Avaya Solution & Interoperability Test Lab

Application Notes for configuring Aura Alliance Client for Skype for Business Softphone Mode with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning the Aura Alliance Client for Skype for Business Softphone mode to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Readers should pay particular attention to the scope of testing as outlined in **Section 2.1**, as well as 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 for provisioning the Aura Alliance Client for Skype for Business Softphone mode to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

The Aura Alliance Client for Skype for Business Softphone mode supports Session Initiation Protocol (SIP) for easy integration with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. It easily integrates with Skype for Business that support audio and video calls with Voice over IP (VoIP) telephony systems that use SIP (Session Initiation Protocol).

2. General Test Approach and Test Results

The interoperability compliance testing evaluates the ability of the Aura Alliance Client for Skype for Business to receive calls from Avaya Digital, H.323 and SIP desk phones as well as PSTN endpoints. The softphone application is registered to Session Manager as a SIP endpoint.

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/Smartphones 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/Smartphones 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.

2.1. Interoperability Compliance Testing

The compliance testing included the test scenarios shown below. Note that when applicable, all tests were performed with Avaya SIP phones, H.323 phones, Digital phones and PSTN endpoints.

- Registration of Aura Alliance Client for Skype for Business Softphone mode.
- Invalid usernames/passwords for registration.
- Basic calls such as hold and retrieve, blind and consultation transfer, and conference calls.
- Video conference call.
- DTMF RFC2833.
- Codec G.711 and G.729 support.
- Serviceability testing.

2.2. Test Results

All test cases passed successfully with the following observation.

- Aura Alliance Client for Skype for business does not support Message Waiting Indicator (MWI) therefore there is no subscriber for MWI sent to Session Manager during the registration.

2.3. Support

Support from Avaya is available by visiting the website http://support.avaya.com and a list of product documentation can be found in **Section 9** of these Application Notes. Technical support for the Aura Alliance Client product can be obtained as follows:

Aura Alliance Limited
Tel: +44 (0)20 3127 7761
http://www.auraalliance.com/global-support/

3. Reference Configuration

Figure 1 shows the network topology during compliance testing an Aura Alliance Client Softphone from Aura Alliance with Communication Manager and Session Manager.

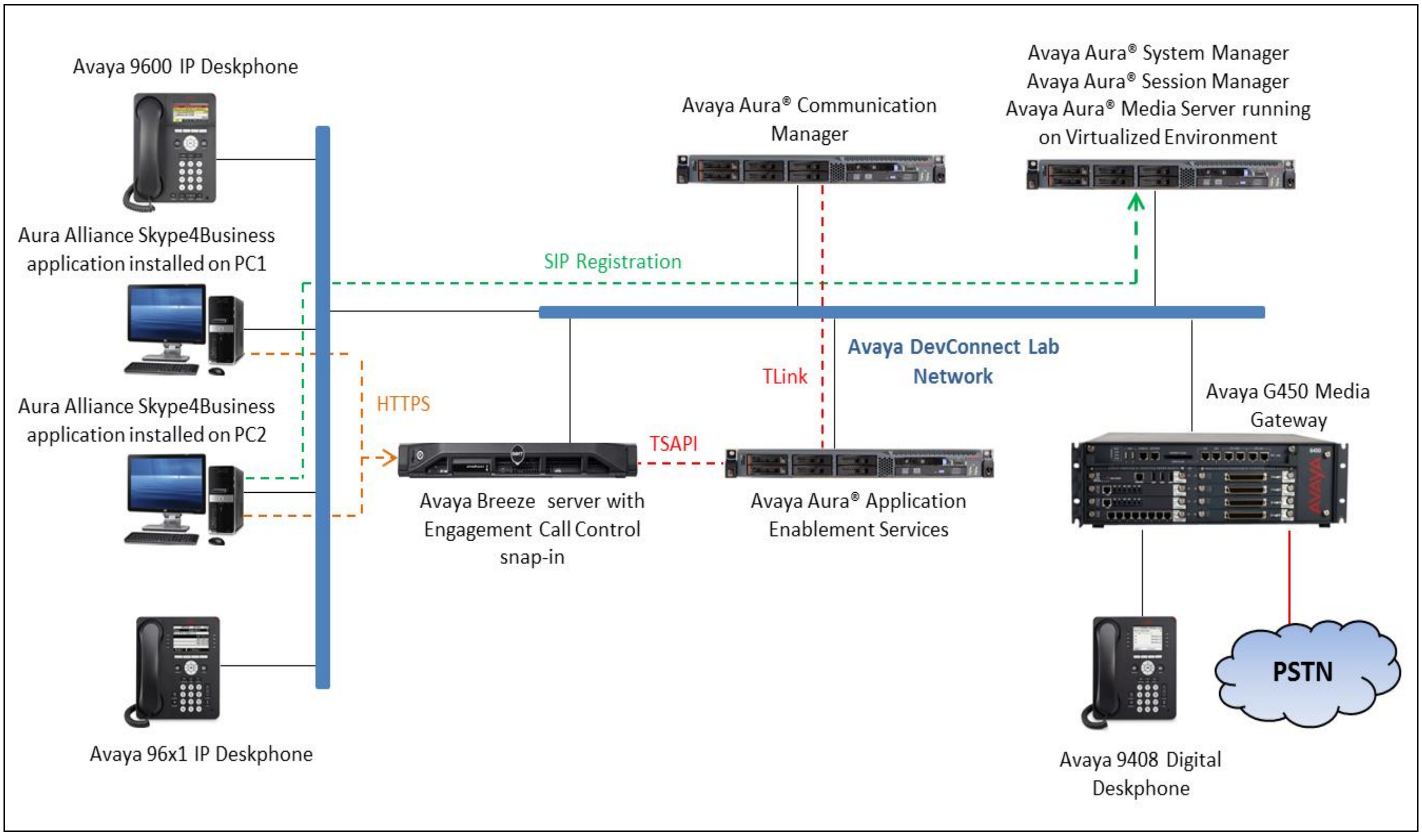


Figure 1: Connection of Aura Alliance Client Softphone mode with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

4. Equipment and Software Validated

The following equipment and software was used for the compliance test.

Equipment/Software	Version/Release
Avaya Aura® Communication Manager running on a virtual platform	R 7.0.1.1.0.441.23169
Avaya Aura® Session Manager running on a virtual platform	R 7.0.1.1.701114
Avaya Aura® System Manager running on a virtual platform	R 7.0.1.2 Revision 7.0.1.2.075662 Service Pack 2
Avaya 9611G Deskphone	H.323 Release 6.6029
Avaya 9611G Deskphone	SIP 7.0
Avaya 9408 Digital Deskphone	V 2.0
Aura Alliance Client for Skype for Business	3.2.51.1

5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of Communication Manager for this solution. It is implied a working system is already in place, including SIP trunks to a Session Manager. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**. The configuration described in this section can be summarized as follows:

- Verify System Capacity
- Define the Dial Plan

Note: Any settings not in **Bold** in the following screen shots may be left as Default.

5.1. Verify System Capacity

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 1**, verify that the **Maximum Off-PBX Telephones** allowed in the system is sufficient. One OPS station is required per SIP device.

```
display system-parameters customer-options                      Page   1 of  10
                                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 290
                                      Maximum Stations: 41000 44
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 41000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 14
                    Maximum Off-PBX Telephones - PBFMC: 41000 0
                    Maximum Off-PBX Telephones - PVFMC: 41000 0
                    Maximum Off-PBX Telephones - SCCAN: 41000 0
                         Maximum Survivable Processors: 313   0

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

On **Page 2** of the **system-parameters customer-options form**, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

```
display system-parameters customer-options                         Page   2 of
10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
                     Maximum Administered H.323 Trunks: 12000 16
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 41000 1
                   Maximum Video Capable IP Softphones: 18000 4
                       Maximum Administered SIP Trunks: 24000 180
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   0
                     Maximum Media Gateway VAL Sources: 250   0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   0
   Maximum Number of Expanded Meet-me Conference Ports: 300   0

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

5.2. Define the Dial Plan

Use the **change dialplan analysis** command to define the dial plan used in the system. This includes all telephone extensions. In the sample configuration, telephone extensions are 4 digits long and begin with **33** and **34**.

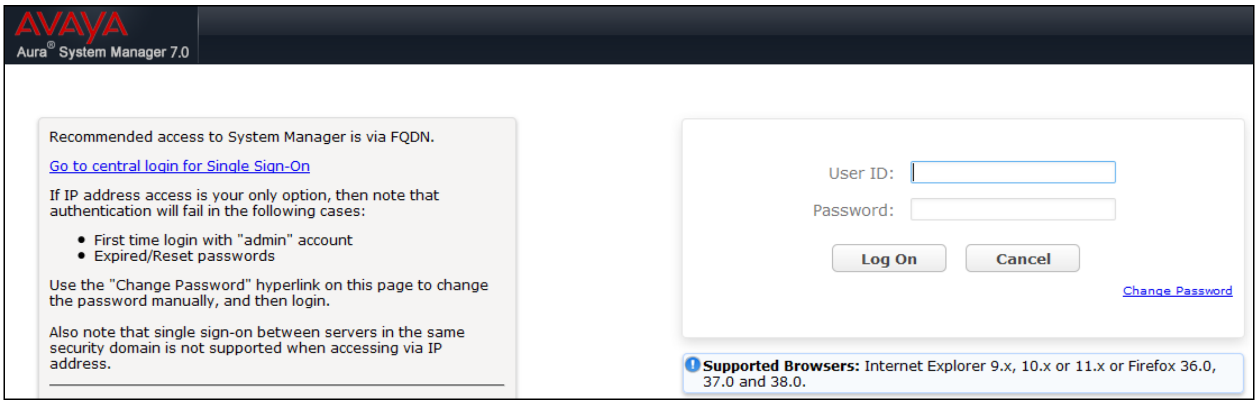
```
change dialplan analysis                                           Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 1

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
   33          4   ext
   34          4   ext
   *           3   fac
   #           3   fac
```

6. Configure Avaya Aura® Session Manager

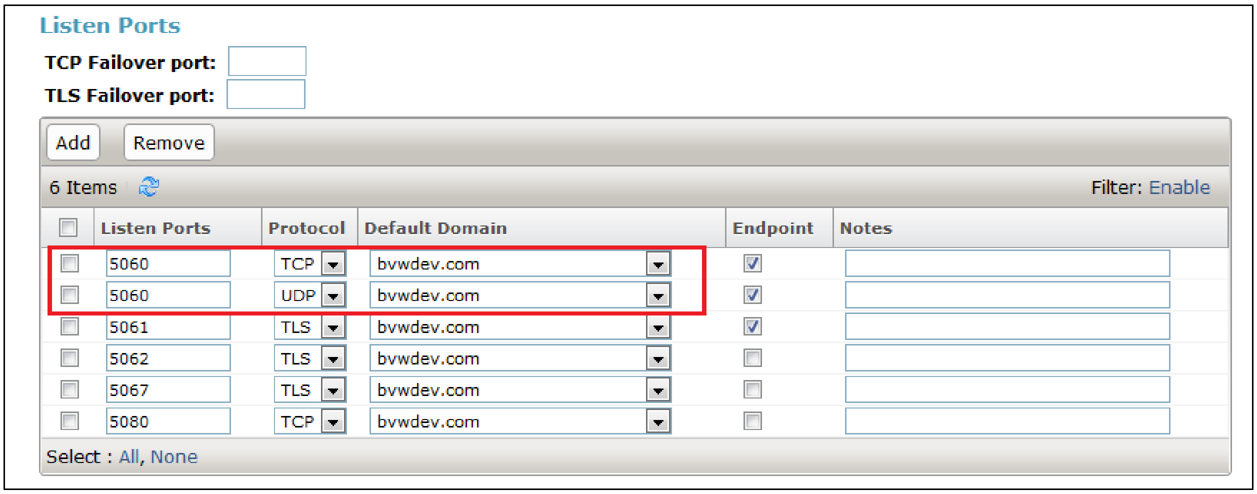
This section describes aspects of the Session Manager configuration required for interoperating with Aura Alliance Client. It is assumed that the Domains, Locations, SIP entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication Manager, Session Manager and Aura Messaging.

Session Manager is managed via System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and click the **Log On** button.



6.1. Check Avaya Aura® Session Manager ports for Aura Alliance Client Registration

Each Session Manager Entity must be configured so that the Aura Alliance Client Softphone can register to it using UDP/TCP. From the web interface click **Routing → SIP Entities** (not shown) and select the Session Manager entity used for registration. Make sure that **TCP** and **UDP** entries are present. The TCP and UDP entries are highlighted below.

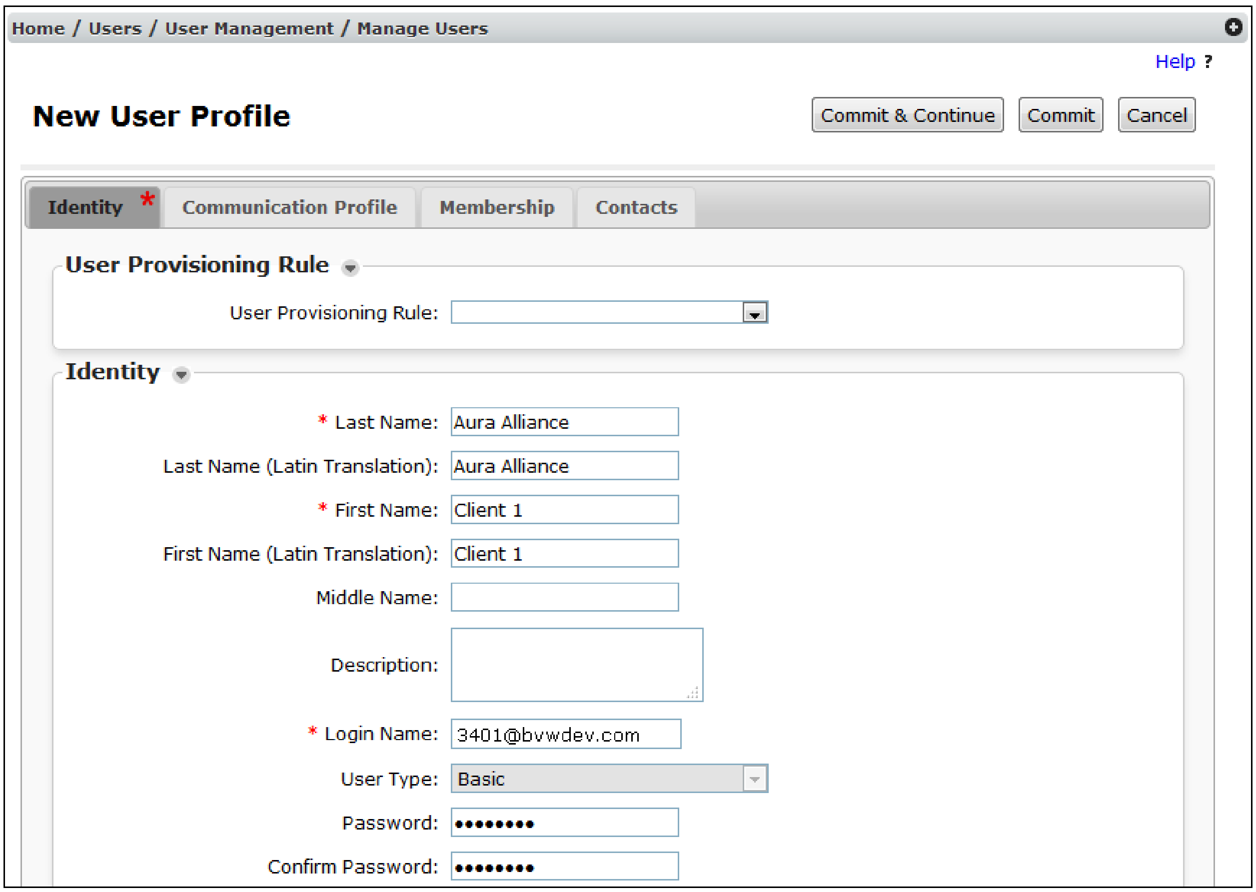


Repeat accordingly on the alternative Session Manager if applicable.

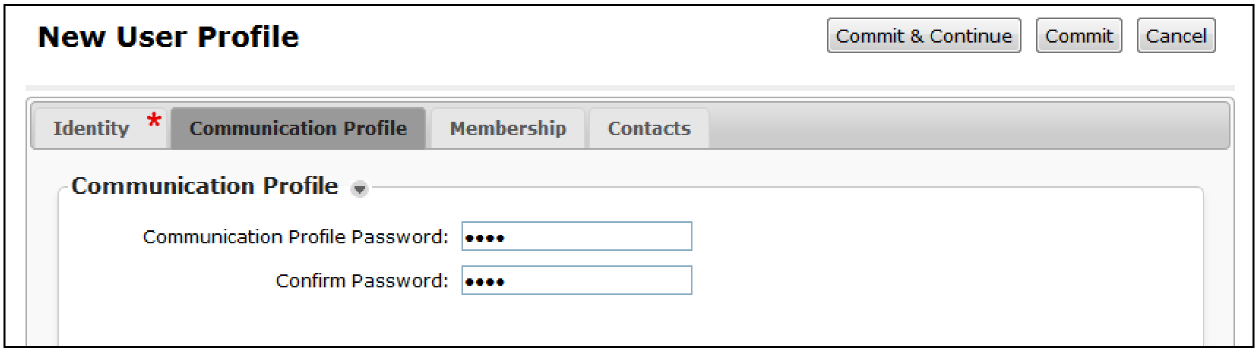
6.2. Add a SIP User

The Aura Alliance Client SIP user must be added as a user. A user must be added for each Aura Alliance Client. Click **User Management → Manage Users → New** (not shown) and configure the following in the **Identity** tab.

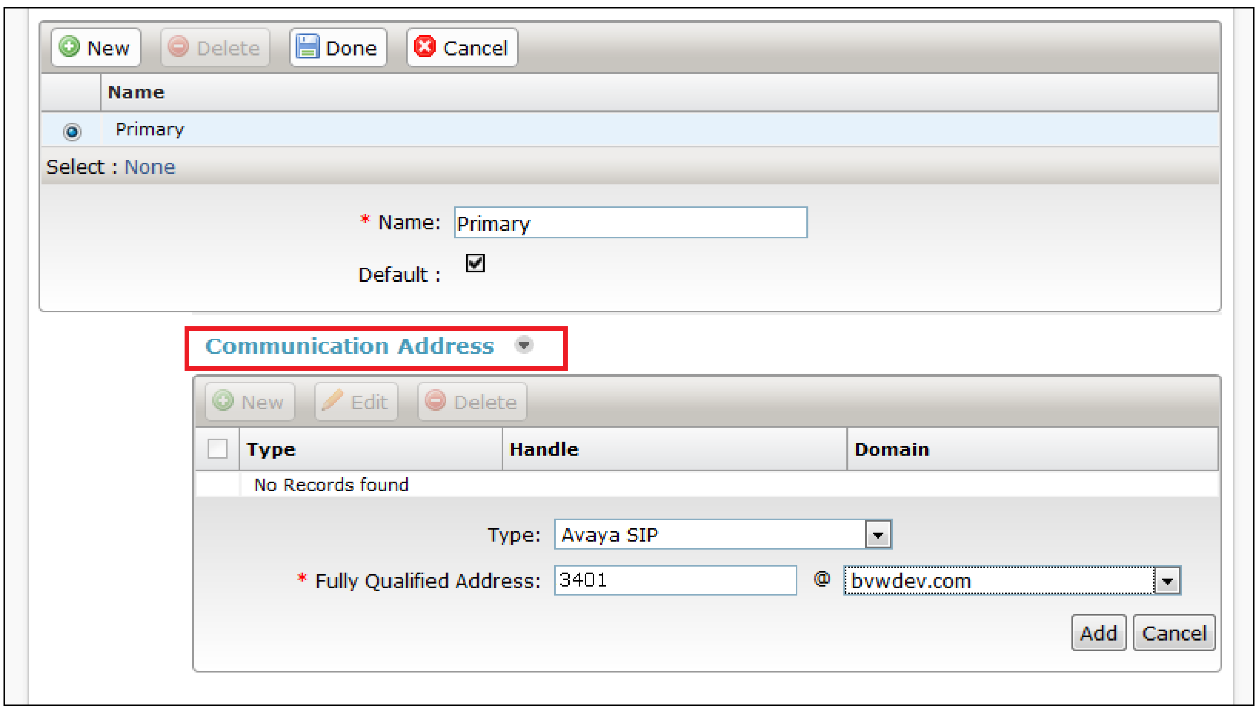
- **First Name** and **Last Name** Enter an identifying name
- **Login Name** Enter the extension number followed by the domain, in this case **3401@bvwdev.com**
- **Authentication Type** Select **Basic** from the drop down list
- **Password** and **Confirm Password** Enter and confirm a password



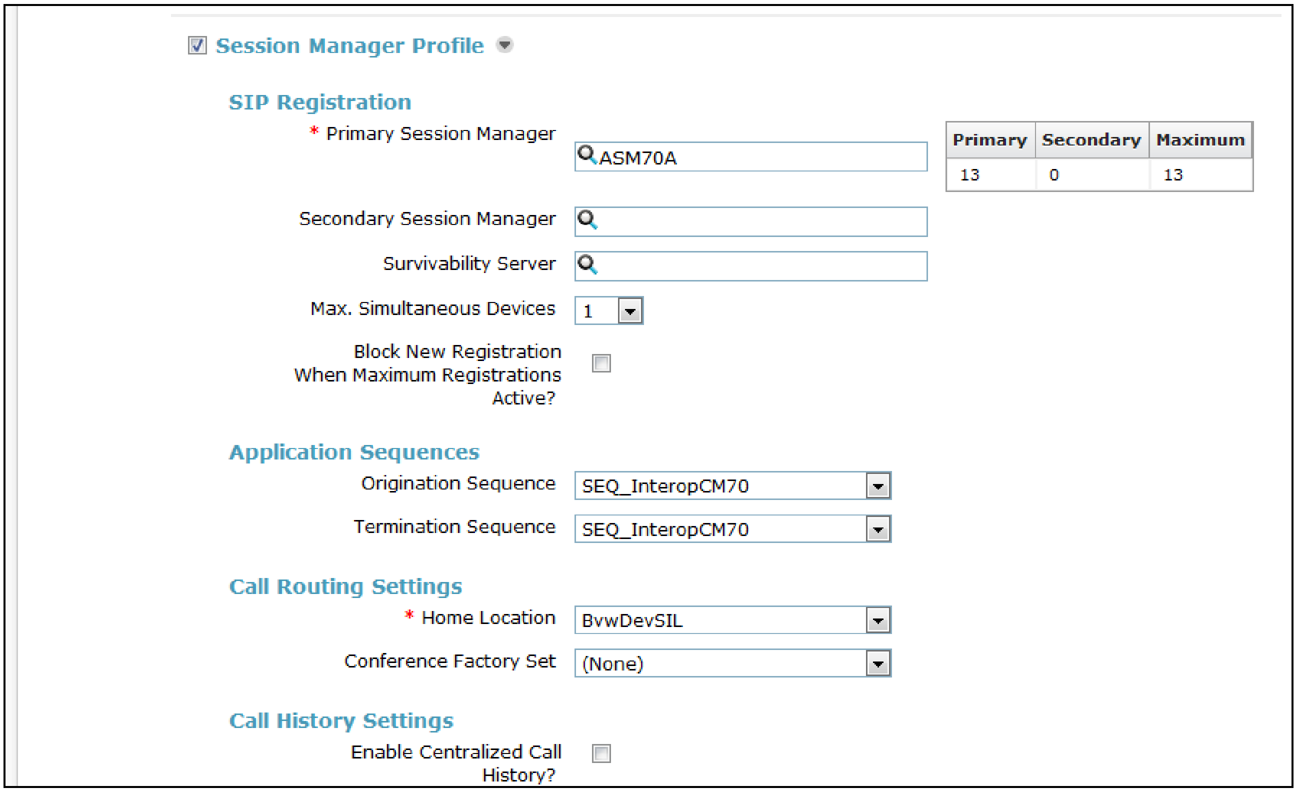
Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the Network Ceiling Speaker during login.



In the **Communication Address** section, for **Type** select **Avaya SIP** from the drop down list. In the **Fully Qualified Address** field enter the extension number as required and select the appropriate **Domain** from the drop down list. Click **Add** when done.



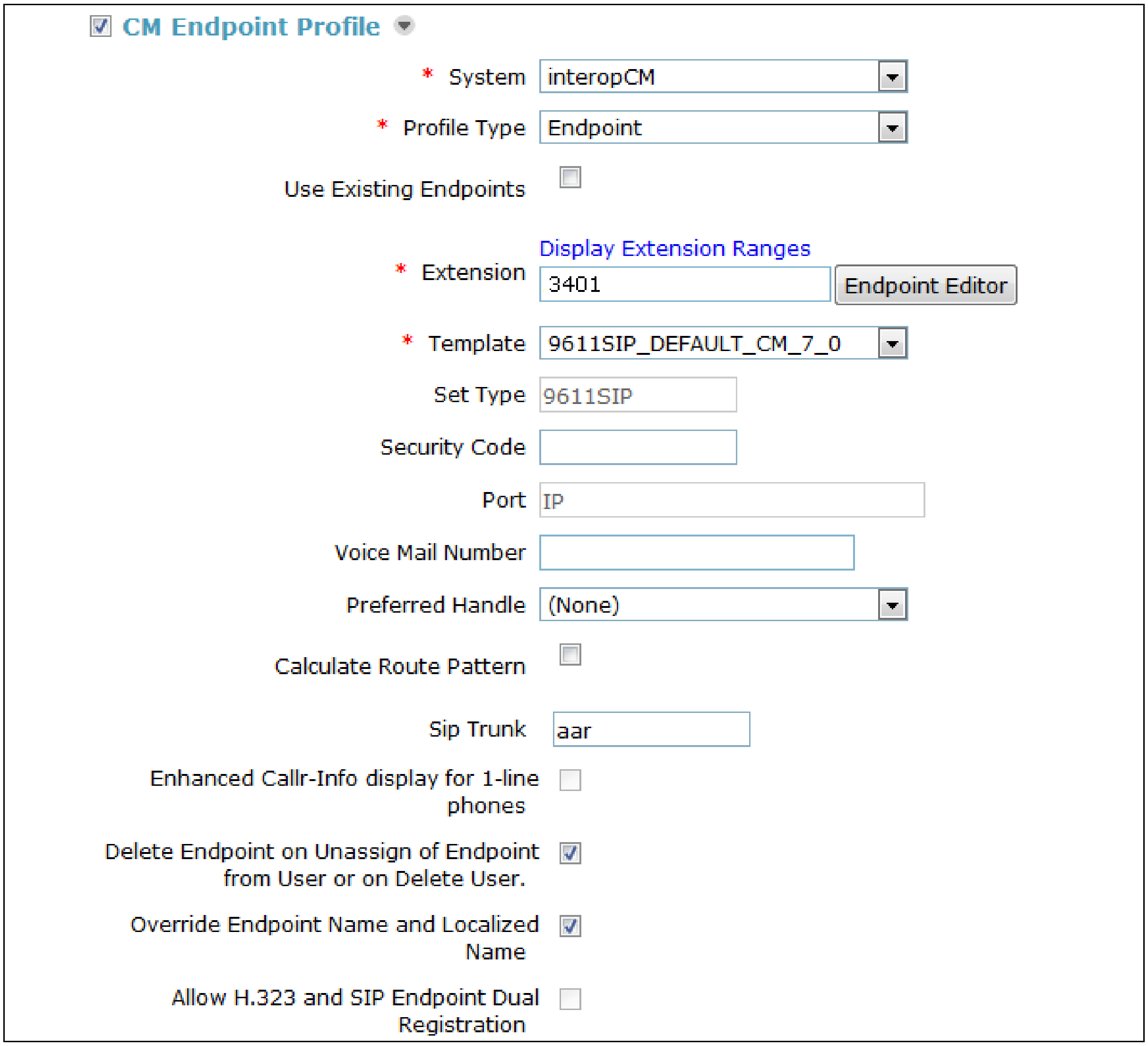
Place a tick in the **Session Manager Profile** check box and configure the **Primary Session Manager, Origination Application Sequence, Termination Application Sequence** and **Home Location,** from the respective drop down lists. The Primary Session Manager used was **ASM70A**.



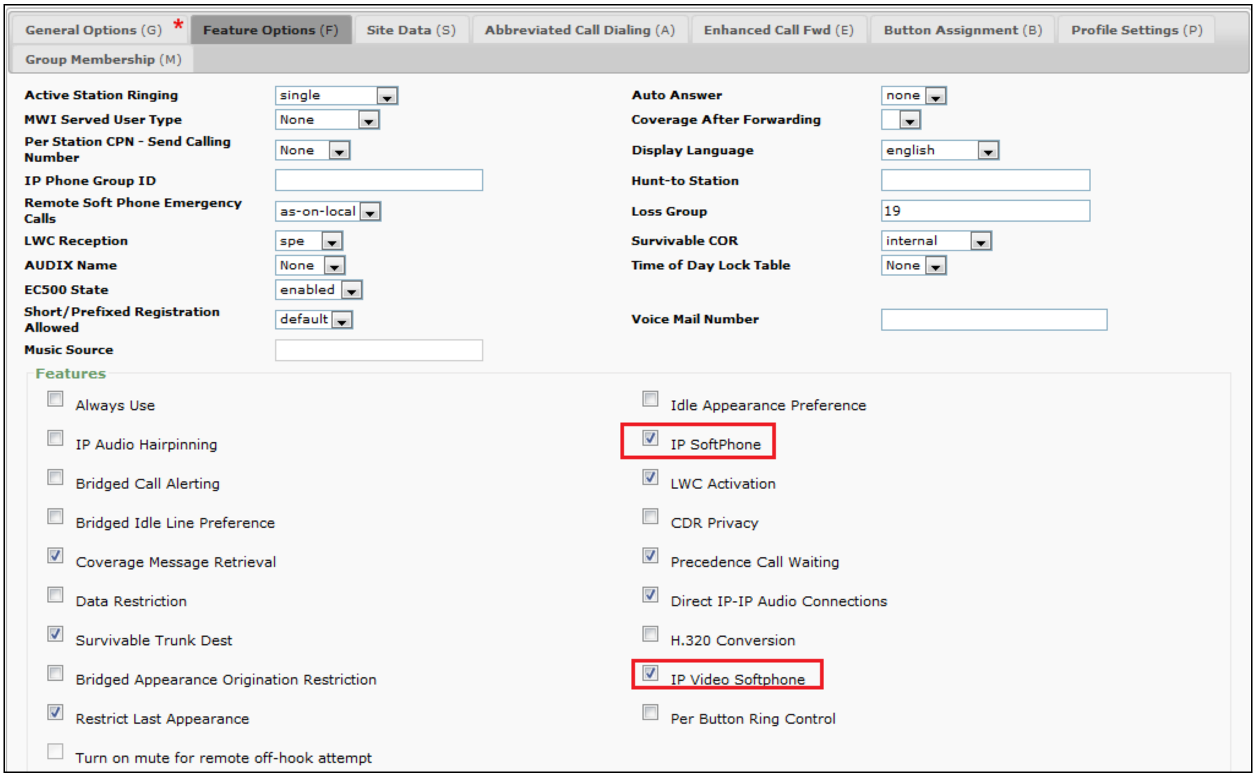
Place a tick in the **CM Endpoint Profile** check box and configure as follows:

- **System** Select the relevant Communication Manager SIP Entity from the drop down list
- **Profile Type** Select **Endpoint** from the drop down list
- **Extension** Enter the required extension number, in this case **3401**
- **Template** Select **9611SIP_DEFAULT_CM_7_0** from the drop down list
- **Port** The “IP” is auto filled out by the system

Click on **Endpoint Editor**.



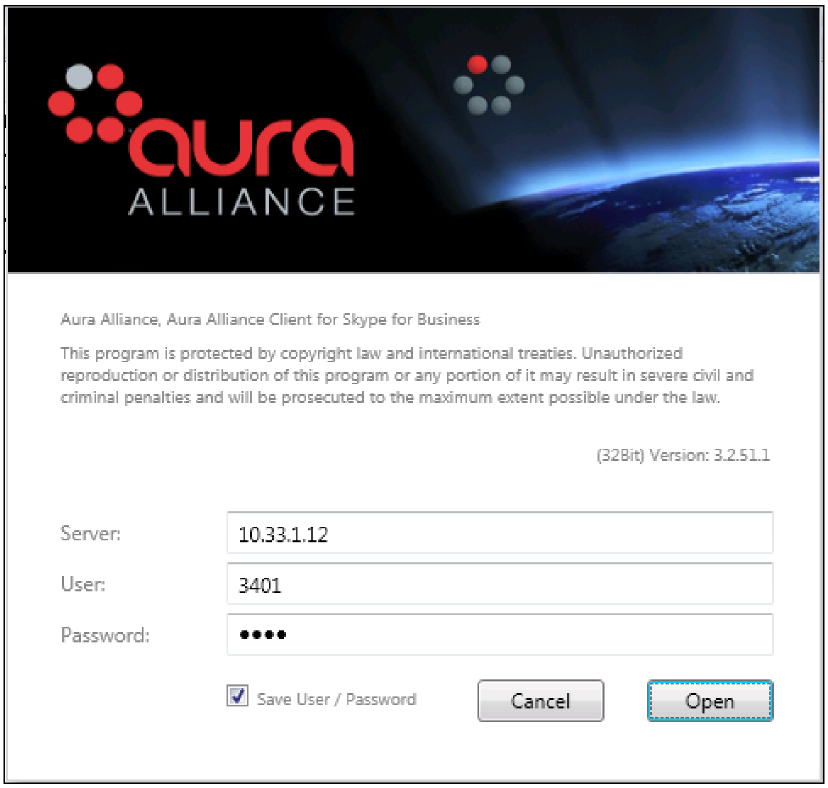
Click on the **Feature Options** tab, the screen shot below shows the Feature options that were used during compliance testing.



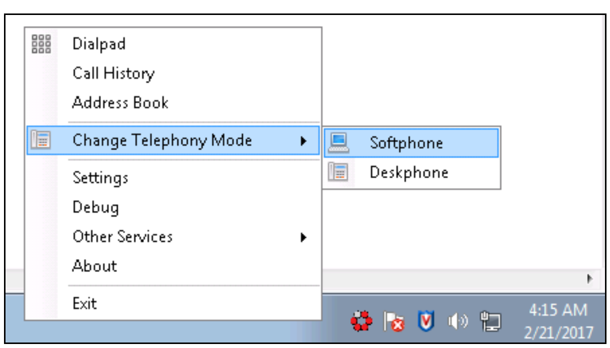
7. Configure Aura Alliance Client

Please refer to Aura Alliance Client documentation listed in **Section 10** of these Application Notes for further information about the Aura Alliance Client configuration. The following sections cover specific settings concerning SIP and the connection to Session Manager.

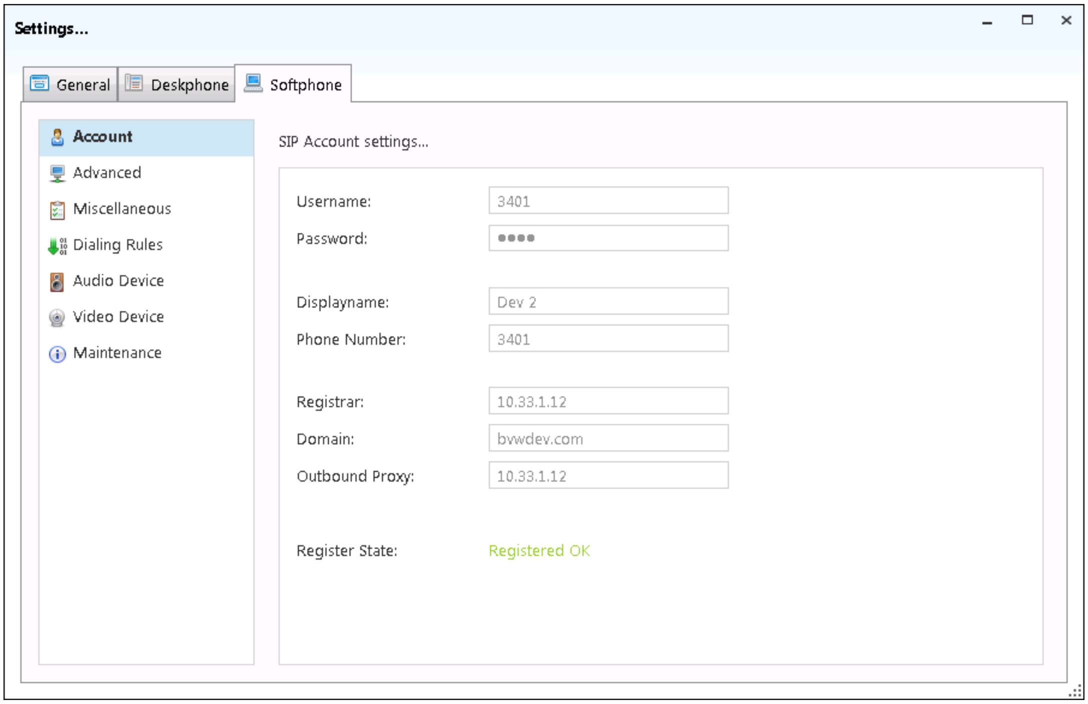
From the PC where Aura Alliance Client for Skype for Business application is installed, run the application from the Start menu. Enter the signalling IP address of Session Manager in the Server field, the SIP user extension in as configured in **Section 6.2** in **User** and **Password** fields. Click **Open** button to start the application.



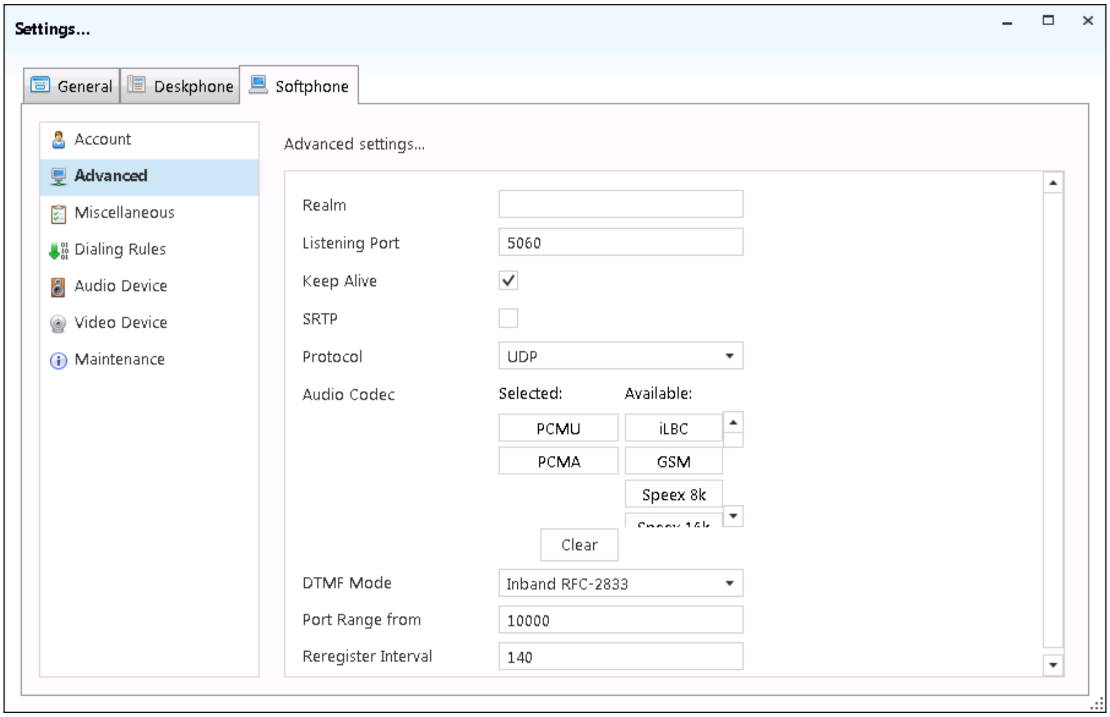
The Aura Alliance Client for Skype for Business application appears in the system tray, right click on the application and select the **Softphone** mode.



Navigate to **Settings** from the context menu above and select the **Softphone** tab, in the **Register State** if the softphone is able to register to Session Manager the state should be in "Registered OK".



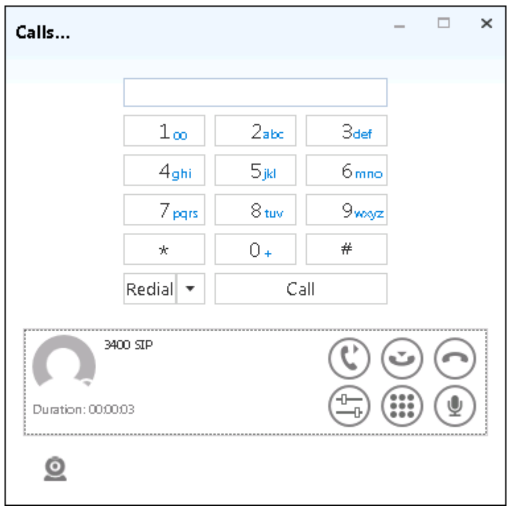
In order to configure the transport protocol, codec and DTMF settings, navigate to the left menu and select **Advanced**. In the compliance test, the UDP protocol, codec G.711U and DTMF RFC2833 were used.



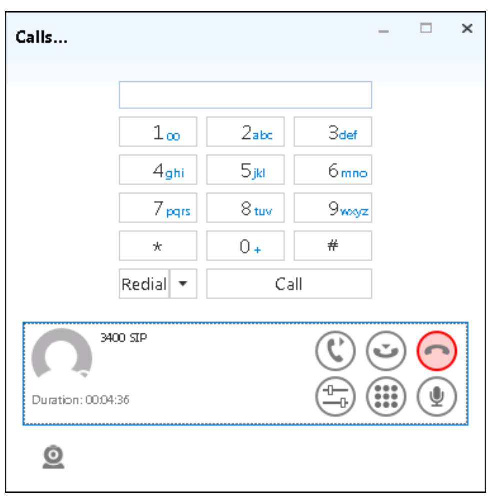
8. Verification Steps

These steps below may be used to verify functions of Aura Alliance Client for Skype for Business softphone with Session Manager using telephony features in Avaya Communication.

1. Place a call from a station to SIP user extension 3401 that the Aura Alliance Client application is registering to.
2. Answer the call on the Aura Alliance Client for Skype for Business application by selecting the **Answer** button (not shown), the **Calls** window below shows the call established.

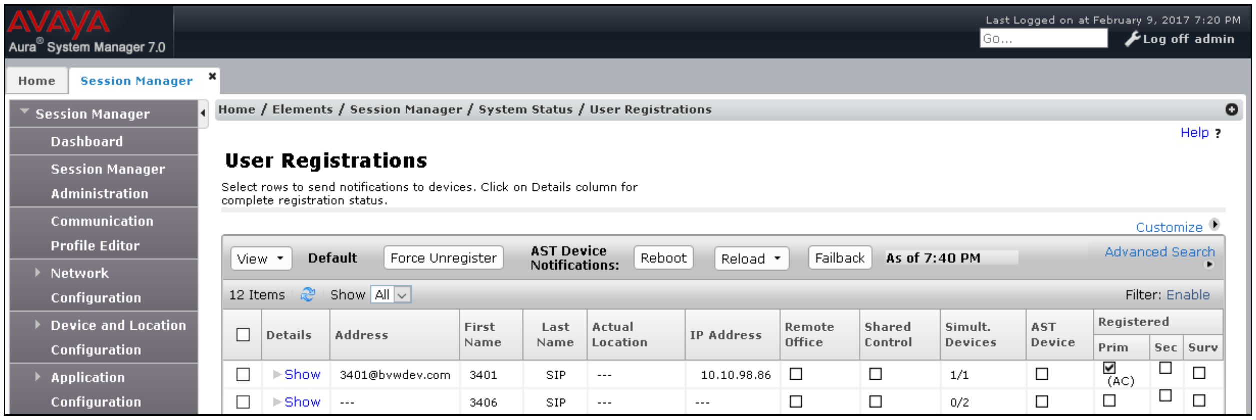


3. From the **Calls** window, hang up the call by select the Hang up button (red circle). The call is released and the softphone returns to idle state.



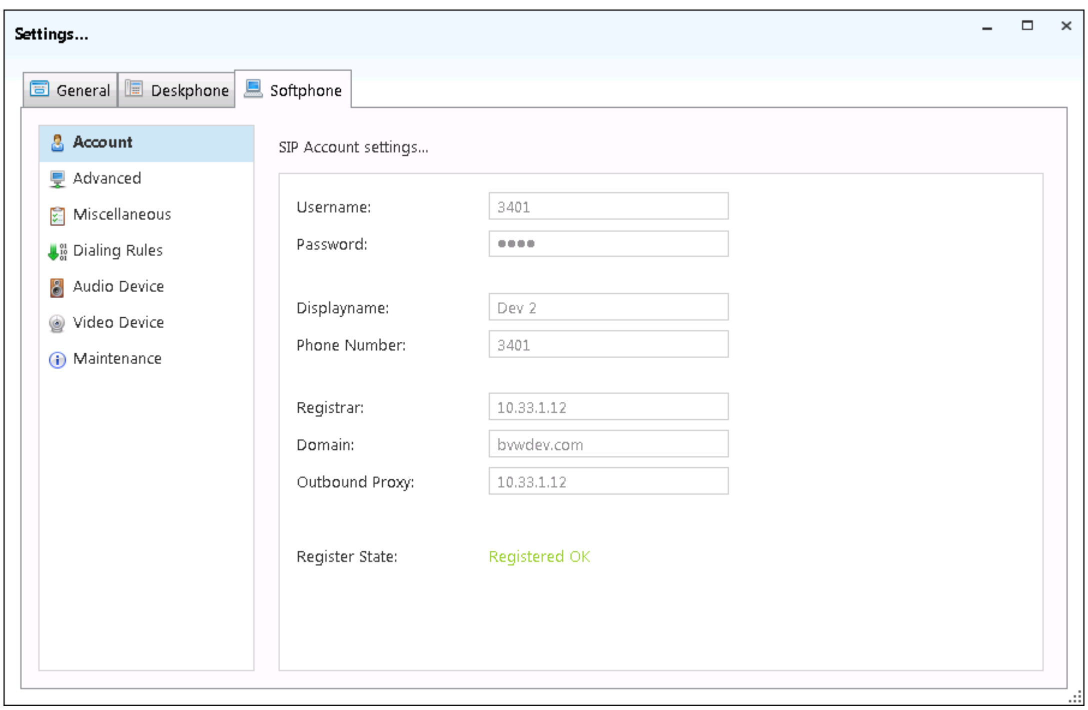
8.1. Verify Registration to Avaya Aura® Session Manager

From the System Manager dashboard select **Session Manager** from the **Elements** section (not shown). From the left hand menu select **System Status→User Registrations** (not shown). The Aura Alliance Client for Skype for Business is listed and a tick under **Registered** for the Session Manager it is registered to.



8.2. Verify Registration from Aura Alliance Client Skype for Business

Navigate to **Settings → Softphone → Account**; the **Register State** should display "Registered OK".



9. Conclusion

These Application Notes describe the configuration steps for provisioning the Skype for Business from Aura Alliance to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Please refer to **Section 2.2** for test results and observations.

10. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com where the following documents can be obtained.

Product documentation for Avaya products may be found at http://support.avaya.com.

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

[2] Avaya Aura® Communication Manager Feature Description and Implementation, Release 7.0, August 2015, Document Number 555-245-205, Issue 1.

[3] Administering Avaya Aura® Session Manager, Release 7.0, Issue 1 August 2015

[4] Administering Avaya Aura® System Manager, Release 7.0, Issue 1, August, 2015

Technical information for the Aura Alliance can be obtained from:
Aura Alliance Limited
Tel: +44 (0)20 3127 7761
http://www.auraalliance.com/global-support/

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.