

Avaya Solution & Interoperability Test Lab

Application Notes for configuring NICE Engage Platform R6.10 to interoperate with Avaya Aura® Communication Manager R8.0 and Avaya Aura® Application Enablement Services R8.0 using Passive Station Side VoIP recording - Issue 1.0

Abstract

These Application Notes describe the configuration steps for the NICE Engage Platform to interoperate with the Avaya solution consisting of an Avaya Aura® Communication Manager R8.0, an Avaya Aura® Session Manager R8.0, and Avaya Aura® Application Enablement Services R8.0 using Passive Station Side VoIP recording with SMS.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps for the NICE Engage Platform R6.10 to interoperate with the Avaya solution consisting of an Avaya Aura® Communication Manager R8.0, an Avaya Aura® Session Manager R8.0 and Avaya Aura® Application Enablement Services R8.0. The NICE Engage Platform was setup to use passive station-side VoIP recording with SMS and the Telephony Services API (TSAPI) via the Application Enablement Services (AES) to capture the audio and call details for call recording on various Communication Manager endpoints, listed in **Section 4**.

Passive Station-Side VoIP Recording (passive recording) uses port mirroring to record the RTP from each phone set. All phone sets that are to be recorded are plugged into the Avaya 4548GT-PWR layer 3 switch where all of these particular ports are mirrored to one port where the NICE Advanced Interactions Recording server is plugged into. All of the RTP information from all of these phone sets will be delivered to the sniffer port on the NICE Advanced Interactions Recording server. An additional Network Interface Card (NIC) is therefore required on the NICE Advanced Interactions Recording (AIR) server. This NIC is not configured to access the IP stack. It will have no IP configuration. This NIC connects into the mirrored port network that allows access to the phone network connection. This is effectively a hub environment. The promiscuous port needs to be on the same physical media path as any telephone endpoint that it is going to record.

NICE Engage Platform provides the ability to record multi-channel interactions across the organization for regulatory compliance and to utilize these interactions for multiple business applications in order to extract insights and gain value. The platform tightly integrates with the telephony environment via CTI, APIs and SIP and stores the metadata in a single recording platform to ensure regulatory adherence and standardized workforce optimization processes across multiple channels. It provides comprehensive search tools and media retrieval, as well as a wide variety of Real-Time capabilities for PCI compliance and advanced applications.

The NICE Engage Platform uses both the Telephony Services Application Programming Interface (TSAPI) and the System Management Service (SMS) connections on AES. The SMS web service provides the ability to discover the status of resources on Communication Manager.

2. General Test Approach and Test Results

The interoperability compliance testing evaluated the ability of the NICE Engage Platform to carry out call recording in a variety of scenarios using passive recording with AES and Communication Manager. A range of Avaya endpoints were used in the compliance testing all of which are listed in **Section 4**.

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 these Application Notes, the interface between Avaya systems and NICE Engage did not include use of any specific encryption features as requested by NICE. The interface between the SIP phones and Session Manager were also unencrypted to allow NICE to capture the IP address information of the phonesets.

NICE used a "Generic SIP Mapper" interface for media location extraction of the SIP Phones that register to Session Manager. In order for this to operate and avoid configuration of fixed IPs, the signaling must be unencrypted. Any TLS messages on the network need to be decoded by the SIP Mapper and in order to decode these messages all TLS protocols use on the AES needed to be ticked, see **Section 6.5.2**.

2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on placing and recording calls in different call scenarios with good quality audio recordings and accurate call records. The tests included:

- **Inbound/Outbound calls** Test call recording for inbound and outbound calls to the Communication Manager to and from PSTN callers.
- **Hold/Transferred/Conference calls** Test call recording for calls transferred to and in conference with PSTN callers.
- Forwarded calls Test call recording for calls that were forwarded to various endpoints.
- **Feature calls** Test call recording for calls that are parked or picked up using Call Park and Call Pickup.
- Calls to Elite Agents Test call recording for calls to Communication Manager agents logged into one-X® Agent.
- **Serviceability testing** The behavior of NICE Engage Platform under different simulated failure conditions.

2.2. Test Results

All functionality and serviceability test cases were completed successfully. There were no issues to report.

2.3. Support

Technical support can be obtained for NICE Engage Platform from the website http://www.nice.com/support-and-maintenance

3. Reference Configuration

The configuration in **Figure 1** was used to compliance test NICE Engage Platform with the Avaya solution using passive recording to record calls. The Avaya 4548GT-PWR switch is configured to mirror ports that the Avaya endpoints are connected to, to one port where the NICE Advanced Interactions recorder sniffer port is connected to.

Note: Any data switch that is capable of port mirroring can be used, the data switch shown in the diagram is that which was used for compliance testing.

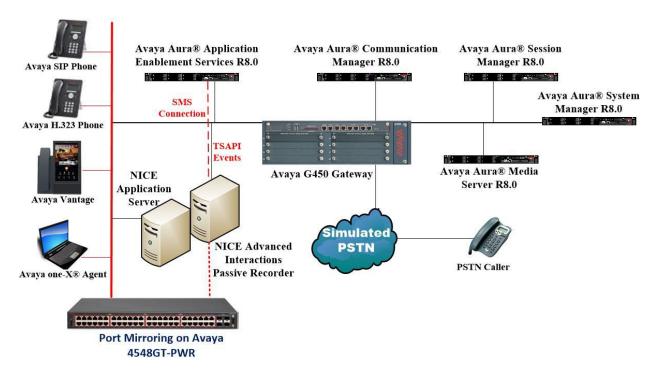


Figure 1: Connection of NICE Engage Platform R6.10 with Avaya Aura® Communication Manager R8.0, Avaya Aura® Session Manager R8.0 and Avaya Aura® Application Enablement Services R8.0

4. Equipment and Software Validated

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

Equipment/Software	Release/Version			
Avaya Aura® System Manager running on Virtual Server	R8.0.0.0.0 Build 8.0.0.0.931077 SW Update Revision No. 8.0.0.0.098174			
Avaya Aura® Session Manager running on Virtual Server	R8.0.0.0.8000035			
Avaya Aura® Communication Manager running on Virtual Server	R8.0 Build 00.0.822.0-24826			
Avaya Aura® Application Enablement Services running on Virtual Server	R8.0 Build No – 8.0.0.0.0.6-0			
Avaya G450 Gateway	41.10.1 /1			
Avaya Media Server running on a Virtual Server	8.0.0.150			
Avaya 4548GT-PWR Ethernet Switch	5.7.3.030			
Avaya 96x1 H323 Deskphone	6.6.115			
Avaya 1616 -I H323 Deskphone	Ha1616ua1_3110A			
Avaya J179 H323 Deskphone	7.002U			
Avaya 96x1 SIP Deskphone	7.1.2.0.14			
Avaya J129 SIP Deskphone	1.0.0.0.43			
Avaya Vantage Equinox	1.0.0.2			
Avaya one-X® Agent	2.5.8			
NICE Engage Platform - NICE Application Server - Advanced Interactions Recorder	6.10			

5. Configure Avaya Aura® Communication Manager

The information provided in this section describes the configuration of Communication Manager relevant to this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

The configuration illustrated in this section was performed using Communication Manager System Administration Terminal (SAT).

5.1. Verify System Features

Use the **display system-parameters customer-options** command to verify that Communication Manager has permissions for features illustrated in these Application Notes. On **Page 3**, ensure that **Computer Telephony Adjunct Links?** is set to **y** as shown below.

```
display system-parameters customer-options
                                                                     Page
                                                                            3 of 11
                                  OPTIONAL FEATURES
    Abbreviated Dialing Enhanced List? y
       previated Dialing Enhanced List? y
Access Security Gateway (ASG)? n
Analog Trunk Incoming Call ID? y
                                                    Audible Message Waiting? y
                                                     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? y
                                                                   DCS (Basic)? y
          ASAI Link Core Capabilities? n
ASAI Link Plus Capabilities? n
                                                            DCS Call Coverage? y
                                                            DCS with Rerouting? y
      Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n Digital Loss Plan Modification? y
                                                                      DS1 MSP? y
              ATM WAN Spare Processor? n
                                   ATMS? y
                                                        DS1 Echo Cancellation? y
                   Attendant Vectoring? y
```

5.2. Note procr IP Address for Avaya Aura® Application Enablement Services Connectivity

Display the procr IP address by using the command **display node-names ip** and noting the IP address for the **procr**.

display node-names	s ip			Page	1 of	2
		IP NODE	NAMES			
Name	IP Address					
SM100	10.10.40.34					
Aes80vmpg	10.10.40.56					
default	0.0.0.0					
g450	10.10.40.15					
procr	10.10.40.59					

5.3. Configure Transport Link for Avaya Aura® Application Enablement Services Connectivity

To administer the transport link to AES use the **change ip-services** command. On **Page 1** add an entry with the following values:

- **Service Type:** Should be set to **AESVCS**.
- **Enabled:** Set to y.
- Local Node: Set to the node name assigned for the procr in Section 5.2
- Local Port: Retain the default value of 8765.

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

Go to **Page 4** of the **ip-services** form and enter the following values:

- AE Services Server: Name obtained from the AES server, in this case aes80vmpg.
- **Password:** Enter a password to be administered on the AES server.
- Enabled: Set to v.

Note: The password entered for **Password** field must match the password on the AES server in **Section 6.2**. The **AE Services Server** should match the administered name for the AES server; this is created as part of the AES installation, and can be obtained from the AES server by typing **uname –n** at the Linux command prompt.

change ip-ser				Page	4 of	4
	AE	Services Adminis	stration			
Server ID	AE Services Server	Password	Enabled	Status		
1: 2: 3:	aes80vmpg	*****	У	idle		

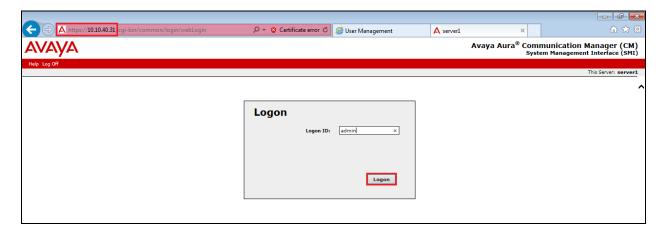
5.4. Configure CTI Link for TSAPI Service

Add a CTI link using the **add cti-link n** command. Enter an available extension number in the **Extension** field. 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		Page	1 of	3
		CTI LINK			
CTI Link:	1				
Extension:	2002				
Type:	ADJ-IP				
				COR:	1
Name:	aes80vmpg				

5.5. Configure System Management Service user on Avaya Aura® Communication Manager

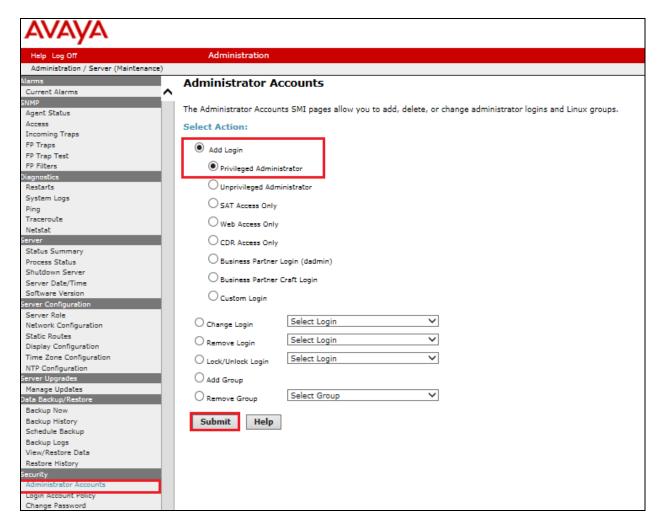
This user is created specifically for the SMS connection that NICE utilise for this specific type of call recording. Using a web browser navigate to the Communication Manager IP Address. Enter the proper credentials and click on Logon.



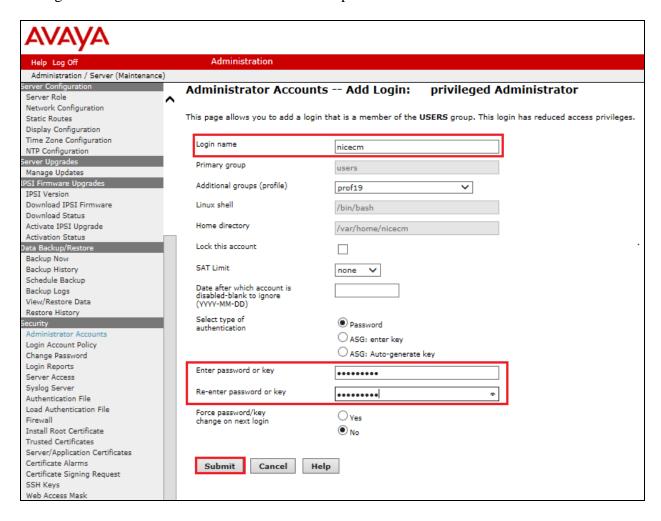
Once logged in click on **Administration** at the top of the page and select **Server (Maintenance)** from the drop-down menu.



In the left window navigate to **Security** \rightarrow **Administrator Accounts**. In the main window select **Add Login** and **Privileged Administrator** as shown below. Click on **Submit** when finished.



Enter a suitable **Login name** and enter a suitable **password**, then click on **Submit** as all other settings can be left as default. Note this name and password will be needed in **Section 7.1**.



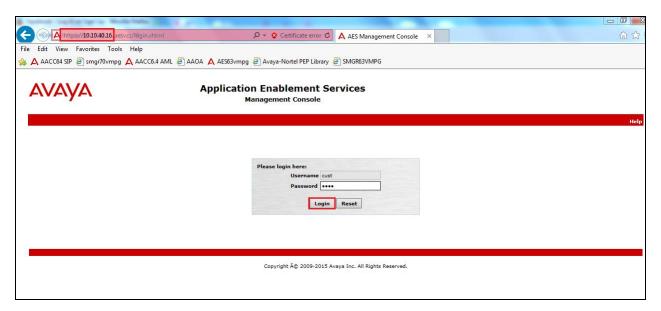
6. Configure Avaya Aura® Application Enablement Services

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

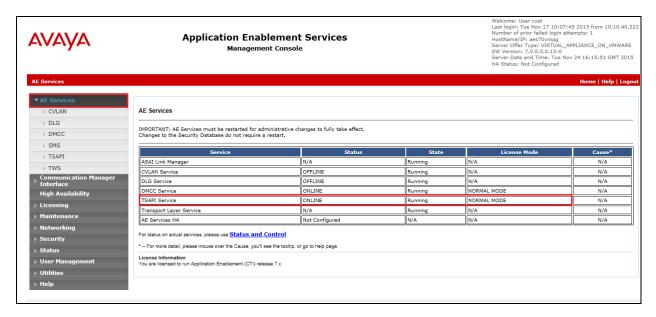
- Verify Licensing
- Create Switch Connection
- Administer TSAPI link
- Identify Tlinks
- Configure Networking Ports
- Create CTI User
- Configure Security Database
- Configure the System Management Service on Avaya Aura® Application Enablement Services

6.1. Verify Licensing

To access the AES Management Console, enter **https://<ip-addr>** as the URL in an Internet browser, where <ip-addr> is the IP address of AES. At the login screen displayed, log in with the appropriate credentials and then select the **Login** button.

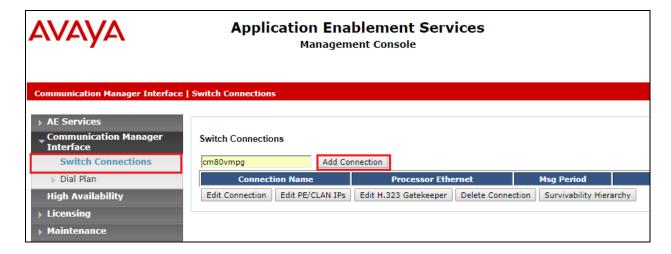


The Application Enablement Services Management Console appears displaying the **Welcome to OAM** screen (not shown). Select **AE Services** and verify that the TSAPI Service is licensed by ensuring that **TSAPI Service** is in the list of **Services** and that the **License Mode** is showing **NORMAL MODE**. If not, contact an Avaya support representative to acquire the proper license for your solution.

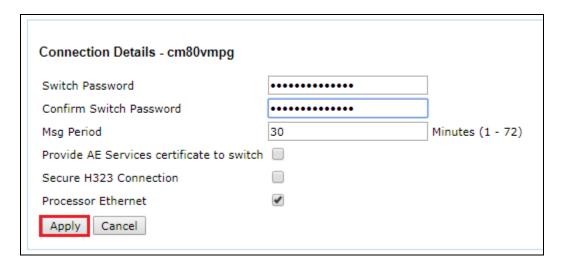


6.2. Create Switch Connection

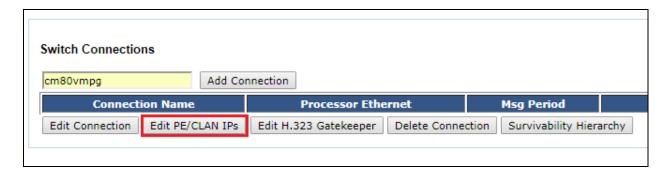
From the AES Management Console navigate to **Communication Manager Interface Switch Connections** to set up a switch connection. Enter a name for the Switch Connection to be added and click the **Add Connection** button.



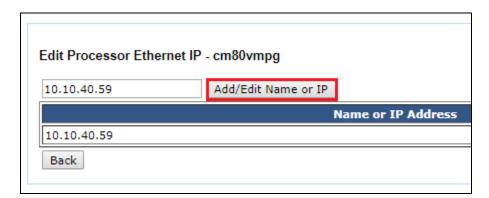
In the resulting screen enter the **Switch Password**; the Switch Password must be the same as that entered into Communication Manager AE Services Administration screen via the **change ipservices** command, described in **Section 5.3**. Default values may be accepted for the remaining fields. Click **Apply** to save changes.



From the **Switch Connections** screen, select the radio button for the recently added switch connection and select the **Edit PE/CLAN IPs** button.

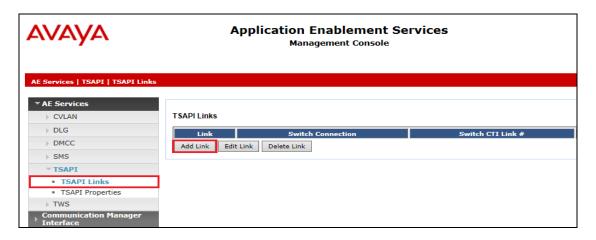


In the resulting screen, enter the IP address of the procr as shown in **Section 5.2** that will be used for the AES connection and select the **Add/Edit Name or IP** button.



6.3. Administer TSAPI link

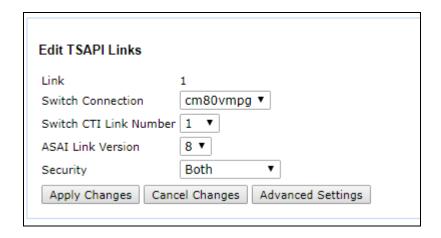
From the Application Enablement Services Management Console, select **AE Services** → **TSAPI** → **TSAPI Links**. Select **Add Link** button as shown in the screen below.



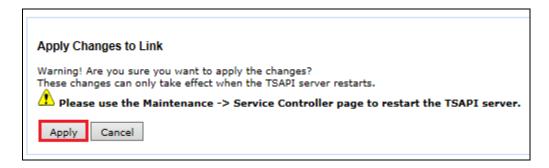
On the **Add TSAPI Links** screen (or the **Edit TSAPI Links** screen to edit a previously configured TSAPI Link as shown below), enter the following values:

- Link: Use the drop-down list to select an unused link number.
- **Switch Connection:** Choose the switch connection **cm80vmpg**, which has already been configured in **Section 6.2** from the drop-down list.
- **Switch CTI Link Number:** Corresponding CTI link number configured in **Section 5.4** which is **1**.
- **ASAI Link Version:** This should correspond with the Communication Manager version.
- **Security:** This can be left at the default value of **both**.

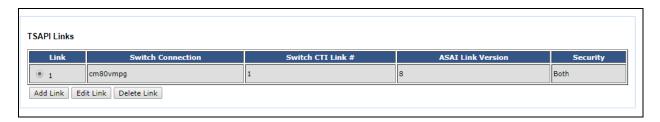
Once completed, select Apply Changes.



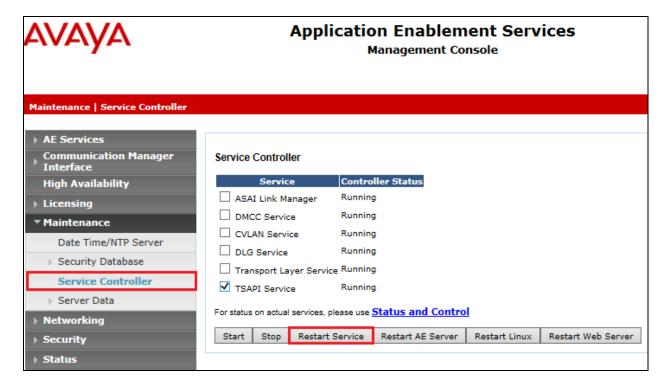
Another screen appears for confirmation of the changes made. Choose **Apply**.



When the TSAPI Link is completed, it should resemble the screen below.

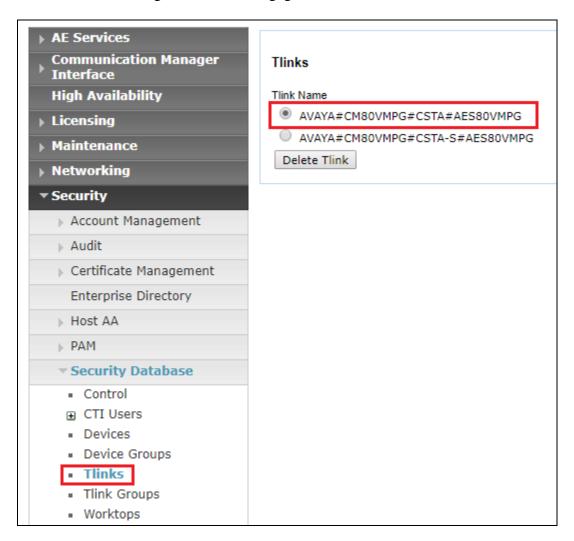


The TSAPI Service must be restarted to effect the changes made in this section. From the Management Console menu, navigate to **Maintenance** → **Service Controller**. On the Service Controller screen, tick the **TSAPI Service** and select **Restart Service**.



6.4. Identify Tlinks

Navigate to **Security** → **Security Database** → **Tlinks**. Verify the value of the **Tlink Name**. This will be needed to configure the NICE Engage Platform in **Section 7.1**.

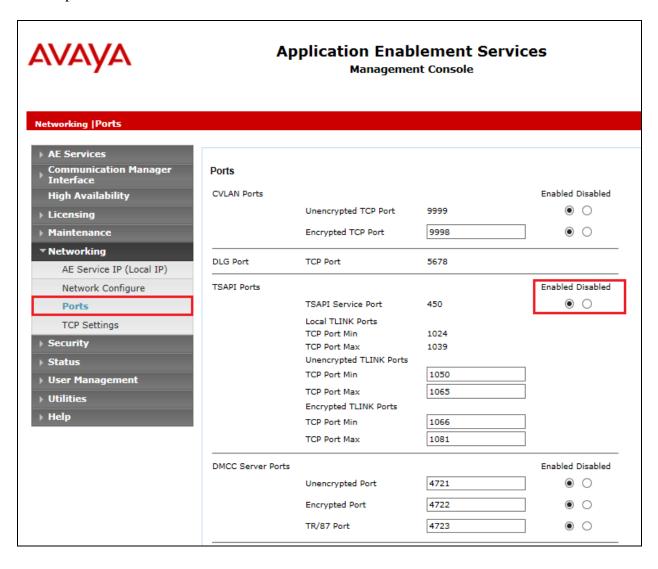


6.5. Configure Networking Ports

Ensure that all ports are enabled for connections to the AES from NICE.

6.5.1. Enable TSAPI Ports

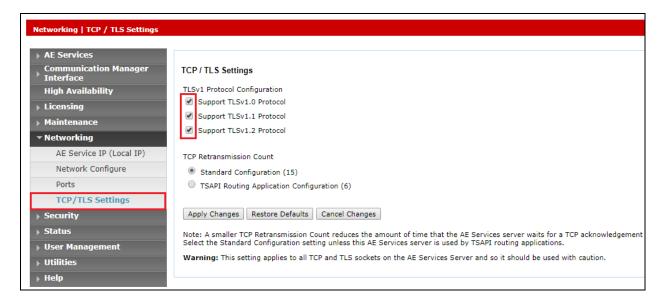
To ensure that TSAPI ports are enabled, navigate to **Networking** \rightarrow **Ports**. Ensure that the TSAPI ports are set to **Enabled** as shown below.



6.5.2. Enable TLS Ports

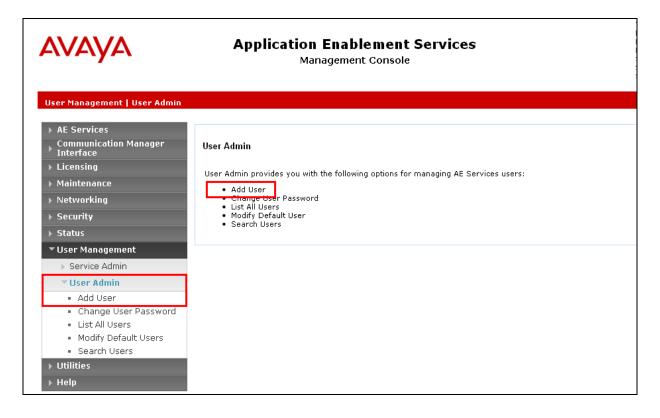
In order to allow the NICE Generic SIP Mapper to decode TLS messages support for all three TLS protocols needed to be ticked.

Navigate to **TCP/TLS Settings** as shown. To ensure that all TLS protocols are supported, tick the boxes as shown below. Click on **Apply Changes**.



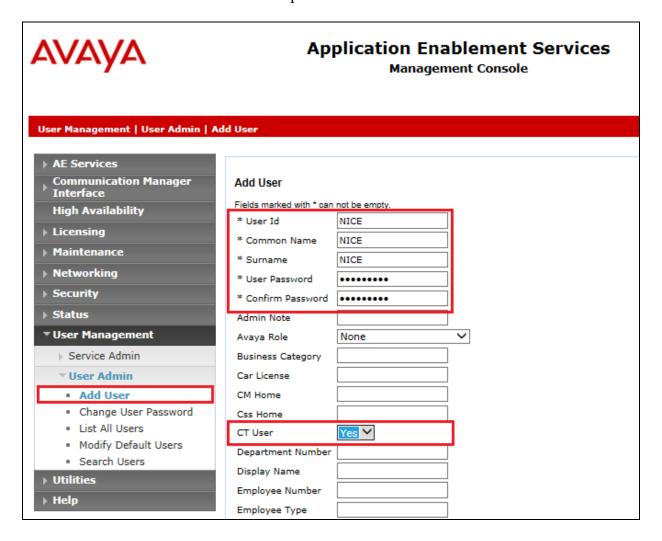
6.6. Create CTI User

A User ID and password needs to be configured for the NICE Engage Platform to communicate with the Application Enablement Services server. Navigate to the **User Management** → **User Admin** screen then choose the **Add User** option.

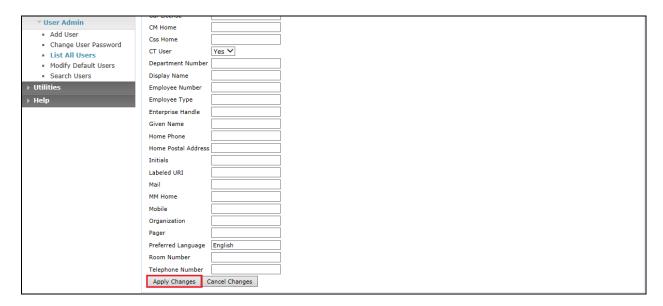


In the **Add User** screen shown below, enter the following values:

- User Id This will be used by the NICE Engage Platform setup in Section 7.1.
- Common Name and Surname Descriptive names need to be entered.
- **User Password** and **Confirm Password** This will be used with NICE Engage Platform setup in **Section 7.1**.
- **CT User -** Select **Yes** from the drop-down menu.



Scroll down and click on Apply Changes.

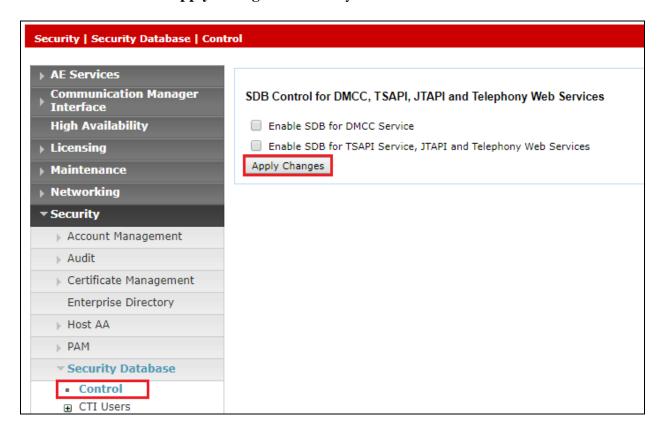


6.7. Configure Security Database

For compliance testing associated with these Application Notes the Security Database was not enabled and the user associated with NICE was given unrestricted access.

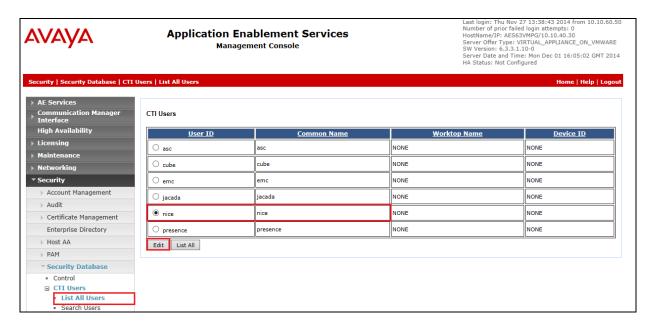
6.7.1. Disable the Security Database Control

Navigate to **Security** → **Security Database** → **Control** as shown below. Ensure that no boxes are ticked and click on **Apply Changes** if necessary.

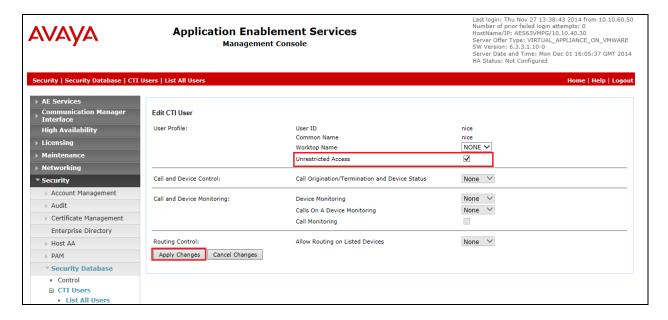


6.7.2. Associate Devices with CTI User

Navigate to Security → Security Database → CTI Users → List All Users. Select the CTI user added in Section 6.6 and click on Edit Users.



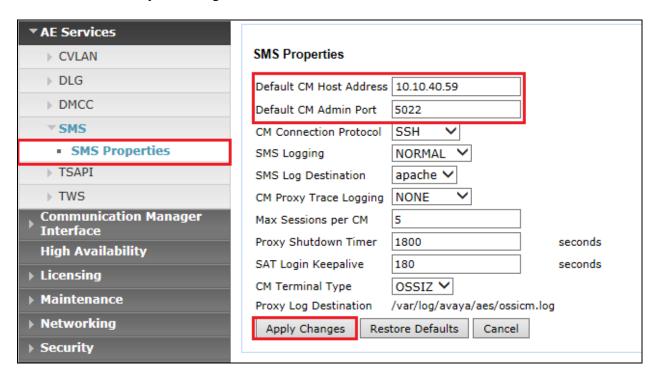
In the main window ensure that **Unrestricted Access** is ticked. Once this is done click on **Apply Changes**.



6.8. Configure the System Management Service on Avaya Aura® Application Enablement Services

From the AE Services Management Console main menu, select **AE Services** \rightarrow **SMS Properties**. The following list describes the SMS configuration settings and provides guidelines for configuring SMS.

- **Default CM Host Address** SMS will attempt to connect to this Communication Manager host address, as long as no host address is explicitly specified in the authorization header of a client request. If this field is blank, all SMS requests must explicitly include the target Communication Manager host address.
- **Default CM Admin Port** By default the System Management Service will use **5022** to connect to a Communication Manager server.
- **CM Connection Protocol** Use the default **SSH** port. The default TUI (or SAT) ports on Communication Manager are **SSH** Port=5022 Telnet Port=5023.
- **SMS Logging** Use the default setting **NORMAL** unless debugging.
- **SMS Log Destination** Use the default **apache**, unless debugging.
- **CM Proxy Trace Logging** Use the default **NONE**, unless debugging.
- Max Sessions per CM This is a safety setting that prevents SMS from consuming all of the TUI processes on Communication Manager. By default the setting is 5.
- **Proxy Shutdown Timer** Use the default **1800** seconds.
- **SAT Login Keepalive** Use the default **180** seconds.
- **CM Terminal Type** Use the default **OSSIZ**.
- **Proxy Log Destination** Use the default destination /var/log/avaya/aes/ossicm.log for the CM Proxy Trace logs on the AE Server.

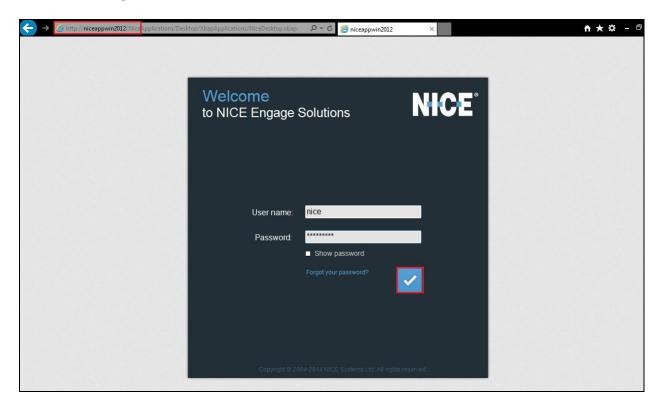


7. Configure NICE Engage Platform

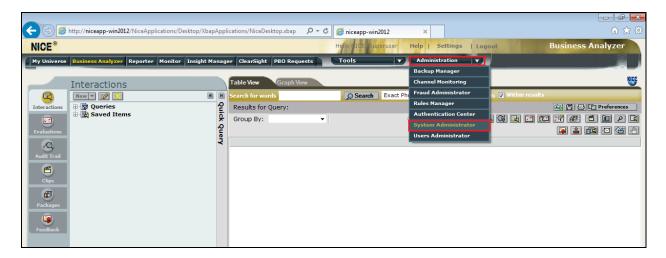
The installation of NICE Engage Platform is usually carried out by an engineer from NICE and is outside the scope of these Application Notes. For information on the installation of the NICE Engage Platform contact NICE as per the information provided in **Section 2.3**.

The following sections will outline the process involved in connecting the NICE Engage Platform to the Avaya solution. All configuration of the NICE Engage Platform for connection with the AES is performed using a web browser connecting to the NICE Engage Application Server. Open a web browser as shown navigate to

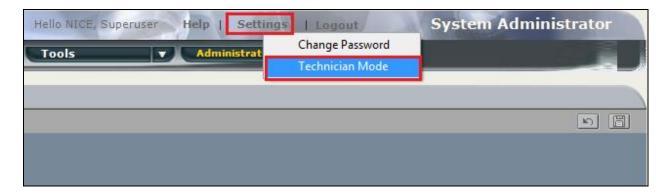
<u>http://<NICEEngageApplicationServerIP>/Nice</u> as shown below and enter the proper credentials and click on **Login**.



Once logged in expand the **Administration** dropdown menu and click on **System Administrator** as highlighted.

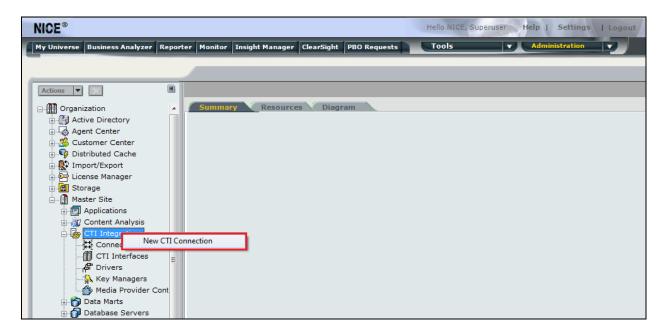


Before any changes can be made, switch to Technician Mode by clicking into Settings at the top of the screen as shown below.

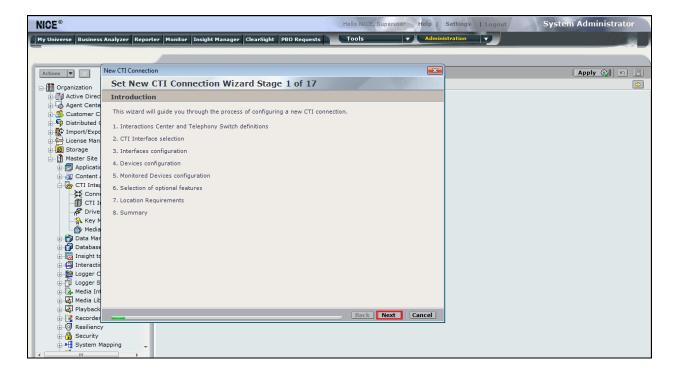


7.1. New CTI Connection

Navigate to **Master Site** → **CTI Integration** in the left window then right-click on CTI Integration and select **New CTI Connection** as shown below.

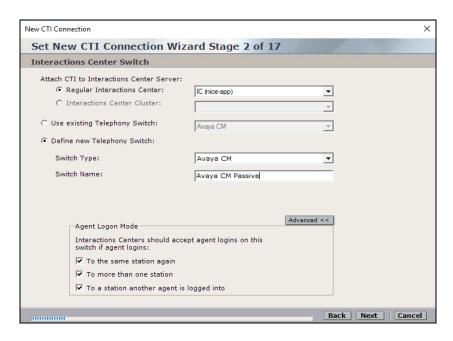


The **New CTI Connection Wizard** is opened, and this will go through the 17 steps required to setup the connection to the AES for Passive Station Side VoIP recording. Click on **Next** to continue.

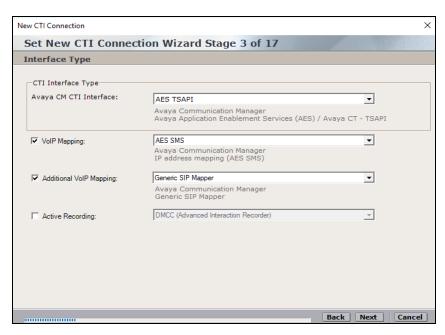


The value for Regular Interactions Center is a value that was already created during the installation of the NICE Engage platform. This value is therefore pre-chosen for the CTI connection being created below.

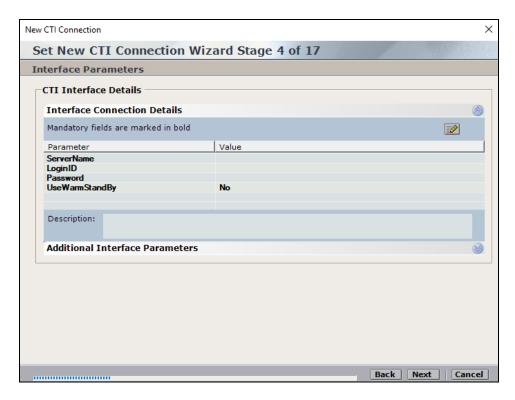
The **Telephony Switch** must be selected, and this will be **Avaya CM**. Enter a suitable name for this **Switch Name**. Click on **Next** to continue.



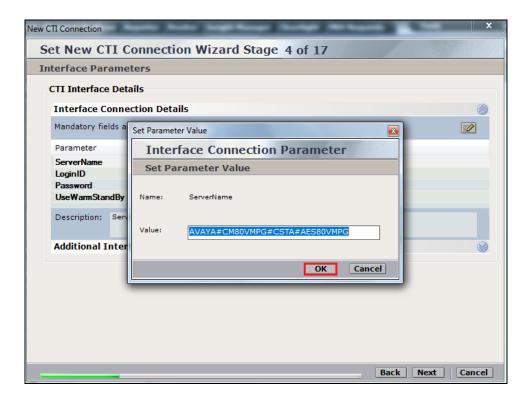
Select **AES TSAPI** for the **Avaya CM CTI Interface**, ensure that **VoIP Mapping** is ticked and select the **AES SMS** from the dropdown menu. Ensure that **Additional VoIP Mapping** is ticked, and that **Generic SIP Mapper** is chosen from the dropdown. Click on **Next** to continue.



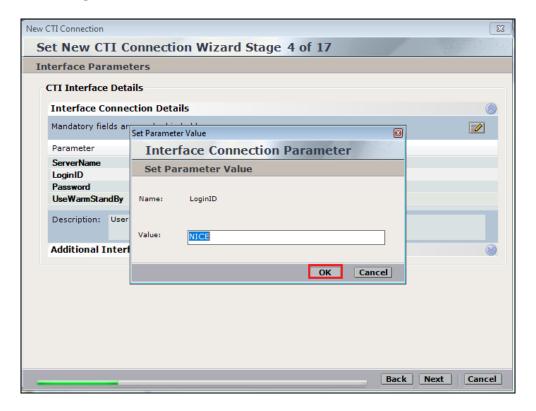
Each of the values below must be filled in. Double-click on each **Parameter** to enter a value for that parameter.



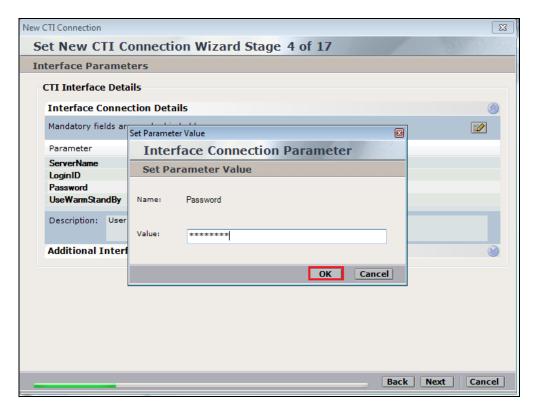
Double-click on ServerName and enter the TSAPI Tlink Value from Section 6.4.



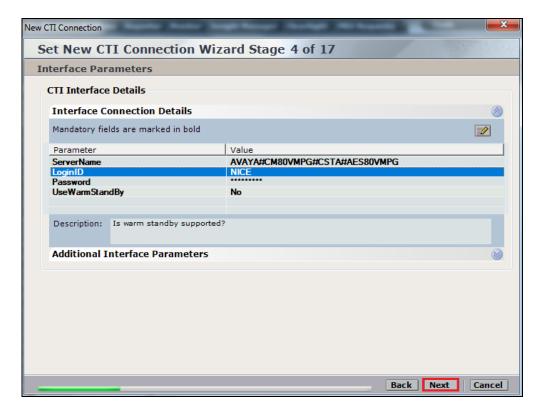
Double-click on **LoginID** and enter the username that was created in **Section 6.6**. Click on **OK**.



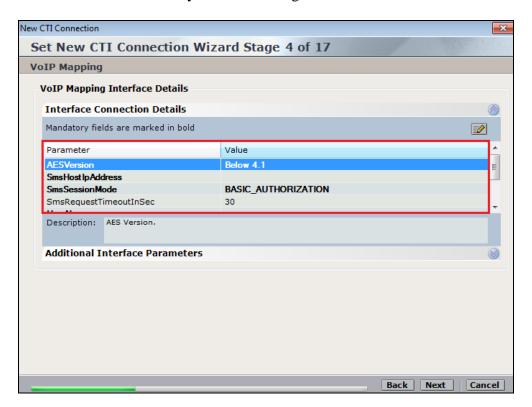
Double-click on password and enter the value for the password that was created in **Section 6.6**.



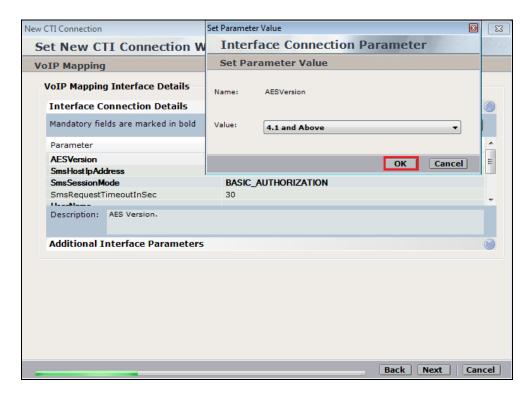
Click on **Next** once these values are all filled in.



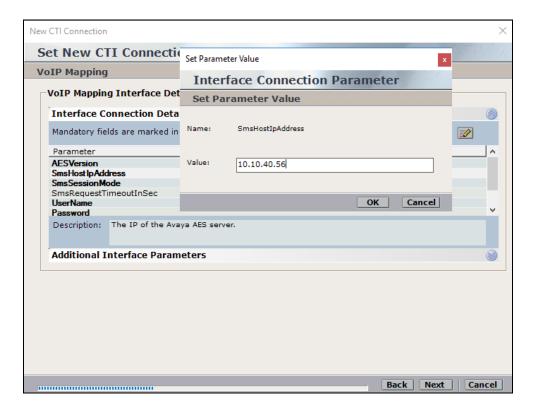
The values below must be filled in by double-clicking on each **Parameter**.



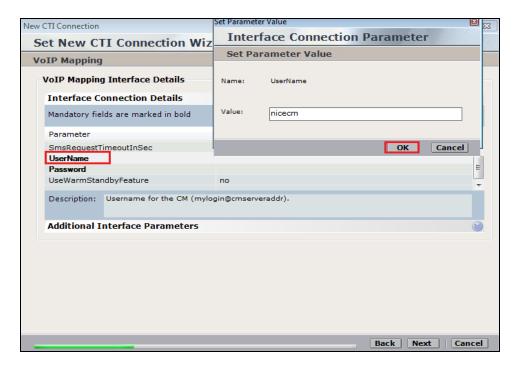
Enter the Value for the AESVersion. Click on OK.



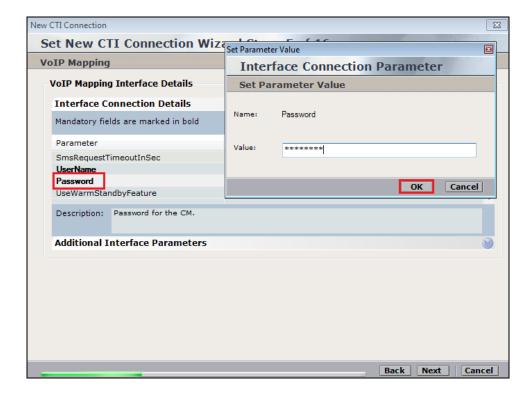
Enter the **Value** for the **SmsHostIpAddress**, note this will be the IP address of the AES in the solution. Click on **OK** to continue.



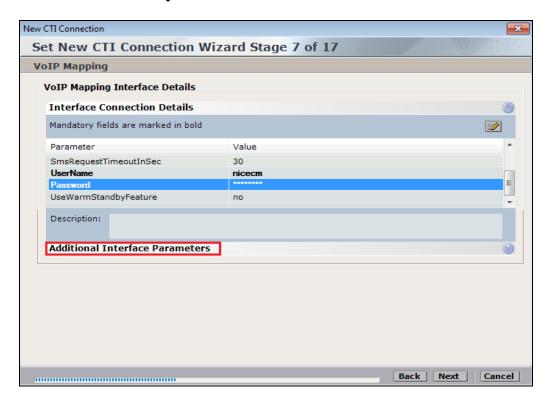
As before, enter the username that was created in **Section5.5** and click on **OK**. The username can be entered as shown below when one Communication Manager has been associated on the SMS properties, see **Section 6.8**. However if there are multiple Communication Manager on site then the username must be in the form login@CMIPADDRESS:port



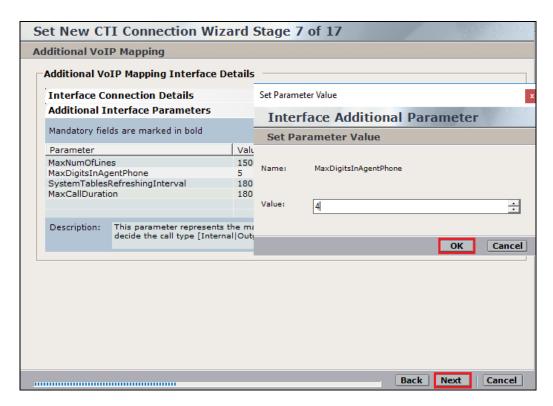
Enter the password that was created in **Section 5.5** and click on **OK**.



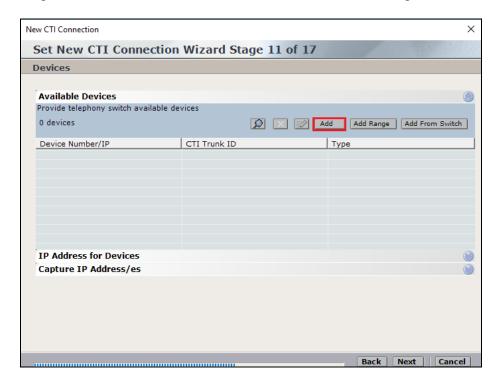
Click on **Additional Interface parameters** to continue.



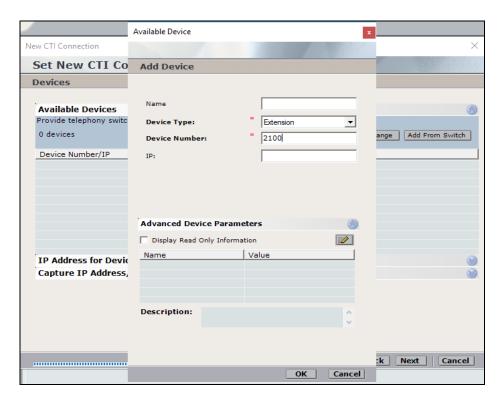
Double-click on **MaxDigitsInAgentPhone** and change the **Value** to **4** as shown below. Click on **Next** at the bottom of the screen.



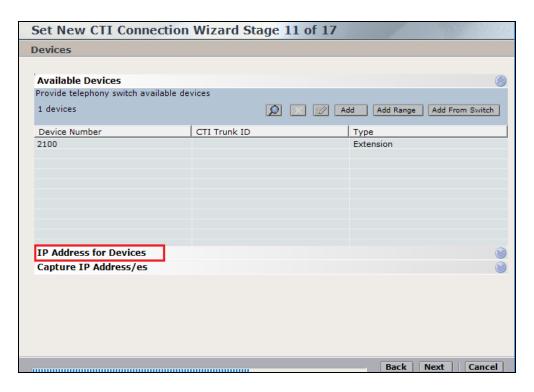
On the following screen, click on Add, to add the Communication Manager devices.



The **Device Type** should be **Extension** and insert the correct extension number. The IP can be left blank if the Generic SIP mapper or the SMS connection will be used to determine the IP address. Click on **OK** to continue.



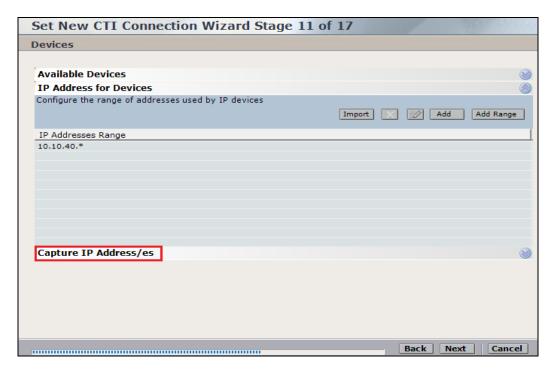
Click on **IP Address for Devices**. This will add the address range for the IP addresses picked up from the SMS connection to the AES.



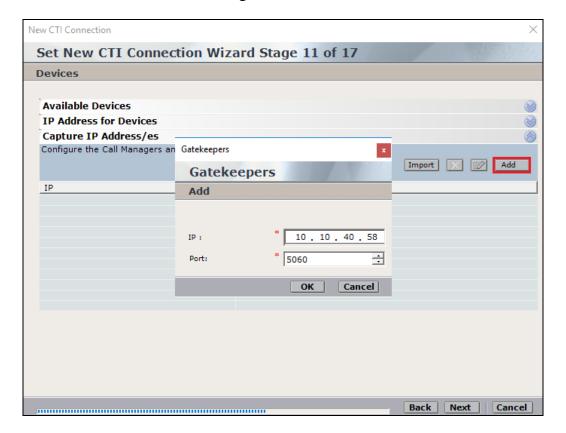
Click on **Add Range** to add the **Device IP Range**. The range is added in the form of x.x.x.* as shown below where the range is from 10.10.40.1 to 10.10.40.254. Click on **OK**.



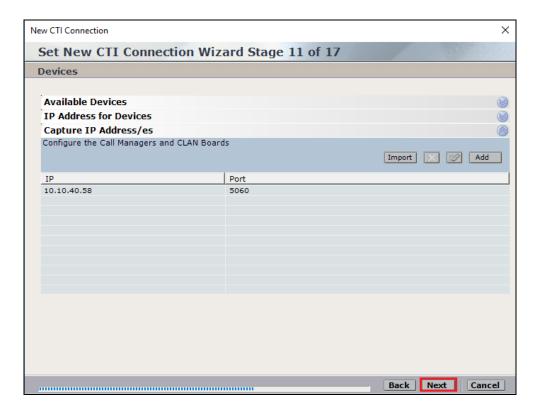
Select **Capture IP Address/es**. This will add the information required for the Generic SIP mapper to capture the IP addresses information of the SIP phones.



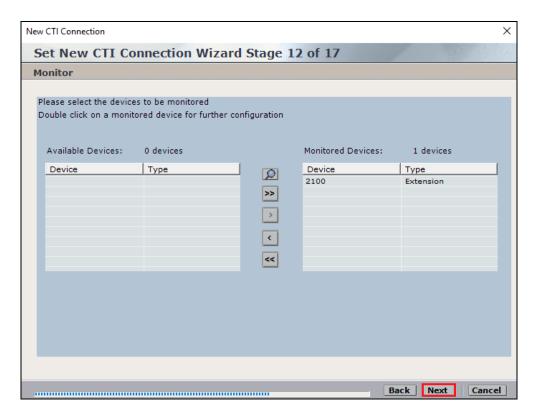
Click on Add and enter the Session Manager's IP address and the SIP Port 5060.



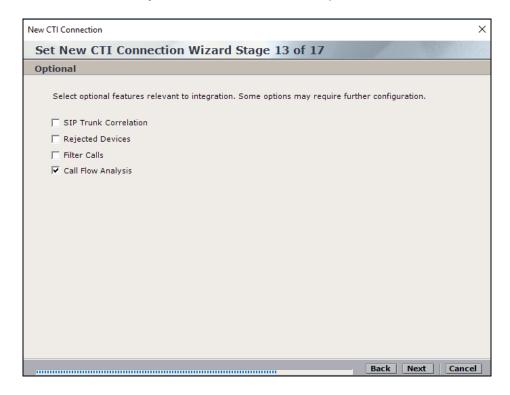
Click on **Next** to continue.



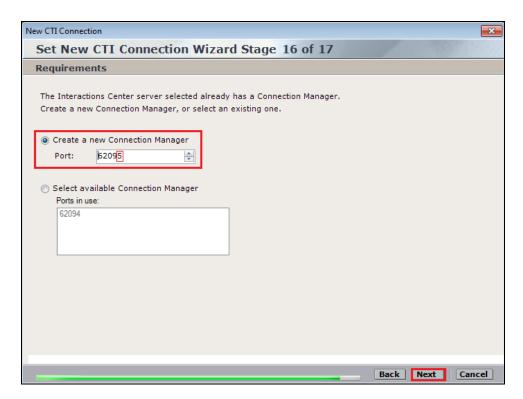
Select the new extension and click on the >> icon as shown. Click on **Next** to continue.



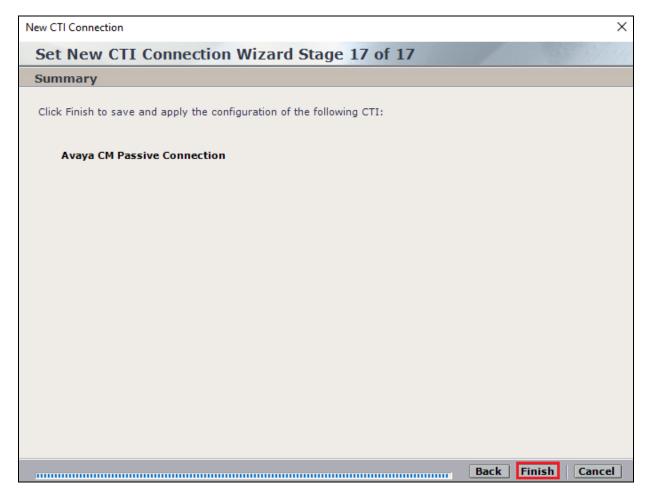
It is optional, but for better analysis tick on Call Flow Analysis and click on Next to continue.



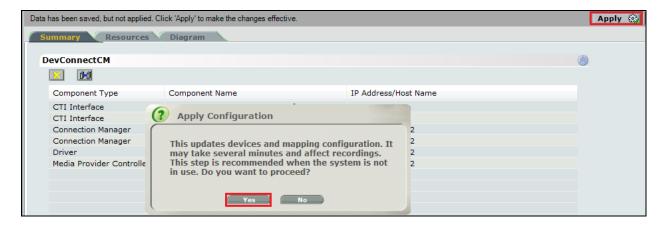
Select a different **Port** number as shown below **62095** is chosen simply because **62094** is already in use.



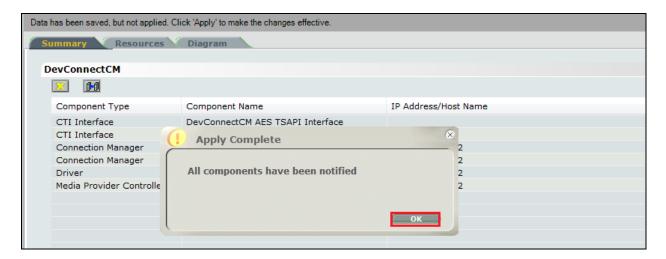
Click on **Finish** to complete the **New CTI Wizard**.



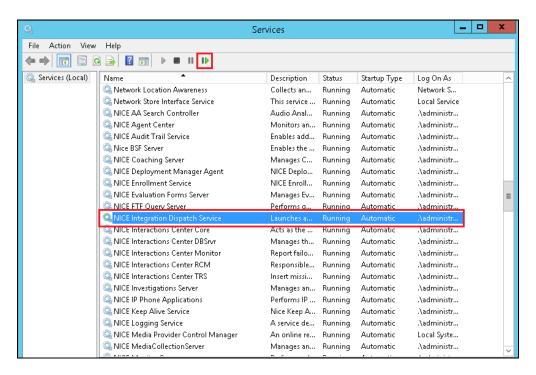
Click on **Apply** at the top right of the screen to save the new connection and click on **Yes** to proceed.



The following shows that the save was successful. Click on **OK** to continue.

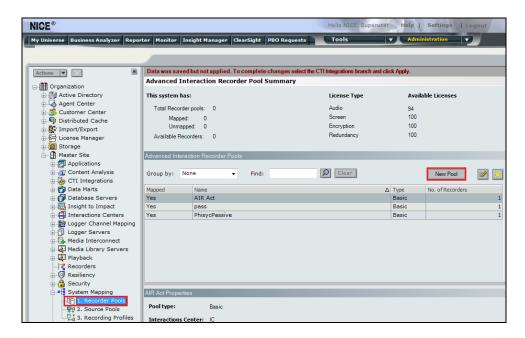


From the NICE Application Server, open **Services** and restart the **NICE Integration Dispatch Service**.

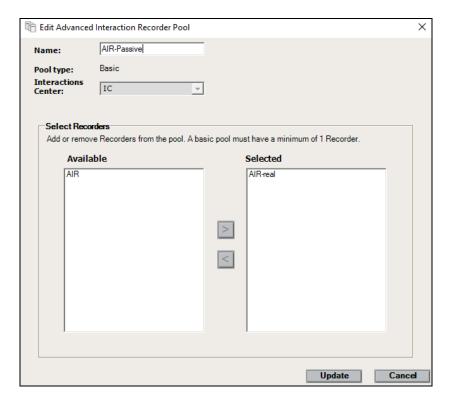


7.2. System Mapping

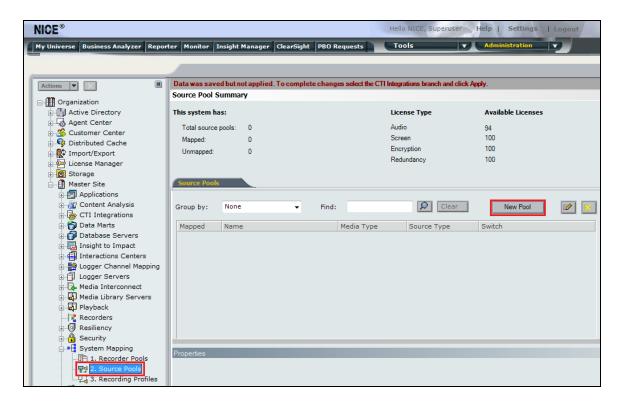
From the web browser navigate to Master Site \rightarrow System Mapping \rightarrow Recorder Pools. In the main window click on New Pool.



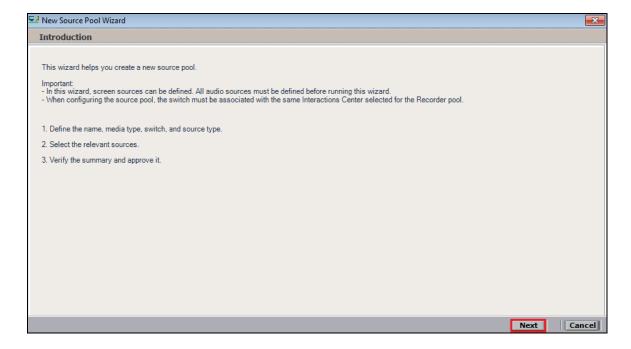
Enter a suitable **Name** for the **Recorder Pool** and select the **AIR-real** from the list of **Available Recorders** and click on **Update** to continue.



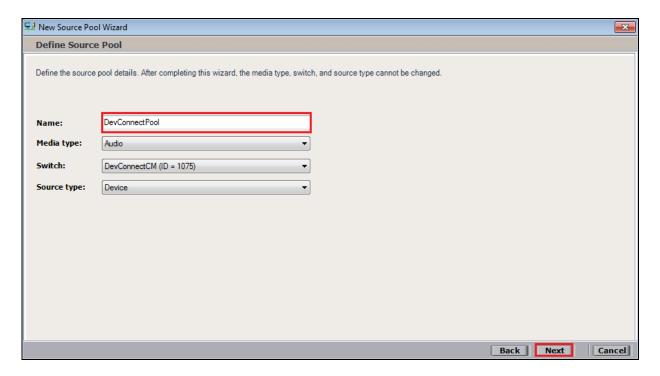
From the left navigation window select **Source Pools** and from the main window click on **New Pool**.



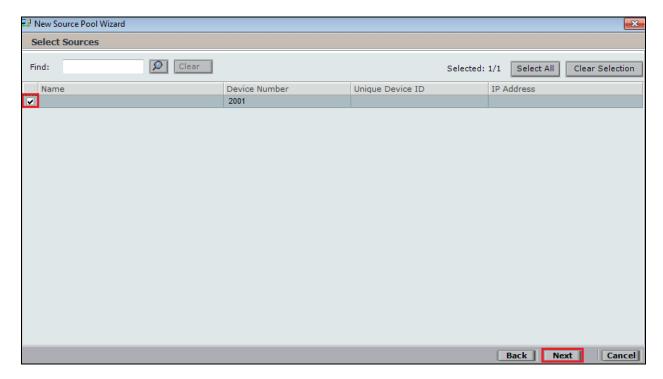
Click on **Next** to continue to add a new **Source Pool.**



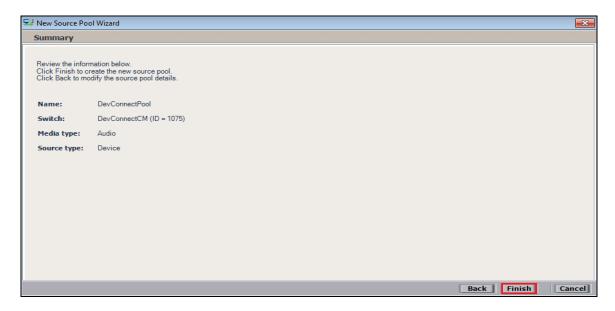
Enter a suitable **Name** and the other values were left as default. Click on **Next** to continue.



Select the extensions that were created in **Section 7.1**, note only one extension number is shown in the example below but this is not typical. Click on **Next** to continue.



Click on **Finish** to complete the **New Source Pool Wizard**.



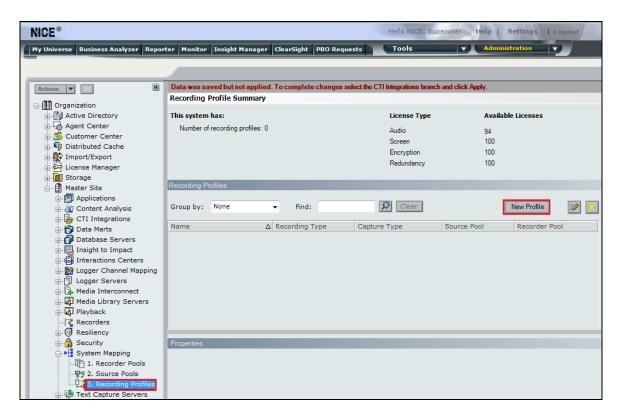
To implement these new changes, navigate to **Master Site** \rightarrow **CTI Integrations** in the left window and in the main window click on **Apply** at the top right of the window.



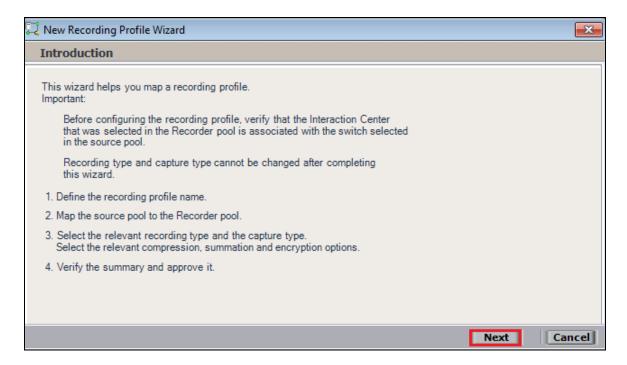
The following screen shows the changes were saved correctly. Click on **OK** to continue.



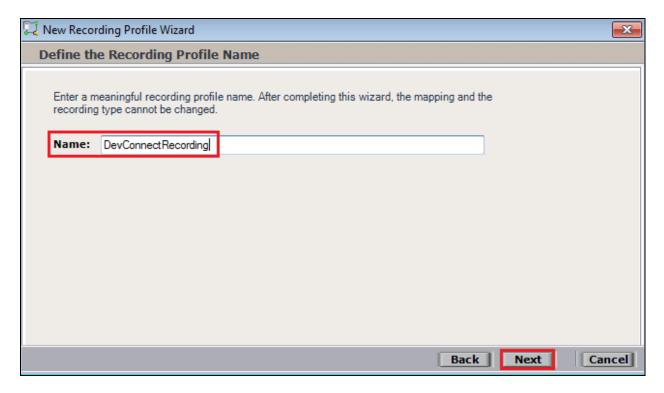
From the left window navigate to **Master Site** \rightarrow **System Mapping** \rightarrow **Recording Profiles** and in the main window click on **New Profile**.



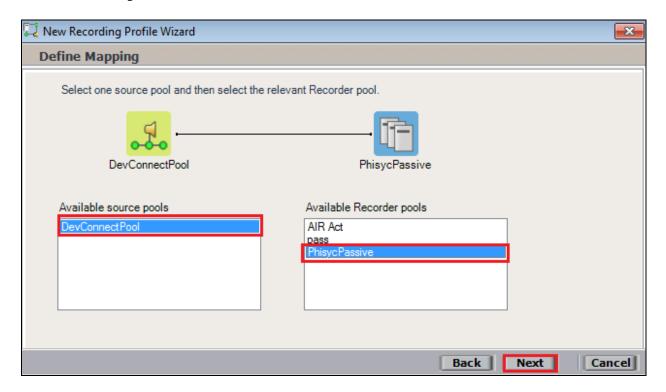
Click on **Next** to continue with the **New Recording Profile Wizard**.



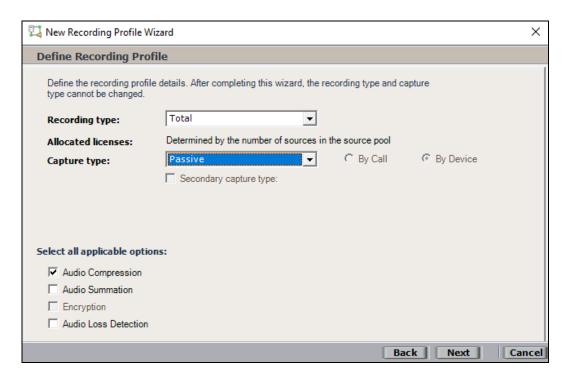
Enter a suitable **Name** for the Recording profile.



Select the correct **source pool** and **Recorder pool**, click **Next** to continue. The recorder pool below shows **Phisyc Passive** but this should be the Recorder pool that was created above and in this case will be **pass**.



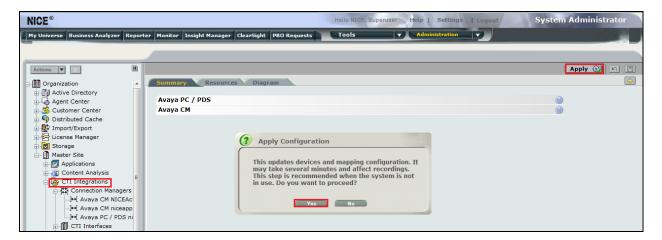
For total recording i.e., the recording of all calls, select **Total** as the **Recording type**. For **Capture type**, ensure that **Passive** is selected from the drop-down box. **Audio Compression** is selected as default and can be left like this. Click on **Next** to continue.



Click on **Finish** to complete the **New Recording Profile Wizard**. The screen below shows that for Total **Passive** recording.



Navigate to **Master Site** \rightarrow **CTI Integrations** and from the main window click on **Apply**. Then click on **Yes** to proceed.



This concludes the setup of the NICE Application Server for Passive Station Side VoIP SMS recording.

8. Verification Steps

This section provides the steps that can be taken to verify correct configuration of the NICE Engage Platform and Avaya Aura® Application Enablement Services.

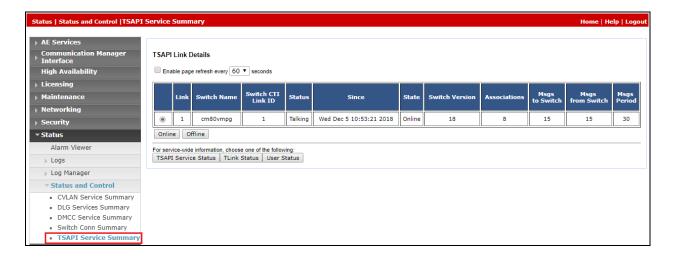
8.1. Verify Avaya Aura® Communication Manager CTI Service State

Before the connection between the NICE Engage Platform and the AES is checked, check the connection between Communication Manager and AES to ensure it is functioning correctly. Check the AESVCS link status by using the command **status aesvcs cti-link**. Verify the **Service State** of the CTI link is **established**.

statu	s aesvcs ct	i-link							
AE SERVICES CTI LINK STATUS									
CTI Link	Version	Mnt Busy	AE Services Server	Service State	Msgs Sent	Msgs Rcvd			
1	4	no	aes80vmpg	established	18	18			

8.2. Verify TSAPI Link

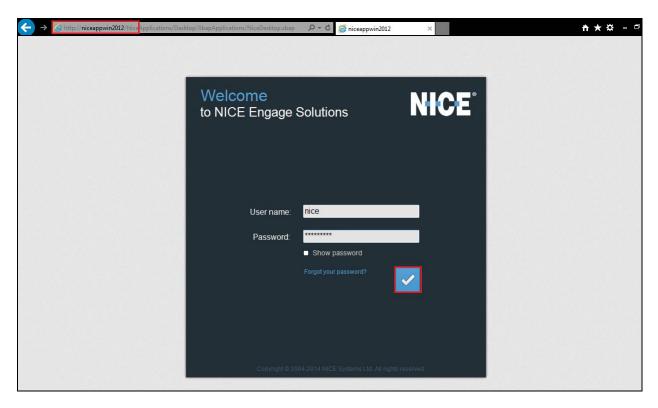
On the AES Management Console verify the status of the TSAPI link by selecting **Status Status and Control TSAPI Service Summary** to display the **TSAPI Link Details** screen. Verify the status of the TSAPI link by checking that the **Status** is **Talking** and the **State** is **Online**.



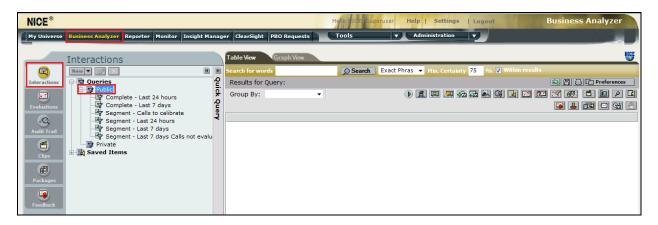
8.3. Verify calls are being recorded

From any of the monitored Avaya endpoints make a series of inbound and outbound calls. Once these calls are completed they should be available for playback through a web browser to the NICE Application Server.

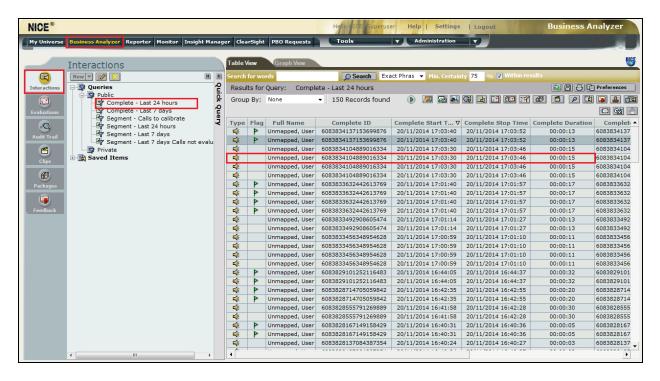
Open a browser session to the NICE Application Server as is shown below. Enter the proper credentials and click on **Login**.



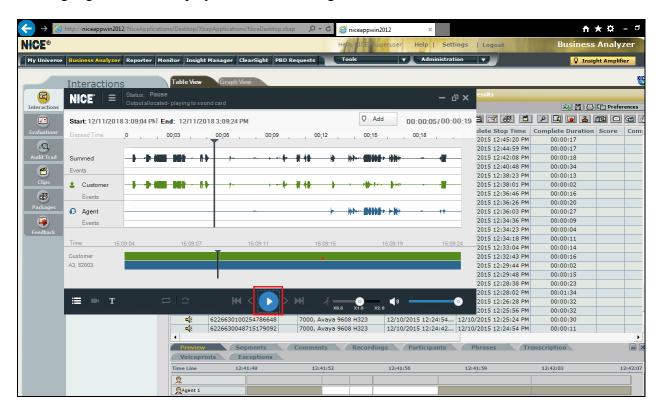
Click on **Business Analyser** at the top of the screen. Select **Interactions** from the left window and then navigate to **Queries** \rightarrow **Public**.



Click on **Complete** – **Last 24 hours**. This should reveal all the recordings that took place over the previous 24 hours. Select the required recording from the list and double-click on this to play the recording.

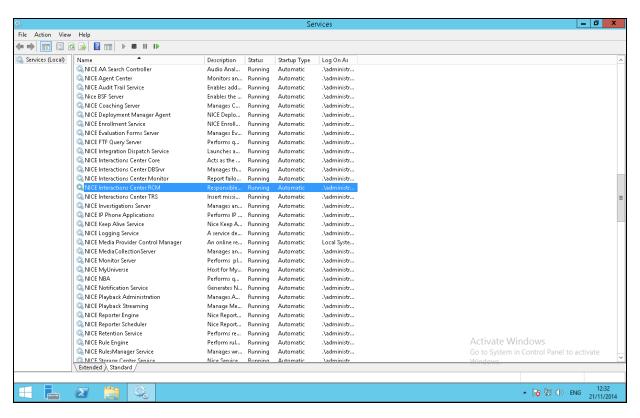


The NICE player is opened and the recording is presented for playback. Click on the **Play/Pause** icon highlighted below to play back the recording.



8.4. Verify NICE Services

If these recordings are not present or cannot be played back the NICE services may not be running or may need to be restarted. There are two separate servers as a part of this NICE Engage Platform. The NICE Application Server and the NICE Passive Logger, both servers can be logged into and checked to ensure all services beginning with NICE are running correctly. As a last resort both servers may need a reboot after the initial configuration.



9. Conclusion

These Application Notes describe the configuration steps required for NICE Engage Platform R6.10 to successfully interoperate with Avaya Aura® Communication Manager R8.0 using Avaya Aura® Application Enablement Services R8.0 to connect to using Passive Station Side VoIP with SMS to record calls. All feature functionality and serviceability test cases were completed successfully with no issues or observations noted in **Section 2.2**.

10. Additional References

This section references the Avaya and NICE product documentation that are relevant to these Application Notes.

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

- [1] Administering Avaya Aura® Communication Manager, Document ID 03-300509
- [2] Avaya Aura® Communication Manager Feature Description and Implementation, Document ID 555-245-205
- [3] Avaya Aura® Application Enablement Services Administration and Maintenance Guide Release 7.0
- [4] Avaya Aura® Session Manager Overview, Doc # 03603323

Product documentation for NICE products may be found at: http://www.extranice.com/

Appendix

Avaya one-X® Agent Softphone

This is a printout of the Avaya one-X® Agent softphone used during compliance testing.

```
display station 2100
                                                                              Page 1 of 5
                                             STATION
     Type: 9630

Port: S00031

Name: one-X Agent1

Lock Messages? n
Security Code: *
Coverage Path 1:
Coverage Path 2:
Hunt-to Station:
Extension: 2100
                                                                                      BCC: 0
                                                                                       TN: 1
                                                                                       COR: 1
                                                                                       cos: 1
                                                                                   Tests? y
STATION OPTIONS
                Location: Time of Day Lock Table:
Loss Group: 19 Personalized Ringing Pattern: 1
        Speakerphone: 2-way Mute Button Enabled? y
Display Language: english Button Modules: 0
able GK Node Name:
                                                          Message Lamp Ext: 2100
 Survivable GK Node Name:
          Survivable COR: internal
                                                         Media Complex Ext:
   Survivable Trunk Dest? y
                                                                 IP SoftPhone? y
                                                         IP Video Softphone? n
                                    Short/Prefixed Registration Allowed: default
                                                        Customizable Labels? Y
```

```
display station 2100
                                                                                    2 of
                                                                            Page
                                           STATION
FEATURE OPTIONS
           LWC Reception: spe Auto Select Any Idle Appearance? n
LWC Activation? y Coverage Msg Retrieval? y
External Calls? n Auto Answer: no
 LWC Log External Calls? n
                                                                          Auto Answer: none
             CDR Privacy? n
                                                                     Data Restriction? n
Redirect Notification? y
Per Button Ring Control? n
Bridged Call Alerting? n
Active Station Division
                                         Idle Appearance Preference? n
Bridged Idle Line Preference? n
                                                           Restrict Last Appearance? y
  Active Station Ringing: single
                                                                   EMU Login Allowed? n
         H.320 Conversion? n Per Station CPN - Send Calling Number?
    Service Link Mode: as-needed
Multimedia Mode: enhanced
MWI Served User Type:
                                                                  EC500 State: enabled
                                                           Audible Message Waiting? n
                                                        Display Client Redirection? n
                                                       Select Last Used Appearance? n
                AUDIX Name:
                                                        Coverage After Forwarding? s
                                                            Multimedia Early Answer? n
 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
  Emergency Location Ext: 2100 Always Use? n IP Audio Hairpinning? n
```

```
display station 2100
                                                               Page
                                                                      3 of 5
                                    STATION
            Conf/Trans on Primary Appearance? n
   Bridged Appearance Origination Restriction? n
              Call Appearance Display Format: disp-param-default
                          IP Phone Group ID:
Enhanced Callr-Info Display for 1-Line Phones? n
                             ENHANCED CALL FORWARDING
                                      Forwarded Destination
                                                                   Active
Unconditional For Internal Calls To: 1000
                  External Calls To: 1000
                                                                       n
         Busy For Internal Calls To:
                                                                       n
                  External Calls To:
     No Reply For Internal Calls To:
                                                                       n
                  External Calls To:
           SAC/CF Override: n
```

```
display station 2100
                                                              Page
                                                                     4 of
                                    STATION
 SITE DATA
                                                      Headset? n
     Room:
      Jack:
                                                      Speaker? n
                                                     Mounting: d
     Cable:
     Floor:
                                                  Cord Length: 0
  Building:
                                                    Set Color:
ABBREVIATED DIALING
    List1:
                             List2:
                                                       List3:
BUTTON ASSIGNMENTS
                                       6: after-call
7: aux-
1: call-appr
                                                           Grp:
2: call-appr
                                                             Grp:
3: call-appr
                                       7: aux-work RC: Grp:
4: auto-in
                     Grp:
                                       8:
   voice-mail
```

Avaya 9608 H.323 Deskphone

This is a printout of the Avaya 9608 H.323 deskphone used during compliance testing.

```
Page 1 of 5
display station 2000
                                     STATION
                                       Lock Messages? n
Security Code: *
Coverage Path 1: 1
Extension: 2000
                                                                        BCC: 0
    Type: 9608
                                                                         TN: 1
                                                                       COR: 1
    Port: S00000
                                       Coverage Path 2:
    Name: Ext2000
                                       Hunt-to Station:
                                                                     Tests? y
STATION OPTIONS
                                           Time of Day Lock Table:
             Loss Group: 19 Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 2000
       Speakerphone: 2-way
Display Language: english
able GK Node Name:
                                              Mute Button Enabled? y
                                                   Button Modules: 0
Survivable GK Node Name:
         Survivable COR: internal Media Complex Ext:
  Survivable Trunk Dest? y
                                                      IP SoftPhone? y
                                               IP Video Softphone? n
                              Short/Prefixed Registration Allowed: yes
                                              Customizable Labels? y
```

```
display station 2000
                                                                Page 2 of 5
                                    STATION
FEATURE OPTIONS
         LWC Reception: spe
LWC Activation? y
External Calls? n
                                   Auto Select Any Idle Appearance? n
                                                   Coverage Msg Retrieval? y
 LWC Log External Calls? n
                                                              Auto Answer: none
           CDR Privacy? n
                                                         Data Restriction? n
                                             Idle Appearance Preference? n
  Redirect Notification? y
 Per Button Ring Control? n
                                           Bridged Idle Line Preference? n
  Bridged Call Alerting? n
                                                 Restrict Last Appearance? y
 Active Station Ringing: single
                                                        EMU Login Allowed? n
       H.320 Conversion? n Per Station CPN - Send Calling Number?
      H.320 Conversion: n

Service Link Mode: as-needed EC500 State: enabled Audible Message Waiting? n
                                           Display Client Redirection? n
   MWI Served User Type: sip-adjunct
                                              Select Last Used Appearance? n
                                               Coverage After Forwarding? s
                                                  Multimedia Early Answer? n
Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
 Emergency Location Ext: 2000 Always Use? n IP Audio Hairpinning? n
```

```
display station 2000
                                                                 Page
                                                                        3 of 5
                                     STATION
             Conf/Trans on Primary Appearance? n
   Bridged Appearance Origination Restriction? n
                                                   Offline Call Logging? y
         Require Mutual Authentication if TLS? n
               Call Appearance Display Format: disp-param-default
                           IP Phone Group ID:
Enhanced Callr-Info Display for 1-Line Phones? n
                              ENHANCED CALL FORWARDING
                                      Forwarded Destination
                                                                    Active
 Unconditional For Internal Calls To:
                  External Calls To:
                                                                        n
         Busy For Internal Calls To:
                                                                        n
                  External Calls To:
                                                                        n
     No Reply For Internal Calls To:
                                                                        n
                  External Calls To:
                                                                        n
            SAC/CF Override: n
```

display station 2000			Page	4 of	5			
STATION								
SITE DATA								
Room:		Headset?	n					
Jack:		Speaker?	n					
Cable:	Mounting:							
Floor:	Cord Length:							
Building:	Set Color:							
ABBREVIATED DIALING								
List1:	List2:	List3:						
BUTTON ASSIGNMENTS								
1: call-appr	5: call-park							
2: call-appr 6:								
3: call-appr	7:							
4: extnd-call	8:							
3. 3. 3. 3. 3. 3. 3. 3. 3. 3. 3. 3. 3. 3								
voice-mail								

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