETUP button and then navigate to **INSTL → SIP → MWI** and verify that the **MSG-SUMMARY SUB** 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**.





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

- **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**.

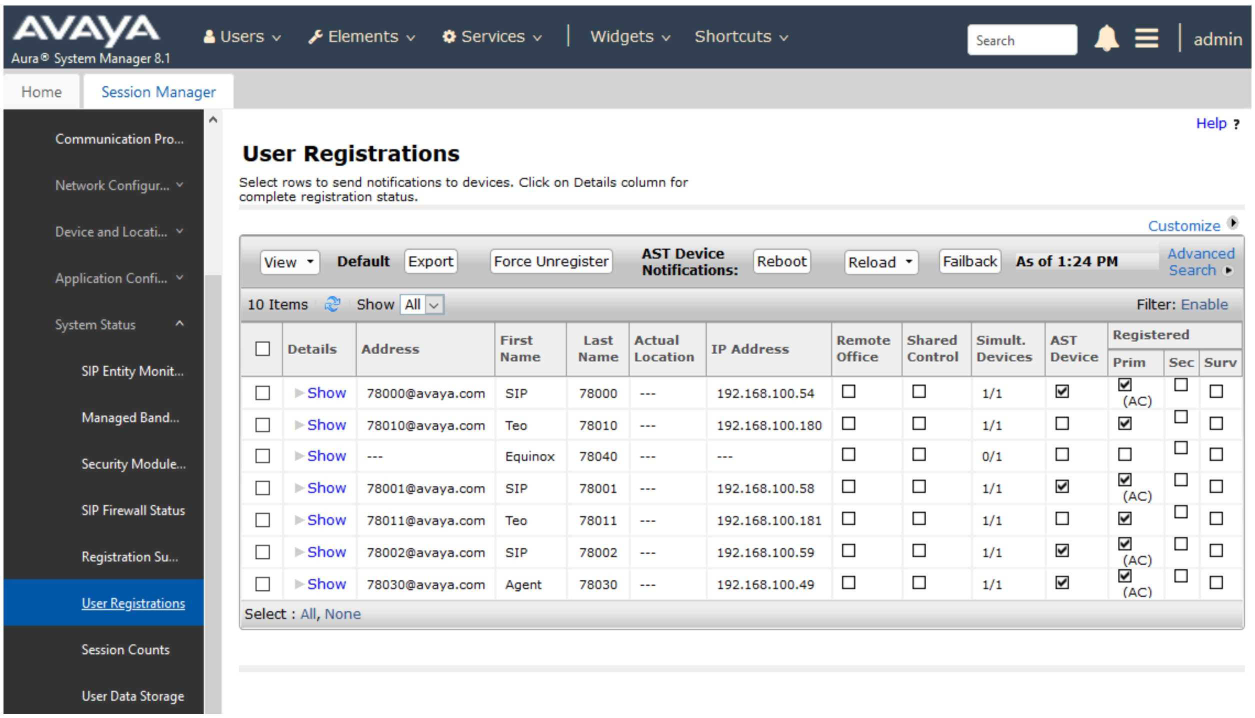
Press the **SETUP** key and select **INSTL → KEYS**, registered keys will light green. Select the key that is to be programmed as **SHARED**. The selected key and any other key associated with it will start to flash red. Select **EDIT** and press the right arrow key once or until the word **SHARE** appears 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 6.3.5**.

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 as shown below.



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 8.1.x, Issue 2, July 2019.
[2] *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019.

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

[3] *IP Phone 7810 TSG Series Installations Instructions*, Document ID 13-280138.
[4] *IP Telephone Network Administration Guide*, Document ID 13-280132.
[5] *IP Phone 7810 Installation Instructions*, Document ID 13-280124, March 2013.
[6] *IP Phone 7810 TSG Series User Guide*, Document ID 14-280211.
[7] *IP Phone 7810 User Guide*, March 2013, Document ID 14-280201.

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.