Avaya Solution & Interoperability Test Lab

Application Notes for configuring Komutel SIT2 IP Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning the Komutel SIT2 IP Console 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 steps required to integrate the Komutel SIT2 IP Console (**S**olution for **I**ntegrated **T**elecommunications) with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. The SIT2 IP Console provides a desktop communications center with enhanced control of call handling features. It provides the ability to handle a high volume of calls and offers tools designed to manage telephony functions. In the compliance test, the SIT2 IP Console successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

2. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing focused on establishing calls between the SIT2 IP Console and Avaya SIP, H.323, and digital stations and exercising common telephony features, such as hold, transfer, and conference.

The serviceability testing focused on verifying that the SIT2 IP Console comes back into service after re-connecting the Ethernet connection or rebooting the PC on which the SIT2 IP Console is running.

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

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.

- Successful registration of the SIT2 IP Console with Session Manager.
- Calls between SIT2 IP Console and Avaya SIP, H.323, and digital stations.
- G.711 codec support.
- Caller ID display on Avaya and Komutel endpoints.
- Proper recognition of DTMF tones.
- Basic telephony features including Hold, Mute, Transfer, and Conference.
- Extended telephony features using Communication Manager Feature Access Code (FAC) such as Call Park and Call Pickup.
- Proper system recovery after a restart of the SIT2 IP Console and loss of IP connectivity.

2.2. Test Results

All test cases passed successfully with the following observation.

- Komutel SIT2 IP Console supports codec G.711 only.
- Komutel SIT2 IP Console does not support Call Forward at the time of testing.

2.3. Support

For technical support on the SIT2 IP Console, contact Komutel Support via phone, email, or website.

- **Phone:** (877) 225-9988
- **Email:** service@komutel.com
- **Web:** http://www.gotoassist.com/ph/komutel

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Avaya Aura® Communication Manager running on a virtualized environment with a G450 Media Gateway. Communication Manager was configured as an Evolution 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.
- Simulated PSTN via PRI/T1 trunk connected to G450 and via SIP Trunk connected to Session Manager.

In addition, a Komutel SIT2 IP Console registered with Session Manager and was configured as Off-PBX Stations (OPS) on the Communication Manager.

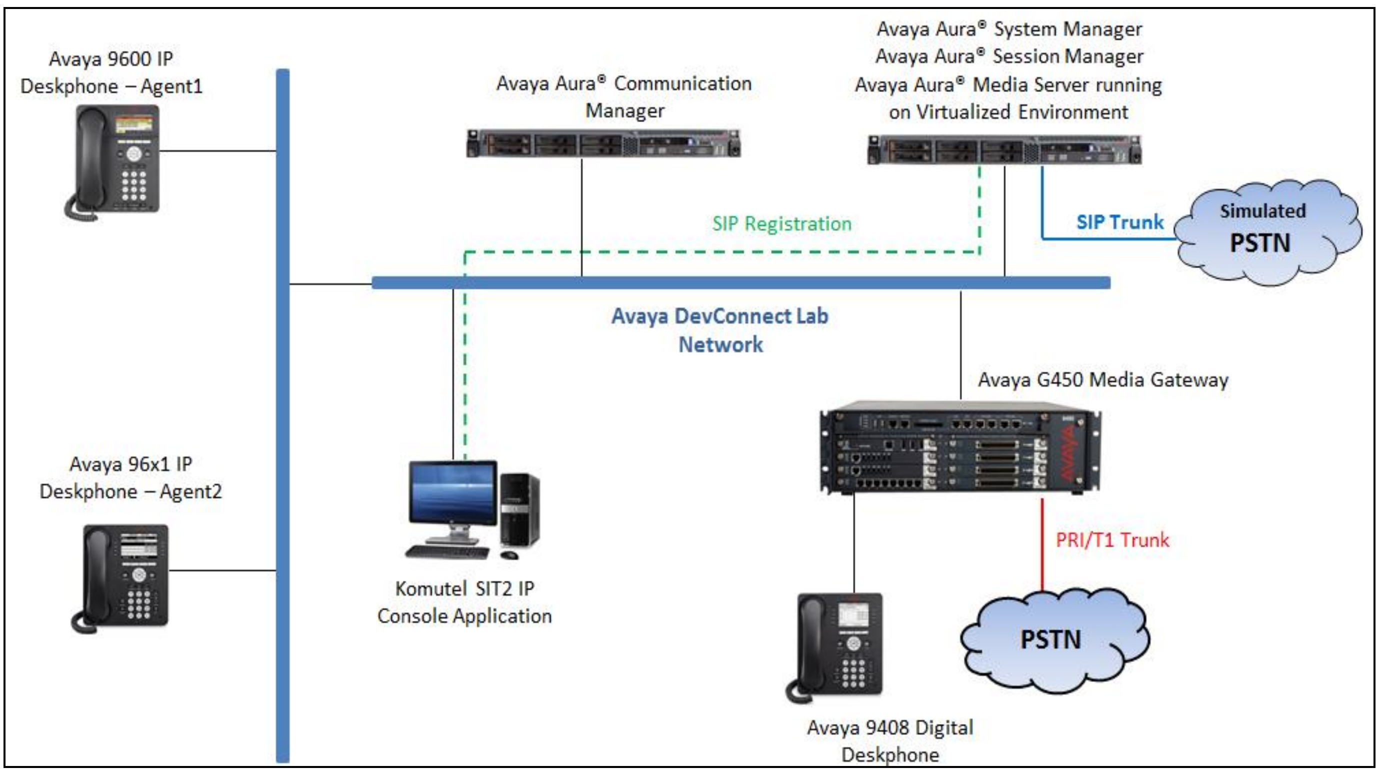


Figure 1: Avaya SIP Network with Komutel SIT2 IP Console

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 7.0.1
Avaya 9408 Digital Deskphone	V 2.0
Komutel SIT2 IP Console with the modTelephony_SIP.dll plugin	2.5.4 1.5.0.33802

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**.

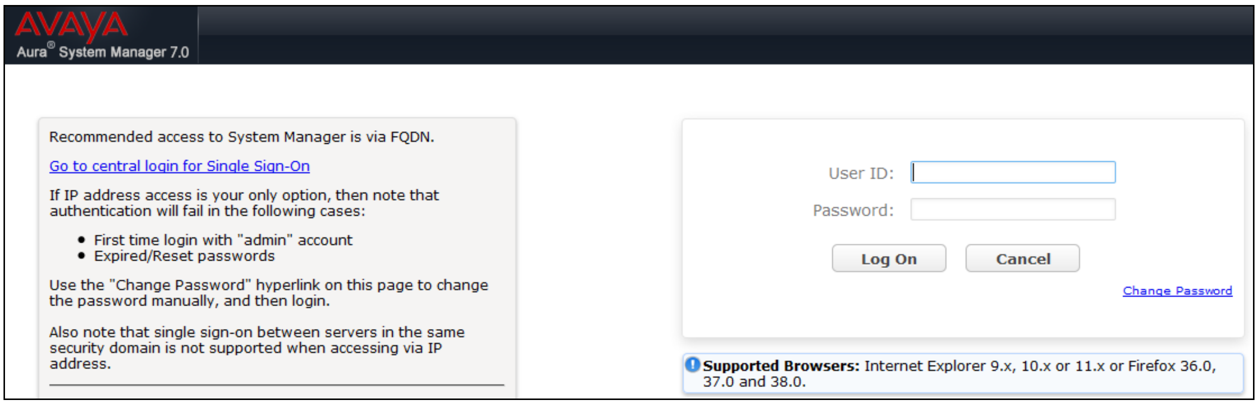
```
change dialplan analysis                                        Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 1

    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

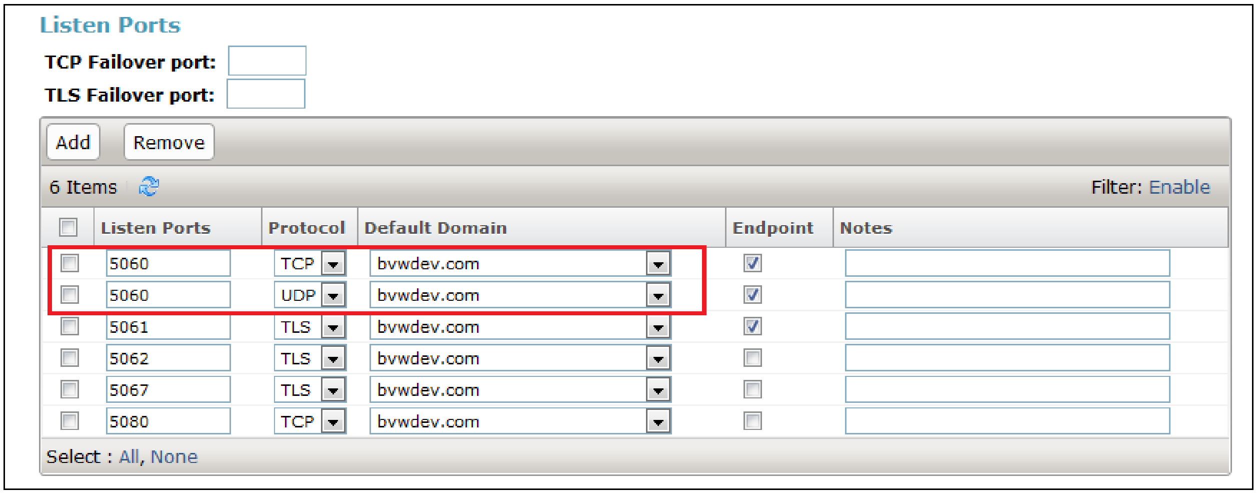
This section describes aspects of the Session Manager configuration required for interoperating with Komutel SIT2 IP Console. It is assumed that the Domains, Locations, SIP entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication 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 SIP Endpoint Registration

Each Session Manager Entity must be configured so that the SIT2 IP Console 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.

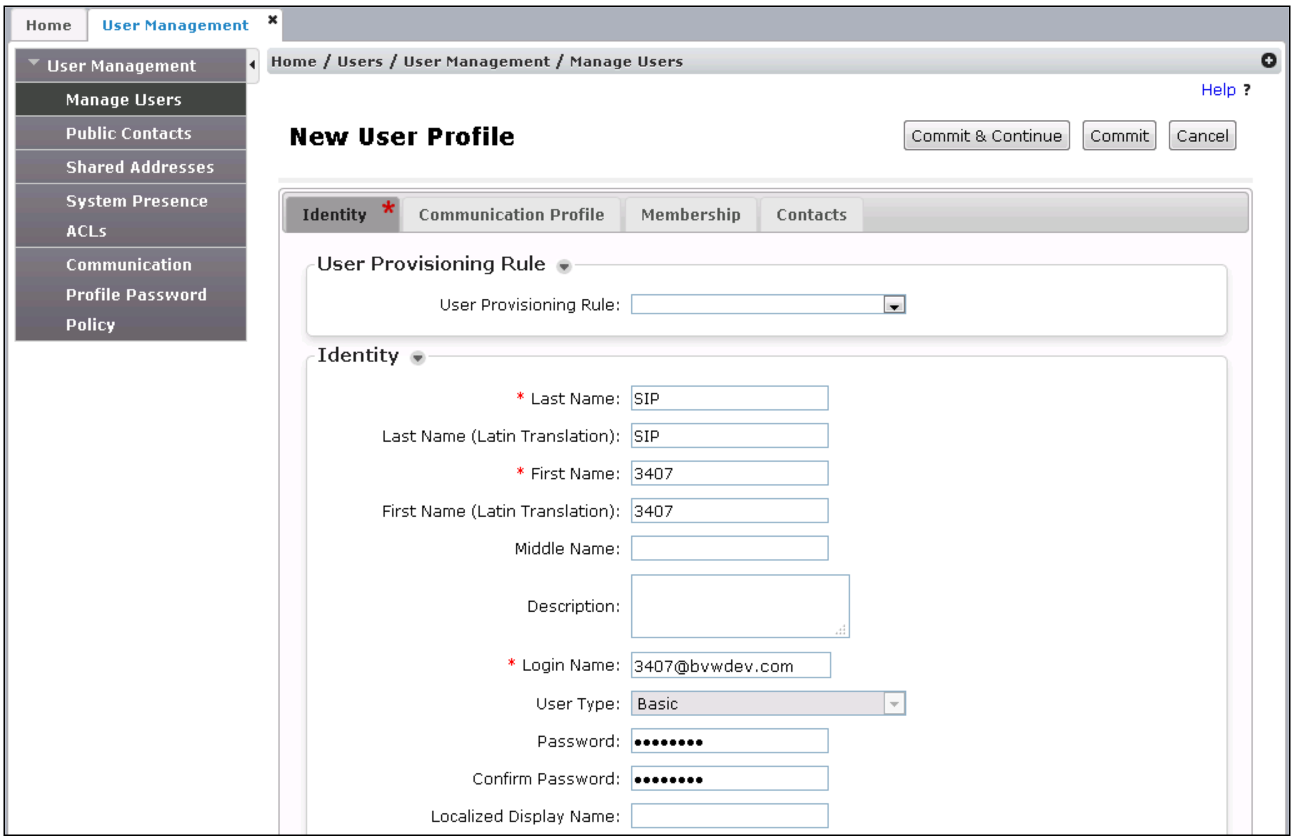


Repeat accordingly on the alternative Session Manager if applicable.

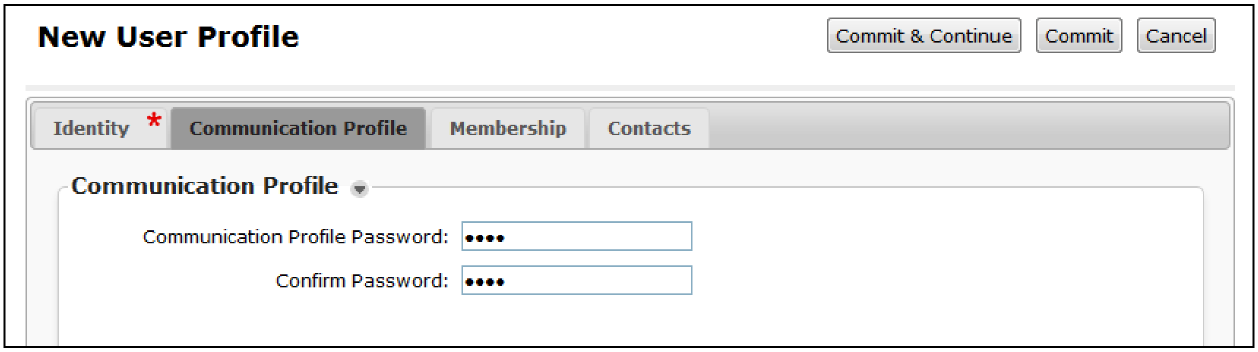
6.2. Add a SIP User

The Komutel SIT2 SIP user must be added as a user. A user must be added for each SIT2 IP Console. Click **User Management → Manage Users → New** (not shown) and configure the following in the **Identity** tab.

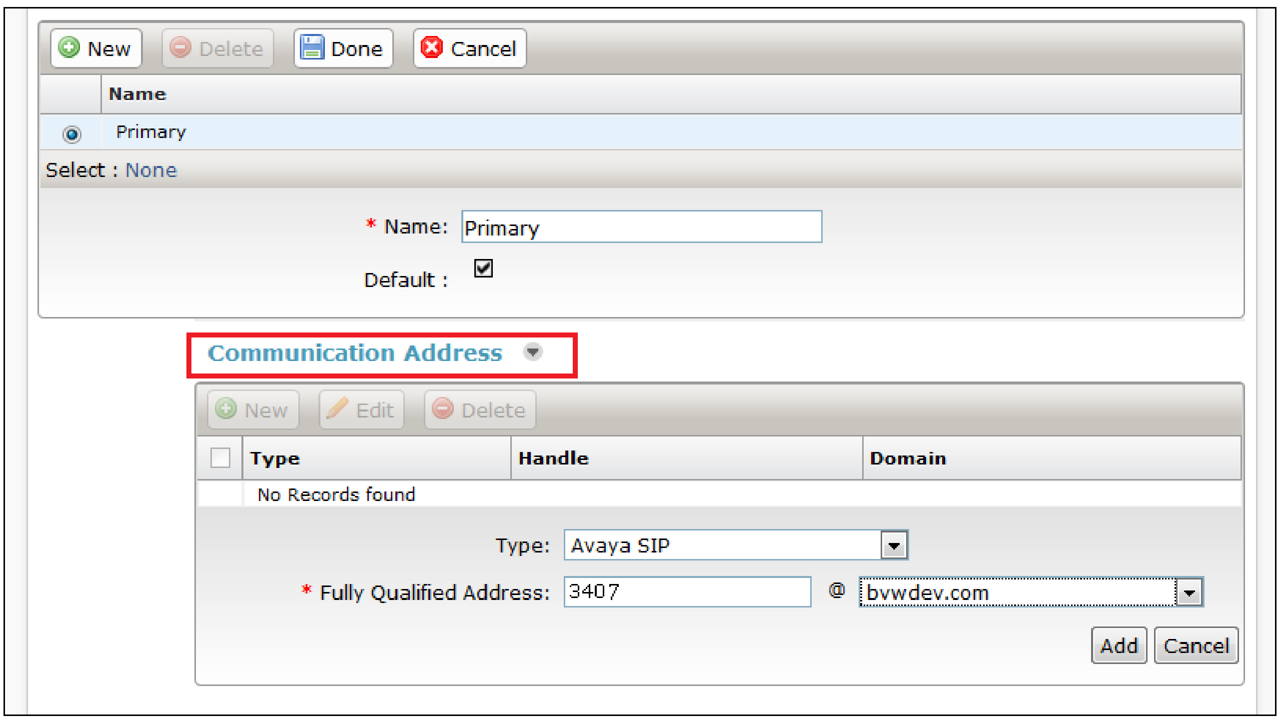
- **First Name** and **Last Name** Enter an identifying name
- **Login Name** Enter the extension number followed by the domain, in this case **3407@bvwdev.com**
- **Authentication Type** Select **Basic** from the drop down list
- **Password** and **Confirm Password** Enter and confirm a password



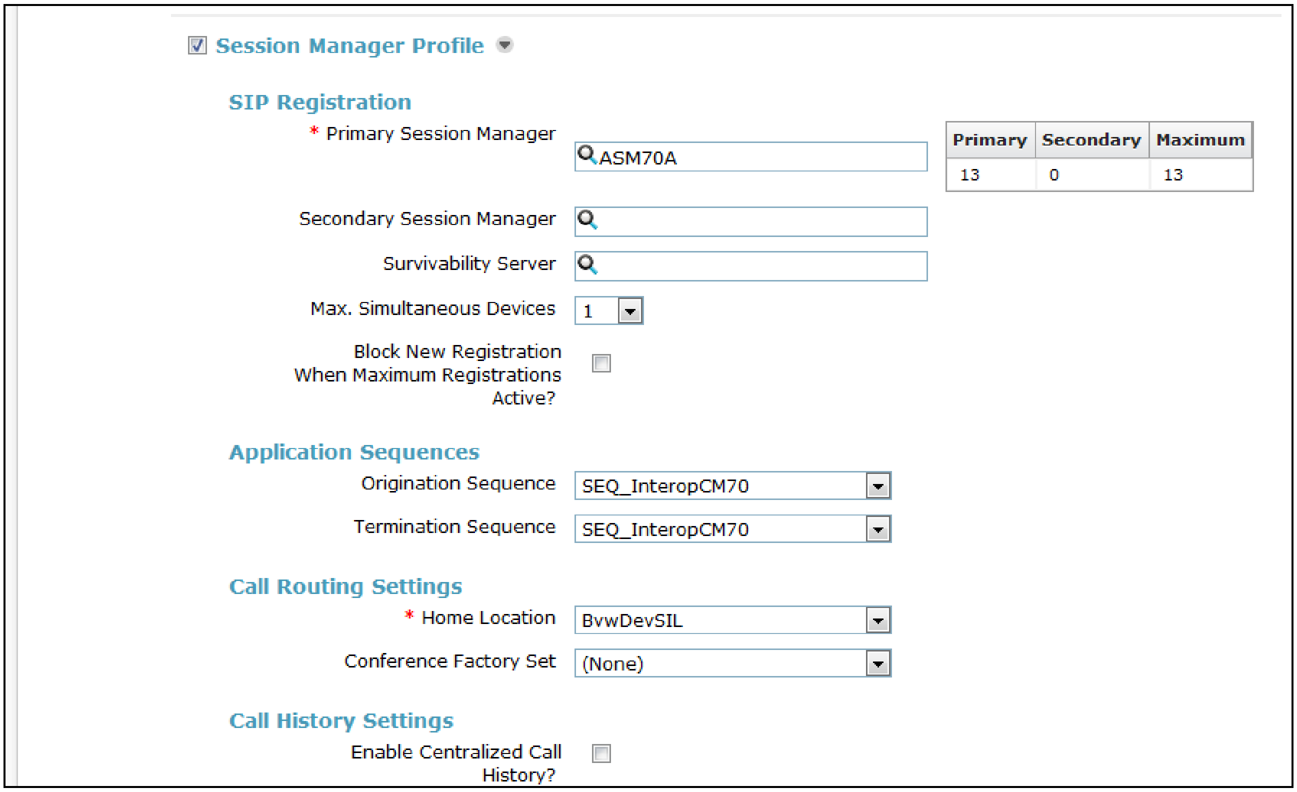
Click the **Communication Profile** tab and in the **Communication Profile Password** and **Confirm Password** fields, enter a numeric password. This will be used to register the SIT2 console during login.



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.



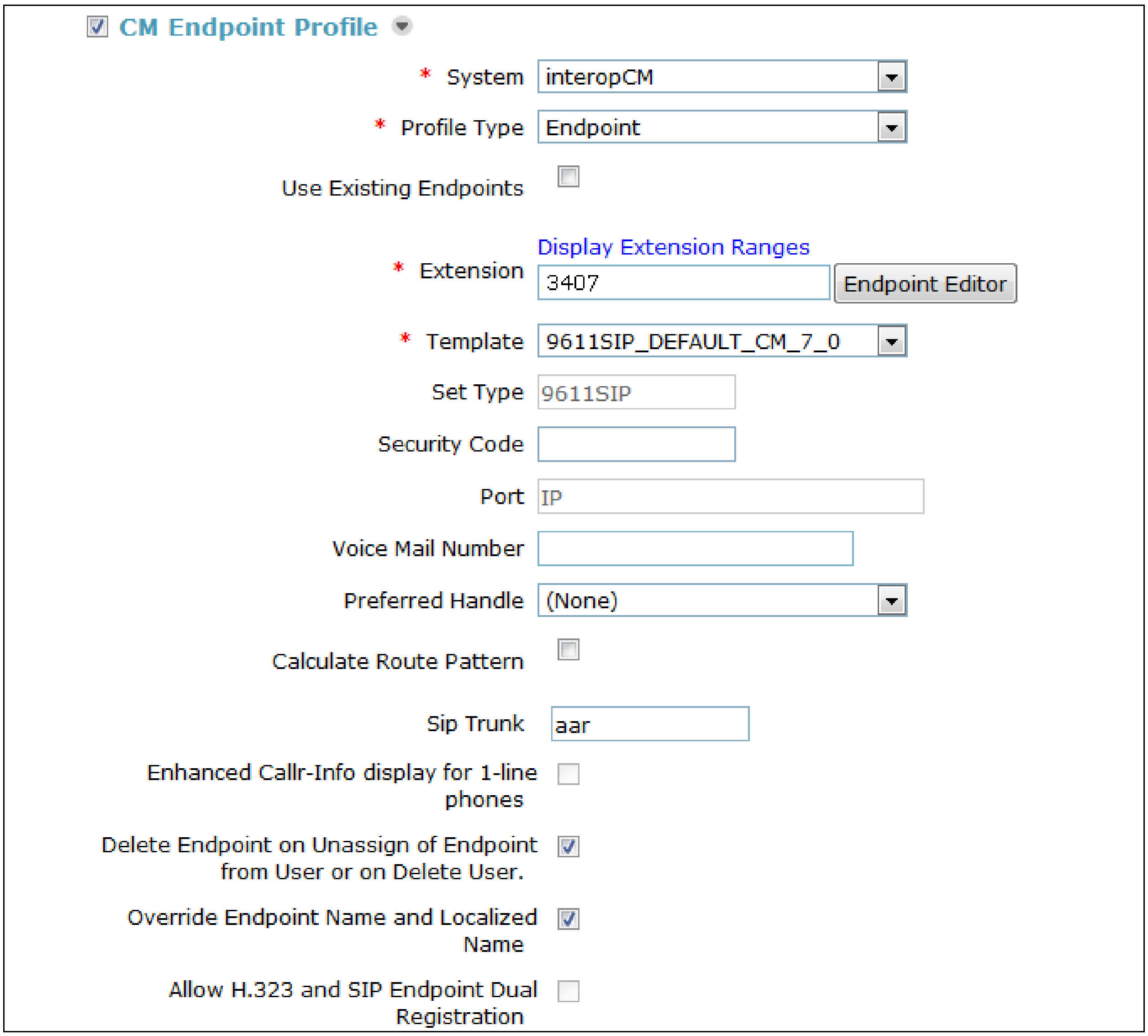
Place a tick in the **Session Manager Profile** check box and configure the **Primary Session Manager, Origination Application Sequence, Termination Application Sequence** and **Home Location,** from the respective drop down lists. The Primary Session Manager used was **ASM70A**.



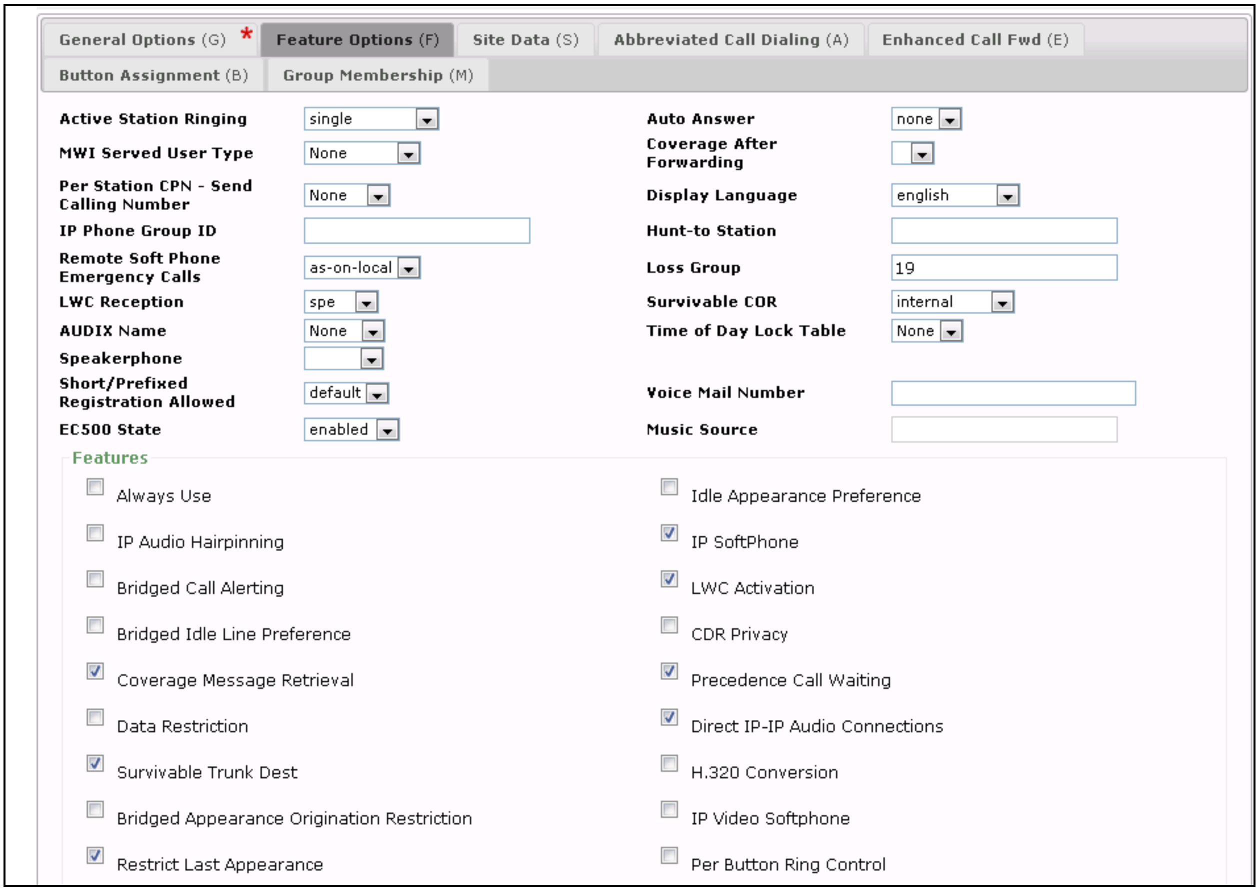
Place a tick in the **CM Endpoint Profile** check box and configure as follows:

- **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 **3407**
- **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.



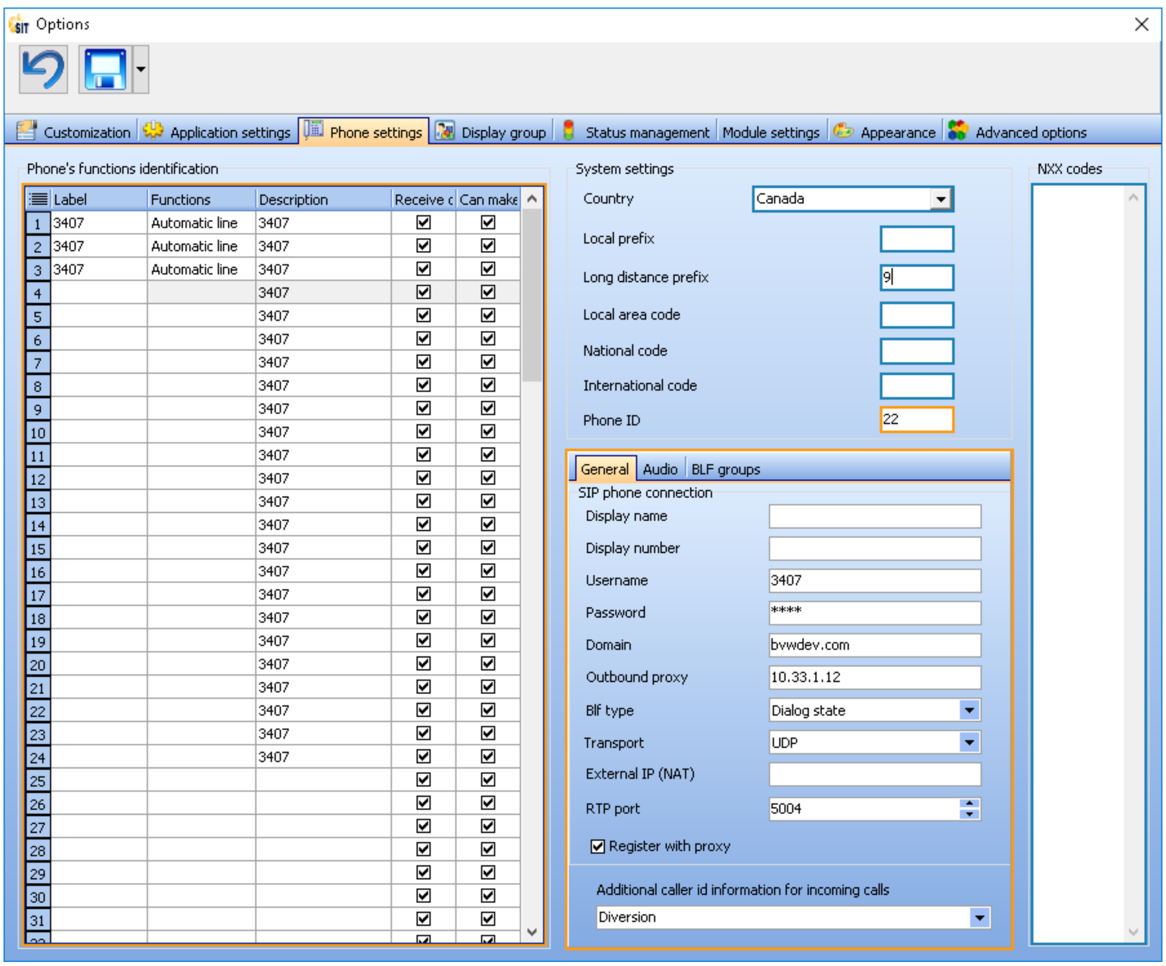
7. Configure Komutel SIT2 IP Console

Launch the SIT application and login in with the appropriate credentials. To configure the console's lines, navigate to the **Tools** menu (not shown) option, and then select the **Phone Settings** tab. Depending on the number of lines that are available, choose *Automatic Line* in the **Functions** column, enter the **DN** in the **Description** column and type the text that will be displayed on the line's button in the **Label** column. As shown below, the console was configured with three line appearances with extension 3407.

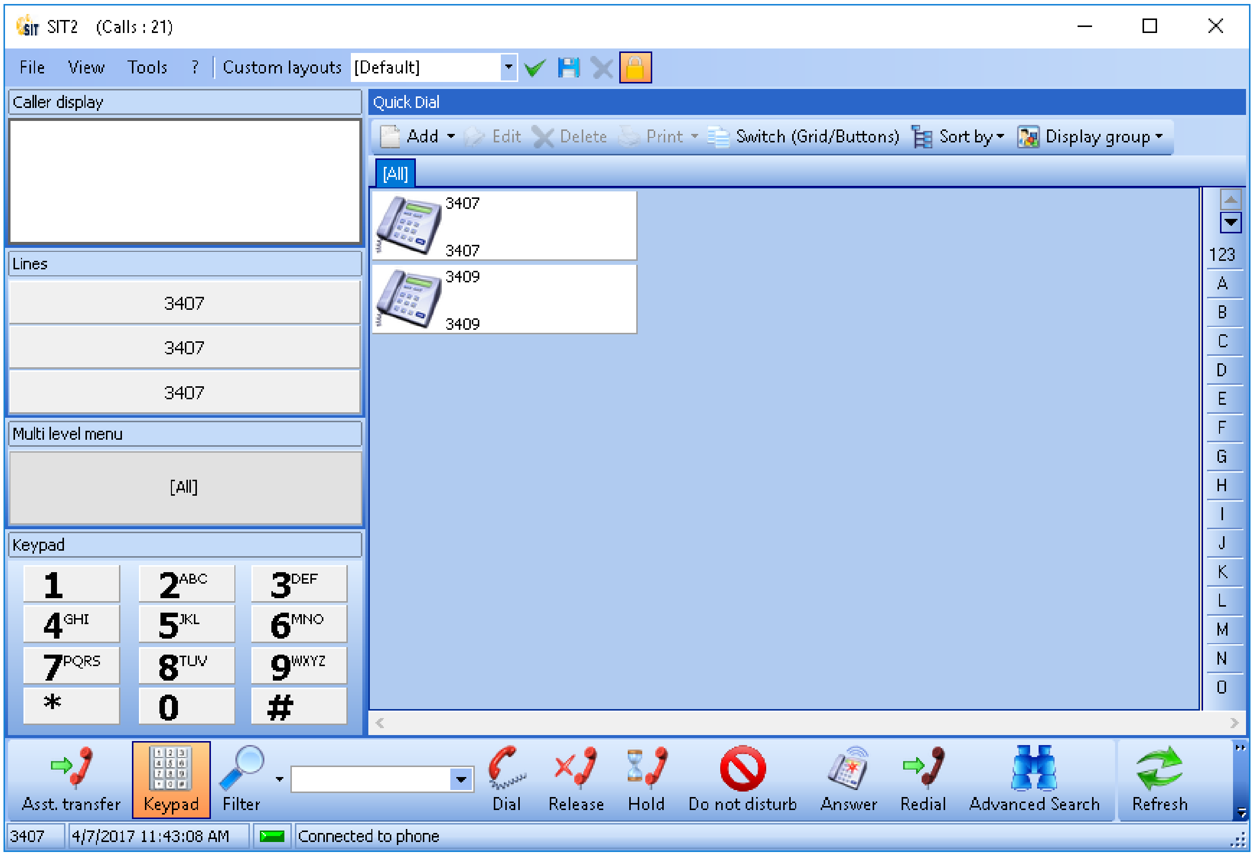
In the **SIP phone connection** section, configure the SIP parameters, including:

- The **Username** and **Password** used to register with Session Manager.
- The SIP **Domain** (e.g., *bvwdev.com*).
- The **Proxy**, which specifies the IP address of the SIP interface of Session Manager.

In the **Audio devices** section (not shown), specify the audio device or headset that will be used with the console. Click the **Save** button at the top of screen.



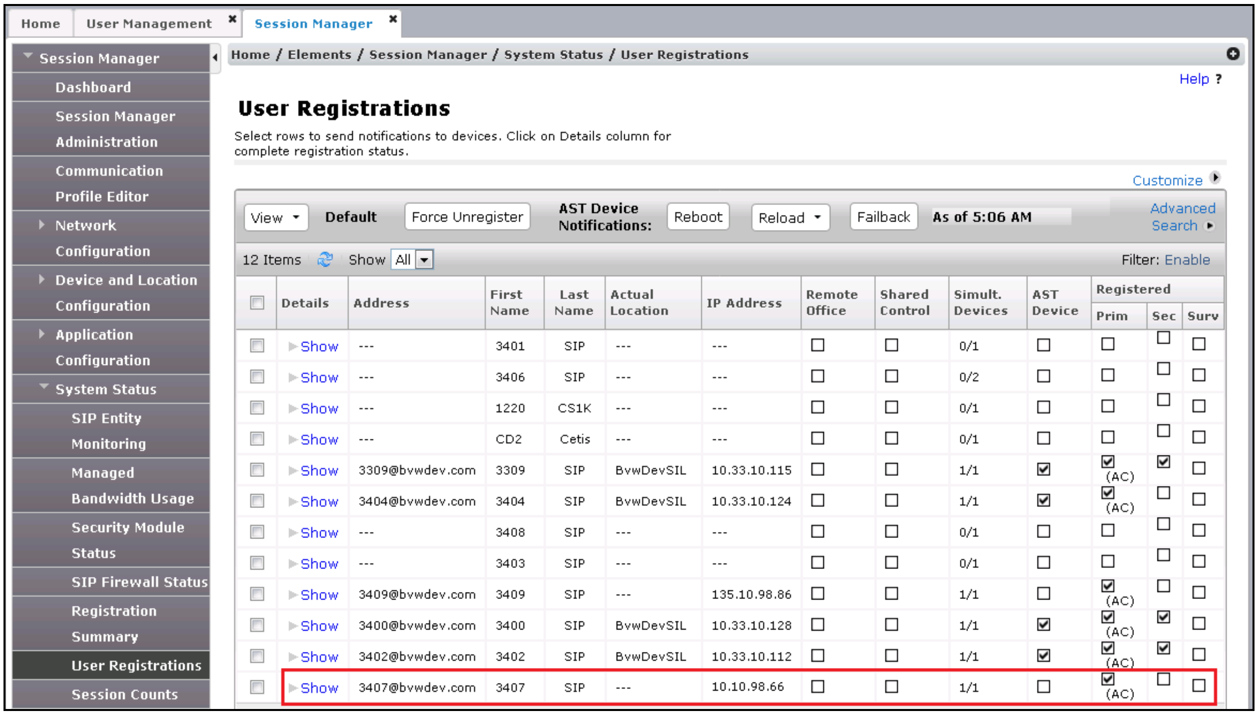
After the configuration is completed, the SIT2 IP Console appears as follows.



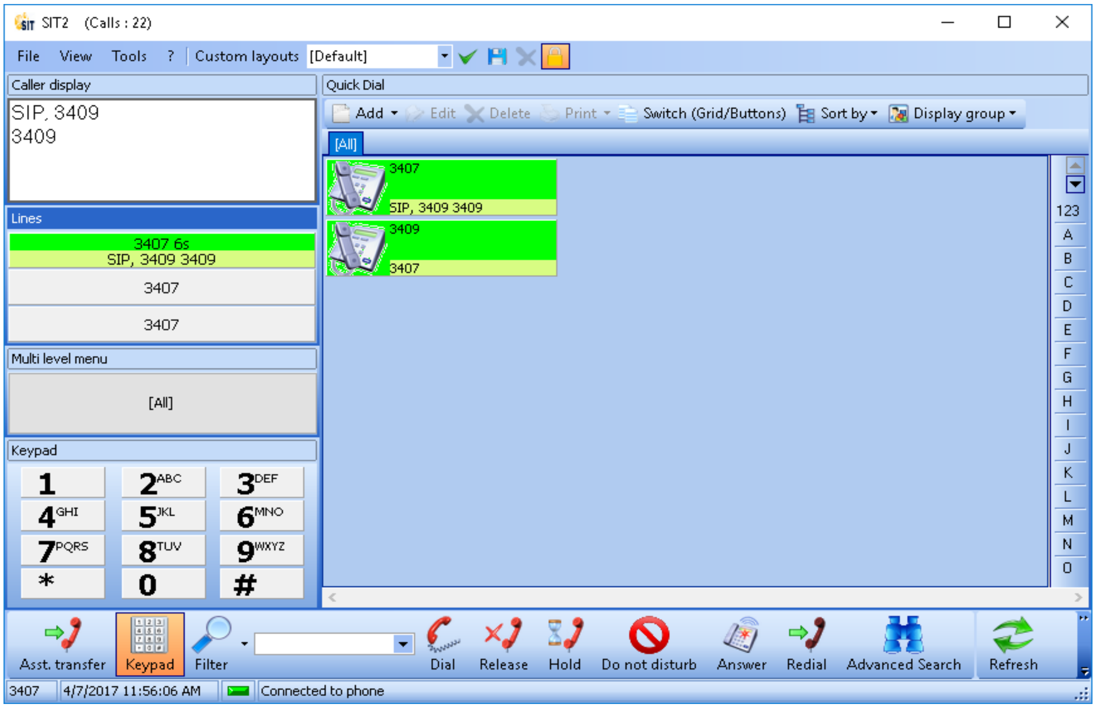
8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Komutel SIT2 IP Console with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

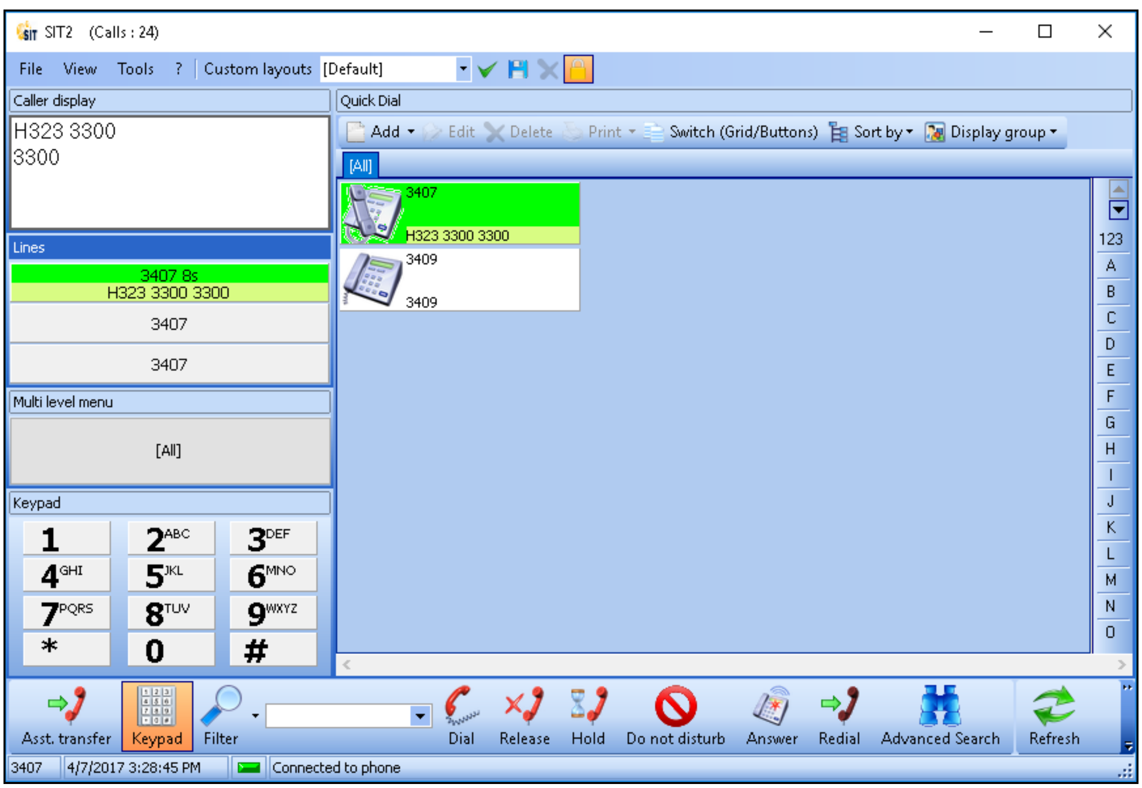
1. Verify that the SIT2 IP Console has successfully registered with Session Manager.



2. Verify basic telephony features by establishing calls between two SIT2 IP Consoles.



3. Verify basic telephony features by establishing calls between a SIT2 IP Console and another endpoint.



9. Conclusion

These Application Notes describe the configuration steps for provisioning the Komutel SIT2 IP Console 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

[5] Komutel User Guide for SIT2 IP Console, Revised 2012-07-16.

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.