DevConnect Program

Application Notes for configuring Komutel Komand911 SIP V3.4.1 with Avaya Aura® Communication Manager R10.1 and Avaya Aura® Session Manager R10.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps for provisioning the Komutel Komand911 SIP to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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 Avaya DevConnect Program.

1. Introduction

These Application Notes describe the steps required to integrate the Komutel Komand911 SIP softphone with Avaya Aura® Session Manager (Session Manager) and Avaya Aura® Communication Manager (Communication Manager). The Komand SIP softphone 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 Komand SIP softphone successfully registered with Session Manager, established calls with other telephones, and executed telephony features such as Hold, Transfer, and Conference.

Komutel SIP registers to Session Manager and is able to work as a SIP agent in Avaya Call Center Elite environment, by using Avaya's Advanced SIP Telephone (AST). Support for this solution is restricted solely to deployments with Canadian Emergency 911 PSAPs and is not supported by Avaya for general enterprise deployments or for use in emergency services call centers in other geographies. Komutel Komand911SIP solution includes Komutel Kore server that is a main portal for Komand web application.

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 Komand SIP softphone 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 Komand SIP softphone comes back into service after re-connecting the Ethernet connection or rebooting the PC on which the Komand SIP softphone 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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products only (private network side). 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 these Application Notes, the interface between Avaya systems and the Komutel SIP softphone did not include use of any specific encryption features as requested by Komutel.

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 Komand SIP softphone with Session Manager.
- Calls between Komand SIP softphone and Avaya SIP, H.323, digital stations and PSTN.
- G.711 and G.722 codecs support.
- Caller ID display on Avaya stations and the Komand SIP softphone.
- Proper recognition of DTMF tones.
- Voicemail coverage, MWI support, and logging into voicemail system to retrieve voice messages.
- Basic telephony features including Hold, Mute, Transfer, and Conference.
- Extended telephony features using Communication Manager Feature Access Code (FAC).
- Contact center features including agent login, agent logout, various agent states and receiving the contact center call.
- Proper system recovery after a restart of the Komand SIP softphone and loss of IP connectivity.

2.2. Test Results

All test cases passed successfully with the following observation.

- Komand SIP softphone does not support Call Forward and Call Park, at the time of testing.

2.3. Support

For technical support on the Komand911 SIP softphone, contact Komutel Support via phone, email, or website.

- **Phone:** +1(877) 225-9988
- **Email:** service@komutel.com
- **Web:** https://www.komutel.com/fr/a-propos-de-komutel/services/

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 alongside an 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.
- Enterprise has SIP trunk connects to PSTN through Avaya Session Border Controller.
- Komutel Komand SIP softphone registered with Session Manager with Advanced SIP Telephony (AST) and acted as a SIP agent for Call Center Elite in Communication Manager.

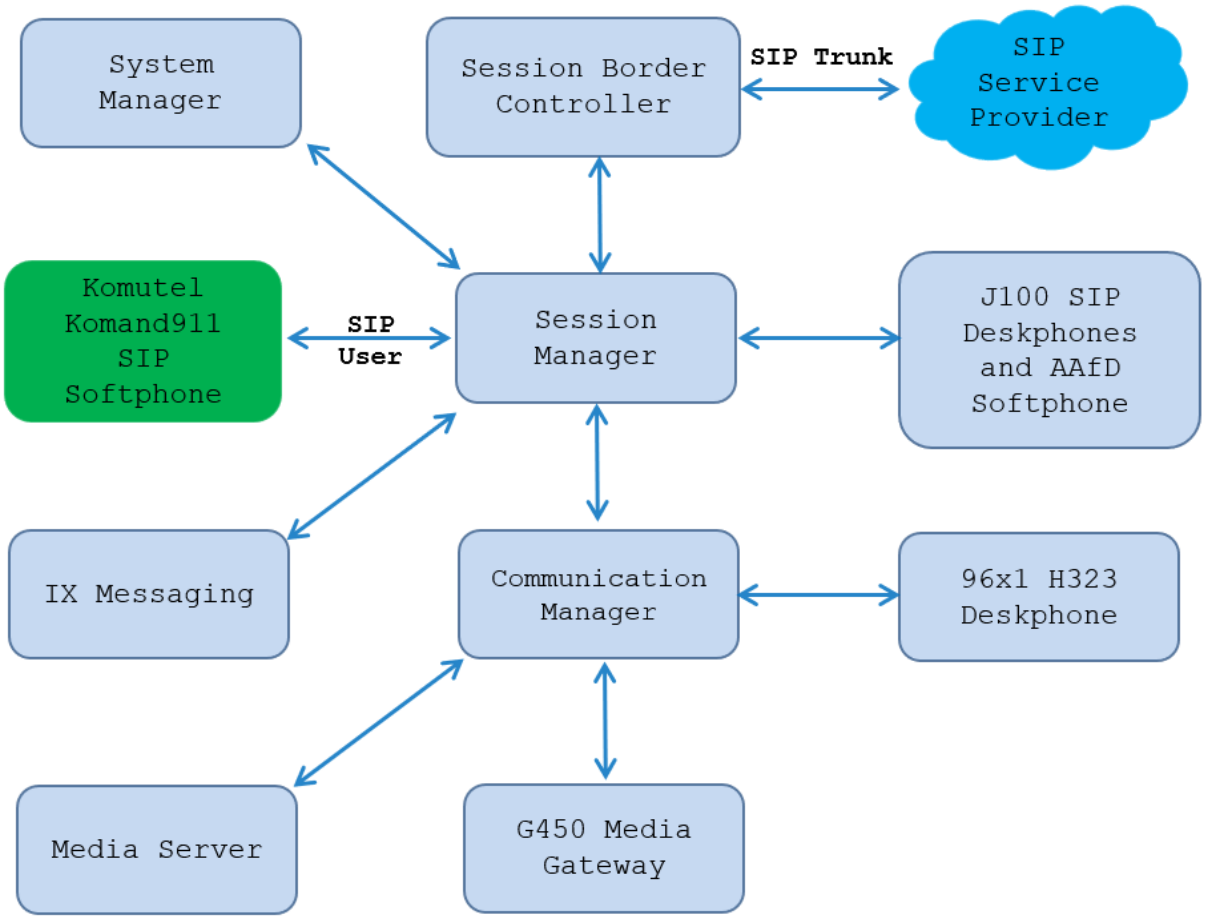


Figure 1: Avaya SIP Network with Komutel Komand911 SIP softphone

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 virtualized environment	10.1.3.1.0-FP3 SP1 01.0.974.0-27937
Avaya IX Messaging	11.1
Avaya Aura® Session Manager running on virtualized environment	10.1.3 10.1.3.1.1013103
Avaya Aura® System Manager running on virtualized environment	10.1.3.1 Feature Pack 3 SP1 10.1.3.1.0716418
Avaya Aura® Media Server running on virtualized environment	10.1.0.154
Avaya Session Border Controller for Enterprise	10.1.2.0-64-23285
Avaya G450 Media Gateway	42.24.0
Avaya IP Deskphones • 9641GS (H.323) • J189 (SIP)	 6.8.5.4 4.1.3.0.6
Avaya Agent for Desktop Softphone	2.0.65
Komutel Kore Server Komutel Komand911 SIP Softphone	2.5.0.5610 3.4.1.1396

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**.

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  12
                              OPTIONAL FEATURES

       G3 Version: V20                          Software Package: Enterprise
         Location: 2                             System ID (SID): 1
         Platform: 28                            Module ID (MID): 1

                                                        USED
                          Platform Maximum Ports:  81000    152
                               Maximum Stations:  41000     89
                          Maximum XMOBILE Stations:  41000      0
                Maximum Off-PBX Telephones - EC500:  41000      1
                Maximum Off-PBX Telephones -   OPS:  41000     21
                Maximum Off-PBX Telephones - PBFMC:  41000      0
                Maximum Off-PBX Telephones - PVFMC:  41000      0
                Maximum Off-PBX Telephones - SCCAN:      0      0
                Maximum Off-PBX Telephones -   EMX:  41000      0
                     Maximum Survivable Processors:  313        1



        (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  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                        USED
                 Maximum Administered H.323 Trunks: 12000     10
       Maximum Concurrently Registered IP Stations: 18000      7
         Maximum Administered Remote Office Trunks: 12000      0
Max Concurrently Registered Remote Office Stations: 18000      0
          Maximum Concurrently Registered IP eCons:  414       0
     Max Concur Reg Unauthenticated H.323 Stations:   100      0
                   Maximum Video Capable Stations:  41000     4
               Maximum Video Capable IP Softphones:  18000    11
                   Maximum Administered SIP Trunks: 40000     30
  Max Administered Ad-hoc Video Conferencing Ports: 24000      0
   Max Number of DS1 Boards with Echo Cancellation: 999    0



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

On **Page 8** of the **system-parameters customer-options form**, verify that the number of **Logged-In ACD Agents** supported by the system is sufficient.

```
display system-parameters customer-options                      Page   8 of  12
                        CALL CENTER OPTIONAL FEATURES

           VDN of Origin Announcement? y                          VuStats? y
               VDN Return Destination? y          VuStats (G3V4 Enhanced)? y









                                          USED
                     Logged-In ACD Agents: 10000     6
                 Logged-In Advocate Agents: 10000     0
              Logged-In IP Softphone Agents: 10000     4
                   Logged-In SIP EAS Agents: 10000     0
        (NOTE: You must logoff & login to effect the permission changes.)
```

5.2. Administer Hunt Group

This section provides the Hunt Group configuration for the call center agents. Agents will log into Hunt Group 1 configured below. Provide a descriptive name and set the **Group Extension** field to a valid extension. Enable the **ACD**, **Queue**, and **Vector** options.

```
add hunt-group 1                                               Page   1 of   4
                               HUNT GROUP

            Group Number: 1                                 ACD? y
              Group Name: Skill-1                         Queue? y
         Group Extension: 3320                           Vector? y
              Group Type: ucd-mia
                      TN: 1
                     COR: 1                     MM Early Answer? n
           Security Code:                 Local Agent Preference? n
 ISDN/SIP Caller Display:

             Queue Limit: unlimited
 Calls Warning Threshold:      Port:
  Time Warning Threshold:      Port:

SIP URI:
```

On **Page 2** of the Hunt Group form, enable the **Skill** option field by selecting "y".

```
add hunt-group 1                                               Page   2 of   4

                                  HUNT GROUP

                    Skill? y      Expected Call Handling Time (sec): 180
                      AAS? n        Service Level Target (% in sec): 80 in 20
                 Measured: none
     Supervisor Extension:


      Controlling Adjunct: none


        VuStats Objective:

   Multiple Call Handling: none


 Timed ACW Interval (sec):          After Xfer or Held Call Drops? n
```

5.3. Administer Vector

Use the command "**change vector n**" while "n" is the vector number from 1-8000. The example of the vector 1 with the basic scripting is shown below. The vector 1 is used for the configuration of VDN in the next step.

```
change vector 1                                              Page   1 of   6
                                CALL VECTOR

    Number: 1                Name: Contact Center
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    10  secs hearing 1100     then silence
02 queue-to     skill 1    pri m
03 wait-time    5   secs hearing ringback
04 check        skill 1    pri m if expected-wait    < 30
05 announcement 1104
06 queue-to     skill 1    pri m
07 stop
```

5.4. Administer VDN

Use the "**add vdn <ext>**" command to add a VDN number. In the **Destination** field, enter **Vector Number** 1 as configured in **Section 5.3** above and keep other fields at their default values.

```
add vdn 3340                                                 Page   1 of   3
                          VECTOR DIRECTORY NUMBER

                            Extension: 3340
                                 Name*: Contact Center 1
                          Destination: Vector Number        1
                  Attendant Vectoring? n
                 Meet-me Conferencing? n
                   Allow VDN Override? n
                                  COR: 1
                                  TN*: 1
                             Measured: both     Report Adjunct Calls as ACD*? n
        Acceptable Service Level (sec): 20
        VDN of Origin Annc. Extension*:
                            1st Skill*:
                            2nd Skill*:
                            3rd Skill*:
```

5.5. Administer Agent Login ID

To add an **Agent LoginID**, use the command "**add agent-loginID <agent ID>**" for each agent. In the compliance test, three agent login IDs **1000**, 1001, and 1002 were created.

```
add agent-loginID 1000                                        Page   1 of   2
                                AGENT LOGINID

                Login ID: 1000                                    AAS? n
                    Name: Agent 1000                            AUDIX? n
                      TN: 1
                     COR: 1
           Coverage Path:                             LWC Reception: spe
           Security Code: 1234                  LWC Log External Calls? n
           Attribute:                          AUDIX Name for Messaging:

                                         LoginID for ISDN/SIP Display? n
                                                             Password:
                                               Password (enter again):
                                                          Auto Answer: station
                                                   MIA Across Skills: system
 AUX Agent Considered Idle (MIA)? system   ACW Agent Considered Idle: system
                                          Aux Work Reason Code Type: system
                                            Logout Reason Code Type: system
                       Maximum time agent in ACW before logout (sec): system
                                           Forced Agent Logout Time:   :
    WARNING:  Agent must log in again before changes take effect
```

On **Page 2** of the **Agent LoginID** form, set the skill number (**SN**) to hunt group 1, which is the hunt group (skill) that the agents will log into.

```
add agent-loginID 1000                                        Page   2 of   2
                                AGENT LOGINID
      Direct Agent Skill:                                Service Objective? n
Call Handling Preference: skill-level                 Local Call Preference? n

    SN   RL SL          SN   RL SL
 1: 1       1       16:
 2:                 17:
 3:                 18:
 4:                 19:
 5:                 20:
 6:
 7:
 8:
 9:
10:
11:
12:
13:
14:
15:
```

5.6. 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**.

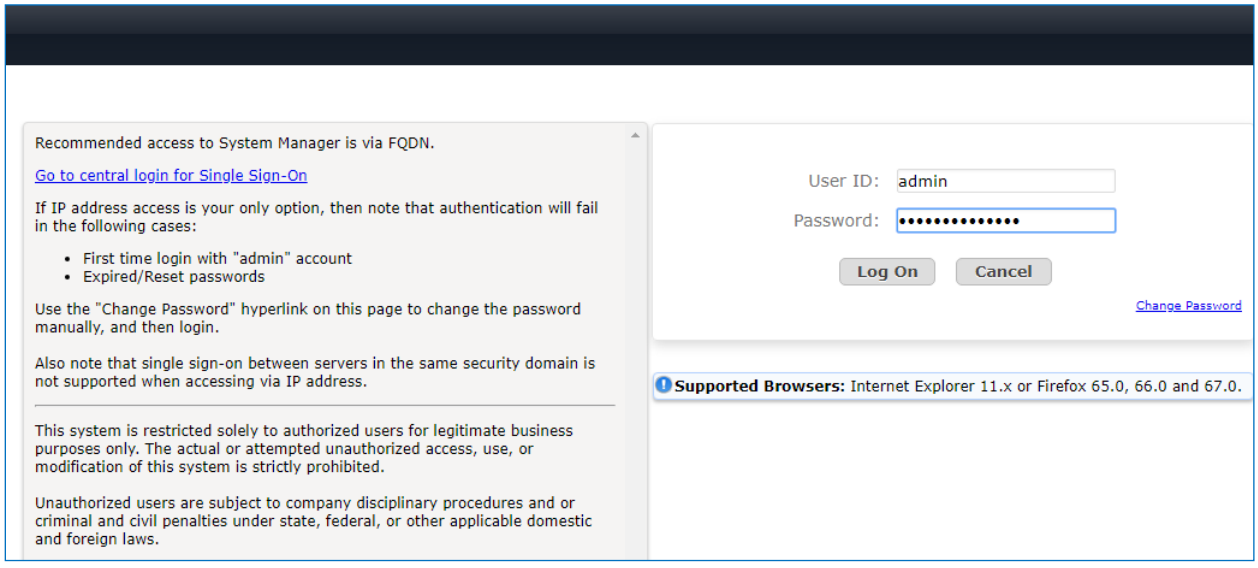
```
change dialplan analysis                                    Page   1 of  12
                          DIAL PLAN ANALYSIS TABLE
                                Location: all            Percent Full: 1

    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

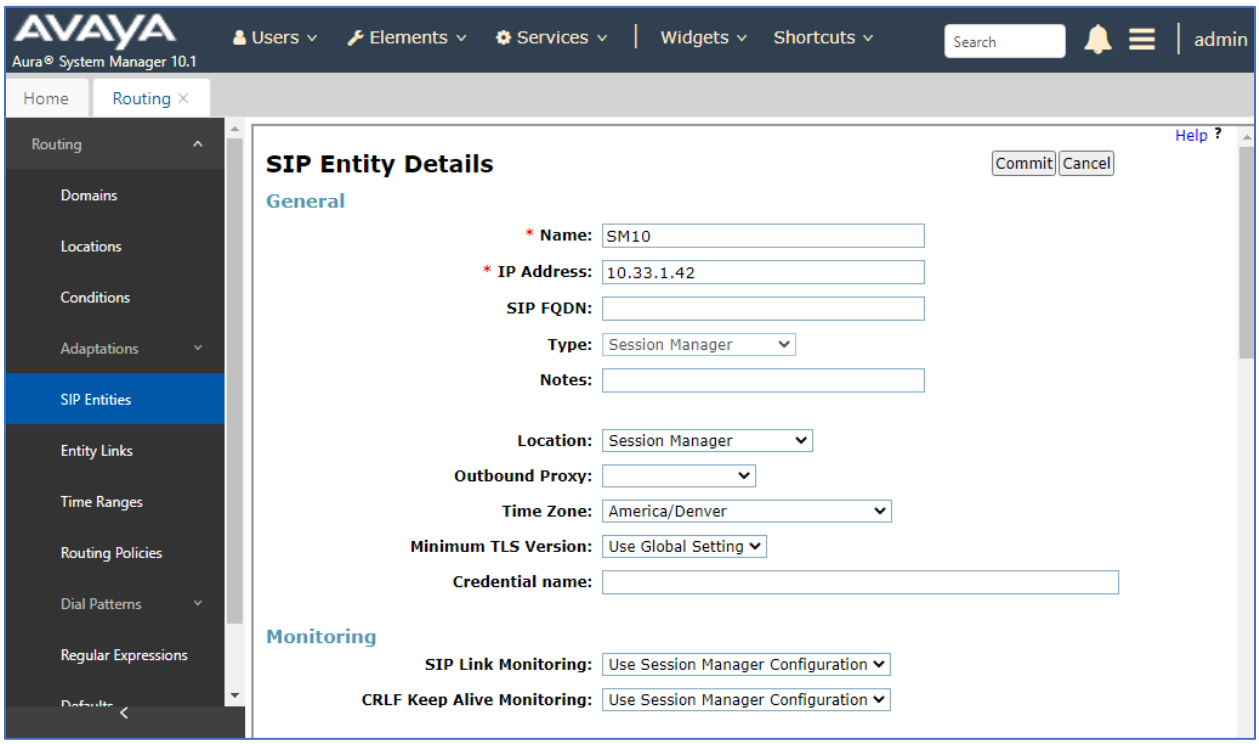
This section describes aspects of the Session Manager configuration required for interoperating with Komutel SIP softphone. It is assumed that the Domains, Locations, SIP entities, Entity Links, Routing Policies, Dial Patterns and Application Sequences have been configured where appropriate for Communication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

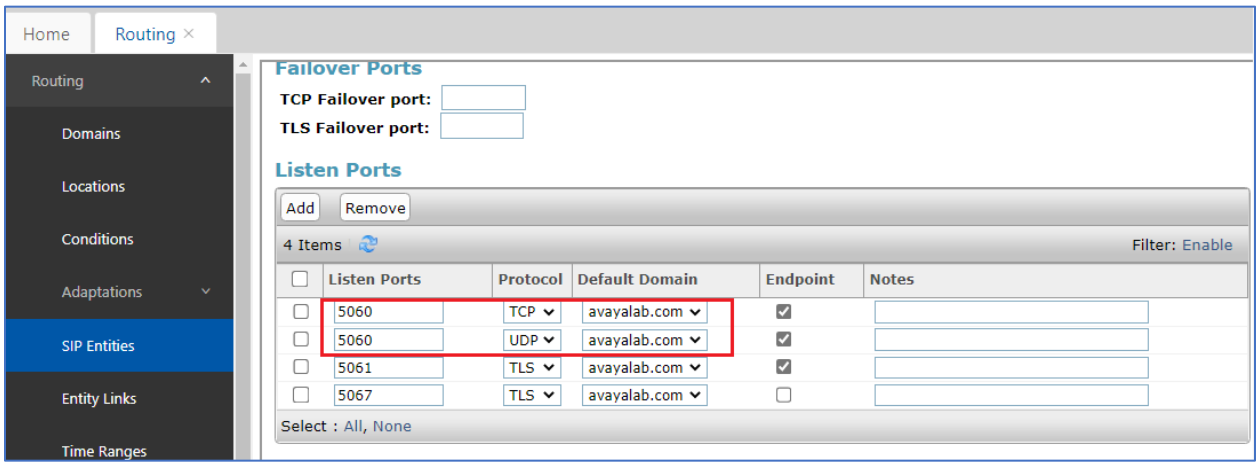
Session Manager is managed via System Manager. Using a web browser, access **https://<ip-addr of System Manager>/SMGR**. In the **Log On** screen, enter appropriate **User ID** and **Password** and click the **Log On** button.



6.1. Check Avaya Aura® Session Manager ports for SIP Endpoint Registration

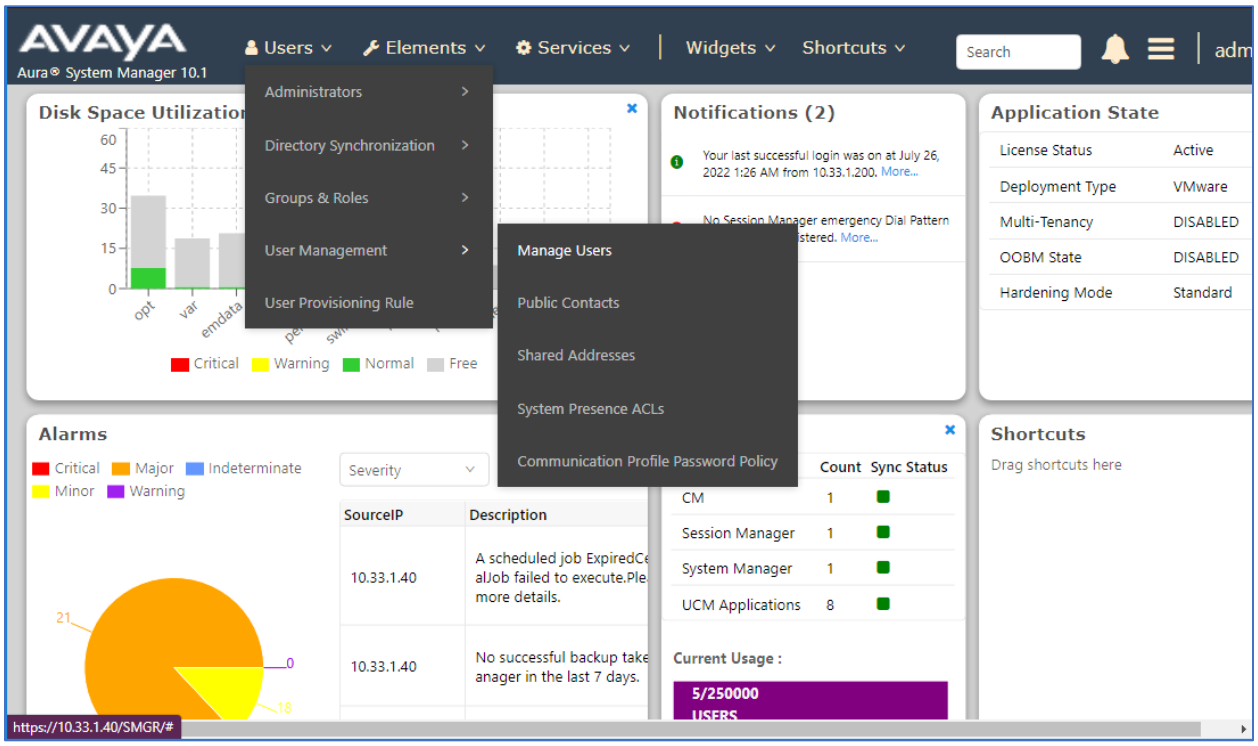
Each Session Manager Entity must be configured so that the Komand SIP softphone can register to it using UDP/TCP. From the web interface click **Routing → SIP Entities** (not shown) and select the Session Manager entity used for registration. Make sure that **TCP** and **UDP** listen ports are present. The TCP and UDP port and SIP domain name are highlighted below.



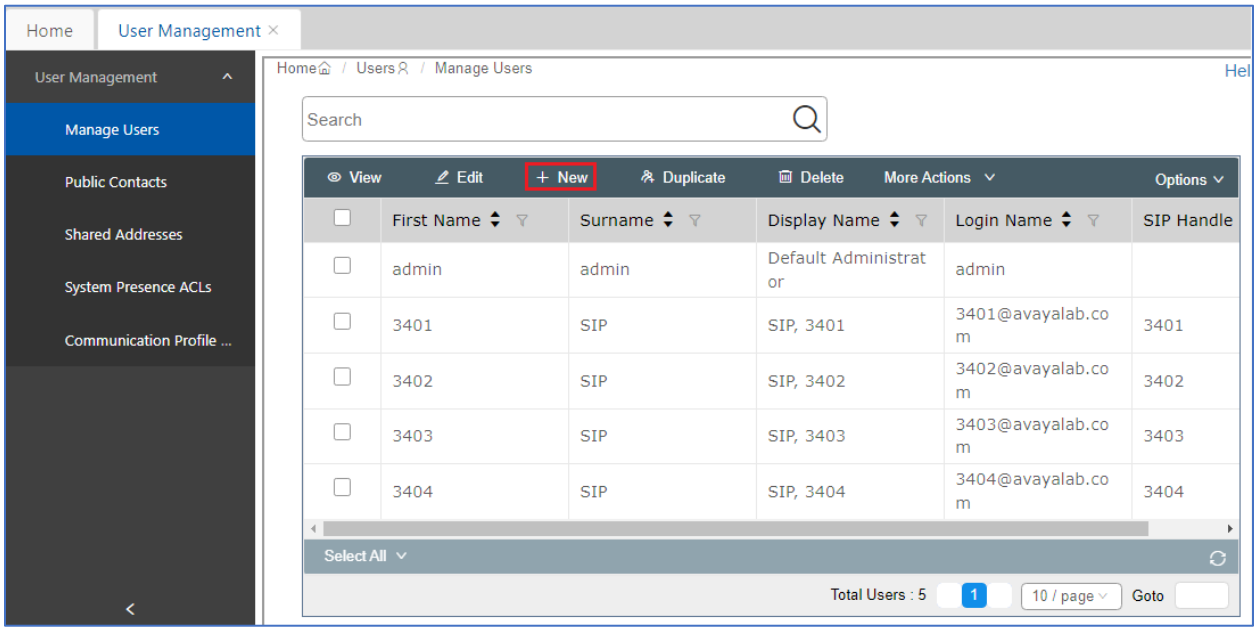


6.2. Administer SIP User

The Komand SIP softphone registers to a SIP user in Session Manager, to create a SIP user navigate to **Users → User Management → Manage Users**.



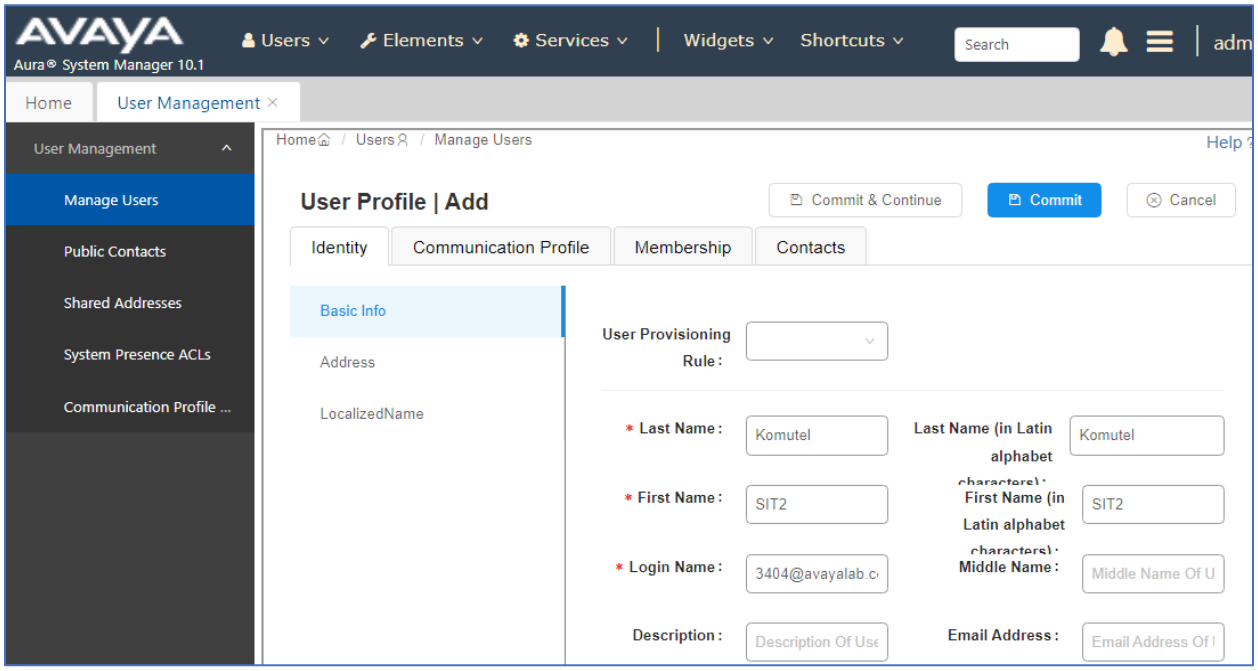
Select **+New** button from the **Manage Users** page in the right-hand side to add a new SIP user.



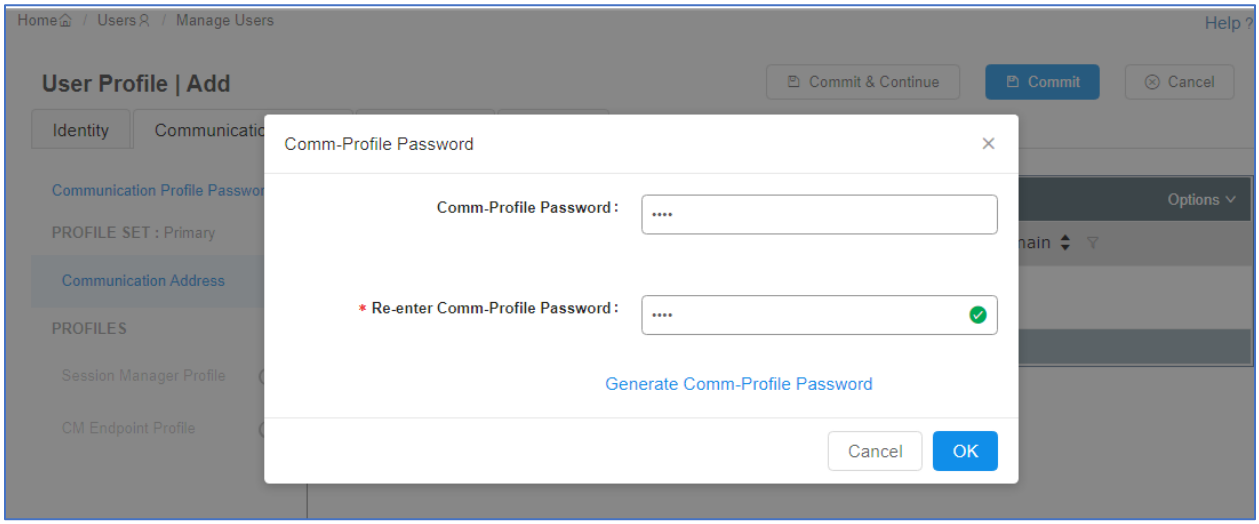
In the **Identity** tab enter the basic information for the SIP user; the fields with asterisk are mandatory.

- **First Name** Enter a descriptive name
- **Last Name** Enter a descriptive name
- **Login Name** Enter the extension number followed by the domain

Leave other fields at default values.



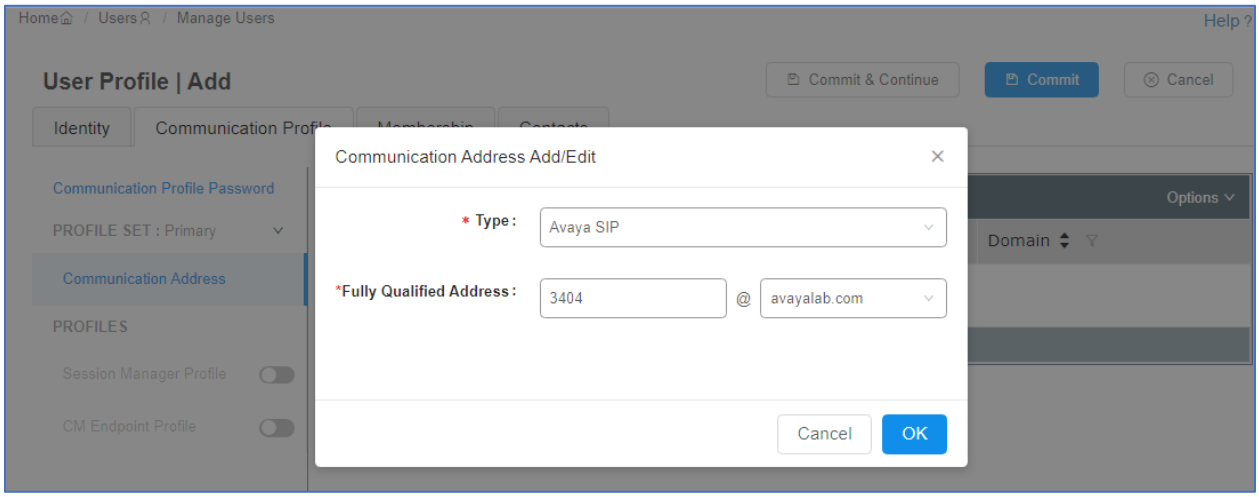
Select **Communication Profile → Communication Profile Password**. The **Comm-Profile Password** window displays, enter the same password in the **Comm-Profile Password** and **Re-enter Comm-Profile Password** fields. Click on **OK** button to save the configuration.



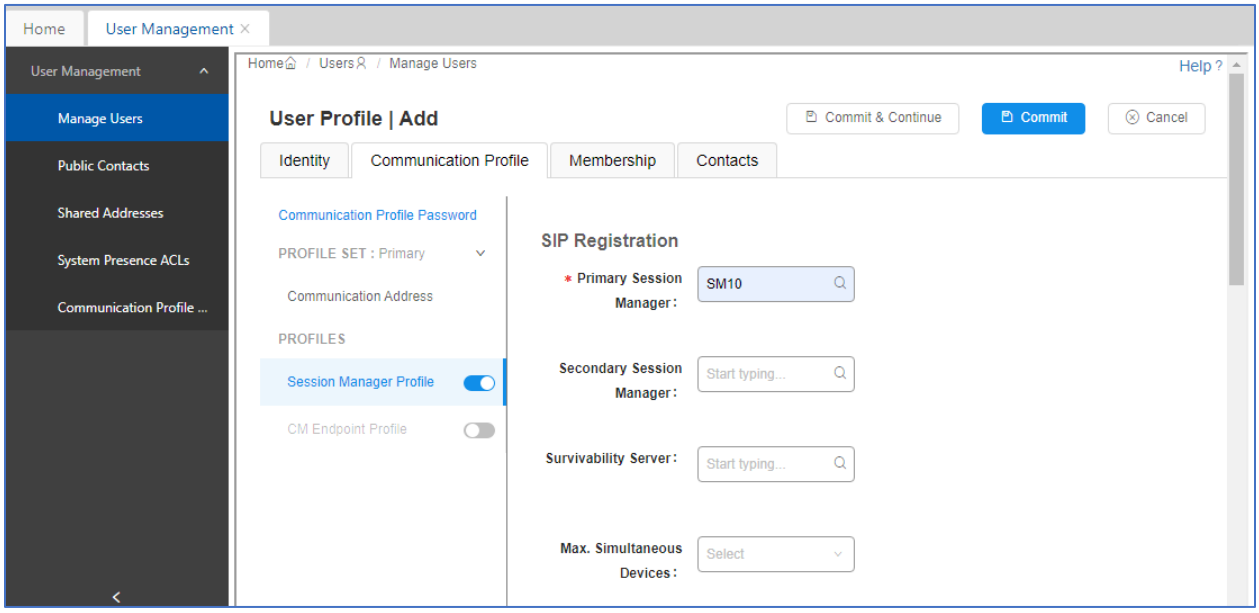
Select **Communication Profile → Communication Address** and then select **+ New** button (not shown) to add a new communication address; enter the information as shown in the picture below.

- **Type** Select **Avaya SIP** in the dropdown menu
- **Fully Qualified Address** Enter the number **3404** and select SIP domain **avayalab.com** in the dropdown menu

Click **OK** button to save the configuration.

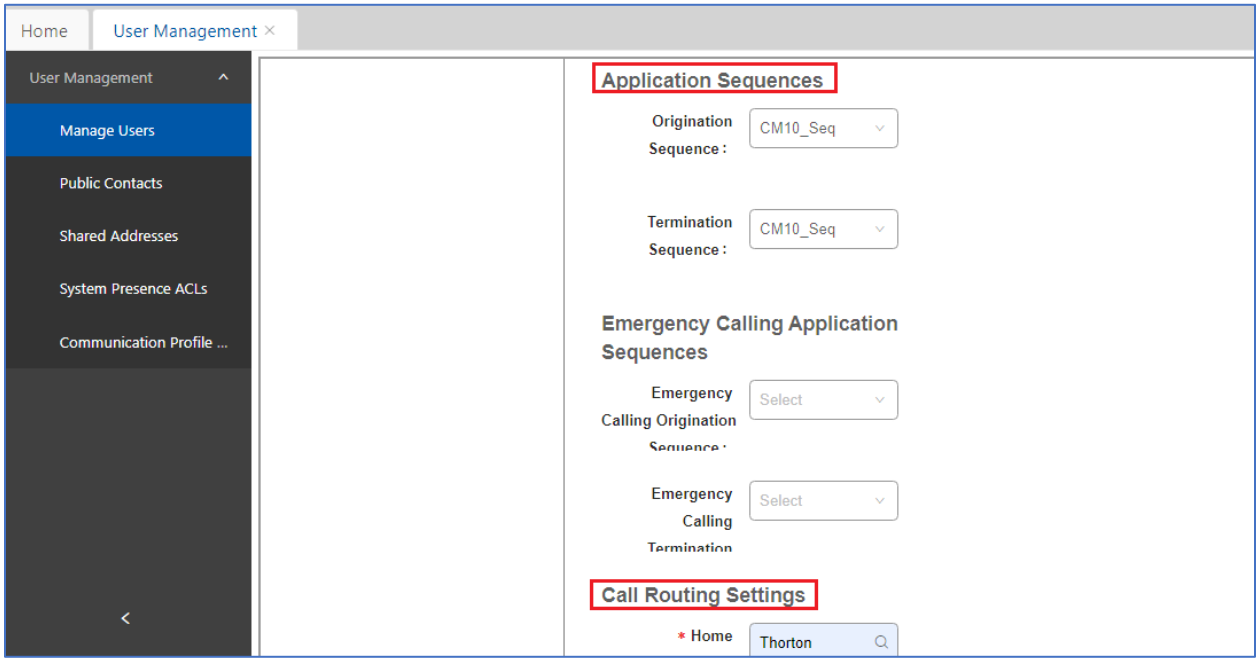


Enable **Session Manager Profile** in the **Communication Profile** tab. In the **SIP Registration** section, select the Session Manager SIP entity **SM10** as shown in the picture below.



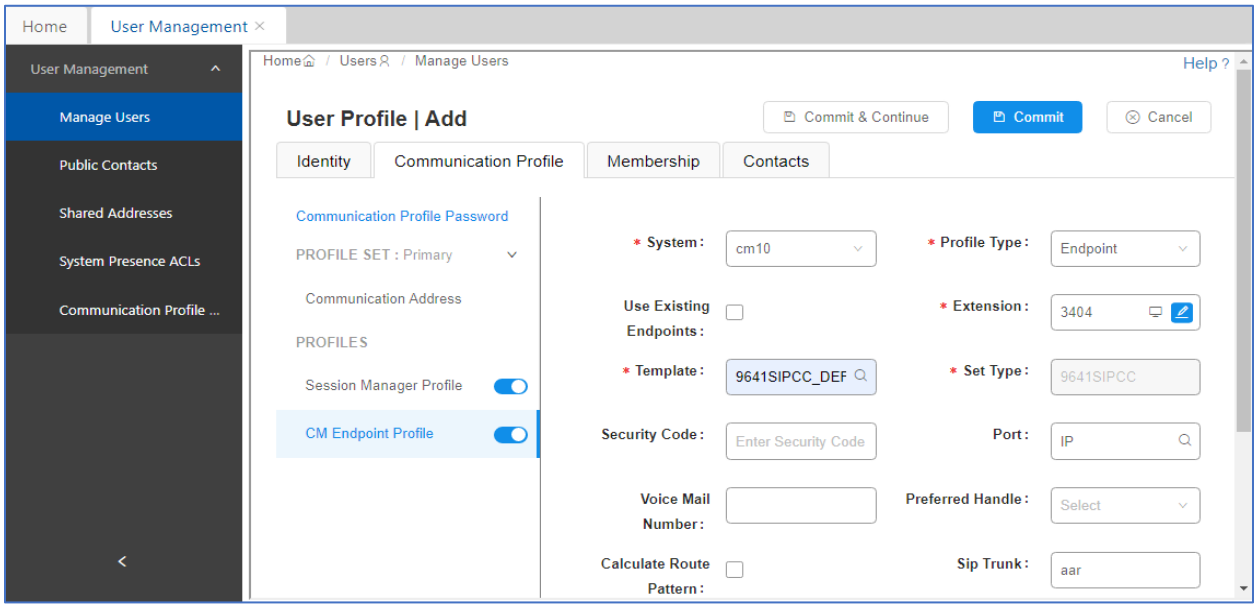
In the **Application Sequences** section of **Session Manager Profile**, select the previously created **CM10_Seq** in the **Origination Sequence** and **Termination Sequence** fields.

In the **Call Routing Settings** section, select the previously created **Thornton** in the **Home Location** field.

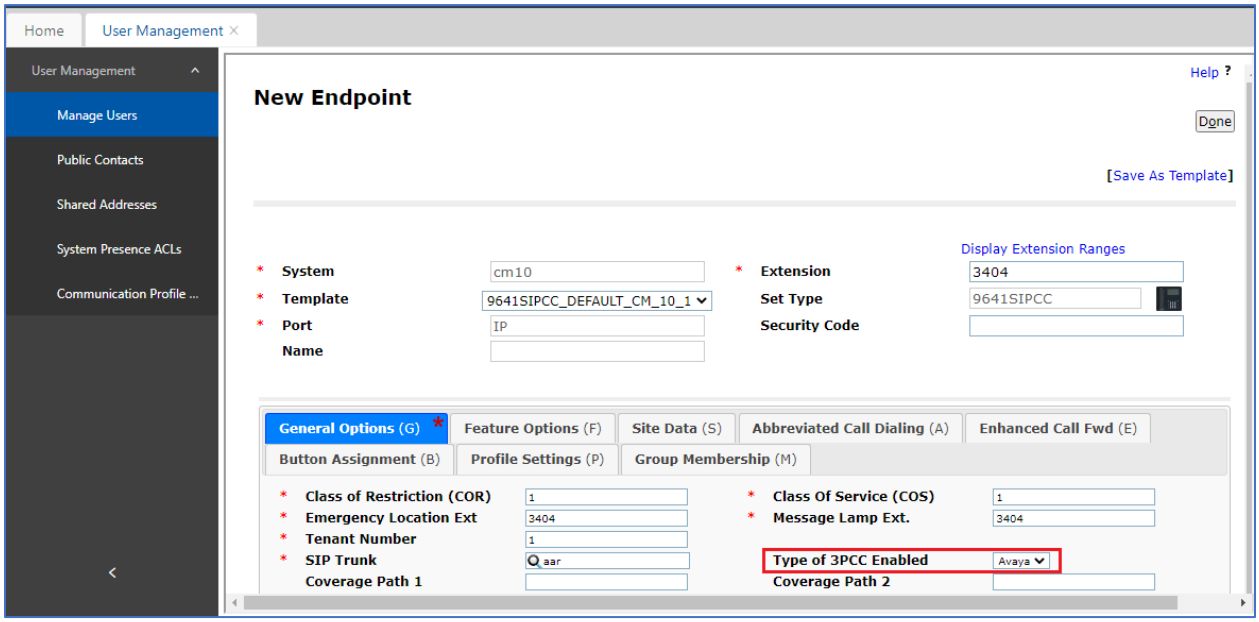


Enable **CM Endpoint Profile** in the **Communication Profile** tab. Enter the following values as shown in the picture below.

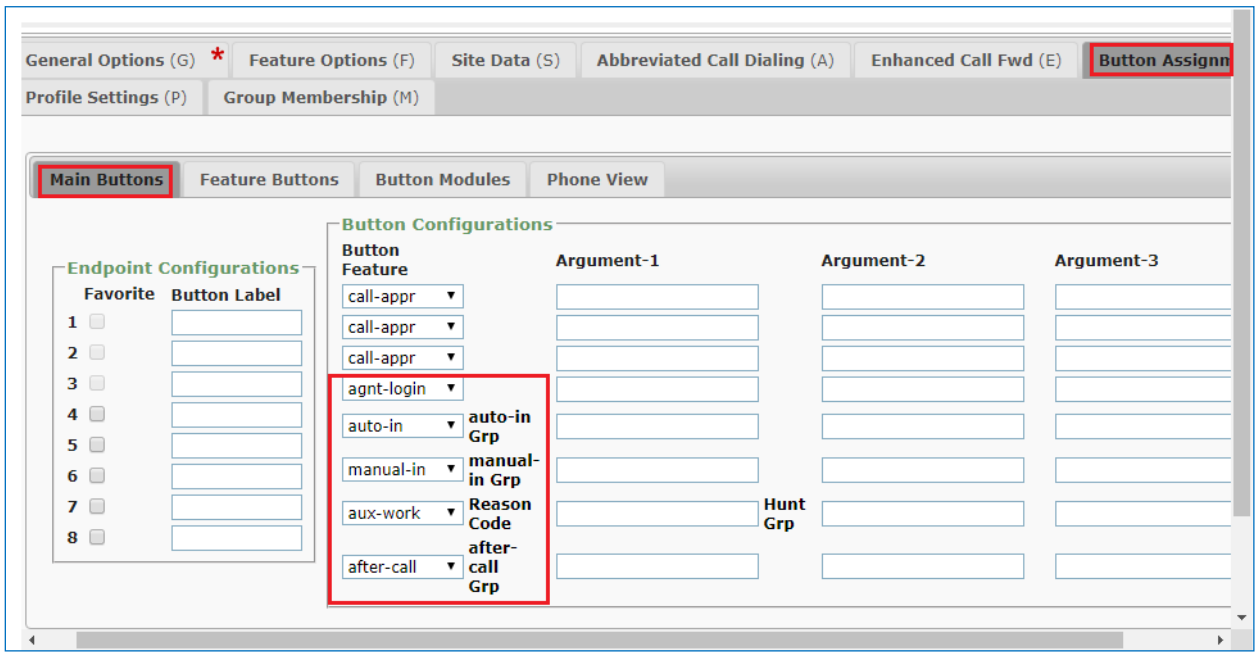
- **System** Select the previously created **cm10**
- **Profile Type** Select **Endpoint**
- **Extension** Enter the extension **3404**
- **Template** Select the template **9641SIPCC_DEFAULT_CM**



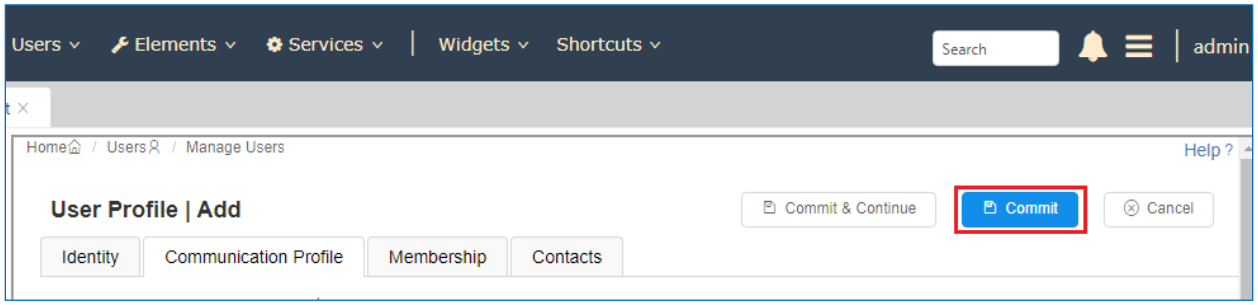
Select **Editor**  icon next to the **Extension** field (see above). The New Endpoint window displays, in the **General Options (G)** tab, make sure the **Type of 3PCC Enabled** is set to **Avaya**.



Select the **Button Assignment (B)** tab, in the Main Buttons sub-tab, add 5 buttons for the contact center agent: **agnt-login**, **auto-in**, **manual-in**, **aux-work**, and **after-call**.



Click on **Done** button (not shown) on the **New Endpoint** window and then click on **Commit** button to save the configuration for the new user created.



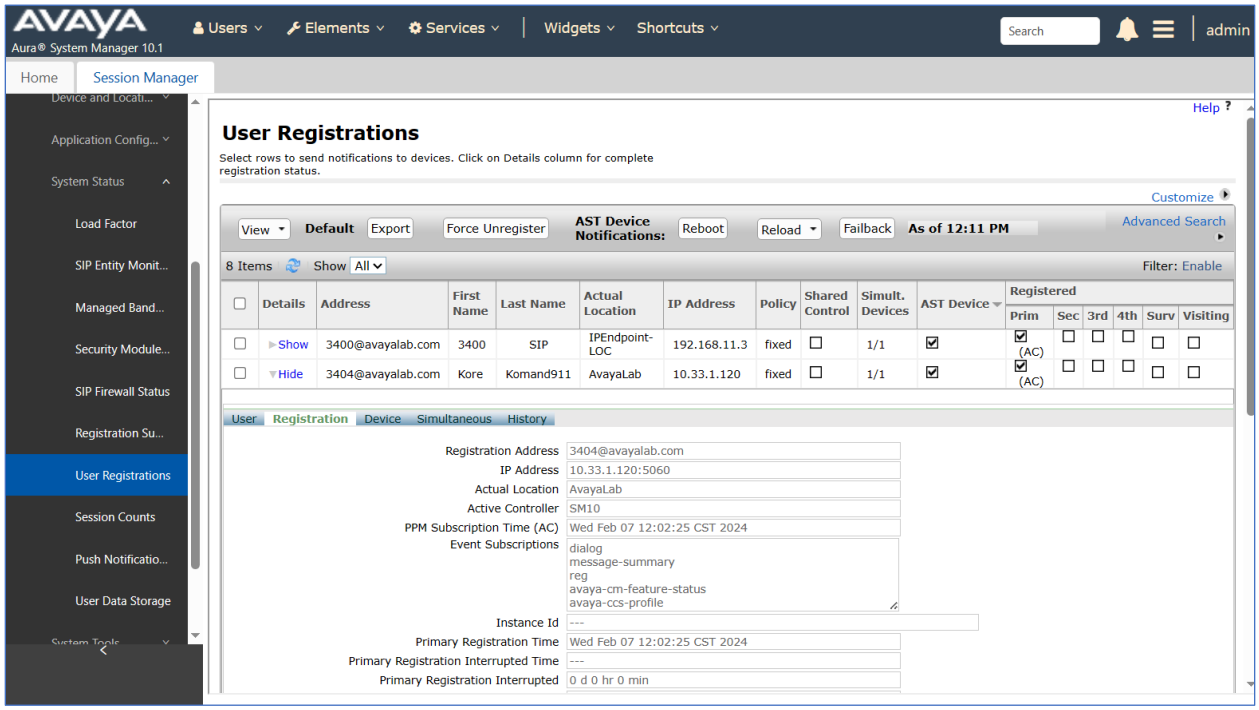
7. Configure Komutel Komand911 SIP Solution

The configuration of Komand911 server and its relevant applications is performed by Komutel technical engineer, and it is not mentioned in these Application Notes.

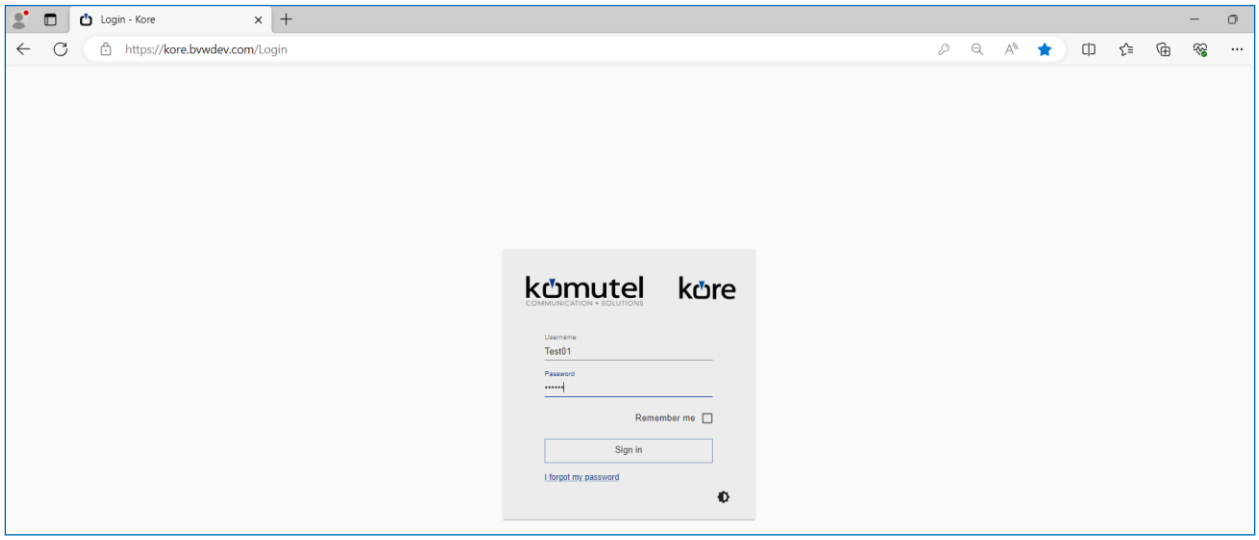
8. Verification Steps

This section provides the procedures that can be performed to verify the configuration of the Komutel Komand SIP softphone with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

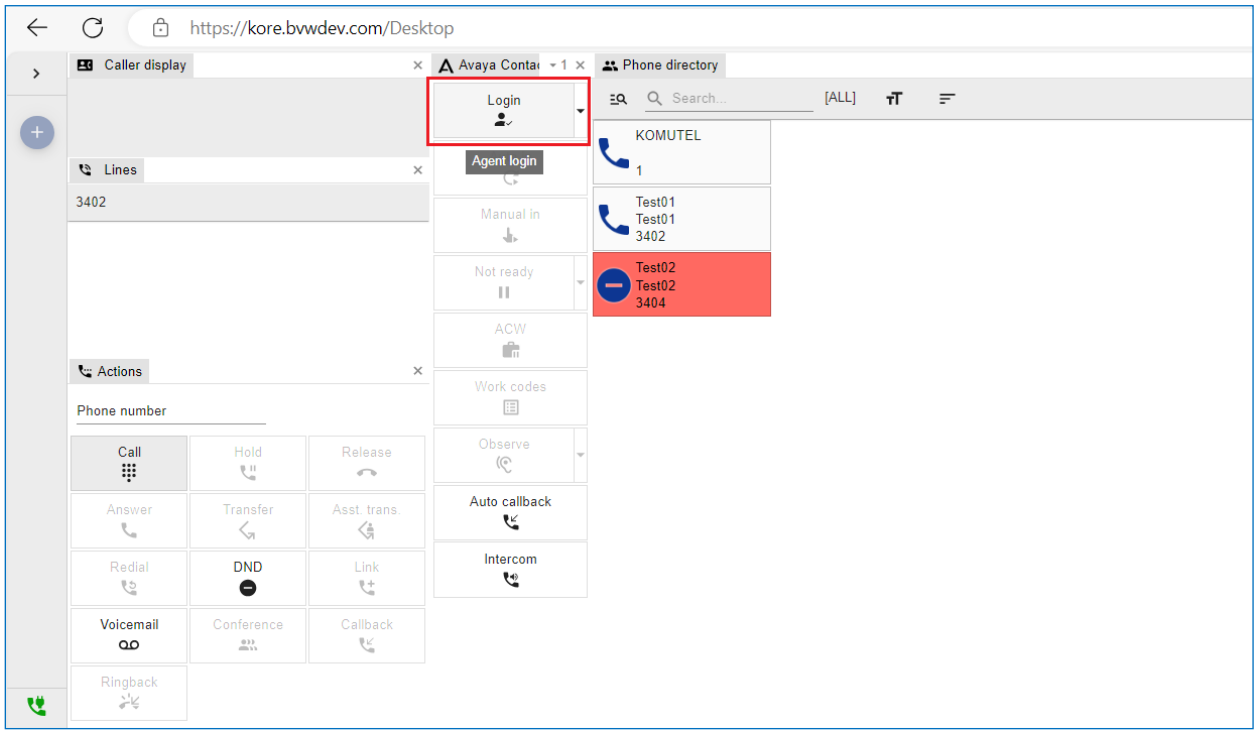
1. Verify that the Komand SIP softphone is successfully registered with Session Manager via a SIP user and acquire the AST feature as well as subscribe the contact center events such as **avaya-cm-feature-status** and **avaya-ccs-profile**.



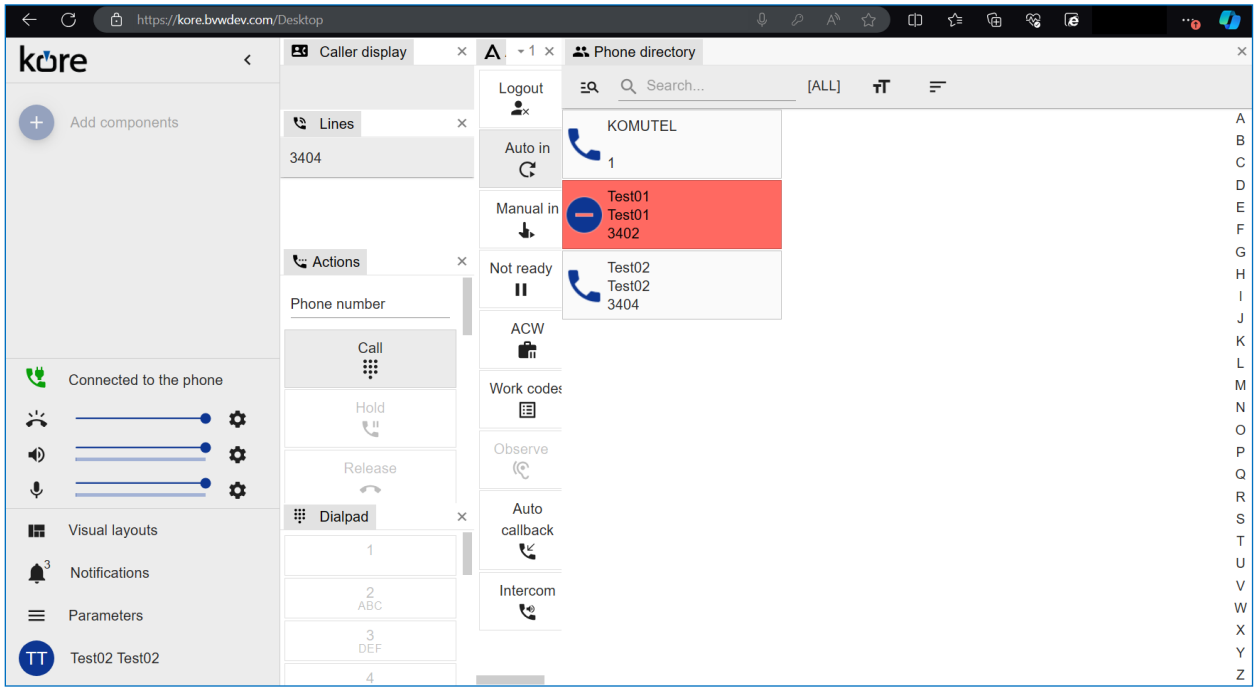
2. Verify the Komand SIP softphone can login a SIP station. To login a SIP station, enter FQDN of the Komand Kore website in the internet browser and enter a configured username and password and select **Sign In**.



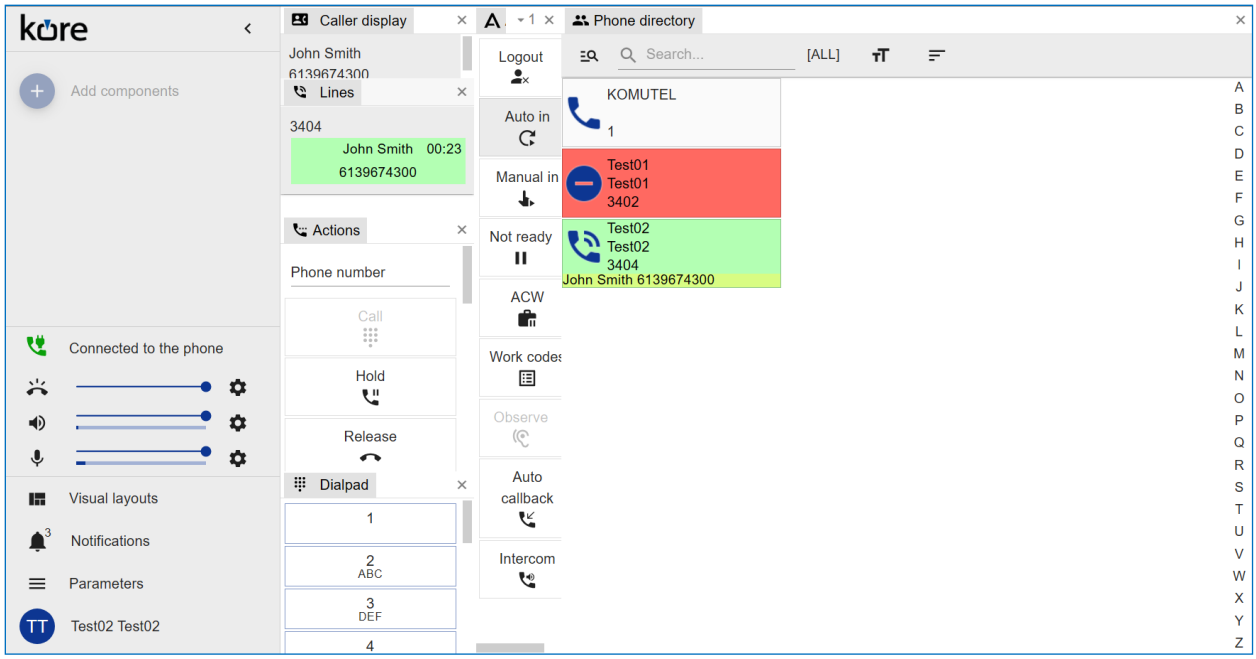
The Komand SIP soft phone is successfully logs on a SIP station. In the **Avaya Contact Center** column, select **Login** button to log in an agent.



To put the agent in the ready mode, select **Auto in**.



3. Verify the Komand SIP softphone can receive an inbound call center ACD call.



9. Conclusion

These Application Notes describe the configuration steps for provisioning the Komutel Komand911 SIP sofphone as a SIP agent in Call Center Elite using Advanced SIP Telephony 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 the product documentation relevant to these Application Notes. Product documentation for Avaya products may be found at http://support.avaya.com

[1] Administering Avaya Aura® Communication Manager, Release 10.1.3, Issue 1, October 2023.
[2] Administering Avaya Aura® Session Manager, Release 10.1.3, Issue 1, May 2023.

The following documentation is available on request from Komutel at the https://www.komutel.com/en/public-safety/call-handling/komand911/

[3] Technical Specifications Kore Server and Databases - NG9-1-1 Edition.pdf
[4] Technical Specifications Komand SIP - Avaya Session Manager CM 10.1.pdf

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.