Avaya Solution & Interoperability Test Lab

Application Notes for Komutel Kontact with Avaya Aura® Session Manager and Avaya Aura® Application Enablement Services - Issue 1.0

Abstract

These Application Notes describe the steps required to integrate the Komutel Kontact (Call Center Solution) with Avaya Aura® Session Manager and Avaya Aura® Application Enablement Services. Komutel Kontact is a call queue and call center (ACD) management tool for managing virtual agents, generating statistics for the proper management of resources, and intelligently distributing calls. In the compliance test, Komutel Kontact successfully registered with Session Manager via SIP, distributed calls to available agents in the appropriate queue, and transferred calls to agents. In addition, Kontact used TSAPI link on Application Enablement Services to obtain status information of monitored stations on Avaya Aura® Communication Manager.

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 Kontact with Avaya Aura® Session Manager and Avaya Aura® Application Enablement Services. Kontact is a call queue and call center (ACD) management tool for managing virtual agents, generating statistics for the proper management of resources, and intelligently distributing calls. In the compliance test, Komutel Kontact successfully registered with Avaya Aura® Session Manager via SIP, distributed calls to available agents in the appropriate queue, and transferred calls to agents. In addition, Kontact used TSAPI to track agent status.

Komutel Kontact consists of two components: BLF Configuration and DAK Configuration. BLF Agent Configuration is used to track the agent status and DAK is basically a SIP phone that is responsible for handling incoming calls and transferring calls.

Komutel Kloud is a virtual next-generation telecom solution available in a cloud-based mode. It provides stability, performance, and extensibility. Kloud is available in either self-hosted local mode or provider-hosted configurations.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

Interoperability compliance testing covered the following features and functionality:

- Successful SIP registration of Kontact with Session Manager.
- Calls routed from Kontact to an available agent.
- Manually transferring calls in a queue by Kontact to a specified agent who is local to Communication Manager or on the PSTN.
- Tracking agent states with TSAPI.
- Playing music on hold by Kontact.
- Playing announcement to queued calls periodically by Kontact.
- Tracking statistics such as abandoned calls and incoming/outgoing calls in the call log.
- Caller ID display on Kontact.
- Proper system recovery after restart of the Kontact server and loss of IP connectivity.

2.2. Test Results

All test cases passed with the following observations noted:

- If a user places a call that is answered by an available agent and then the PC with Komutel Kloud installed is restarted, the agent's status will be set to Not Ready Kontact PC.
- If a user places a call that is answered by an available agent, and then the agent's PC is restarted, after the agent logs in to Kontact again the agent's status will be set to logged out. The user has to manually change to the proper status.

2.3. Support

For technical support on Kontact, 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:

- Communication Manager running on an Avaya S8300 Server with a G450 Media Gateway.
- Session Manager connected to Communication Manager via a SIP trunk and acting as a Registrar/Proxy for SIP telephones.
- System Manager used to configure Session Manager.
- Application Enablement Services used to monitor the availability of stations via a TSAPI link.

In addition, a Komutel Kontact registered with Session Manager and was configured SIP user. The Komutel BLF Agent was also installed and configured to support the BLF feature on the console.

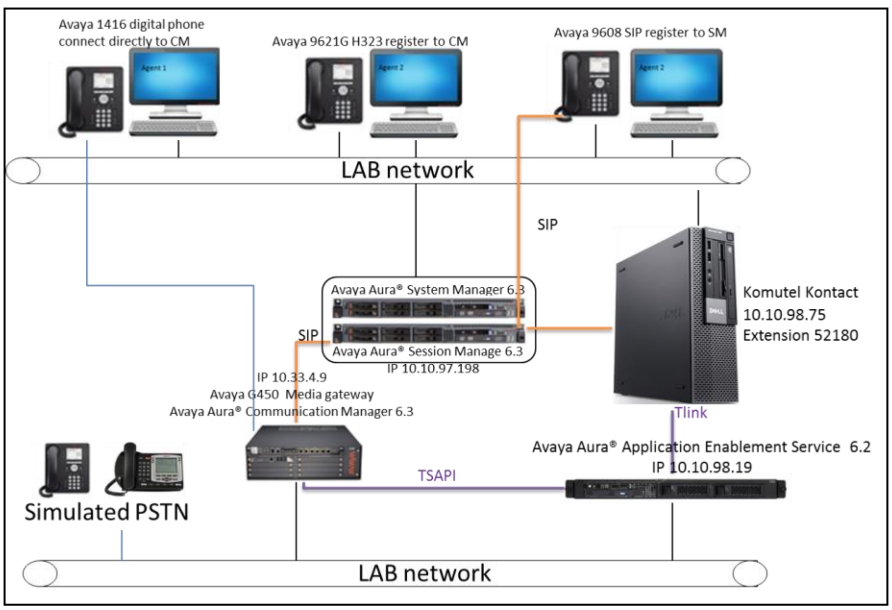


Figure 1: Avaya SIP Network with Komutel Kontact

4. Equipment and Software Validated

The following equipment and software were used for the sample configuration provided:

Hardware Component	Version
Avaya S8300D Media Server with Avaya G450 Media Gateway	Avaya Aura® Communication Manager 6.3 SP3
Avaya Aura® System Manager running on S8800 Server	Avaya Aura® System Manager 6.3.4
Avaya Aura® Session Manager running on S8800 Server	Avaya Aura® Session Manager 6.3SP4
Avaya Aura® Application Enablement Services VM on Avaya S8800 Server	6.2
Avaya 9600 Series IP Telephones	6.2.3 (H.323) 6.2.2 (SIP)
Avaya Digital Telephones 1416	NA
Komutel Kloud	1.3.9.15677
Komutel BLF Agent	2.0.7.22990
Komutel DAK	2.0.0.11107

5. Configure Avaya Aura® Communication Manager

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

- Verify Communication Manager license
- Administer CTI link between Communication Manager and Application Enablement Services for use with the BLF feature on the Kontact

A SIP trunk is created between Communication Manager and Session Manager. It is assumed the general installation of Communication Manager on Avaya G450 Media Gateway and Session Manager has been previously completed such as ip-network-regions, SIP signaling groups, SIP trunk, etc.

The Communication Manager configuration was performed using Communication Manager System Access Terminal (SAT) interface. Some screens in this section have been abridged and highlighted for brevity and clarity in presentation.

Please note that in the sample screenshots listed below the "display" command was used instead of the "change" or "add" commands, this is because all necessary changes were already in place when the screenshots were taken.

5.1. Verify OPS and SIP Trunk Capacity

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

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

```
display system-parameters customer-options                      Page   1 of  11
                                OPTIONAL FEATURES

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

                                                                USED
                                Platform Maximum Ports: 65000 170
                                      Maximum Stations: 41000 77
                              Maximum XMOBILE Stations: 41000 0
                    Maximum Off-PBX Telephones - EC500: 36000 0
                    Maximum Off-PBX Telephones -   OPS: 41000 6
                    Maximum Off-PBX Telephones - PBFMC: 36000 0
                    Maximum Off-PBX Telephones - PVFMC: 36000 0
                    Maximum Off-PBX Telephones - SCCAN: 0     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 SIP trunks supported by the system is sufficient.

```
display system-parameters customer-options                 Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 40
           Maximum Concurrently Registered IP Stations: 18000 17
             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: 18000 0
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 30
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
                             Maximum TN2501 VAL Boards: 128   1
                     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.)
```

On **Page 3**, verify that the **C**omputer Telephony Adjunct Links option is enabled.

```
display system-parameters customer-options                 Page   3 of  11
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? n                         CAS Main? n
Answer Supervision by Call Classifier? y                Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? y
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                      DCS (Basic)? y
          ASAI Link Core Capabilities? y                DCS Call Coverage? y
          ASAI Link Plus Capabilities? y               DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? n
              ATM WAN Spare Processor? n                          DS1 MSP? y
                                 ATMS? n            DS1 Echo Cancellation? y
                  Attendant Vectoring? y

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

5.2. Configuring AE Services

Enabling AE Services refers to administering the transport link between Communication Manager and AE Services.

Enter command **change ip-services**. Complete Page 1 of the **ip-services** form as follows:

- In the **Service Type** field, type **AESVCS**.
- In the **Enabled**, enter **y**.
- In the **Local Node** field, type **procr**.
- In the **Local Port** field, accept the default (**8765**).

```
change ip-services                                         Page   1 of   3
                                           IP SERVICES
 Service     Enabled     Local        Local       Remote      Remote
  Type                   Node         Port        Node        Port
  AESVCS        y        procr        8765
```

Complete Page 3 of the **ip-services** form as follows:

- In the **AE Services Server** field, type the name of the AES Server, for example: **DevAES62**.
- Enter **Password**, see example in screenshot below.
- Set the **Enabled** field to **y**.

```
change ip-services                                         Page   3 of   3
                            AE Services Administration

   Server ID    AE Services        Password          Enabled    Status
                   Server
        1:      DevAES62         DevConnect123         y         in use
```

5.3. Configure CTI Link

Add a CTI link using the “add cti-link n” command, where “n” is an available CTI link number. Enter an available extension number in the **Extension** field. Note that the CTI link number and extension number may vary. Enter “ADJ-IP” in the **Type** field, and a descriptive name in the **Name** field. Default values may be used in the remaining fields.

```
display cti-link 7                                              Page   1 of   3
                                    CTI LINK
 CTI Link: 2
Extension: 52102
     Type: ADJ-IP
                                                                     COR: 1
     Name: DevAES62
```

6. Configure Avaya Aura® Session Manager

It is assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms. All the configuration items required as part of the initial Session Manager installation have already been defined. This includes items such as certain SIP domains, locations, SIP entities and Session Manager itself.

This section provides the procedures for configuring SIP User on Session Manager.

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL "https://<*ip-address*>/SMGR", where <*ip-address*> is the IP address of System Manager. Log in with the appropriate credentials.

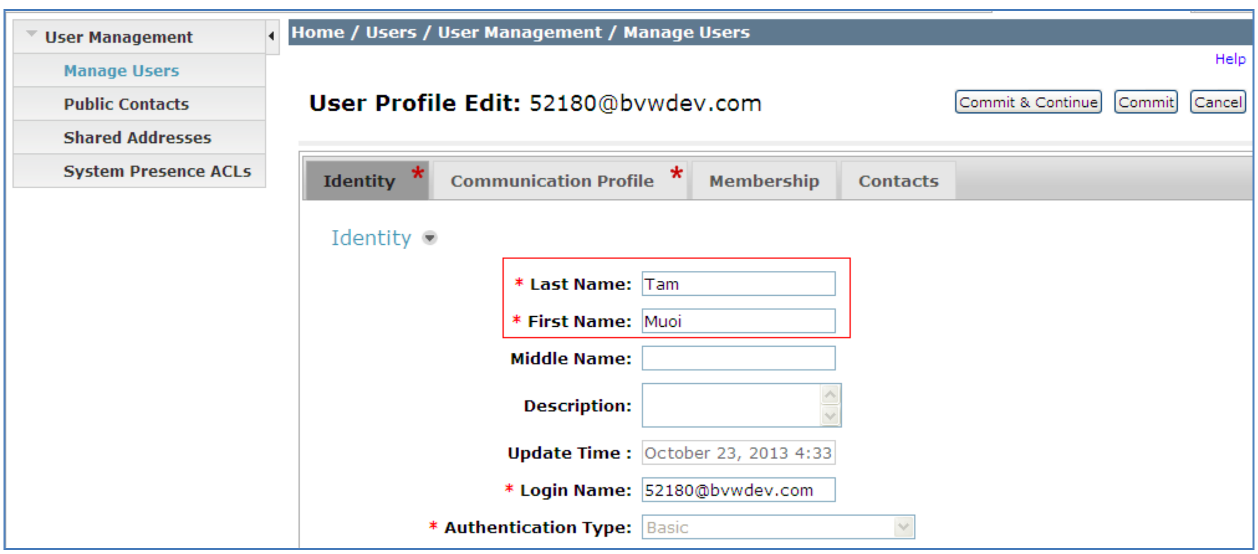
6.1. Add SIP Users

Add a SIP user for the Kontact and use the option to automatically generate the SIP stations on Communication Manager when adding a new SIP user for Session Manager.

To add new SIP users, expand **Users Management** and select **Manage Users** from left and select the **New** button (not shown) on the right. Enter values for the following required attributes for a new SIP user in the **General** section of the new user form.

- **Last Name:** Enter the last name of the user.
- **First Name:** Enter the first name of the user.
- **Login Name:** Enter *<extension>@<sip domain>* of the user (e.g., 52180@bvwdev.com).
- **Authentication Type:** Select *Basic*.
- **SMGR Login Password:** Enter the password which will be used to log into System Manager
- **Confirm Password:** Re-enter the password from above.
- **Shared Communication Profile Password:** Enter the password which will let the SIP phone log into Session Manager.
- **Confirm Password:** Re-enter the password from above.

The screen below shows the information of a SIP user created for the compliance test:



Click on the **Communication Profile** tab, in the **Communication Profile** section, enter:

- **Communication Profile password:** Enter the password used to login SIP device.

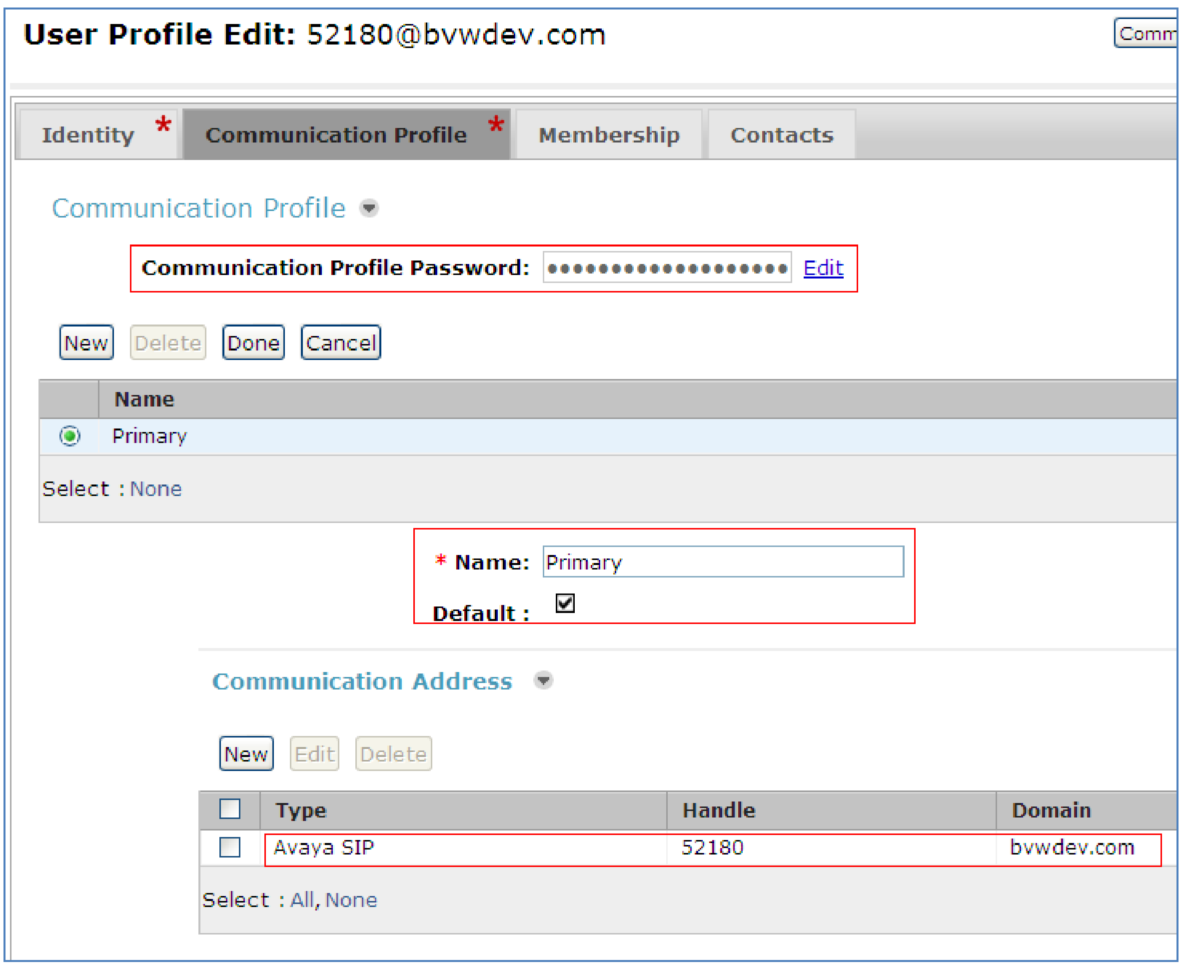
Verify there is a default entry identified as the **Primary** profile for the new SIP user. If an entry does not exist, select **New** (not shown) and enter values for the following required:

- **Name:** Enter **Primary**.
- **Default:** Enter ☑

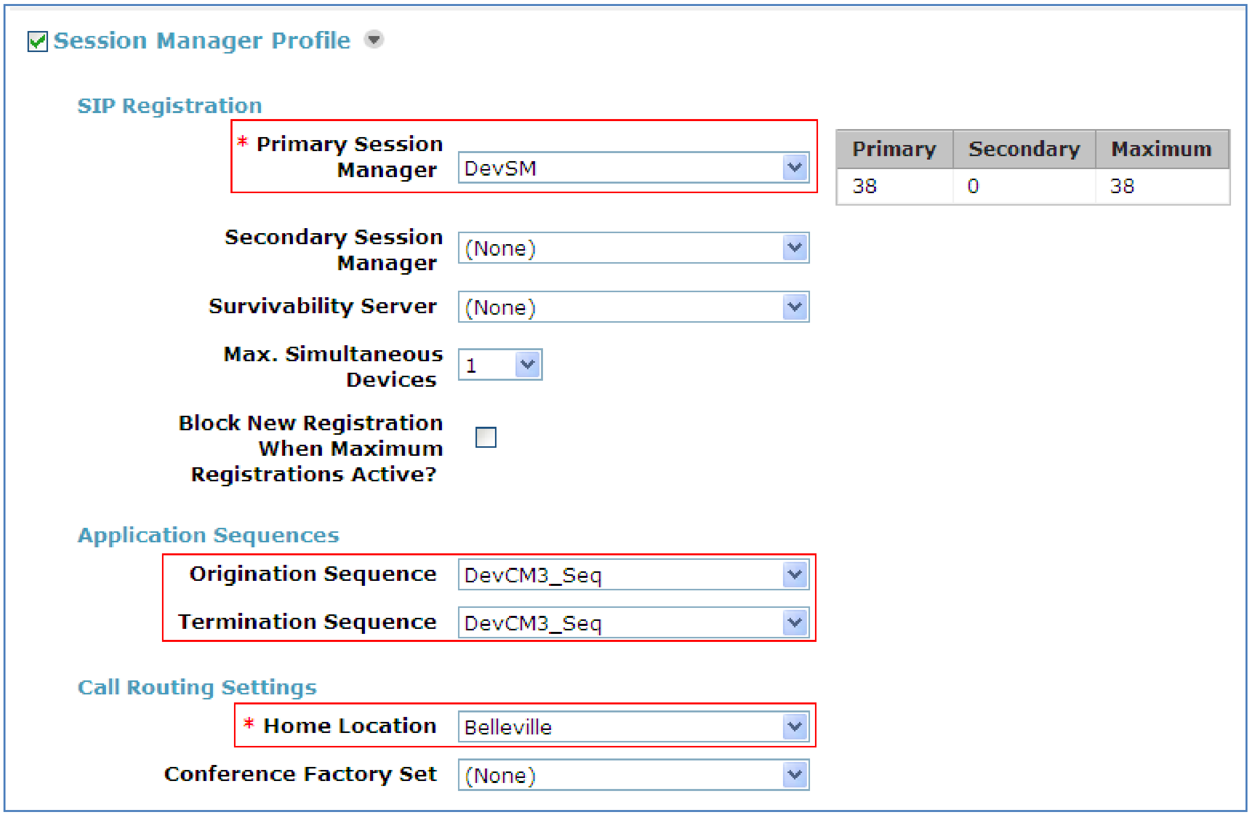
Click **New** to define a **Communication Address** for the new SIP user. Enter values for the following required fields:

- **Type:** select *Avaya SIP*.
- **Handle:** SIP extension.
- **Domain:** select SIP domain.

The screen below shows the information when adding a new SIP user to the sample configuration. Click **Add** (not shown).



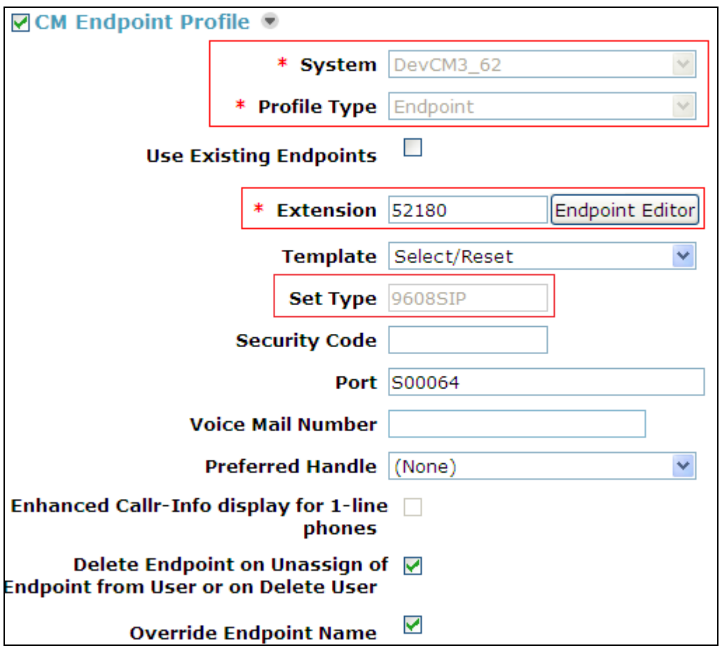
In the **Session Manager Profile** section, specify the Session Manager entity for **Primary Session Manager** and assign the **Application Sequence** to the new SIP user. The same Application Sequence can be used for both the originating and terminating sequence. Set the **Home Location** field. Below is an example used during the compliance test:



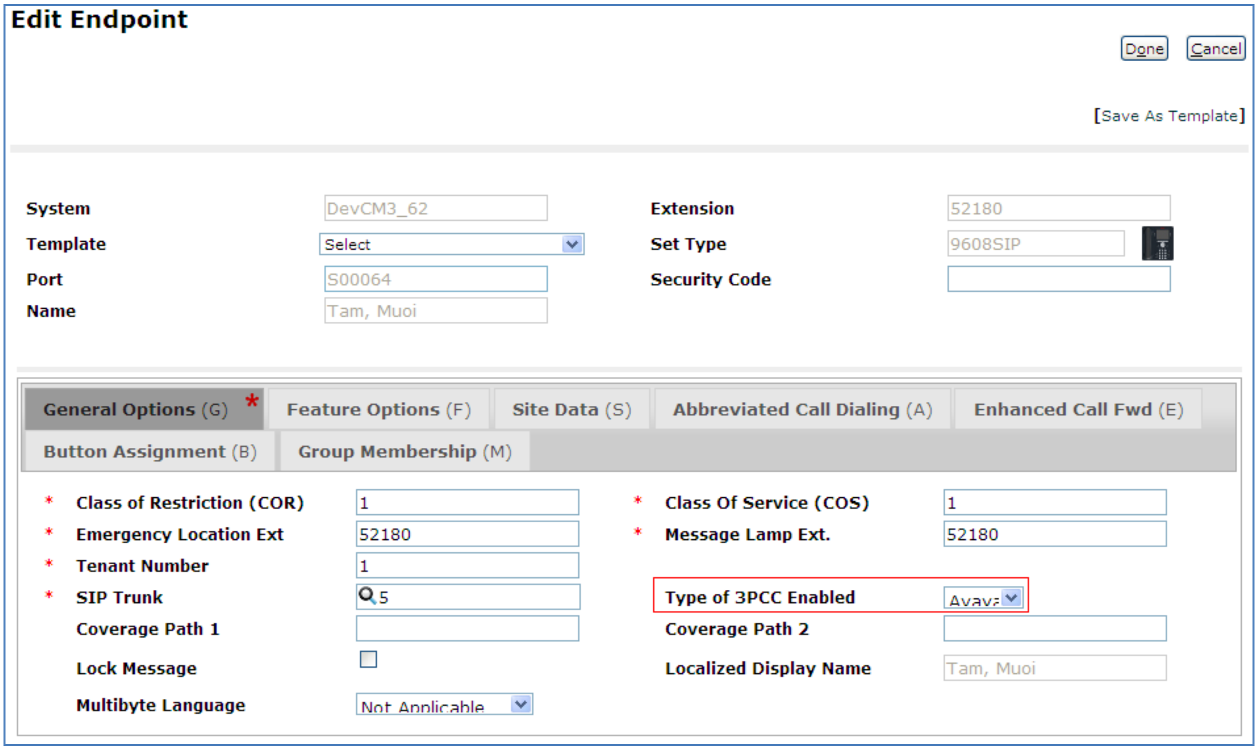
In the **CM Endpoint Profile** section, fill in the following fields:

- **System:** Select the managed element corresponding to Communication Manager.
- **Profile Type:** Select **Endpoint.**
- **Extension:** Enter extension number of SIP user.
- **Set Type:** The value of this read only field depends on the selected template. In the example as shown below, the type is **9608SIP** (when this user was created, the template **9608SIP_DEFAULT_CM_6_3** was selected).

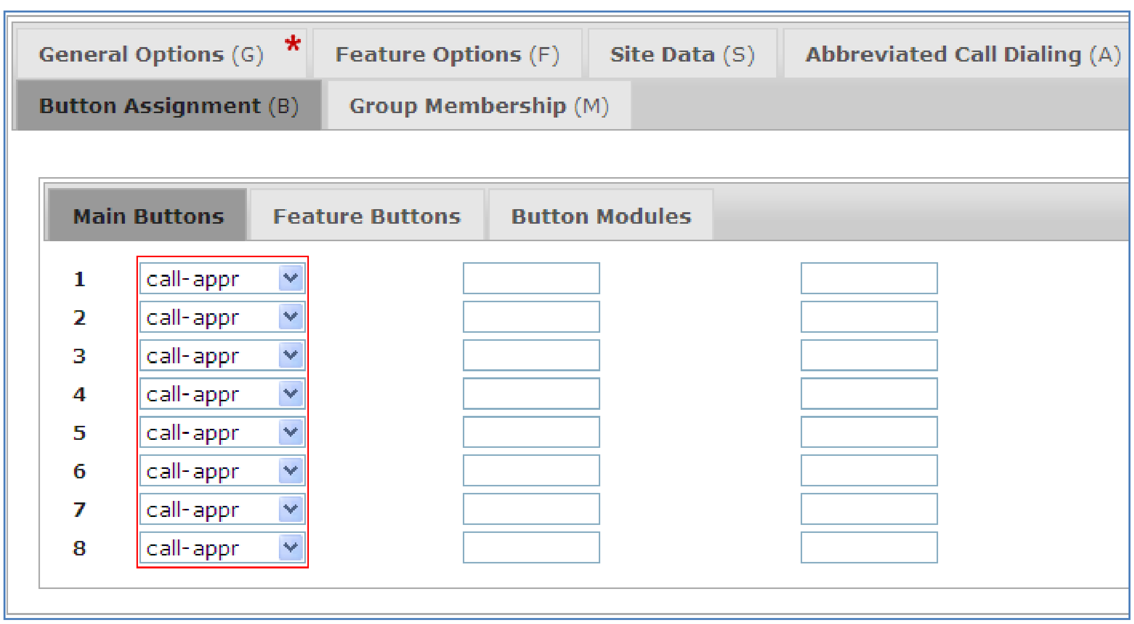
Leave other fields with default value. The screen below shows the information when adding a new SIP user to the sample configuration.



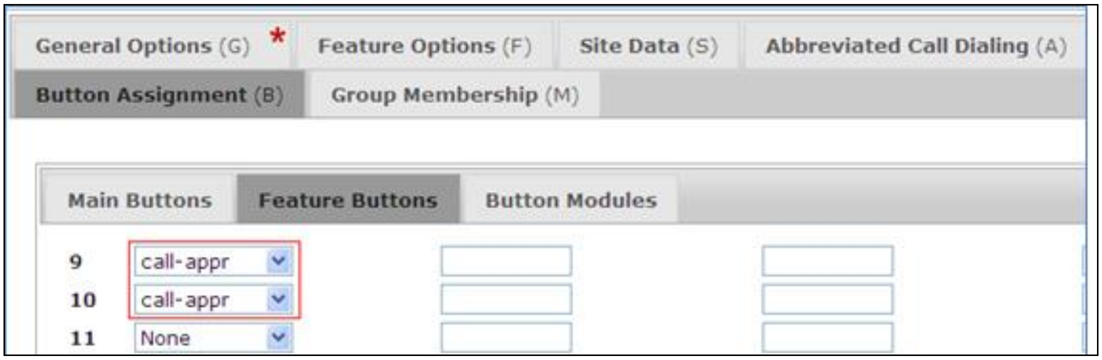
Click the **Endpoint Editor** button beside **Extension** field (shown above) to modify endpoint details. In the **General Options** tab, make sure **Type of 3PCC Enable** is set to **Avaya.**



In the **Feature Options** tab, make sure **IP Softphone** option is checked (not shown). In the **Button Assignment** tab, maximize the allowed of **call-appr** are 10 lines (8 lines in **Button Assignment** tab:



Button assigment for **call-appr** continue if needed on the **Feature Buttons** tab:



7. Configure Avaya Aura® Application Enablement Services

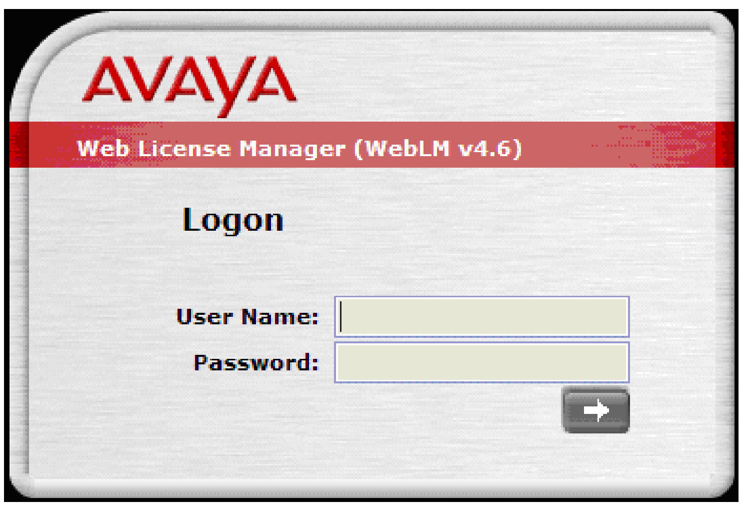
This section provides the procedures for configuring Avaya Aura® Application Enablement Services. The procedures include the following areas:

- Verify TSAPI license
- Launch OAM interface
- Administer TSAPI link
- Disable security database
- Restart TSAPI service
- Obtain Tlink name
- Administer user for Komutel BLF Agent

7.1. Verify TSAPI License

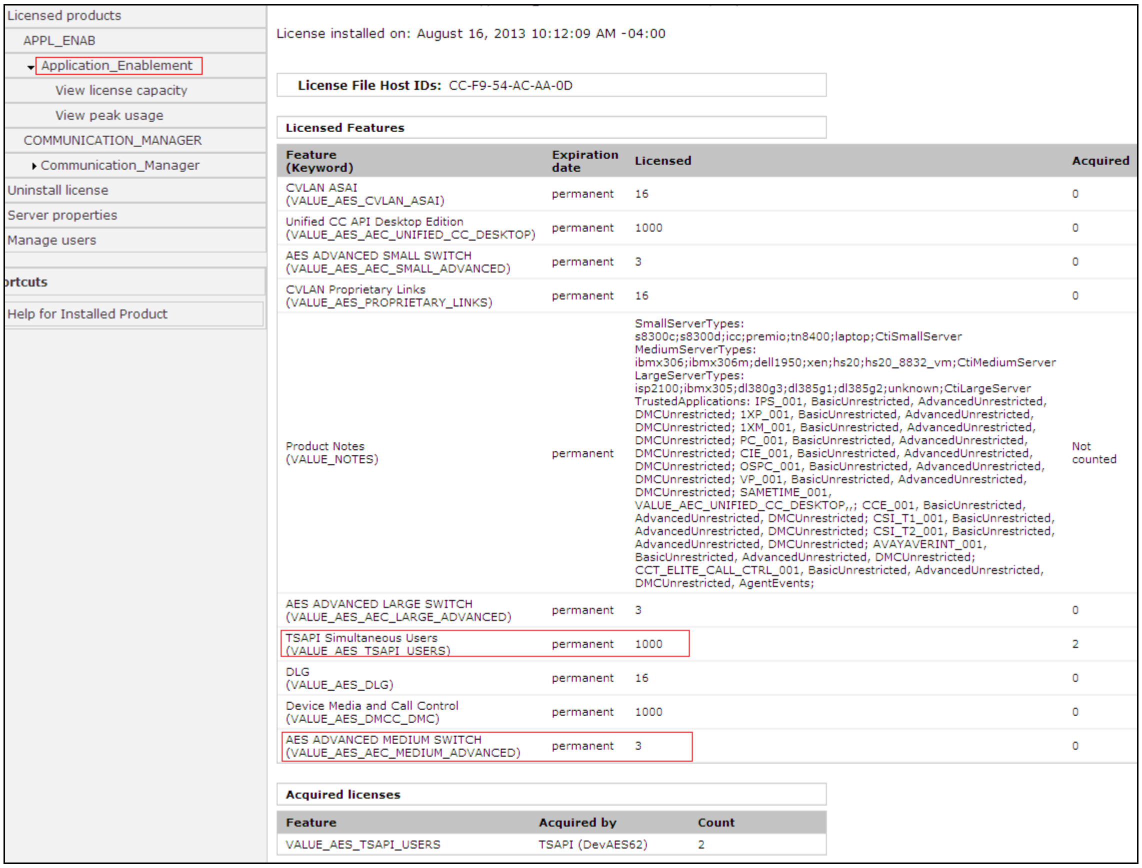
Access the Web License Manager interface by using the URL "https://<*ip-addr*>/WebLM/" in an Internet browser window, where <*ip-addr*> is the IP address of the Application Enablement Services server.

The **Web License Manager** screen is displayed. Log in using the appropriate credentials.



The **Web License Manager** screen is displayed. Select **Licensed products →APPL_ENAB→ Application_Enablement** in the left pane to display the **Licensed Features** screen in the right pane.

Verify that there are sufficient licenses for **TSAPI Simultaneous Users** as shown below. Also verify that there is an applicable advanced switch license, in this case **AES ADVANCED MEDIUM SWITCH** for the Avaya S8800 Server.

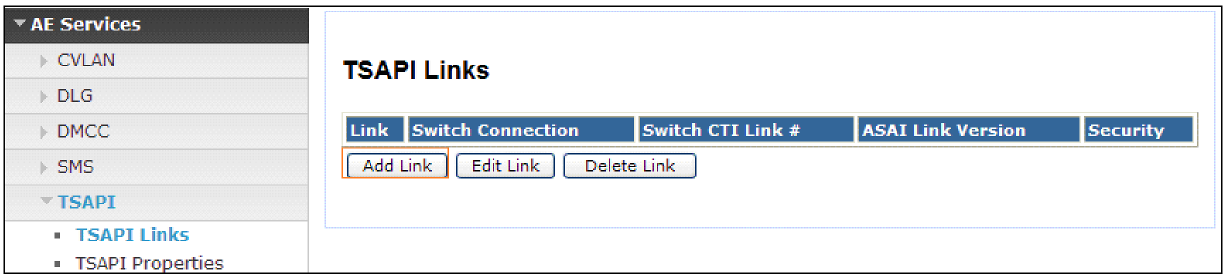


7.2. Launch OAM Interface

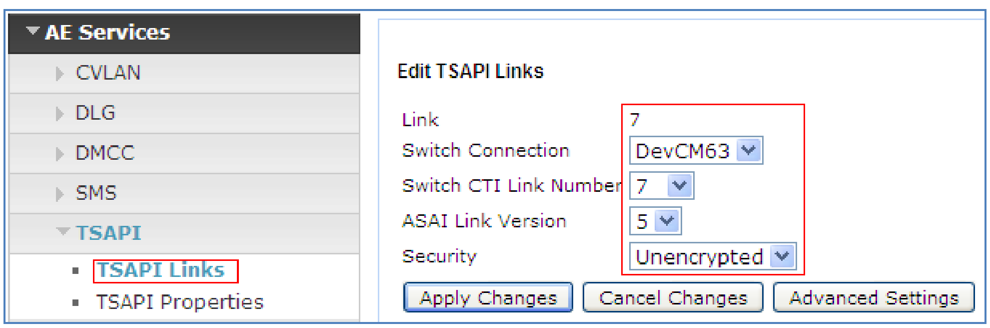
Access the OAM web-based interface by using the URL "https://<*ip-addr*>" in an Internet browser window, where <*ip-addr*> is the IP address of the Application Enablement Services server. Log in using the appropriate credentials. (not shown)

7.3. Administer TSAPI Link

To administer a TSAPI link, select **AE Services→TSAPI→TSAPI Links** from the left pane. The **TSAPI Links** screen is displayed as shown below. Click **Add Link**.

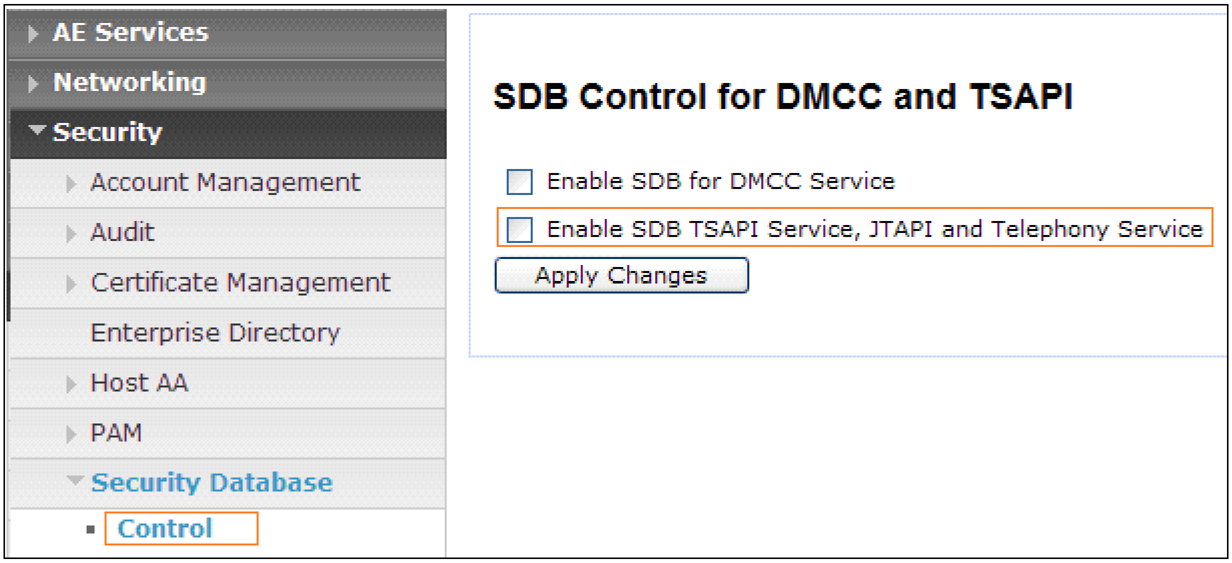


The **Add TSAPI Links** screen is displayed next. The **Link** field is only local to the Application Enablement Services server and may be set to any available number. For **Switch Connection**, select the relevant switch connection from the drop-down list. In this case, the existing switch connection **DevCM63** is selected. For **Switch CTI Link Number**, select the CTI link number from **Section 5.3**. Retain the default values in the remaining fields and click **Apply Changes**.



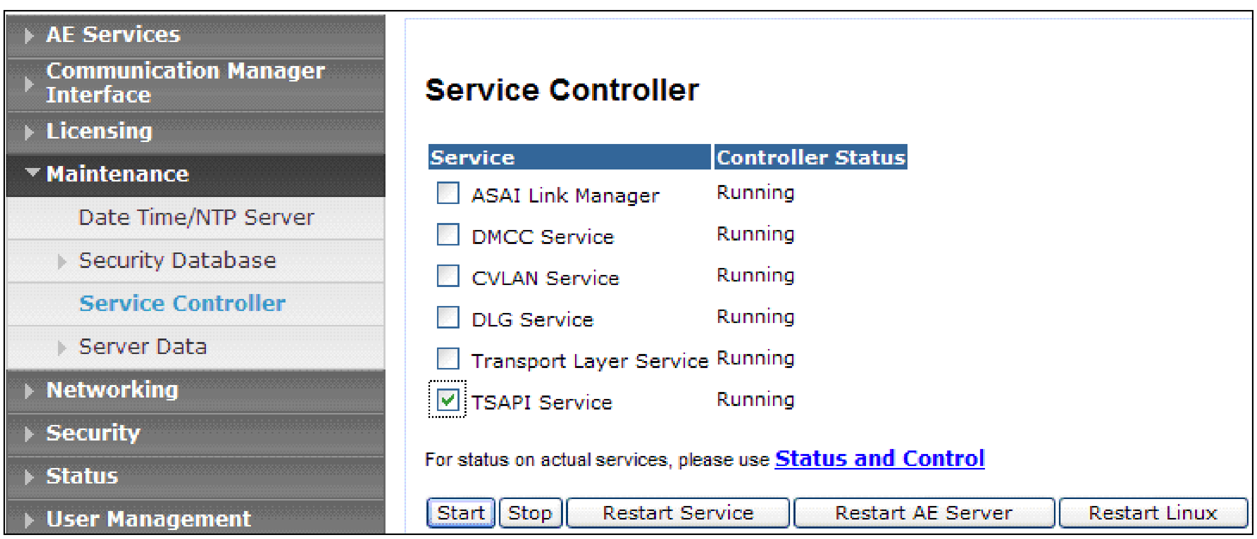
7.4. Disable Security Database

Select **Security→Security Database→Control** from the left pane to display the **SDB Control for DMCC and TSAPI** screen. Uncheck **Enable SDB TSAPI Service, JTAPI and Telephony Service** and click **Apply Changes**.



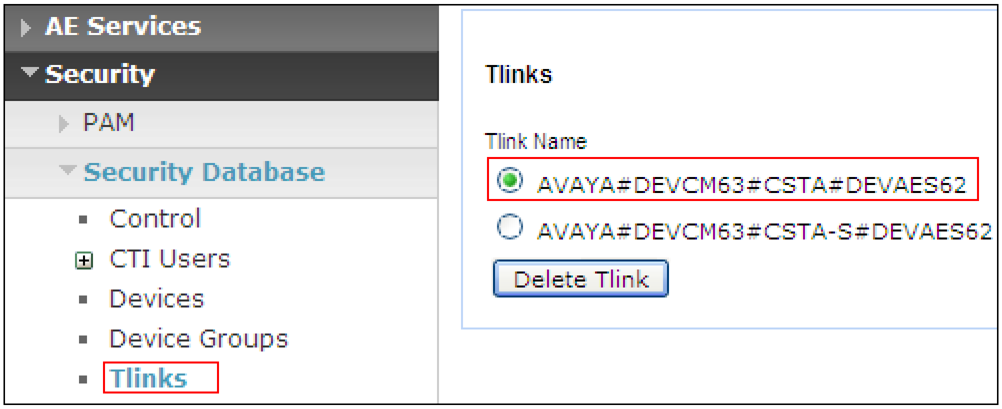
7.5. Restart TSAPI Service

Select **Maintenance→Service Controller** from the left pane to display the **Service Controller** screen. Check the **TSAPI Service** and click **Restart Service**.



7.6. Obtain Tlink Name

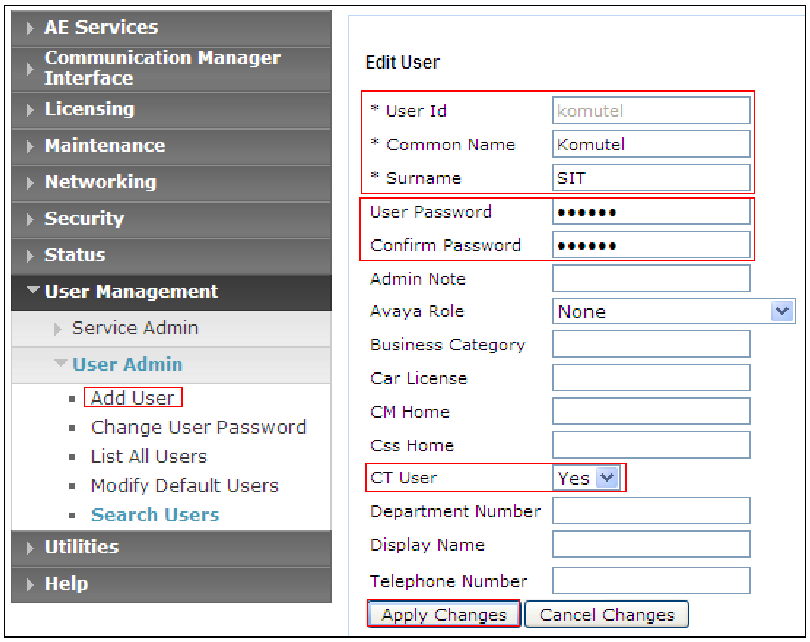
Select **Security→Security Database→Tlinks** from the left pane. The **Tlinks** screen shows a listing of the Tlink names. A new Tlink name is automatically generated for the TSAPI service. Locate the Tlink name associated with the relevant switch connection, which would use the name of the switch connection as part of the Tlink name. Make a note of the associated Tlink name, which will be used later for configuring the Komutel BLF Agent. See the screen below for an example of tlink name:



7.7. Administer User for Komutel Kontact

Select **User Management→User Admin→Add User** from the left pane to display the **Add User** screen.

Enter desired values for **User Id**, **Common Name**, **Surname**, **User Password**, and **Confirm Password**. For **CT User**, select "Yes" from the drop-down list. Retain the default value in the remaining fields. Click **Apply** at the bottom of the screen:



8. Configure Komutel Kontact

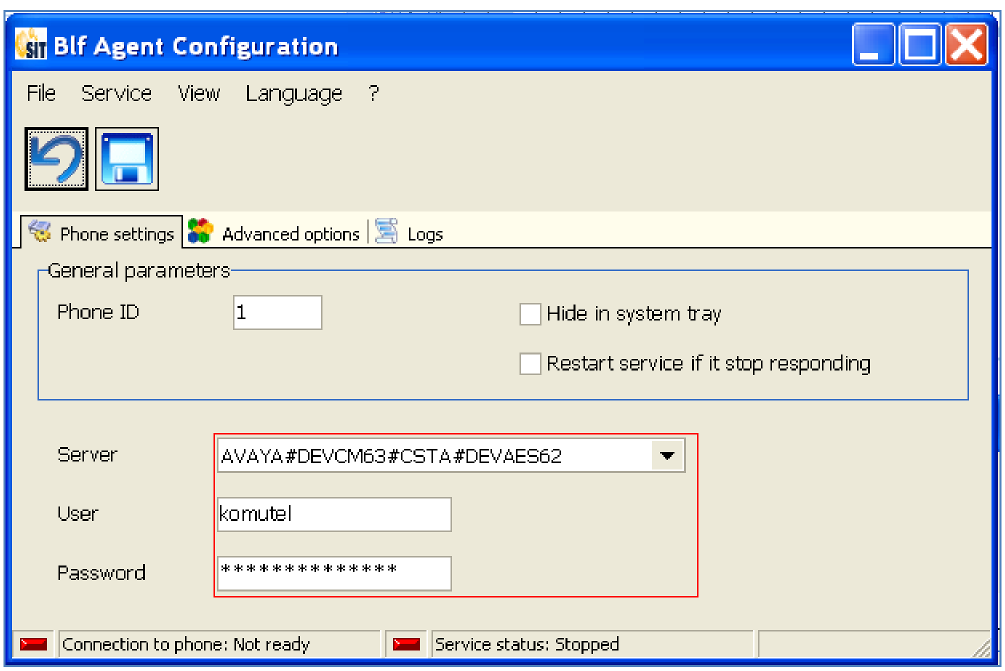
This section provides the procedures for configuring Komutel Kontact. The procedures include the following areas:

- Configure Kontact through **BLF Agent** and **DAK Configuration** applications
- Verify Phone System
- Configure Queues
- Configure Agents

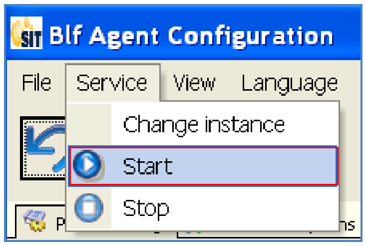
Komutel Kontact consists of two components: BLF Agent Configuration and DAK Configuration. BLF Agent Configuration is used to track agent status and DAK is a SIP phone responsible for handling incoming calls and transferring calls.

8.1. Configure BLF Agent

The BLF Agent is used to track the line status of monitored stations on Communication Manager using a TSAPI link configured on Application Enablement Services. In the **BLF Agent Configuration** application, set the **Server** field to the Tlink name of the TSAPI link obtained in **Section 7.6.** The **Server** field should automatically discover the Tlink name. Set the **User** and **Password** fields according to the CT User configured in **Section 7.77.7**. Set the **Phone ID** field, which will group stations from a particular Communication Manager system. This Phone ID will be used for all users when configuring BLF buttons.

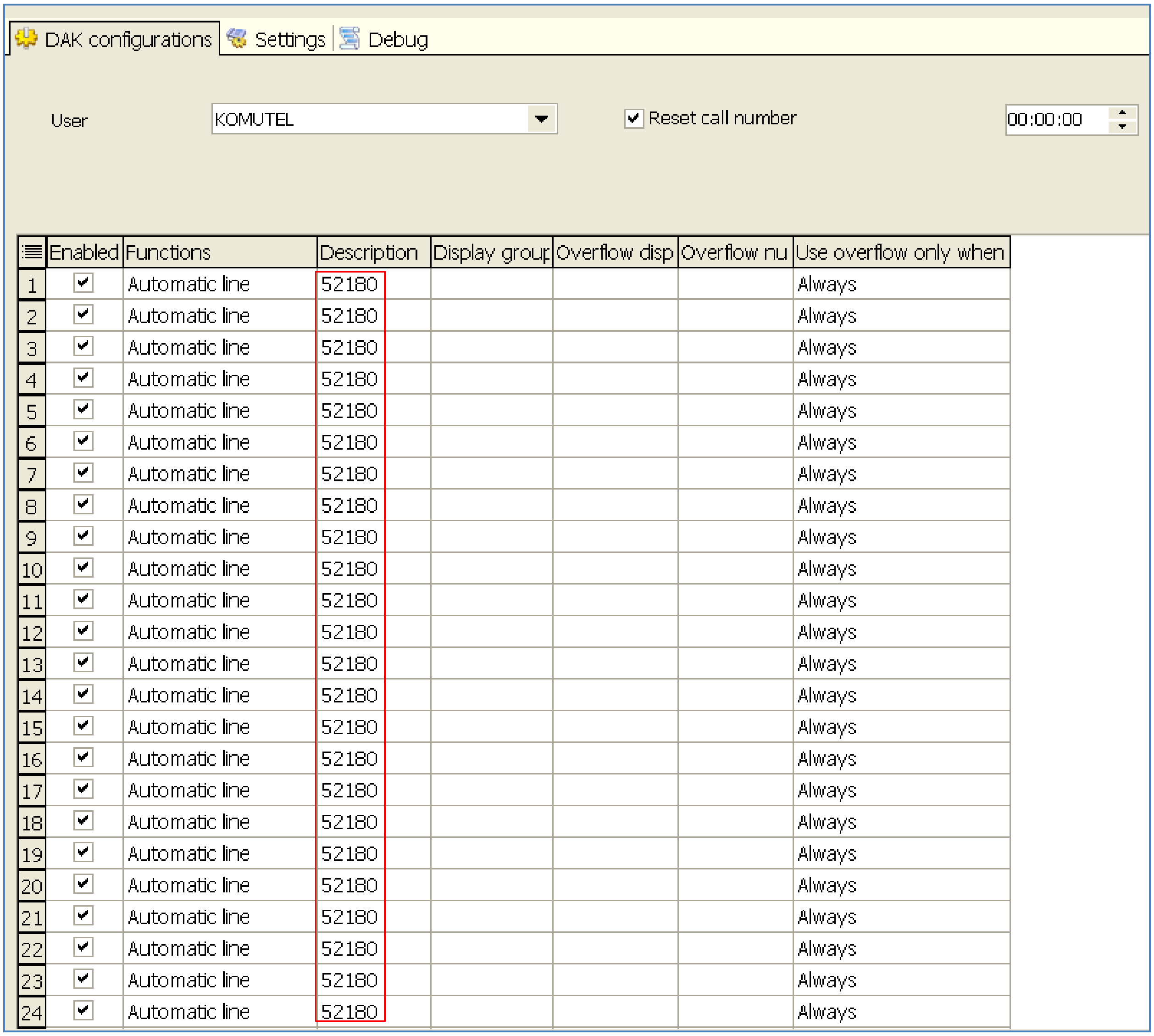


To start the BLF Agent service, select **Service→Start** from the menu of the **BLF Agent Configuration** application as shown below.



8.2. Configure DAK Configuration

Use the **DAK Configuration** application to configure the console's lines. In the **DAK configurations** tab, depending on the number of lines that are available, choose *Automatic Line* in the **Functions** column and enter the extension number in the **Description** column. As shown below, the console was configured with 24 line appearances with extension 52180.



	Enabled	Functions	Description	Display group	Overflow disp	Overflow nu	Use overflow only when
1	☑	Automatic line	52180				Always
2	☑	Automatic line	52180				Always
3	☑	Automatic line	52180				Always
4	☑	Automatic line	52180				Always
5	☑	Automatic line	52180				Always
6	☑	Automatic line	52180				Always
7	☑	Automatic line	52180				Always
8	☑	Automatic line	52180				Always
9	☑	Automatic line	52180				Always
10	☑	Automatic line	52180				Always
11	☑	Automatic line	52180				Always
12	☑	Automatic line	52180				Always
13	☑	Automatic line	52180				Always
14	☑	Automatic line	52180				Always
15	☑	Automatic line	52180				Always
16	☑	Automatic line	52180				Always
17	☑	Automatic line	52180				Always
18	☑	Automatic line	52180				Always
19	☑	Automatic line	52180				Always
20	☑	Automatic line	52180				Always
21	☑	Automatic line	52180				Always
22	☑	Automatic line	52180				Always
23	☑	Automatic line	52180				Always
24	☑	Automatic line	52180				Always

In the **Setting** tab, enter the following information:

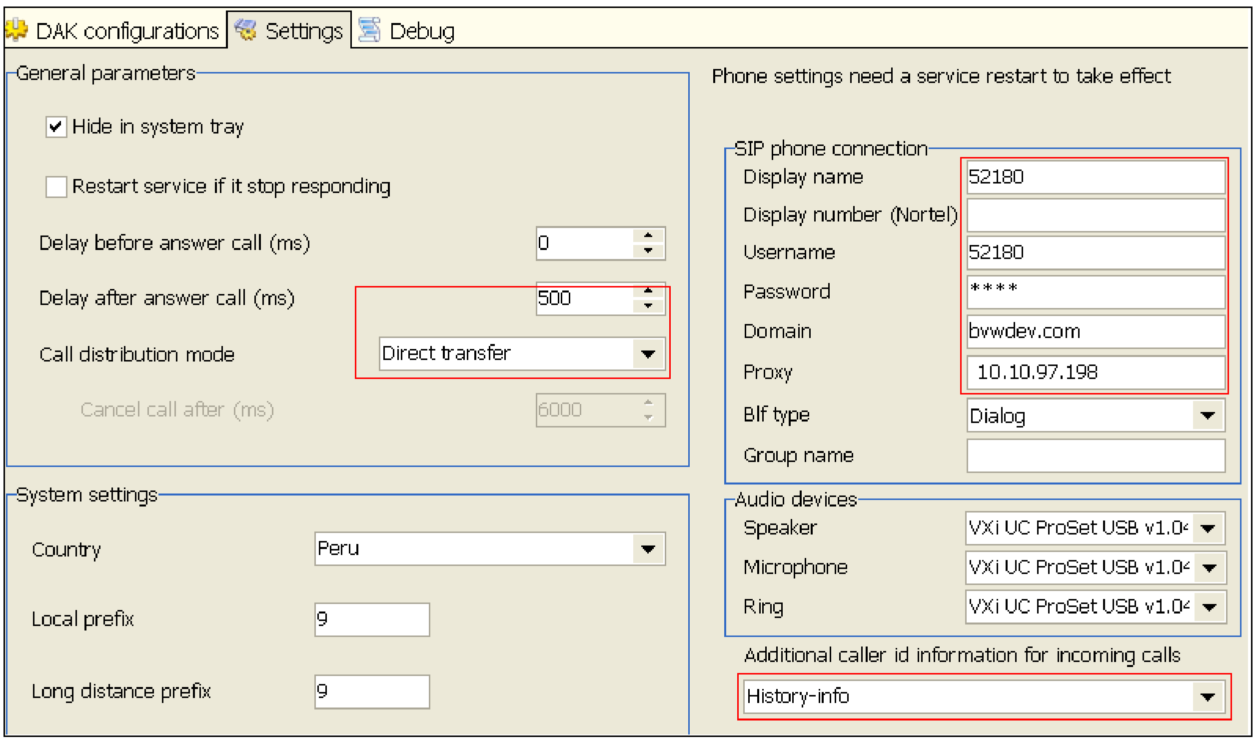
In the **General parameter** section:

- **Delay after answer call (ms)**: select **500**.
- **Call distribution mode**: select **Direct transfer**.

In the **SIP phone connection** section, enter:

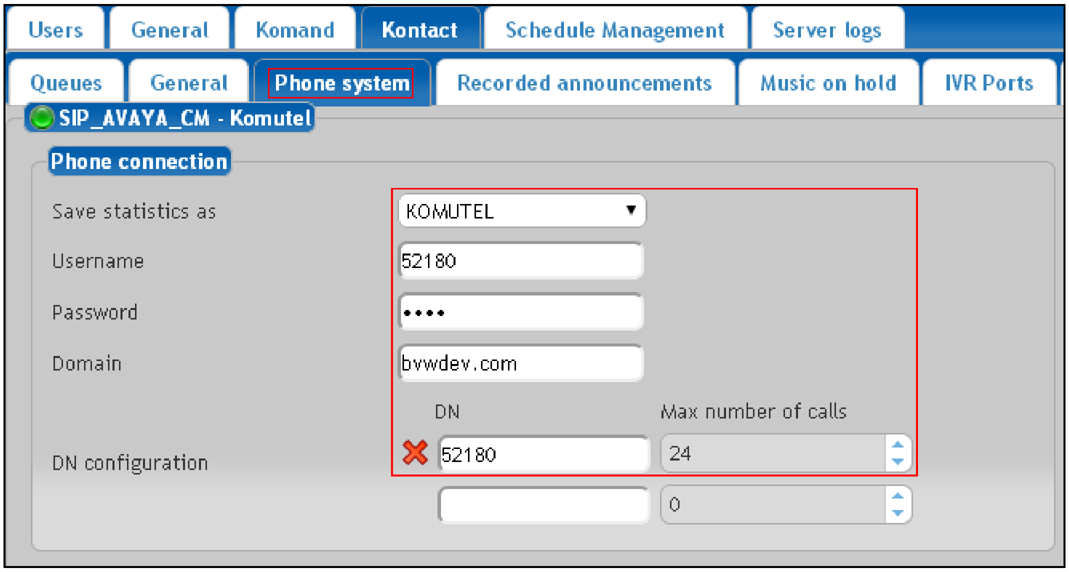
- A descriptive **Display name** (e.g., *52180*).
- The **Username** and **Password** used to register with Session Manager specified in **Section 6.1**.
- The SIP **Domain** (e.g., *bvwdev.com*).
- The **Proxy**, which specifies the IP address of the SIP interface of Session Manager.

In the **Audio devices** section, specify the audio device or headset that will be used with the console. And select **History-info** option for **Additional caller id information for incoming calls**. Click the **Save** button at the top of screen.



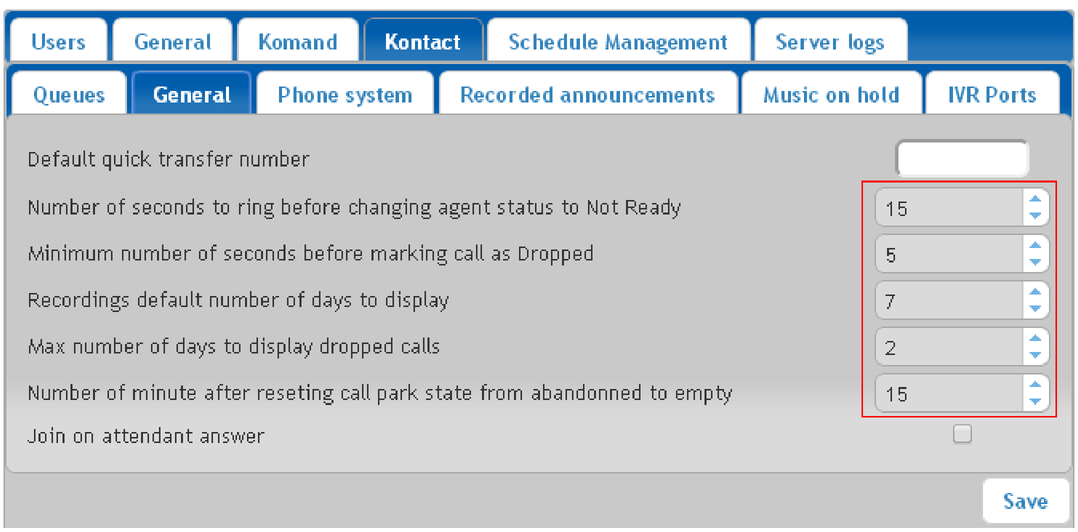
8.3. Verify Kontact Phone system

From a web browser access the page127.0.0.0 with the appropriate login credentials. This will launch the Komutel Kloud admin web interface. Select the **Configuration** tab then click on the **Configurations** link. In the configuration page, select **Kontact → Phone System**. It shows detail of SIP extension that was configured in **DAK** application.



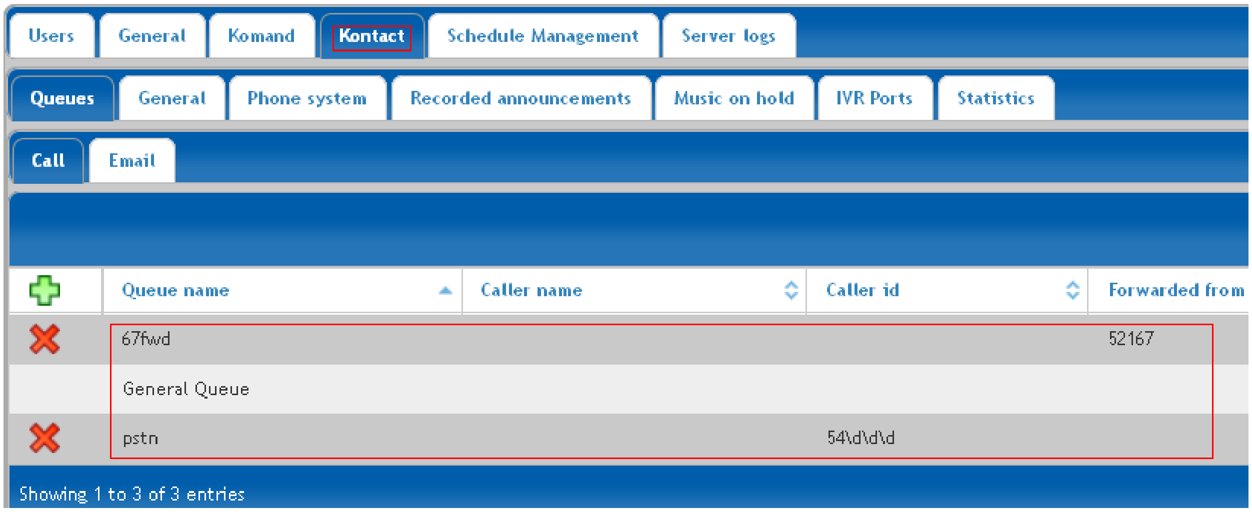
Click on **Recored announcements** and **Music on hold** tab to manager the audio file for annoucement and music on hold so that music would be played while a caller is in queue with a message played periodically. This configuration is not shown in these Application Notes.

Select **Kontact → General** to manager general information for Kontact, below is the setup used for the compliance test:



8.4. Configure queues

Select **Kontact → Queues**. This tab shows the current list of queues in the system. Click on the "+" plus sign to add new queue.

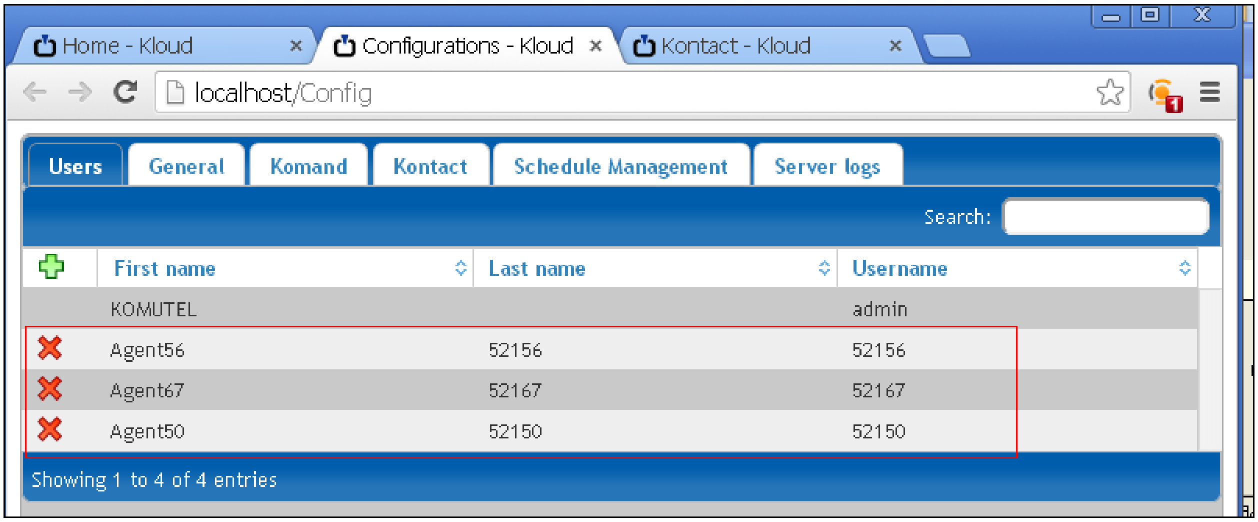


Here is the example of a queue named **pstn** used during compliance testing. When the caller with extension start with 54 dial 52180, Kontact will route the call to an available agent logged into this queue or queue the call if there is no available agent.

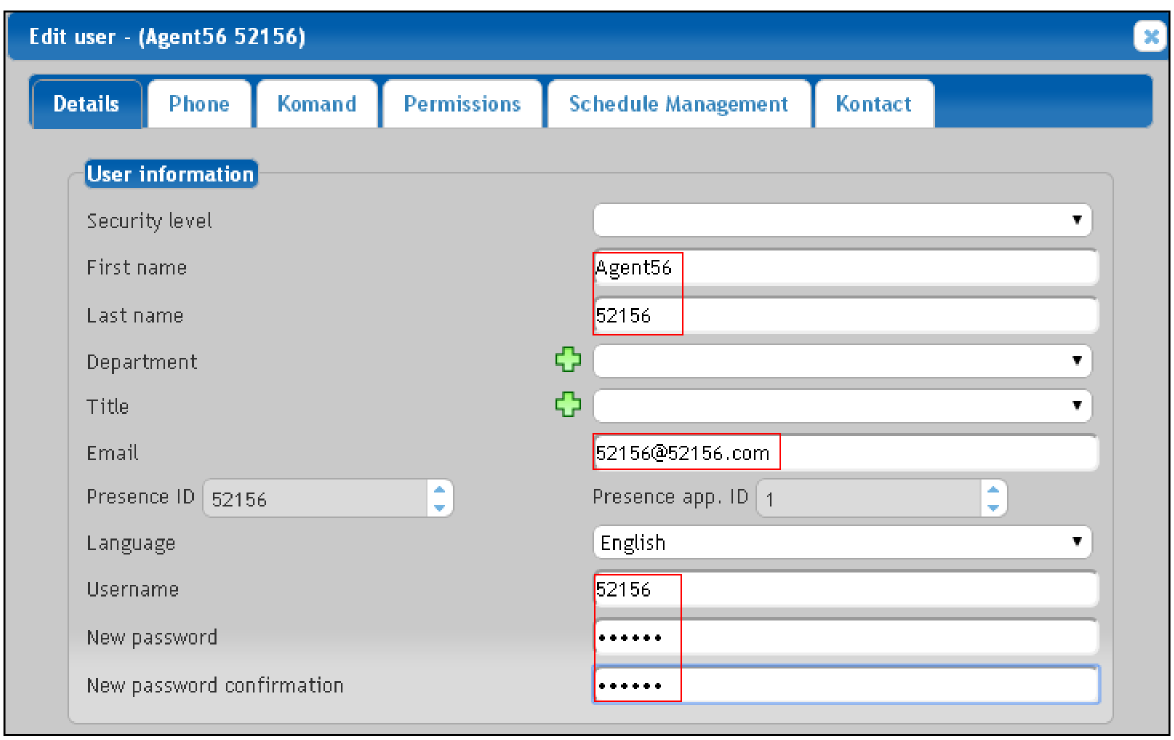
8.5. Configure agents

Next, configure agent and its extension that will be monitored by the Komutel Kontact. Repeat these steps for each agent and extension to be monitored.

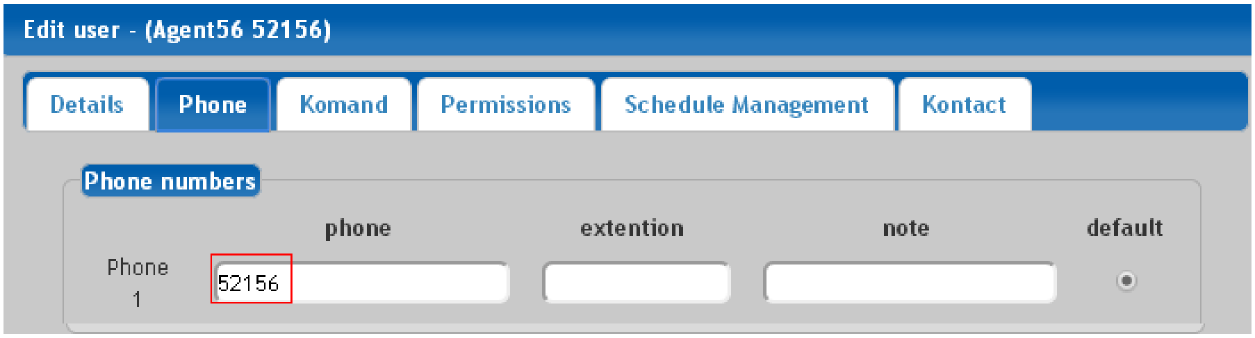
In the **Configurations** page, select **User** tab. It shows the current list of agents in the system.



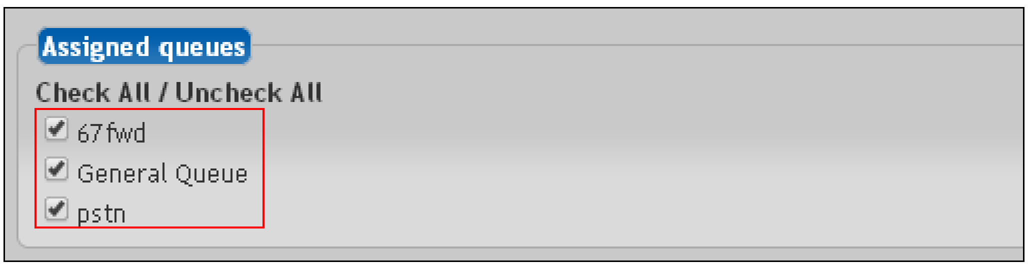
Click on the "+" plus sign to add new agent. The following screen shows the **User information** fields populated for an agent previously configured.



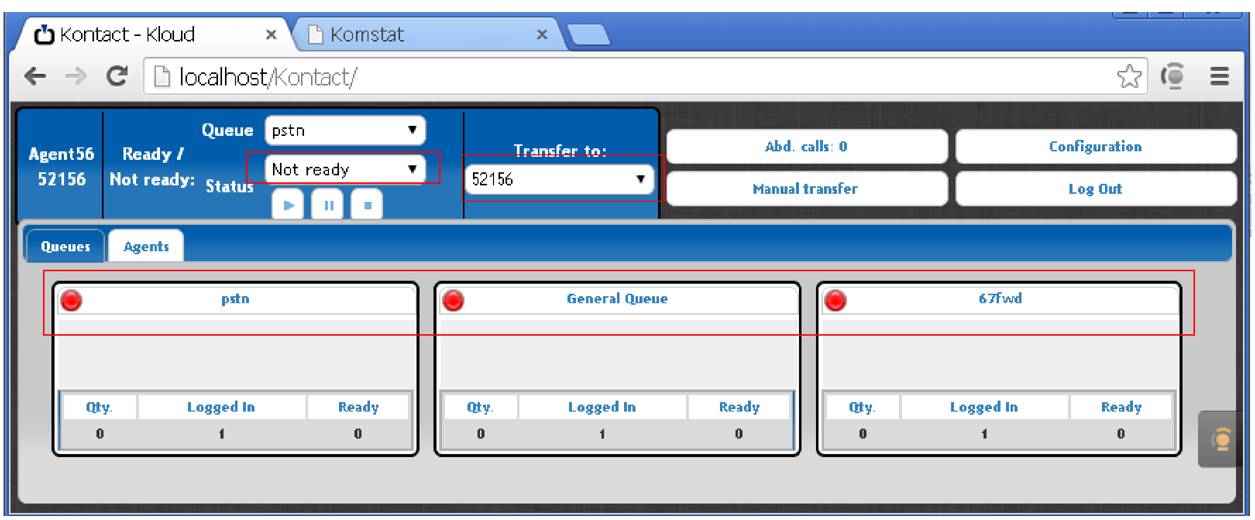
Select the **Phone** tab to add phone number for this agent:



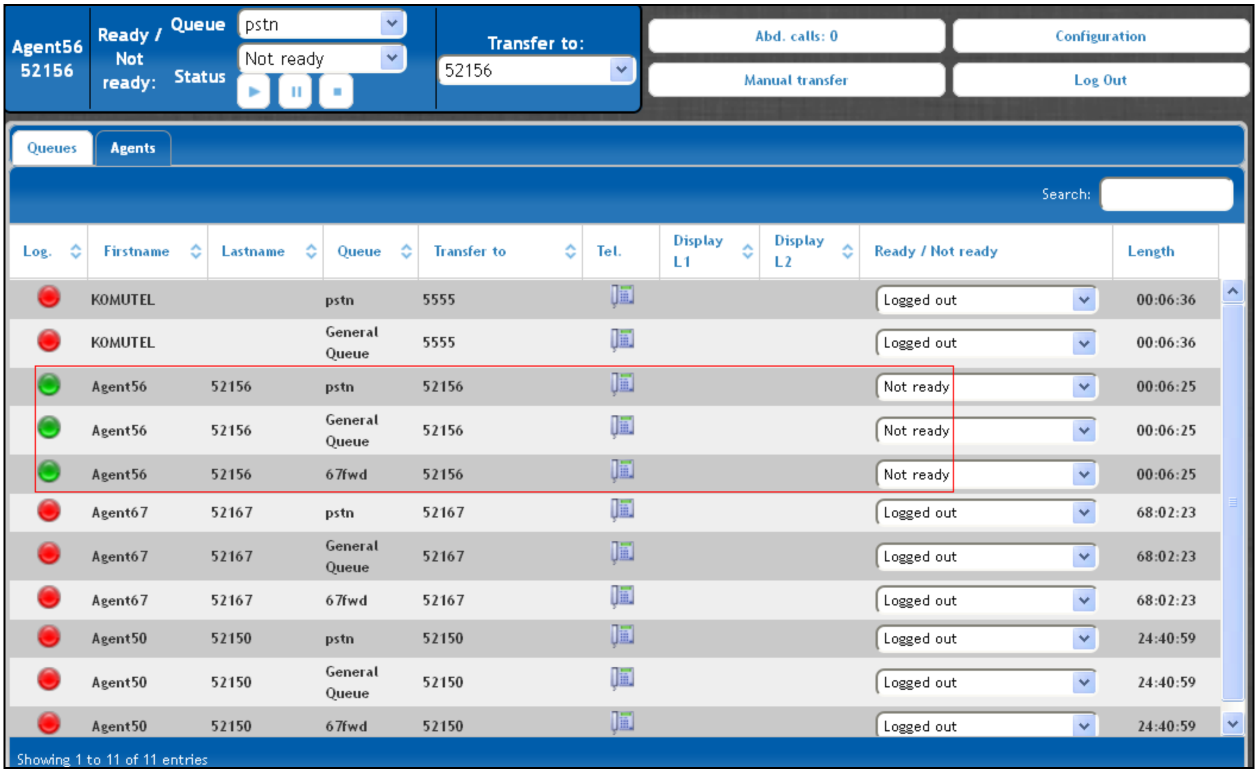
In the **Kontact** tab, assign queue to this agent, in the example below this **52156** agent is assigned to 3 available queues in the sytem:



Log into **Komutel Kloud**, then select the **Kontact** link. In the **Queues** tab, all of the assigned queues will be displayed for the logged in agent. In this figure, the 3 queues **pstn**, **General** and **67fwd** are displayed, this 52156 agent is not ready, and all of the queues are in the **Not Ready** state as indicated by the Red circle by the queue name.



Click on the **Agent** tab, all of the configured agents that will be monitored are displayed as shown below. In order for an agent to become available to handle an incoming call, the agent's state must be set to *Ready*. In this example, all of the agents are in the *Not Ready* state and only 52156 is logged in.



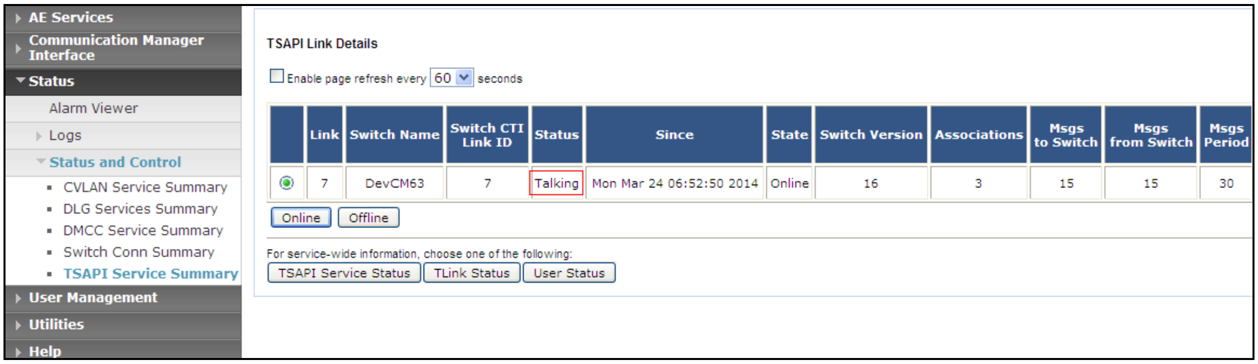
9. Verification Steps

This section provides the tests that can be performed to verify proper configuration of the Komutel Kontact with Communication Manager, Session Manager, and Application Enablement Services.

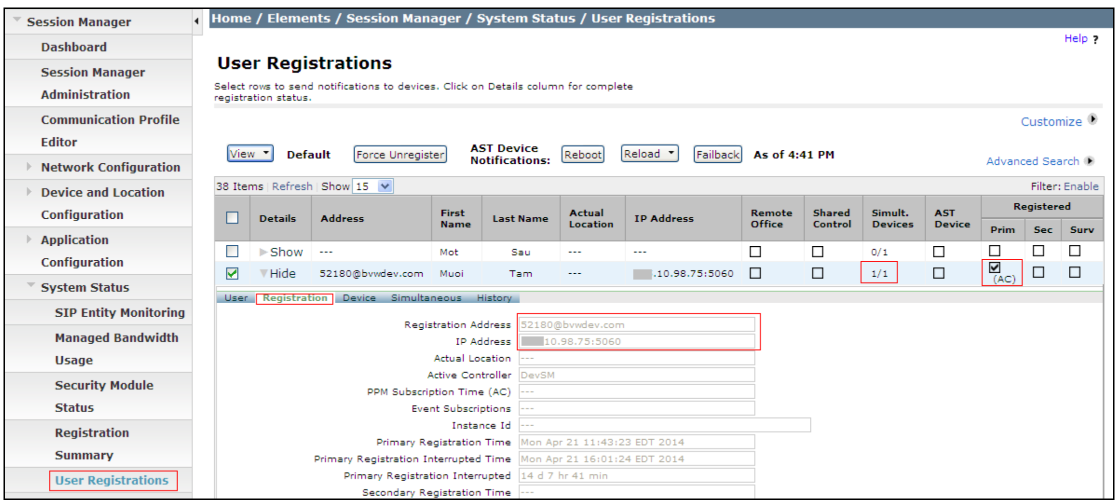
1. On Communication Manager, verify the status of the administered CTI link by using the "status aesvcs cti-link" command. Verify that the **Service State** is "established" for the configured CTI link.

```
status aesvcs cti-link

                          AE SERVICES CTI LINK STATUS

CTI   Version  Mnt   AE Services      Service       Msgs      Msgs
Link           Busy  Server           State         Sent      Rcvd

7     5        no    DevAES62        established   15        15
```

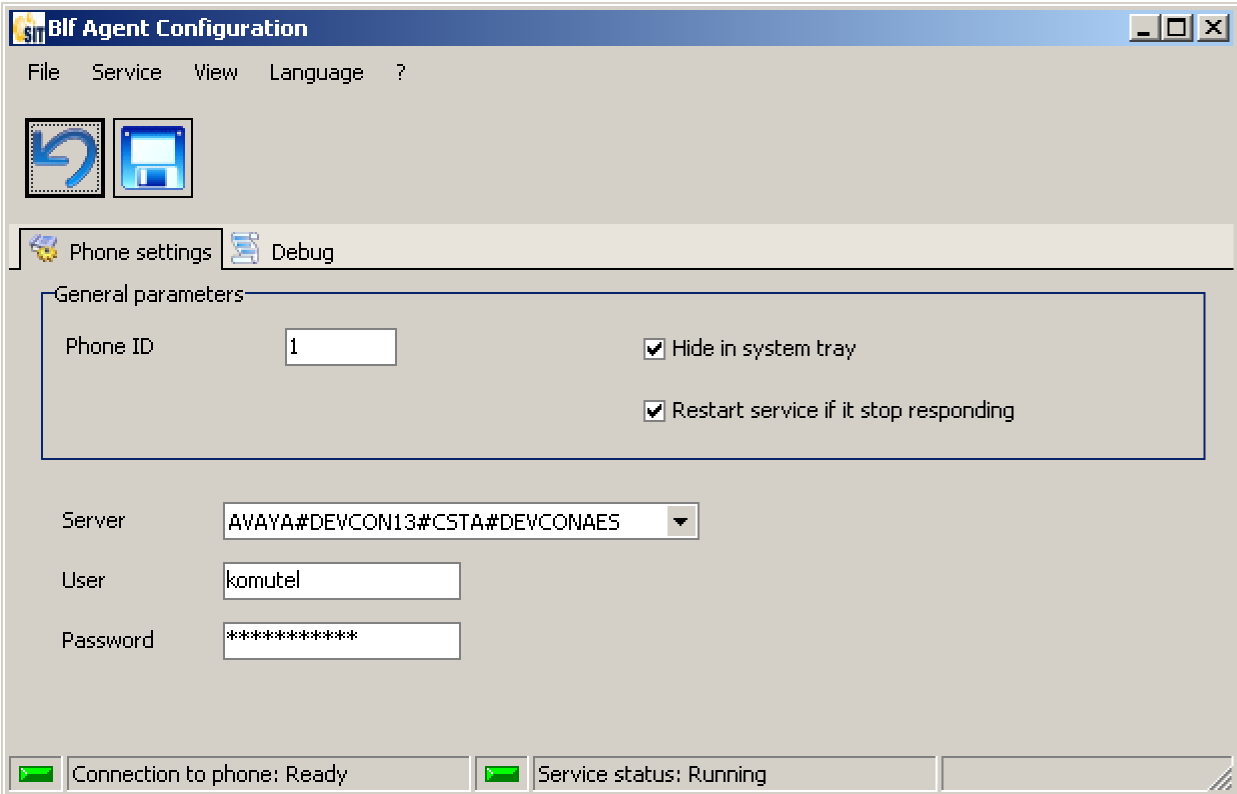
2. On Application Enablement Services, verify the status of the TSAPI link by selecting **Status→Status and Control→TSAPI Service Summary** from the left pane. The **TSAPI Link Details** screen is displayed. Verify the **Status** is "Talking" for the TSAPI link.



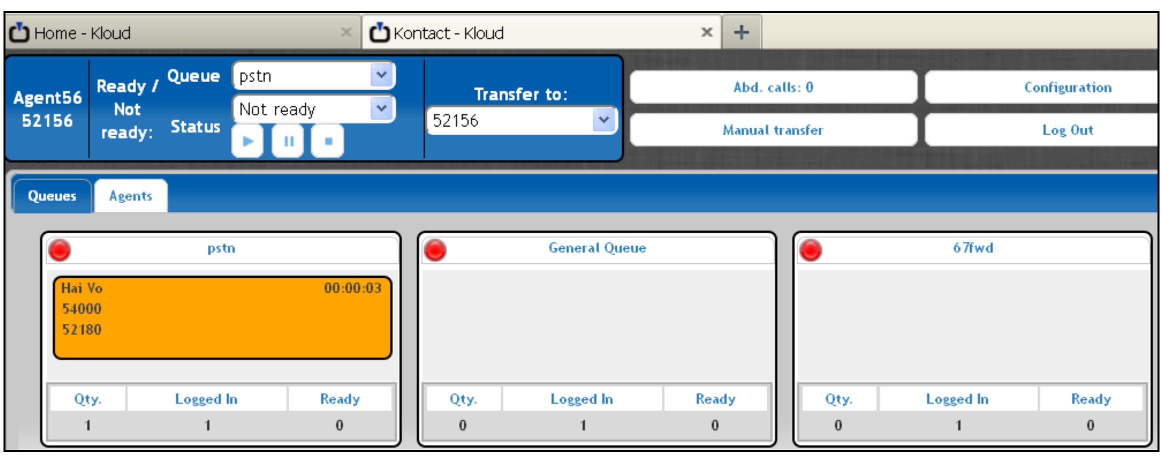
3. Verify that the Kontact has successfully registered with Session Manager. The System Manager screen below shows the successful registration of Kontact SIP user.



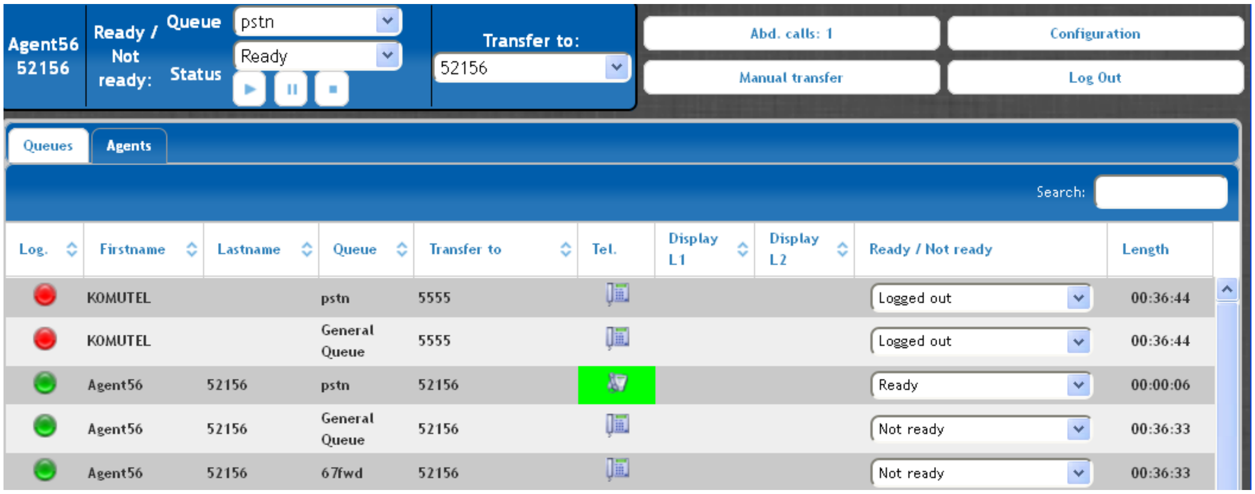
4. Verify that the BLF Agent service is running. Open the **Komutel BLF Agent Configuration** application and check the status bar at the bottom of the window. The **Connection to phone** status should display *Ready* and the **Service status** should display *Running* as shown below.



5. Place a call to a queue that has no available agents and verify that the call is successfully queued as shown below in the **Queues** tab in **Komutel Kloud**. In this example, a call is queued for the **pstn** queue and the caller ID is also displayed. The queued call can also be transferred to an agent by specifying the agent's number in the **Transfer to** field and clicking on the queued call in orange.



6. Place a call to a queue with an available agent and verify that when the call is answered by the agent, the agent's status is updated as shown below in the **Agents** tab in **Komutel Kloud**. In this example, the call was answered by *Agent56* and the green highlight indicates that the agent is on a call. In addition, the caller ID is also displayed.



10. Conclusion

These Application Notes have described the administration steps required to integrate the Komutel Kontact with Avaya Aura® system. Kontact was able to successfully register as SIP User on Session Manager and incoming calls were routed to an available agent, queued, or transferred to an agent as expected. All test cases passed with observations noted in **Section** Error! Reference source not found..

11. References

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

[1] *Administering Avaya Aura® Communication Manager*, May 2013, Release 6.3, Document Number 03-300509.

[2] *Administering Avaya Aura® Session Manager*, June 2013, Release 6.3

[3] *Administering Avaya Aura® System Manager*, May 2013, Release 6.3.

[4] *Implementing Avaya Aura® Application Enablement Services for a Bundled Server Upgrade*, August 2010, Issue 9, Release 5.2, Document Number 02-300356.

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.