

DevConnect Program

Application Notes for Cogito Emotion AI with Avaya Aura® Application Enablement Services 10.1 and Avaya Session Border Controller 10.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Cogito Emotion AI to interoperate with Avaya Aura® Application Enablement Services and Avaya Session Border Controller using TLS and SRTP. Cogito Emotion AI is a cloud-based SIPREC call recording and analysis solution.

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 configuration steps required for Cogito Emotion AI to interoperate with Avaya Aura® Application Enablement Services and Avaya Session Border Controller (Avaya SBC) using TLS and SRTP. Cogito Emotion AI is a cloud-based SIPREC call recording and analysis solution.

In the compliance testing, Cogito Emotion AI used the Java Telephony API (JTAPI) client to access the Telephony Services Application Program Interface (TSAPI) from Avaya Aura® Application Enablement Services (AES) to monitor call center agents on Avaya Aura® Communication Manager. The SIPREC call recording capabilities of the Avaya SBC are used to capture the media associated with the monitored agents as they are on call with a PSTN customer through SIP trunking.

2. General Test Approach and Test Results

The general test approach was to verify the features and serviceability of the Cogito Emotion AI successfully integrate with Application Enablement Services using JTAPI and utilize SIPREC in the Avaya SBC for call recording.

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. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

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

For the testing associated with this Application Note, the interface between Avaya systems and the Cogito recording server utilizes the secure SIP Transport Layer Security (TLS) and secure RTP.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the third-party solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of

the third-party solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

Readers should be aware that network behaviors (e.g., jitter, packet loss, delay, speed, etc.) can vary significantly from one location to another and may affect the reliability or performance of the overall solution. Different network elements (e.g., session border controllers, soft switches, firewalls, NAT appliances, etc.) can also affect how the solution performs.

If a customer is considering implementation of this solution in a cloud environment, the customer should evaluate and discuss the network characteristics with their cloud service provider and network organizations and evaluate if the solution is viable to be deployed in the cloud.

The network characteristics required to support this solution are outside the scope of these Application Notes. Readers should consult the appropriate Avaya and third-party documentation for the product network requirements. Avaya makes no guarantee that this solution will work in all potential deployment configurations.

2.1. Interoperability Compliance Testing

To verify the monitor events and call recording on the agent devices, the following features and functionalities were exercised during the compliance test.

- Verifying connection of Cogito JTAPI client to Application Enablement Services using TSAPI services.
- Response to SIP OPTIONS messages.
- Caller ID Presentation.
- Call recording of inbound calls from SIP trunk to elite call center queue and then answered by an available agent.
- Call recording of inbound calls from SIP trunk directly to agent.
- Call recording of outbound calls from agents over SIP trunk.
- Call recording of inbound call from SIP trunk to SIP agent remote worker.
- Call recording of mute, hold and transfer calls on the agent endpoints.
- Serviceability testing The behavior of Cogito recording server under different failure conditions.

Note: A SIP Agent remote worker was tested as part of this solution. The configuration necessary to support the SIP remote worker is beyond the scope of these Application Notes and is not included in this document.

2.2. Test Results

The compliance test of the Cogito recording solution was completed successfully with the exception of the observations or limitations described below.

- Current design of Cogito Emotion AI only records SIP trunk calls from/to monitored agent endpoints. The SIP trunk calls from to regular endpoints were not recorded.
- Calls between an internal agent endpoint and a SIP agent remote worker agent were not recorded or not supported by Cogito.
- Cogito stops recording as the agent places a call on hold and creates a new recording as the agent resumes the call. Therefore, there is no recording during the time that the agent holds the call.
- Cogito does not record a conference call between SIP trunk and two agents.

2.3. Support

Technical support on Cogito Emotion AI can be obtained through the following:

• Phone: (617) 580-3101

• Email: avayasupport@cogitocorp.com

3. Reference Configuration

The **Figure 1** below illustrates the test configuration diagram for the compliance test. In the test diagram, the SIP trunk was configured in the Avaya SBC to connect to service provider for calls from PSTN to enterprise and vice versa. The Cogito Emotion AI solution established a connection to Application Enablement Services using TSAPI services with a JTAPI client and receives SIP messages and audio call recording from Avaya SBC.

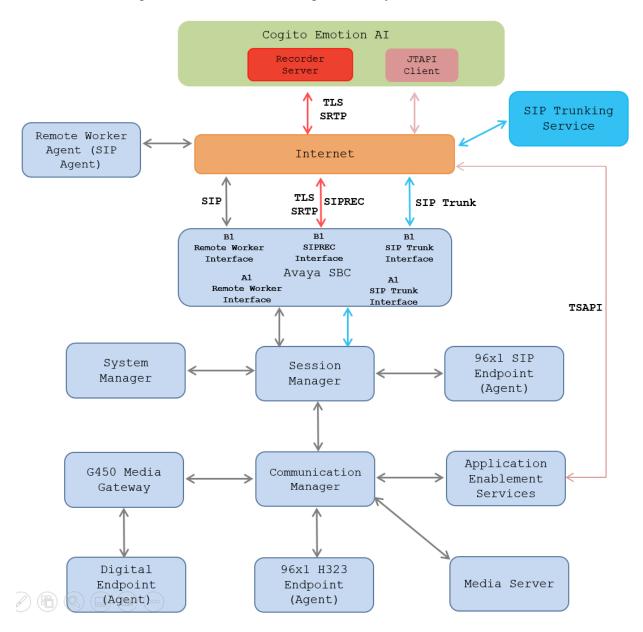


Figure 1: Test Configuration Diagram for Cogito Emotion AI

The following table indicates the IP addresses that were assigned to the systems in the test configuration diagram:

Description	IP Address
System Manager	10.33.1.40
Session Manager	10.33.1.41
Communication Manager	10.33.1.43
Application Enablement Services	10.33.1.14
Session Border Controller	10.33.10.102
Media Server	10.33.1.30
G450 Media Gateway	10.33.1.8
H.323 Endpoints	192.168.11.10-12
SIP Endpoints	192.168.11.8-9
Cogito Recording Server	192.218.23.33
Cogito JTAPI Client	192.232.32.110

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	10.1.3.0
running on Virtualized Environment	CM 10.1.3.0.1.974.27893
Avaya Aura® System Manager running on	10.1.2.0
Virtualized Environment	Software No: 10.1.2.0.0715476
Avaya Aura® Session Manager running on	10.1.2.0
Virtualized Environment	10.1.2.0.1012016
Avaya Aura® Application Enablement	10.1.2.0
Services	10.1.2.0.0.12-0
Avaya Session Border Controller	10.1.2.0
	10.1.2.0-64-23285
Avaya Aura® Media Server running on	10.1.0.154
Virtualized Environment	
Avaya G450 Media Gateway	42.7.0
Avaya 96x1 IP Deskphones	6.8.5.4.10 (H.323)
Avaya J189 Deskphone	4.1.0.0.9 (SIP)
Avaya Agent for Desktop (SIP)	2.0.6.25 (SIP)
Avaya 9408 Digital Deskphone	2.0 SP8 (R19)
Cogito Emotion AI	
- Core	1.6.0
- Compute	1.2.6
- Telephony	1.105.1
- Web	1.14.1

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager.

5.1. Verify License

Log into the System Access Terminal to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command to verify that the **Computer Telephony Adjunct Links** customer option is set to "y" on **Page 4**. If this option is not set to "y", then contact the Avaya sales team or business partner for a proper license file.

```
display system-parameters customer-options
                                                            Page
                                                                   4 of 12
                              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? y
                                                                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? n
                                                            DCS (Basic)? y
         ASAI Link Core Capabilities? n
                                                      DCS Call Coverage? y
         ASAI Link Plus Capabilities? n
                                                      DCS with Rerouting? y
```

5.2. Administer 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.

add cti-link 1

CTI Link: 2

Extension: 3331

Type: ADJ-IP

COR: 1

Name: AES50

Unicode Name? n

5.3. Administer System Parameters Features

Use the "change system-parameters features" command to enable **Create Universal Call ID** (**UCID**), which is located on **Page 5**. For **UCID Network Node ID**, enter an available node ID.

```
change system-parameters features
                                                                     5 of 19
                                                              Page
                        FEATURE-RELATED SYSTEM PARAMETERS
SYSTEM PRINTER PARAMETERS
 Endpoint:
                        Lines Per Page: 60
SYSTEM-WIDE PARAMETERS
                                     Switch Name:
           Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? n
                              COR to Use for DPT: station
               EC500 Routing in Survivable Mode: dpt-then-ec500
MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? n
                                         MCT Voice Recorder Trunk Group:
     Delay Sending RELease (seconds): 0
SEND ALL CALLS OPTIONS
     Send All Calls Applies to: station Auto Inspect on Send All Calls? n
              Preserve previous AUX Work button states after deactivation? n
UNIVERSAL CALL ID
    Create Universal Call ID (UCID)? y
                                           UCID Network Node ID: 1
    Copy UCID for Station Conference/Transfer? y
```

Navigate to **Page 13**, and enable **Send UCID to ASAI**. This parameter allows for the universal call ID to be sent to ASAI and it will be used by the TJAPI application.

```
Page 13 of 20
change system-parameters features
                       FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER MISCELLANEOUS
          Callr-info Display Timer (sec): 10
                         Clear Callr-info: next-call
       Allow Ringer-off with Auto-Answer? n
   Reporting for PC Non-Predictive Calls? n
           Agent/Caller Disconnect Tones? n
         Interruptible Aux Notification Timer (sec): 3
             Zip Tone Burst for Callmaster Endpoints: double
 ASAI
                   Copy ASAI UUI During Conference/Transfer? y
               Call Classification After Answer Supervision? y
                                          Send UCID to ASAI? y
                 For ASAI Send DTMF Tone to Call Originator? y
         Send Connect Event to ASAI For Announcement Answer? n
 Prefer H.323 Over SIP For Dual-Reg Station 3PCC Make Call? n
```

5.4. Administer AE Services

To administer the transport link to AES, use the command "change ip-services". On **Page 1**, add an entry with the following values. Service Type should be set to **AESVCS**, enter "y" in the **Enabled**, "procr" in the **Local Node** and 8765 in the **Local Port**.

change ip-s	services					Page	e 1 of	4
			IP	SERVICES				
Service	Enabled	Local		Local	Remote	Remo	ote	
Type		Node		Port	Node	Port	5	
AESVCS	У	procr		8765				

Go to **Page 4**, enter the following values. **AE Services Server** should be the AES host name, enter a password in the **Password** field and select "y" in the **Enabled** field.

Note: The password entered for **Password** field must match the password on the AES server in the Switch Connection in **Section 6.3**. The **AE Services Server** should match the host name of the AES server. To obtain the host name of AES server, use the command "uname –n" in the AES server Linux command prompt.

change ip-ser	vices			Page 4 of	4
AE Services Administration					
Server ID	AE Services Server	Password	Enabled	Status	
1:	aes10	*	У	in use	
2:	aes50	*	У	in use	

5.5. Administer Hunt Group

This section provides the Hunt Group configuration for the call center agents. Agents will log into the 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. This hunt group will be specified in the **Agent LoginIDs** configured in **Section 5.8**.

```
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
                                     Local Agent Preference? n
          Security Code:
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.

```
add hunt-group 1

Skill? y

AAS? n

Measured: both
Supervisor Extension:

Controlling Adjunct: none

VuStats Objective:

Multiple Call Handling: none

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

5.6. Administer Vector

Use the command "change vector n" while "n" is the vector number from 1-8000. The example of the vector 1 with a basic scripting to route calls to a skill group is shown below. Vector 1 is used for the configuration of the VDN in the next step.

```
Change vector 1

CALL VECTOR

Number: 1

Name: Contact Center

Multimedia? 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
```

5.7. 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 5.6** above and keep other fields at their default values.

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

5.8. 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
                                     LWC Log External Calls? n
          Security Code: 1234
          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
                                                   Local Call Preference? n
Call Handling Preference: skill-level
   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:
```

5.9. Configure SIP Trunk

Use the command "change trunk-group n" where "n" is number of the trunk group that is previously configured to connect to Avaya SBC. Go to **Page 3**, select "shared" in the **UUI Treatment** field. With the selection of shared UUI, the **Send UCID** field is present and select "y" in this field.

```
change trunk-group 3
                                                                     3 of
                                                              Page
                                                                            5
TRUNK FEATURES
         ACA Assignment? n
                                      Measured: none
                                                          Maintenance Tests? y
  Suppress # Outpulsing? n Numbering Format: private
                                                UUI Treatment: shared
                                              Maximum Size of UUI Contents: 128
                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y
                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
               Send UCID? y
Show ANSWERED BY on Display? y
```

On **Page 4**, enter the value "1" in the **Universal Call ID** (**UCID**) field and keep other fields at default values.

```
Change trunk-group 3

SHARED UUI FEATURE PRIORITIES

ASAI:

Universal Call ID (UCID): 1

MULTI SITE ROUTING (MSR)

In-VDN Time: 3

VDN Name: 4

Collected Digits: 5
Other LAI Information: 6
Held Call UCID: 7
ECD UUI: 8
```

6. Configure Avaya Aura® Application Enablement Services

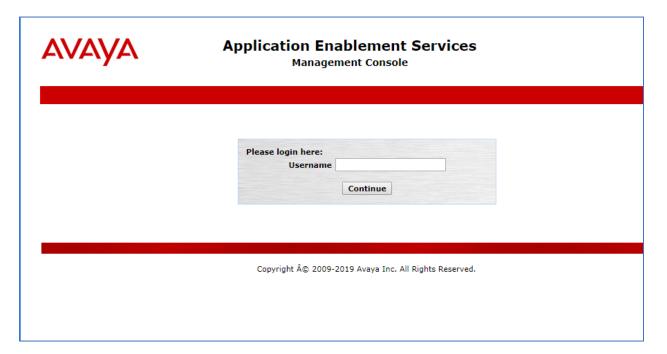
This section provides the procedures for configuring Application Enablement Services. The procedures include the following areas:

- Launch AE web interface
- Verify license
- Administer Switch Connection
- Administer TSAPI link
- Administer CTI user
- Administer Security Database
- Administer ports
- Restart services

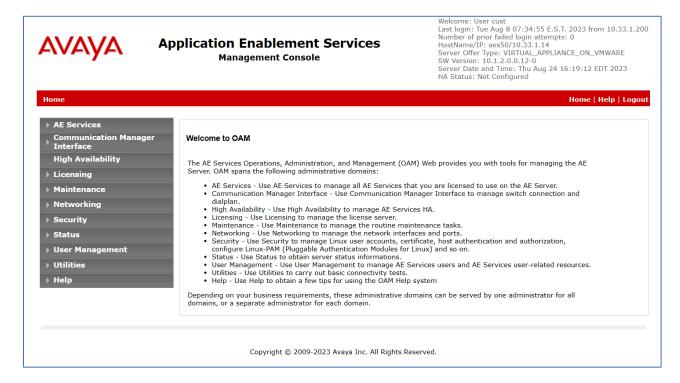
6.1. Launch AE web Interface

Access the AE web-based interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the Application Enablement Services server.

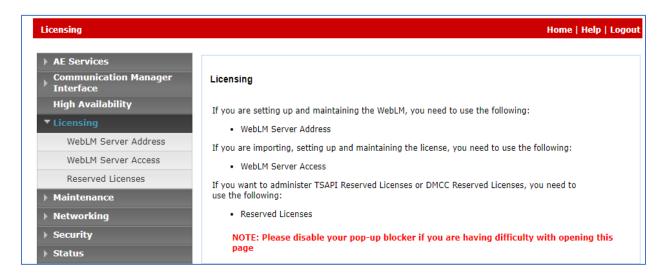
The **Please login here** screen is displayed. Log in using the appropriate credentials.



The Welcome to OAM screen is displayed next.

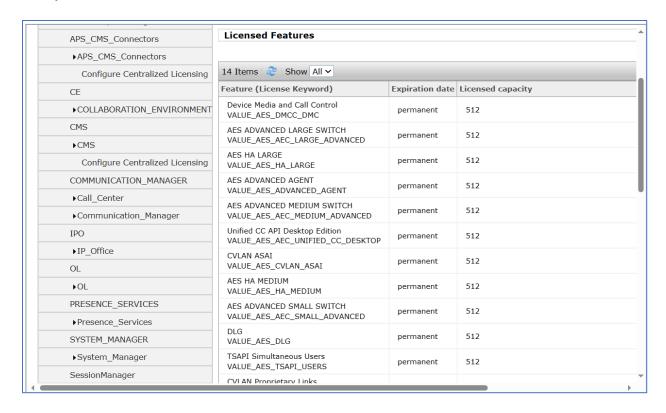


6.2. Verify License



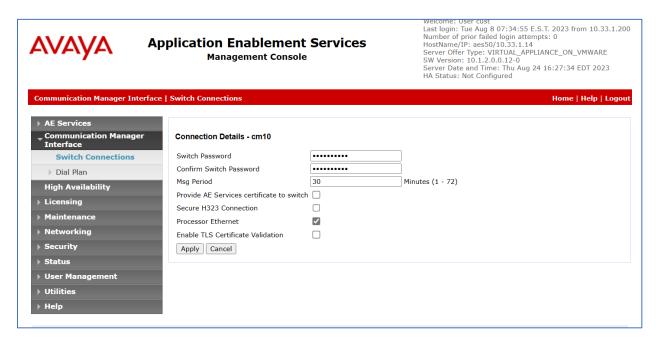
Select Licensed products \rightarrow APPL_ENAB \rightarrow Application_Enablement in the left pane, to display the Application Enablement (CTI) screen in the right pane.

Verify that there are sufficient licenses for **TSAPI Simultaneous Users**, as shown below.

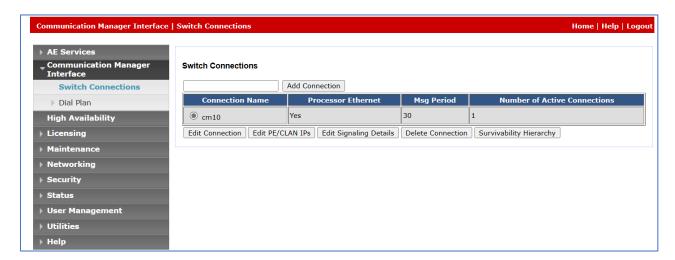


6.3. Administer Switch Connection

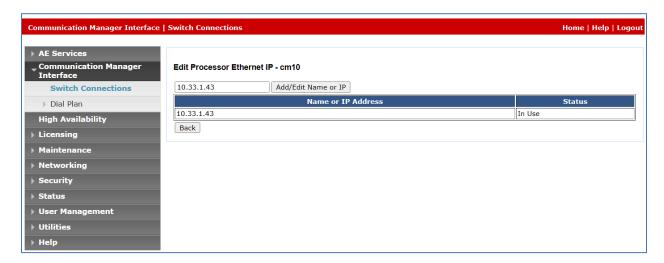
Select Communication Manager Interface → Switch Connections from the left pane of the Management Console, enter a name in the Switch Connection box and click the Add button (not shown). Enter the password as configured in Section 5.4 in the Switch Password and Confirm Switch Password and check on Processor Ethernet field if the Processor Ethernet is used in Communication Manager. Click the Apply button to save the configuration.



Select the cm10 switch connection has been added above and selects Edit PE/CLAN IPs to add the IP address of the switch connection.



Enter the IP address of the Processor Ethernet of Communication Manager in the box and click the **Add/Edit Name of IP** button to add the IP.

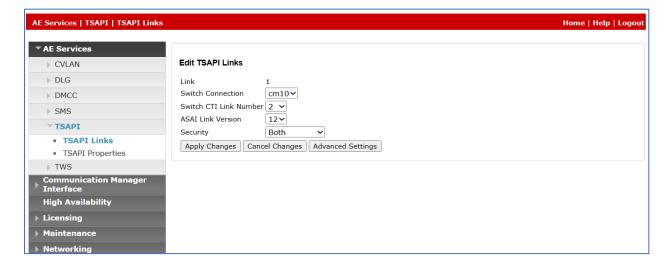


6.4. Administer TSAPI Link

Select **AE Services** → **TSAPI** → **TSAPI Links** from the left pane of the **Management Console**, to administer a TSAPI link. The **TSAPI Links** screen is displayed, as shown below. Click **Add Link**.

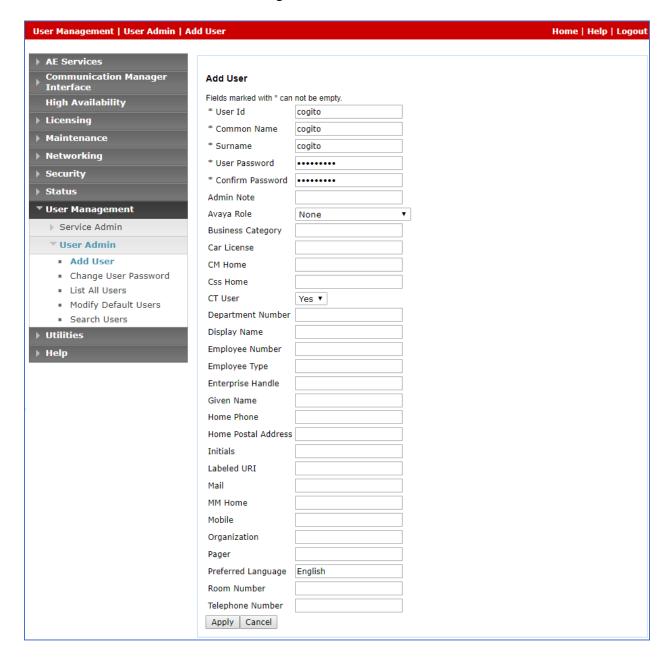


The **Add TSAPI Links** screen is displayed in the right side. The **Link** field is only local to the Application Enablement Services server and may be set to any available number. For **Switch Connection**, select the relevant switch connection from the drop-down list. In this case, the existing switch connection "**cm10**" which is added in the step above. For **Switch CTI Link Number**, select the CTI link number 2 from **Section 5.2**, select **Both** in the **Security** dropdown menu to support both unencrypted and encrypted TSAPI link. Retain the default values in the remaining fields.



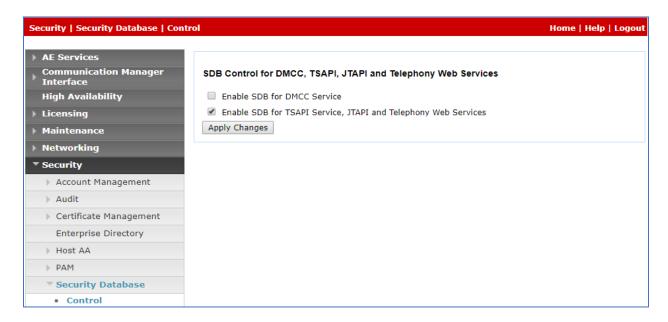
6.5. Administer CTI User

Select User Management \rightarrow User Admin \rightarrow Add User from the left pane, to display the Add User screen in the right pane. Enter the 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.

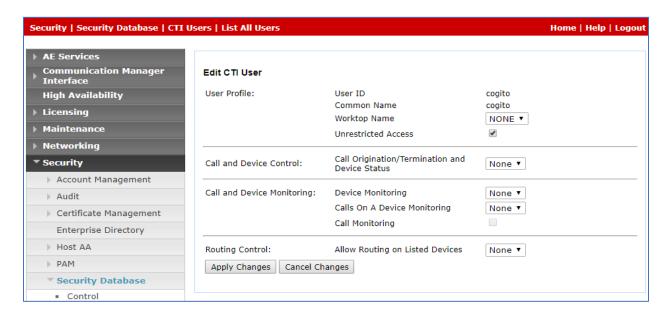


6.6. Configure Security Database

Select Security → Security Database → Control from the left pane, to display the SDB Control for DMCC, TSAPI, JTAPI and Telephony Web Services screen in the right pane. Leave it as default as checked on Enable SDB for TSAPI Service, JTAPI and Telephony Web Services.

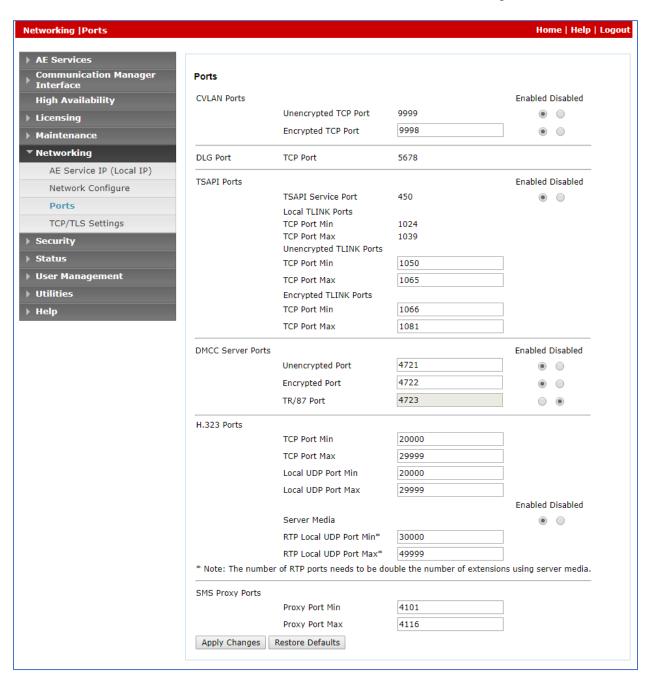


Select Security \rightarrow Security Database \rightarrow CTI Users \rightarrow List All Users and select the "cogito" CTI user which is created in Section 6.5 and select Edit button (not shown). In the Edit CTI User, select the check box Unrestricted Access and click Apply Changes to save the configuration.



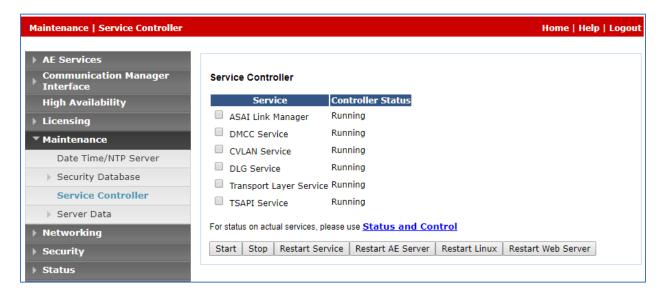
6.7. Administer Ports

Select **Networking** \rightarrow **Ports** from the left pane, to display the **Ports** screen in the right pane. In the **TSAPI Ports** section, select the radio button for **TSAPI Service Port 450** under the **Enabled** column, as shown below. Retain the default values in the remaining fields.



6.8. Restart Services

Select Maintenance \rightarrow Service Controller from the left pane, to display the Service Controller screen in the right pane. Click Restart AE Server.



7. Configure Avaya Session Border Controller

This section describes the configuration of the Avaya SBC. It is assumed that the initial installation of the Avaya SBC has been completed, including the assignment of a management IP address. The management interface **must** be provisioned on a different subnet than either the Avaya SBC private or public network interfaces (e.g., A1 and B1).

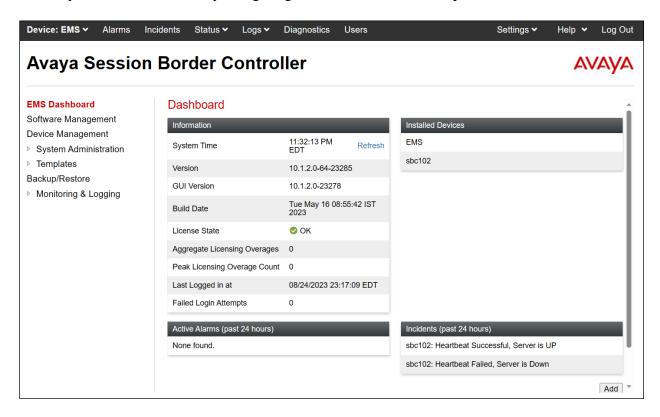
On all screens described in this section, it is assumed that parameters are left at their default values unless specified otherwise.

7.1. Access the Management Interface

Use a web browser to access the web interface by entering the URL https://<ip-addr>, where <ip-addr> is the management IP address assigned during installation. The Avaya SBC login page will appear as shown below. Log in with appropriate credentials.

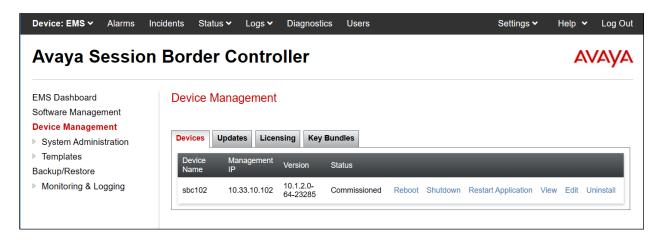
AVAYA	Log In Username: ucsec Continue		
	WELCOME TO AVAYA SBC		
Avaya Session Border Controller	Unauthorized access to this machine is prohibited. This system is for the use authorized users only. Usage of this system may be monitored and recorded by system personnel.		
	Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide the evidence from such monitoring to law enforcement officials.		
	© 2011 - 2023 Avaya Inc. All rights reserved.		

After logging in, the Dashboard screen will appear as shown below. All configuration screens of the Avaya SBC are accessed by navigating the menu tree in the left pane.

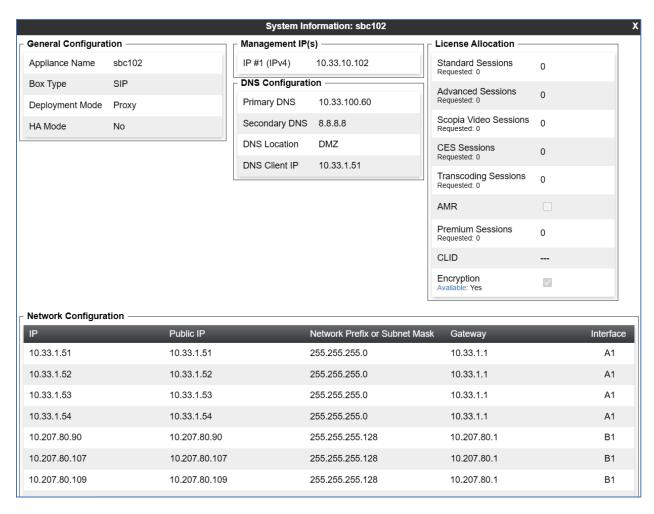


7.2. Verify Network Configuration and Enable Interfaces

To view the network information provided during installation, navigate to **Device Management**. In the right pane, click **View**.



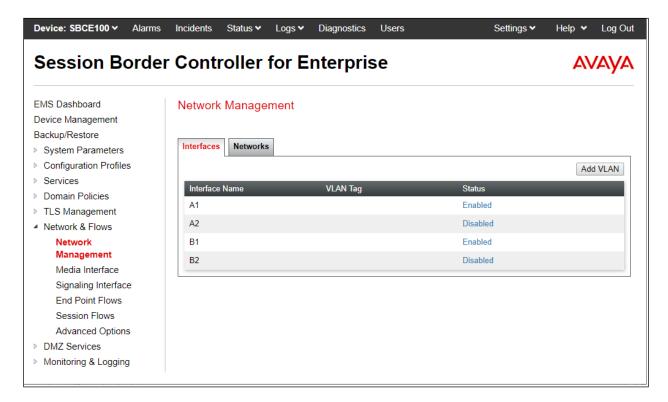
A System Information page will appear showing the information provided during installation. The **Appliance Name** field is the name of the device (**sbc102**). This name will be referenced in other configuration screens. Interface **A1** and **B1** represent the private and public interfaces of the Avaya SBC respectively. Each of these interfaces must be enabled after installation.



From the right top corner of the window, select **Device** dropdown menu and select the SBC system, e.g., **sbc102**, the administration is displayed in the right pane.



To enable the interfaces, first navigate to **Network & Flows** → **Network Management** in the left pane. In the right pane, click on the **Interfaces** tab. Verify the **Status** is **Enabled** for both the **A1** and **B1** interfaces. If not, click the status **Enabled/Disabled** to toggle the state of the interface.



7.3. TLS Management

Note – Testing was done with System Manager signed identity certificates for Cogito recording server and Avaya SBC. The procedure to create and obtain these certificates is outside the scope of these Application Notes.

In the reference configuration, TLS transport is used for the communication between Session Manager and Avaya SBC and between Avaya SBC and Cogito recording server. The following procedures show how to create the client and server profiles.

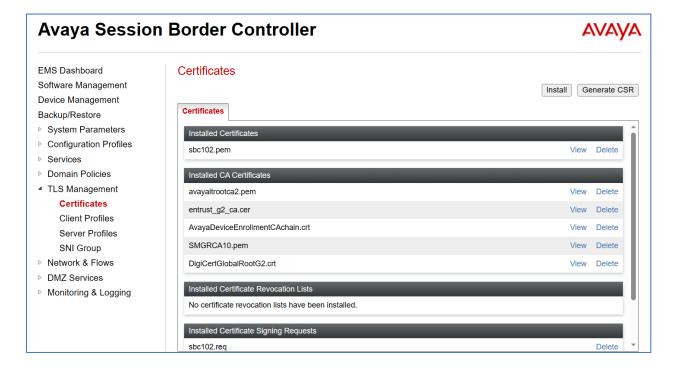
7.3.1. Verify TLS Certificates – Avaya Session Border Controller for Enterprise

To access the SBC configuration menus, select the SBC device from the top navigation menu.



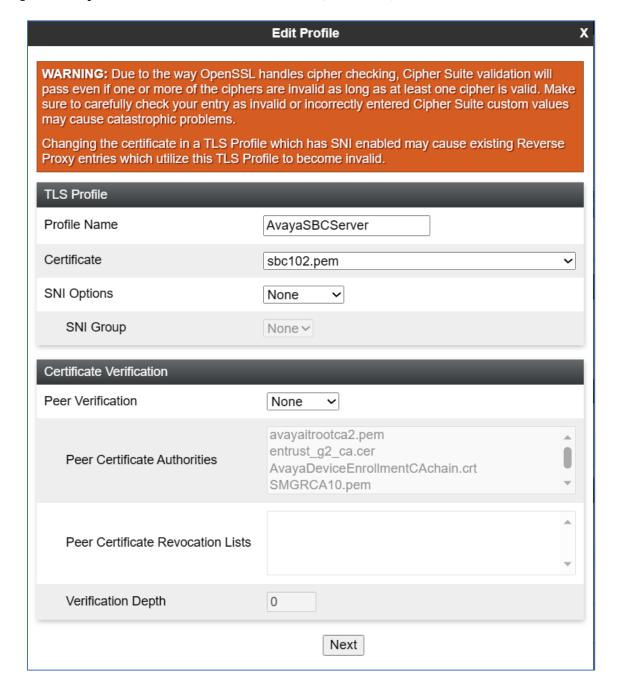
Step 1 - Select **TLS Management** → **Certificates** from the left-hand menu. Verify the following:

- System Manager CA certificate is present in the **Installed CA Certificates** area.
- System Manager CA signed identity certificate is present in the **Installed Certificates** area.
- Private key associated with the identity certificate is present in the **Installed Keys** area.

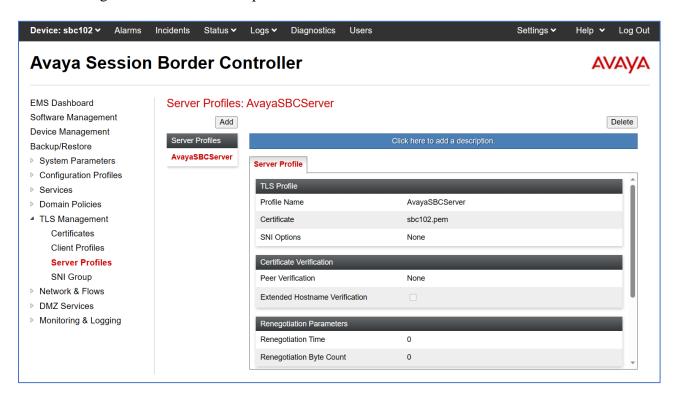


7.3.2. Server Profiles

- **Step 1** Select **TLS Management** → **Server Profiles** and click on **Add**. Enter the following:
 - **Profile Name:** enter a descriptive name. (e.g., **AvayaSBCServer**).
 - **Certificate:** select the identity certificate, e.g., **sbc102.pem**, from pull down menu.
 - Peer Verification = None.
 - Click Next.
- **Step 2** Accept default values for the next screen (not shown) and click **Finish**.



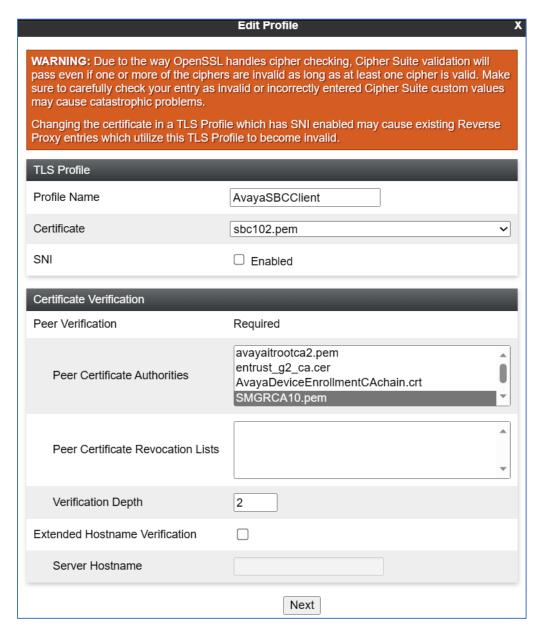
The following screen shows the completed TLS **Server Profile** form:



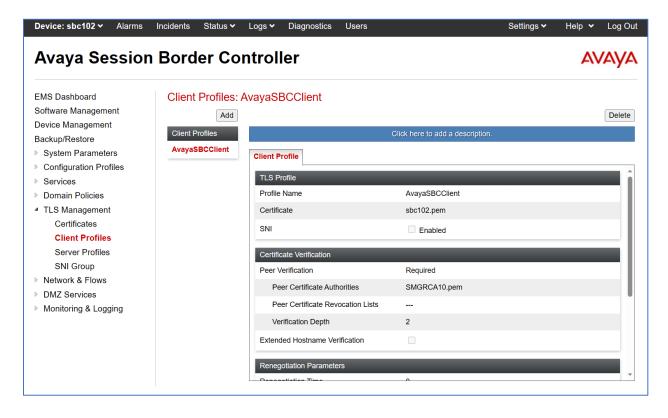
7.3.3. Client Profiles

Step 1 - Select **TLS Management** → **Server Profiles** and click on **Add** (not shown). Enter the following:

- **Profile Name:** enter a descriptive name (e.g., **AvayaSBCClient**)
- **Certificate:** select the identity certificate, e.g., **sbc102.pem**, from pull down menu.
- Peer Verification = Required.
- **Peer Certificate Authorities**: select the CA certificate used to verify the certificate received from Session Manager, e.g., **SMGRCA10.pem**.
- Enter 1 under **Verification Depth**. Click **Next**.
- **Step 2** Accept default values for the next screen (not shown) and click **Finish**.



The following screen shows the completed TLS Client Profile form:

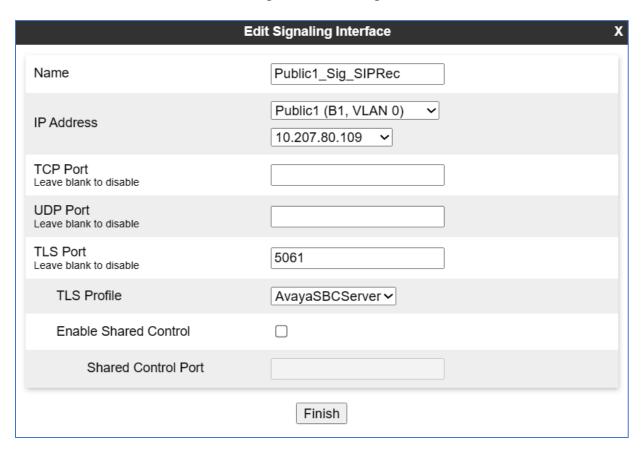


7.4. Signaling Interface

A signaling interface defines an IP address, protocols and listen ports that the Avaya SBC can use for signaling. Create a signaling interface for both the internal and external sides of the Avaya SBC.

To create a new interface, navigate to **Network & Flows** → **Signaling Interface** in the left pane. In the center pane, select the Avaya SBC device (**sbc102**) to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by one or more pop-up windows in which the interface parameters can be configured. Once complete, the settings are shown in the far-right pane.

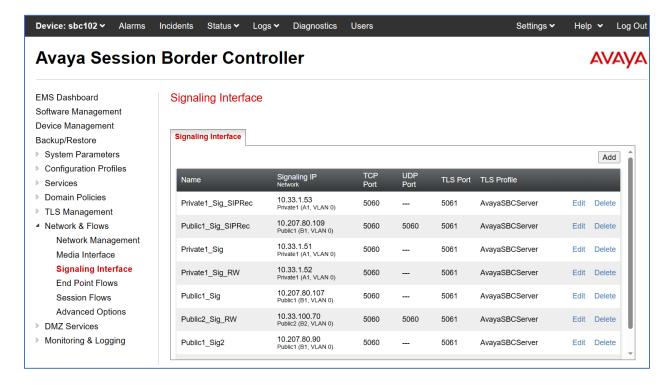
- Name: enter a descriptive name.
- For the internal interface, set the **IP Address** to the IP address associated with the private interface (A1) defined in **Section 7.2**. For the external interface, set the **IP Address** to the IP address associated with the public interface (B1) defined in **Section 7.2**.
- In the **UDP Port**, **TCP Port** and **TLS Port** fields, enter the port Avaya SBC will listen on for each transport protocol. For the internal interface, the Avaya SBC was configured to listen for TLS on port 5061. For the external interface, the Avaya SBC was configured to listen for TLS on port 5061.
- TLS Profile: select the server TLS profile in the dropdown menu.



For the testing, the list of signaling interfaces in the table below created:

Name	IP address	Description
Private1_Sig	10.33.1.51	The private signaling interface connects to
		Session Manager
Public1_Sig	10.50.207.107	The public signaling interface connects to
		Service Provider
Private1_Sig_RW	10.33.1.52	The private signaling interface for SIP remote
		worker connects to Session Manager
Public1_Sig_RW	10.50.207.108	The public signaling interface for SIP remote
		worker connects to SIP remote worker endpoint
Public1_Sig_SIPREC	10.50.207.109	The public signaling interface connects to Cogito
		recording server

The screenshot below shows the list of signaling interfaces with TLS Server Profiles assigned that were used during the compliance test.

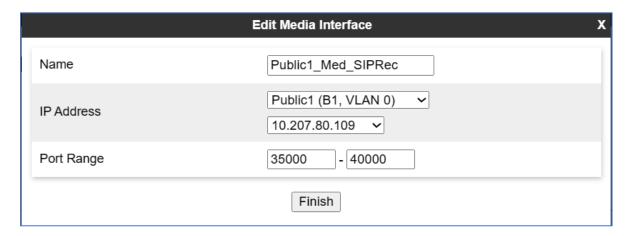


7.5. Media Interface

A media interface defines an IP address and port range for transmitting media. Create a media interface for both the internal and external sides of the Avaya SBC.

To create a new interface, navigate to **Network &Flows** → **Media Interface** in the left pane. In the center pane, select the Avaya SBC device (**sbc102**) to be managed. In the right pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new interface, followed by one or more pop-up windows in which the interface parameters can be configured. Once complete, the settings are shown in the far-right pane.

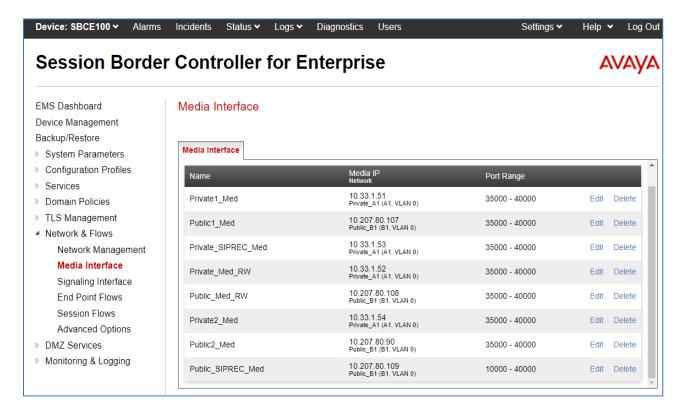
- Name: enter a descriptive name.
- For the internal media interface, set the **IP Address** to the IP address associated with the private interface (A1) defined in **Section 7.2**. For the external interface, set the **IP Address** to the IP address associated with the public interface (B1) defined in **Section 7.2**.
- Set **Port Range** to a range of ports acceptable to both the Avaya SBC and the far-end. For the testing, the default port range was used for the SIPREC public media interface.



For the testing, list of media interfaces was added and shown in the table below.

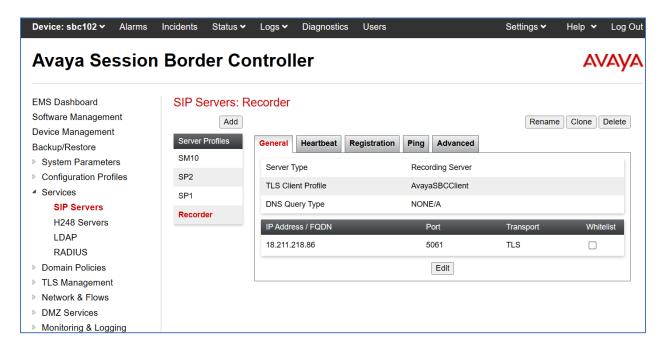
Name	IP address	Description
Private1_Med	10.33.1.51	The private media interface connects to enterprise
		endpoints such as media gateway and agent
		endpoints
Public1_Med	10.207.80.107	The public media interface connects to media
		gateway of Service Provider
Private_Med_RW	10.33.1.52	The private media interface for SIP remote worker
		connects to enterprise endpoints
Public1_Med_RW	10.207.80.108	The public media interface for SIP remote worker
		connects to SIP remote worker endpoint
Public1_SIPREC_Med	10.207.80.109	The public media interface for SIPREC sends
		media to Cogito SIP recording server

The screenshot below shows the list of media interface used for the testing.



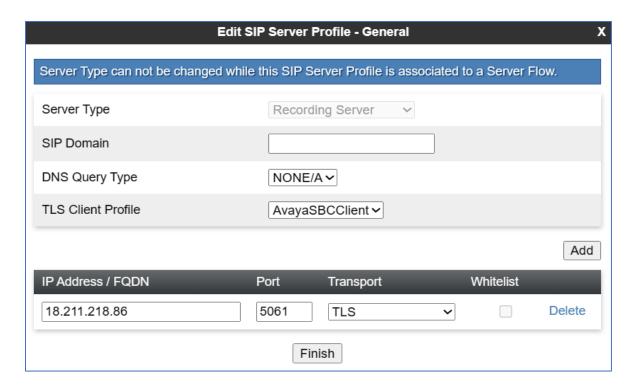
7.6. Server Configuration

A server configuration profile defines the attributes of the physical server. To create a new profile, navigate to **Services** \rightarrow **SIP Servers** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by one or more pop-up windows in which the profile parameters can be configured.



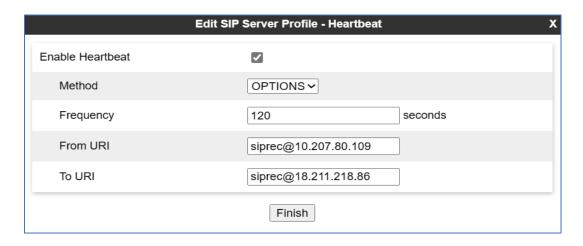
The screenshot shows the **Edit SIP Server Profile** - **General** tab parameters as follow.

- Set Server Type to Recording Server.
- Leave blank for SIP Domain and DNS Query.
- Set TLS Client Profile to the TLS profile for client as defined in Section 7.3.3.
- Enter a valid combination of **IP Address / FQDN**, **Port** and **Transport** that the Cogito recording server will use to listen for SIP requests. The standard SIP TLS port is 5061.

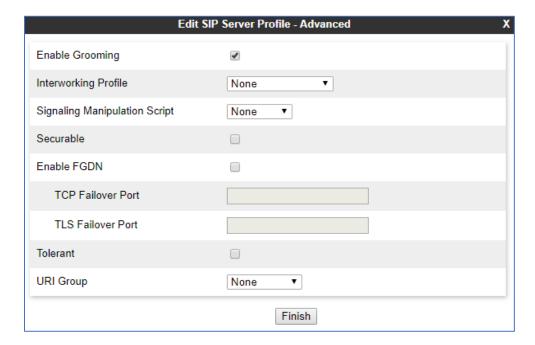


In the **Heartbeat** tab, enter following parameters as shown in the screenshot below.

- Enable Heartbeat: checked.
- Method: select OPTIONS in the dropdown menu.
- **Frequency**: enter an interval for the Avaya SBC sending out OPTIONS to the Cogito recording server.
- **From URI**: enter the uri format as user@domain or user@ipaddress. In the testing, the public IP for SIPREC was used in "**From**" header in OPTIONS message sent to Cogito.
- **To URI**: enter the uri format as user@ipaddress with the IP address of the Cogito recording server.



In the Advanced tab, check on the Enable Grooming checkbox and keep other fields as default.



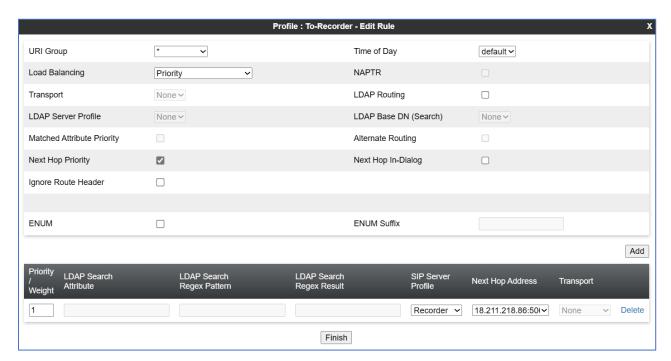
7.7. Routing Configuration

A routing profile defines where traffic will be directed based on the contents of the Request-URI. To create a new profile, navigate to **Configuration Profiles** \rightarrow **Routing** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new profile, followed by one or more pop-up windows in which the profile parameters can be configured.

For the compliance test, routing profile **To-Recorder** was created for the Cogito recording server. The screenshot bellows shows the parameters for the routing profile to Cogito.

- Set the **URI Group** to the wild card "*" to match on any URI.
- Set **Load Balancing** to **Priority** from the pull-down menu.
- Click **Add** to enter the following for the Next Hop Address:
 - o For **SIP Server Profile**, select the SIP server profiles **Recorder** (**Section 7.6**) from the pull-down menu. The **Next Hop Address** will be filled-in automatically.
- Set Priority/Weight to "1".
- Set **SIP Server Profile** to the Cogito SIP server.

Keep other parameters as default and click **Finish**.



7.8. Media Rules

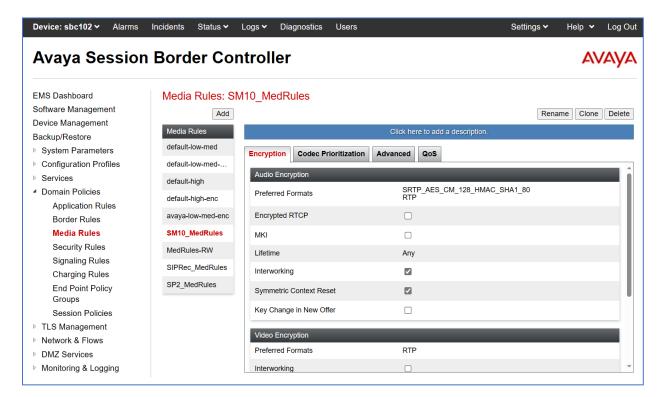
A media rule defines the processing to be applied to the selected media. For the compliance test, media rules were created for Session Manager, Service Provider and Cogito recording server to use SRTP.

To define the Media Rule for Session Manager, Service Provider, and the recording server, navigate to **Domain Policies** → **Media Rules** in the main menu on the left-hand side. Click on **Add** and enter details in the Media Rule pop-up box (not shown)

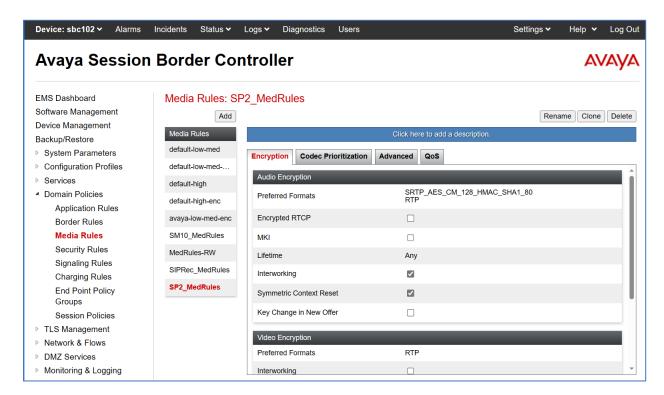
- In the **Rule Name** field enter a descriptive name such as **SM10_MedRules**.
- Set Preferred Format #1 to SRTP_AES_CM_128_HMAC_SHA1_80.
- Set Preferred Format #2 to RTP.
- Uncheck **Encrypted RTCP**.

Default values were used for all other fields. Click **Finish** (not shown).

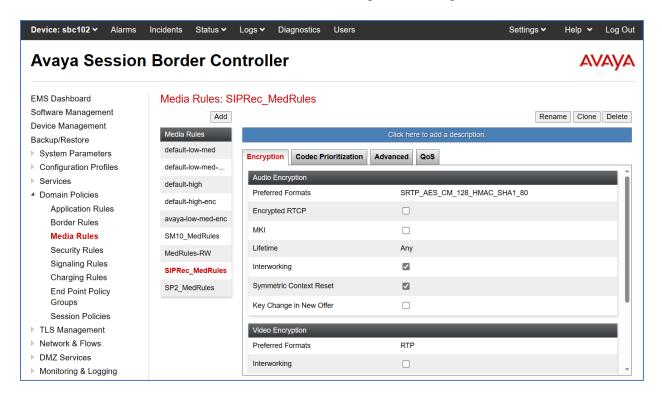
The screenshot below shows the media rule for the Session Manager.



The screenshot below shows the media rule for the Service Provider.



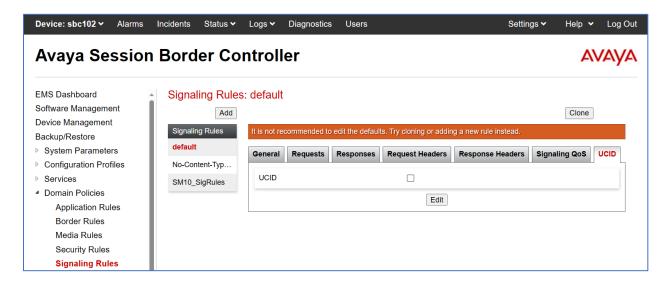
The screenshot below shows the media rule for the Cogito recording server.



7.9. Signaling Rules

A signaling rule defines the processing to be applied to the selected signaling traffic. A signaling rule is one component of the larger endpoint policy group defined in **Section 7.10**. A specific signaling rule was created for Session Manager, Service Provider, and the Cogito recording server.

To create a new rule, navigate to **Domain Policies** → **Signaling Rules** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new rule, followed by one or more pop-up windows in which the rule parameters can be configured. Note that the signaling rules can be also cloned from the default signaling rules by select the **default** in the **Signaling Rules** central column and then click on **Clone** button.



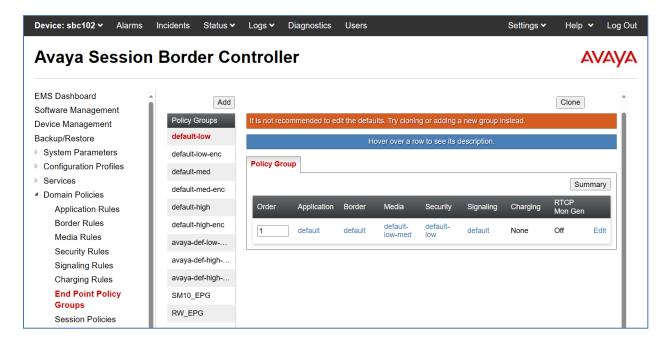
In the testing, there is one signaling rule created for **SM10_SigRules**, the default signaling rule is used for SIP trunks to service provider and the Cogito recording server. The signaling rules for Session Manager must have UCID enabled and set the ID number as the same number as the UCID configured in Communication Manager in **Section 5.9**. The screenshot below shows the signaling rules of Session Manager with UCID enabled. Note that UCID in the Service Provider and SIPREC don't need to be enabled; UCID only needs to be enabled for Session Manager Signaling.



7.10. End Point Policy Groups

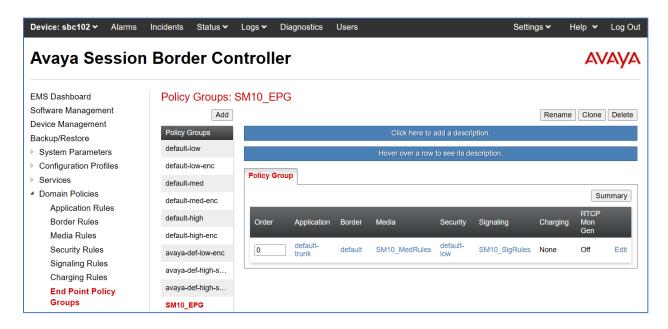
An endpoint policy group is a set of policies that will be applied to traffic between the Avaya SBC and an endpoint (connected server). Thus, an endpoint policy group must be created for Session Manager, Service Provider and the Cogito recording server.

To create a new group, navigate to **Domain Policies** → **End Point Policy Groups** in the left pane. In the center pane, select **Add**. A pop-up window (not shown) will appear requesting the name of the new group, followed by one or more of pop-up windows in which the group parameters can be configured.

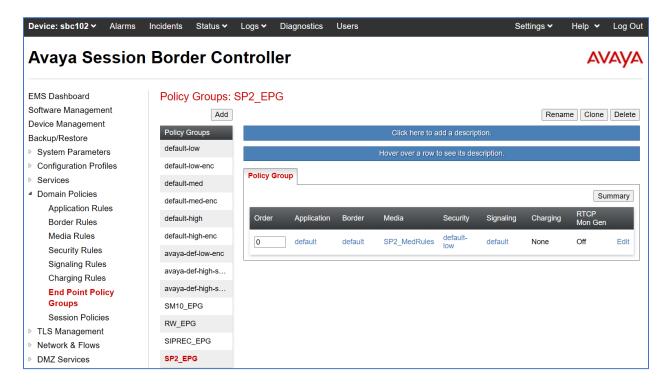


In the testing, there are three end point policy groups created: **SM_EPG** and **SP2_EPG** are previously created for the SIP trunk, and **SIPREC_EPG** is created for the Cogito recording server.

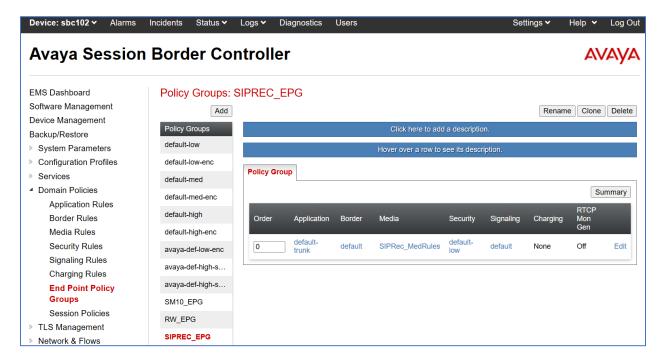
The screenshot below shows the end point policy groups used for Session Manager, SM10_EPG. The policy group uses the SM10_MedRules and SM10_SigRules created in Section 7.8 and 7.9 above.



The screenshot below shows the end point policy groups used for Service Provider, **SP2_EPG**. The policy group uses the **SP2_MedRules** created in **Section 7.8** above.

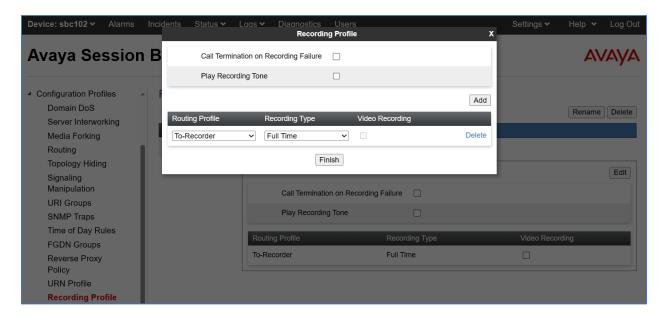


The screenshot below shows the end point policy groups used for Service Provider, **SIPREC_EPG**. The policy group uses the **SIPRec_MedRules** created in **Section 7.8** above.



7.11. Recording Profiles

To create a recording profile, from the left pane, navigate to **Configuration Profiles** → **Recording Profile**. Select **Add** button (not shown) to add a new recording profile. In the Recording Profile window, select the routing profile as created in **Section 7.7** and select "Full Time" in the **Recording Type**. Select **Finish** to complete.



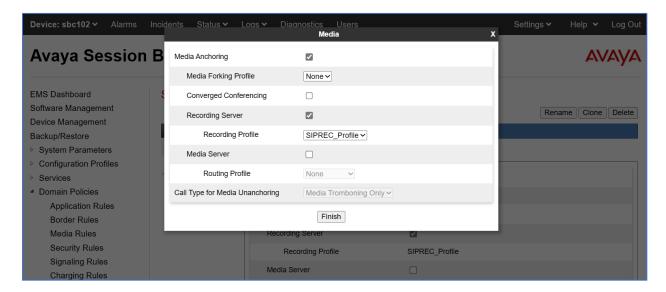
7.12. Session Policies

To create a new session policy group, navigate to **Domain Policies** → **Session Policies** in the left pane. In the center pane, select **Add** (not shown). A pop-up window will appear requesting the name of the new group, followed by one or more of pop-up windows in which the group parameters can be configured.

In the testing, the session policy **SIPREC_SessPolicies** is created with configuration as shown below.

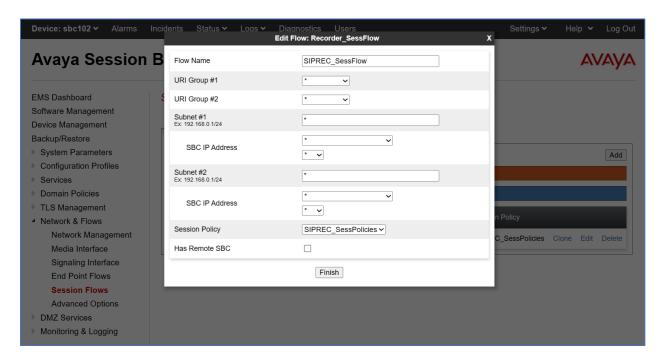
- Media Anchoring: checked.
- **Recording Server**: checked and in the **Recoding Profile**: select **SIPREC_Profiles** as created in **Section 7.11** in the dropdown menu.

Keep other fields at default and click **Finish** to complete.



7.13. Session Flows

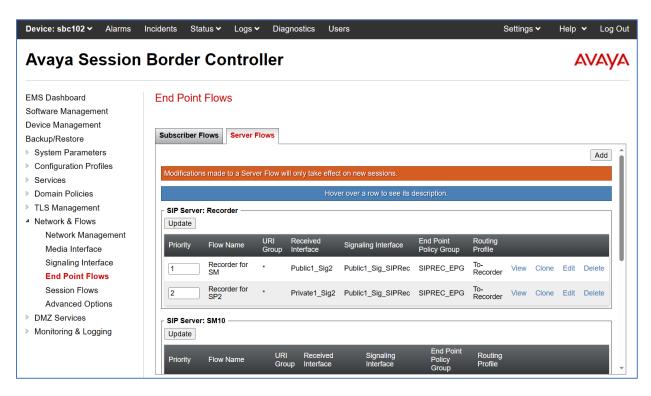
To create a new flow, navigate to **Network & Flows** → **Session Flow** in the left pane. In the center pane, select **Add** (not shown). A pop-up window displays, enter a descriptive name **SIPREC_SessFlow** in the **Flow Name** field and select the **SIPREC_SessPolicies** as created in **Section 7.12**. Click **Finish** to complete.



7.14. End Point Flows

Endpoint flows are used to determine the endpoints (connected servers) involved in a call in order to apply the appropriate policies. When a packet arrives at the Avaya SBC, the content of the packet (IP addresses, URIs, etc.) is used to determine which flow it matches. Once the flow is determined, the flow points to policies and profiles which control processing, privileges, authentication, routing, etc. Once routing is applied and the destination endpoint is determined, the policies for the destination endpoint are applied.

To create a new flow for a server endpoint, navigate to **Network & Flows** → **End Point Flows** in the left pane. In the right pane, select the **Server Flows** tab and click the **Add** button. A popup window (not shown) will appear requesting the name of the new flow and the flow parameters.

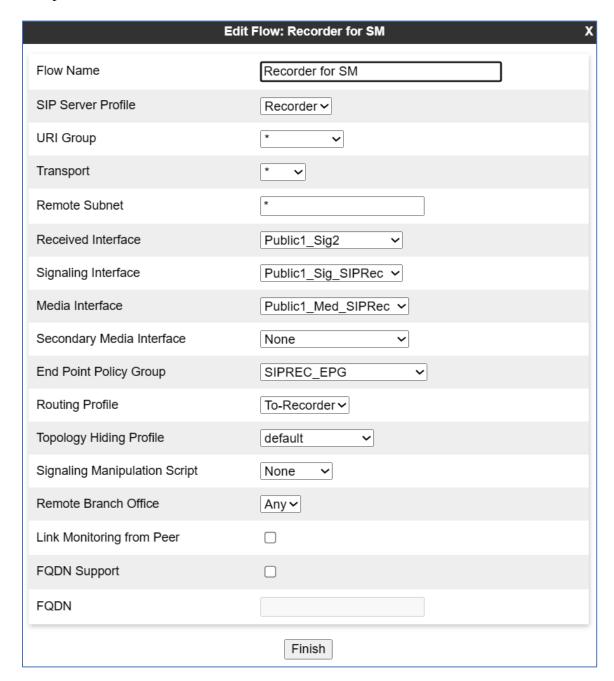


In the testing, there were totally two server flows created for the Cogito recording server to record both ways from the PSTN to the enterprise (agent phone) and from the enterprise (agent phone) to the PSTN via the SIP trunk.

The screenshot below shows the configuration for the Cogito Recorder server flow from Session Manager toward the service provider, *Recorder1 For SM*:

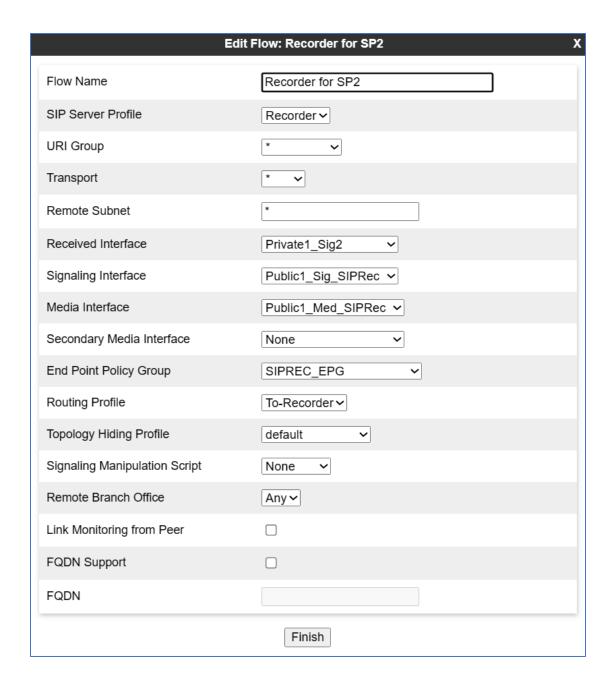
- Flow Name: enter a descriptive name, e.g., Recorder for SM.
- SIP Server Profile: select *Recorder* as configured in Section 7.6.
- **Received Interface**: select *Public1_Sig2* in the list. This is the interface receiving the signaling for the server flow from Session Manager to the service provider.
- Signaling Interface: select *Public1 SIPREC Sig* as configured in Section 7.4.

- Media Interface: select *Public1_Med_SIPRec* as configured in Section 7.5.
- End Point Policy Group: select SIPREC_EPG as configured in Section 7.10.
- **Routing Profile**: select *To-Recorder* as configured in **Section 7.7**.
- Keep other fields at the default values.



The screenshot below shows the configuration for the Cogito Recorder server flow from the Service Provider toward Session Manager, *Recorder For SP*:

- Flow Name: enter a descriptive name, e.g., Redorder For SP.
- SIP Server Profile: select *Recorder* as configured in Section 7.6.
- **Received Interface**: select *Privarte1_Sig2* in the list. This is the interface receiving the signaling for the server flow from the service provider toward to Session Manager.
- **Signaling Interface**: select *Public1_Sig_SIPRec* as configured in **Section 7.4**.
- Media Interface: select *Public1_Med_SIPRec* as configured in Section 7.5.
- End Point Policy Group: select SIPREC_EPG as configured in Section 7.10.
- **Routing Profile**: select *To-Recorder* as configured in **Section 7.7**.
- Keep other fields at the default values.



8. Configure Cogito Emotion Al

The Cogito Emotion AI solution is installed and deployed in the cloud. The configuration of the Cogito recording server and its related applications are done by Cogito technical engineers; therefore, it is not documented in the Application Notes. For more information about the Cogito recording solution, please contact Cogito Support directly.

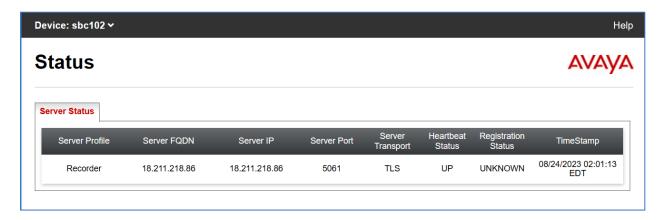
For configuring TLS, the certificate authority (CA) of System Manager is used to create the certificate for the Cogito SIP recording server.

9. Verification Steps

This section provides verification steps that may be performed in the field to verify that the solution is configured properly.

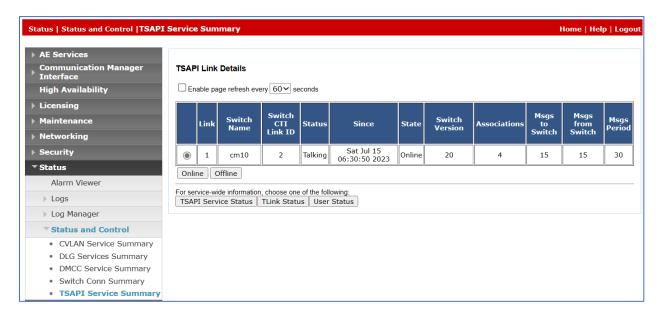
9.1. Verify Server Status in SBC

Verify the status of the Cogito recording servers in the Avaya SBC, from the horizontal menu navigate to **Status Server Status** (not shown). The status in the **Heartbeat Status** column should display as "**UP**".

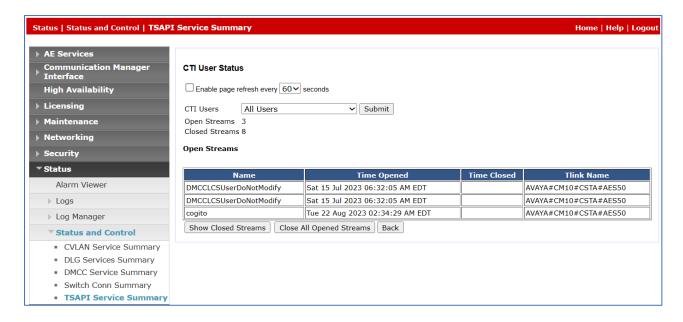


9.2. Verify AES Connection

Verify the status of the **TSAPI Service Summary** service by selecting **Status** → **Status** and **Control** → **TSAPI Service Summary** from the left pane. The **TSAPI Link Details** is displayed in the right pane. The status should be in "**Talking**" in the **Status** column.



Select the **User Status** button in the **TSAPI Link Details** page above to show the status of CTI user used for TSAPI service. The **CTI User Status** displays the *cogito* CTI username with the time of the connection established.



9.3. Verify Status of Agent in CM

Use the command "**list monitored-station**" to verify the Cogito JTAPI client is able to establish a connection with Application Enablement Services using TSAPI service and monitor agent extensions in Communication Manager. The CTI link number should be matched with the CTI link as configured in **Section 5.2**.

list monitored-station																
MONITORED STATION																
Associations:	CTI	1	СТІ	2	СТІ	3	СТІ	4	СТІ	5	СТІ	6	СТІ	7	СТІ	8
Station Ext	-	CRV		CRV		CRV		CRV		CRV		CRV		CRV		CRV
3301	2	0007														
3302	2	0006														
3401	2	8000														
3402	2	0005														

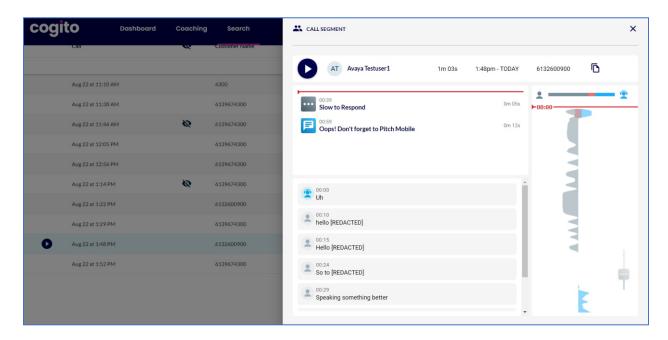
Use the command "**list agent-loginID**" to verify the status of agent. Note that the agents need to be logged in for Cogito recording server to trigger the recording.

list agent-loginID										
AGENT LOGINID										
Login ID	Name		Extensi	ion	Dir Agt	AAS/AUI	AUD COR AgPr SO			
	Skil/Lv	Skil/Lv								
1000	Agent	1000	3301				1	lvl		
	1/01	/	/	/	/	/	/	/		
1001	Agent	1001	3401				1	lvl		
	1/01	/	/	/	/	/	/	/		
1002	Agent	1002	3402				1	lvl		

9.4. Verify SIPREC

The steps to verify SIPREC are:

- 1. Place a call from PSTN to contact center queue via the SIP trunk through the Avaya SBC and Session Manager and the call arrives to an available agent.
- 2. Answer the contact center call on the agent.
- 3. Verify the Cogito recording server receives a live recording call from the Avaya SBC as shown in the screen below.



Disconnect the contact center call from the PSTN user. Verify the Avaya SBC sends Bye
message to the Cogito recording server and receive responses from Cogito to end the
recording call.

10. Conclusion

These Application Notes describe the configuration steps required for Cogito Emotion AI to successfully interoperate with Avaya Aura® Application Enablement Services and Avaya Session Border Controller. All feature and serviceability test cases were completed with observations noted in **Section** Error! Reference source not found..

11. Additional References

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

- [1] Deploying Avaya Aura® applications from System Manager, Release 10.1, August 2023
- [2] Deploying Avaya Aura® Communication Manager, Release 10.1, June 2023
- [3] Administering Avaya Aura® Communication Manager, Release 10.1, June 2023
- [4] Deploying Avaya Aura® Session Manager, Release 10.1 February 2023
- [5] Upgrading Avaya Aura® Session Manager Release 10.1, February 2023
- [6] Administering Avaya Aura® Session Manager Release 10.1, February 2023
- [7] Deploying Avaya Session Border Controller e Release 10.1, September 2023
- [8] Upgrading Avaya Session Border Controller Release 10.1, September 2023
- [9] Administering Avaya Session Border Controller Release 10.1, September 2023

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