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 DMCC Service Observation and Single Step Conference - 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.

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. NICE Engage Platform uses either Communication Manager's Service Observation feature or Conference feature via the Application Enablement Services (AES) Device, Media, and Call Control (DMCC) interface and the Telephony Services API (TSAPI) to capture the audio and call details for call recording on various Communication Manager endpoints, listed in **Section 4**.

Device Media Call Control (DMCC) works by allowing software vendors to create soft phones, in memory on a recording server, and use them to monitor and record other phones. This is purely a software solution and does not require telephony boards or any wiring beyond a typical network infrastructure.

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 the Communication Manager feature "Service Observe" to observe an extension on a call; this way the call is recorded and can be played back at a later time. NICE can also conference into the call and record the call using this method. Both methods of call recording use virtual stations on Communication Manager in order to observe or conference into existing calls to record them.

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 DMCC Service Observation and Single Step Conference 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 Recording did not include use of any specific encryption features as requested by NICE.

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.
- **EC500 Calls/Forwarded calls** - Test call recording for calls terminated on Avaya DECT handsets using EC500.
- **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

Most functionality and serviceability test cases were completed successfully. The following observations were noted.

- On occasion when calling into a SIP phone the beginning (2-3 secs) of the call is missed, probably due to the timing of the single step conference (SSC) with a SIP phone.
- An issue was observed when a SIP phone made a supervised transfer using all monitored phones where the "Service Observer" was not dropped from the call upon completion of the transfer. The call scenario is A calls to B (SIP Phone) and B then transfers A to C. A and C are now talking, when A hangs up the display on C changes to that of the "virtual station" and same if C hangs up first then the display of A shows the "virtual station". The call is cleared then when the second person (A or C) hangs up their phone. This is not the same if a H.323 makes the transfer. This issue appears when using Service Observation where all extensions are being observed constantly. Avaya are investigating this issue.

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 DMCC Service Observe and Single Step Conference to record calls. The NICE Application Server is setup for DMCC Service Observe and Single Step Conference mode and connects to the AES.

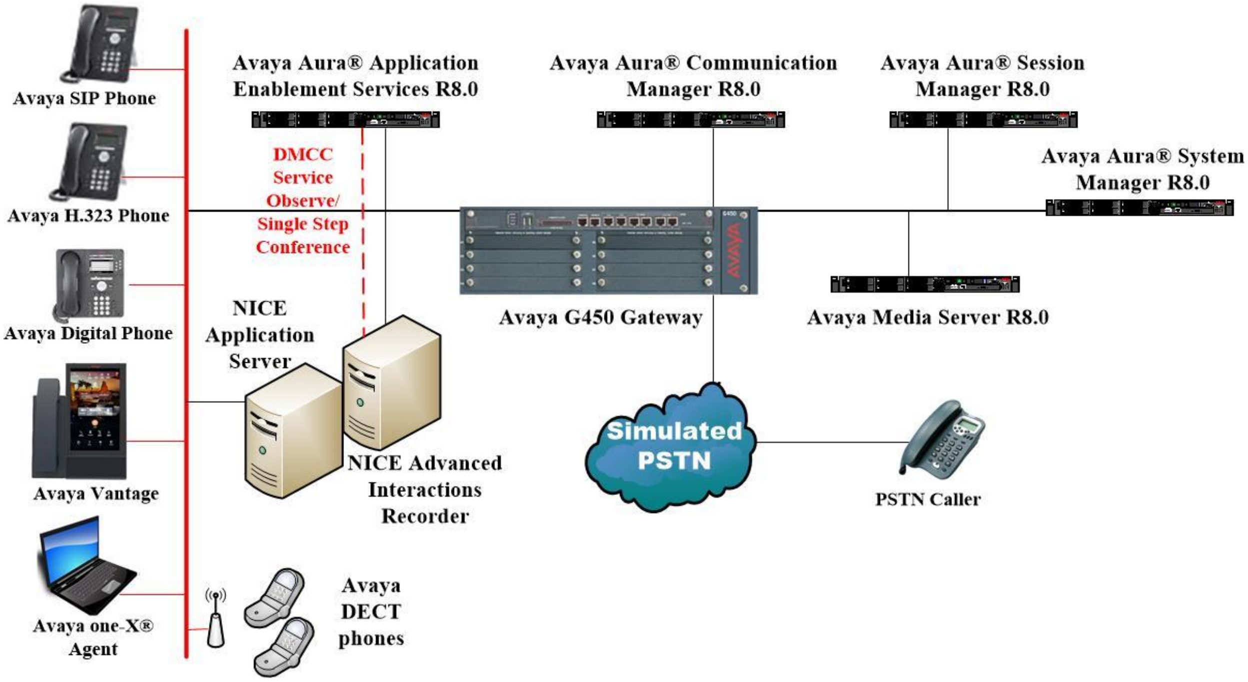

Avaya SIP Phone
Avaya H.323 Phone
Avaya Digital Phone
Avaya Vantage
Avaya one-X® Agent
Avaya Aura® Application Enablement Services R8.0
DMCC Service Observe/ Single Step Conference
NICE Application Server
NICE Advanced Interactions Recorder
Avaya DECT phones
Avaya Aura® Communication Manager R8.0
Avaya G450 Gateway
Simulated PSTN
Avaya Aura® Session Manager R8.0
Avaya Aura® System Manager R8.0
Avaya Media Server R8.0
PSTN Caller

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 Aura® Media Server running on a Virtual Server	8.0.0.150
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	1.0.0.0.0.43
Avaya Vantage Equinox	1.0.0.2
Avaya 9408 Digital Deskphone	2.0
Avaya one-X® Agent	2.5.8
Avaya DECT Handsets	3725 DH4 (R3.3.11) 3720 DH3 (R3.3.11)
NICE Engage Platform - 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          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? y                      DCS (Basic)? y
          ASAI Link Core Capabilities? n                DCS Call Coverage? y
          ASAI Link Plus Capabilities? n               DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                          DS1 MSP? y
                                 ATMS? y            DS1 Echo Cancellation? y
                  Attendant Vectoring? y
```

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

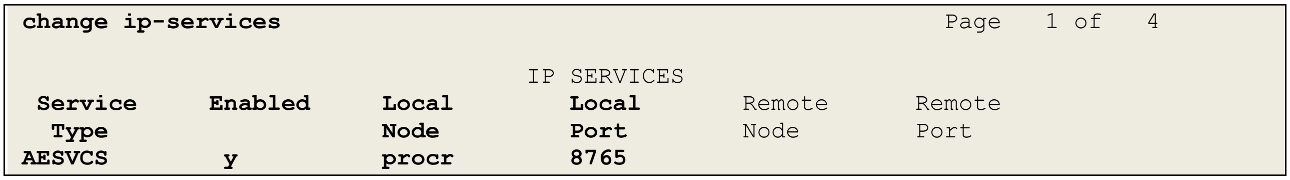
Display the procr IP address by using the command **display node-names ip** and note the IP address for the **procr**.

```
display node-names 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**.

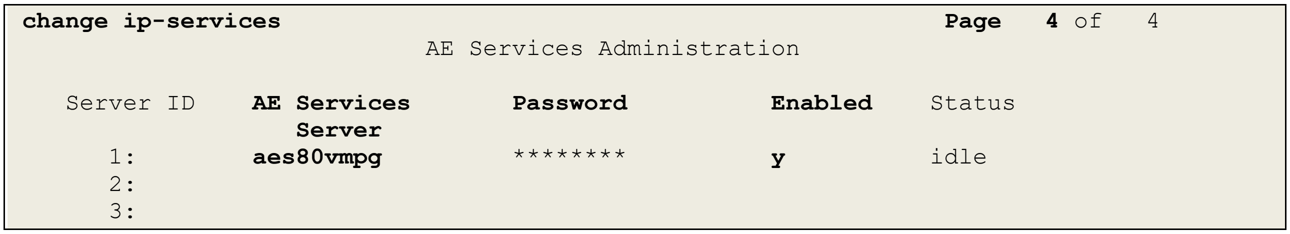


```
change ip-services                                            Page   1 of   4

                                          IP SERVICES
 Service     Enabled     Local        Local       Remote      Remote
  Type                   Node         Port        Node        Port
AESVCS         y         procr        8765
```

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 **y**.

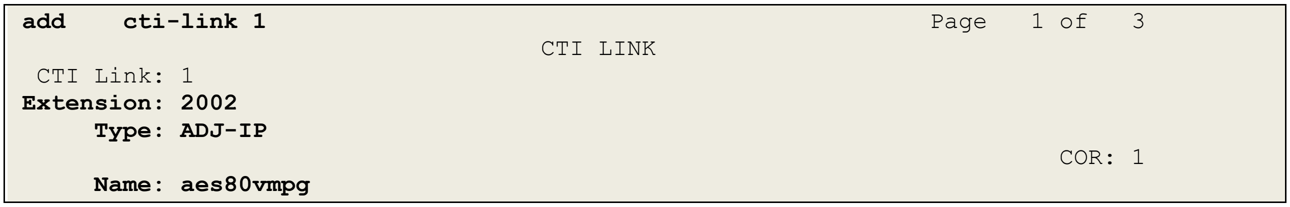
Note: The password entered for **Password** field must match the password on the AES server in **Section 6.2**. The **AE Services Server** should match the administered name for the AES server; this is created as part of the AES installation, and can be obtained from the AES server by typing **uname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



```
change ip-services                                            Page   4 of   4
                            AE Services Administration

   Server ID    AE Services        Password          Enabled    Status
                   Server
      1:        aes80vmpg          ********          y          idle
      2:
      3:
```

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 Communication Manager for Service Observation

Type display cor x, where x is the COR number in the screen above, to check the existing Class of Restriction. Ensure that **Can be Service Observed** is set to **y**, if not type **change cor x** to make a change to the Class or Restriction. This value needs to be enabled in order for Service Observe to work for call recording.

```
display cor 1                                                   Page 1 of 23
                             CLASS OF RESTRICTION
               COR Number: 1
          COR Description:

                      FRL: 0                                APLT? y
  Can Be Service Observed? y           Calling Party Restriction: all-toll
Can Be A Service Observer? y            Called Party Restriction: none
        Time of Day Chart: 1       Forced Entry of Account Codes? n
         Priority Queuing? n                Direct Agent Calling? y
     Restriction Override: all        Facility Access Trunk Test? n
     Restricted Call List? n                 Can Change Coverage? n
   Unrestricted Call List: 1
            Access to MCT? y            Fully Restricted Service? n
Group II Category For MFC: 7            Hear VDN of Origin Annc.? n
         Send ANI for MFE? n             Add/Remove Agent Skills? n
            MF ANI Prefix:              Automatic Charge Display? n
Hear System Music on Hold? y   PASTE (Display PBX Data on Phone)? n
                        Can Be Picked Up By Directed Call Pickup? y
                                    Can Use Directed Call Pickup? y
                                    Group Controlled Restriction: inactive
```

Type **change system-parameters features**, on **Page 11** ensure that **Allow Two Observes in Same Call** is set to **y**.

```
change system-parameters features                              Page  11 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
  EAS
         Expert Agent Selection (EAS) Enabled? y
        Minimum Agent-LoginID Password Length:
          Direct Agent Announcement Extension:                   Delay:
    Message Waiting Lamp Indicates Status For: station

  VECTORING
                    Converse First Data Delay: 0      Second Data Delay: 2
               Converse Signaling Tone (msec): 100         Pause (msec): 70
                     Prompting Timeout (secs): 10
                 Interflow-qpos EWT Threshold: 2
    Reverse Star/Pound Digit For Collect Step? n
          Available Agent Adjustments for BSR? n
                            BSR Tie Strategy: 1st-found
   Store VDN Name in Station's Local Call Log? n
  SERVICE OBSERVING
              Service Observing: Warning Tone? y     or Conference Tone? n
 Service Observing/SSC Allowed with Exclusion? n
             Allow Two Observers in Same Call? y
```

Type **change feature-access-codes** to access the feature codes on Communication Manager. Scroll to **Page 5** in order to view or change the **Service Observing** access codes. Note the **Service Observing Listen Only Access Code** is **#43**; this will be required in **Section 7.1** during the setup of NICE Engage Platform.

```
change feature-access-codes                                     Page   5 of  10
                               FEATURE ACCESS CODE (FAC)
                                 Call Center Features
  AGENT WORK MODES
                           After Call Work Access Code: #36
                                    Assist Access Code:
                                   Auto-In Access Code: #38
                                  Aux Work Access Code: #39
                                     Login Access Code: #40
                                    Logout Access Code: #41
                                 Manual-in Access Code: #42
  SERVICE OBSERVING
             Service Observing Listen Only Access Code: #43
             Service Observing Listen/Talk Access Code: #44
                 Service Observing No Talk Access Code:
   Service Observing Next Call Listen Only Access Code:
 Service Observing by Location Listen Only Access Code:
 Service Observing by Location Listen/Talk Access Code:

  AACC CONFERENCE MODES
                     Restrict First Consult Activation:      Deactivation:
                    Restrict Second Consult Activation:      Deactivation:
```

5.6. Configure H323 Stations for Service Observation

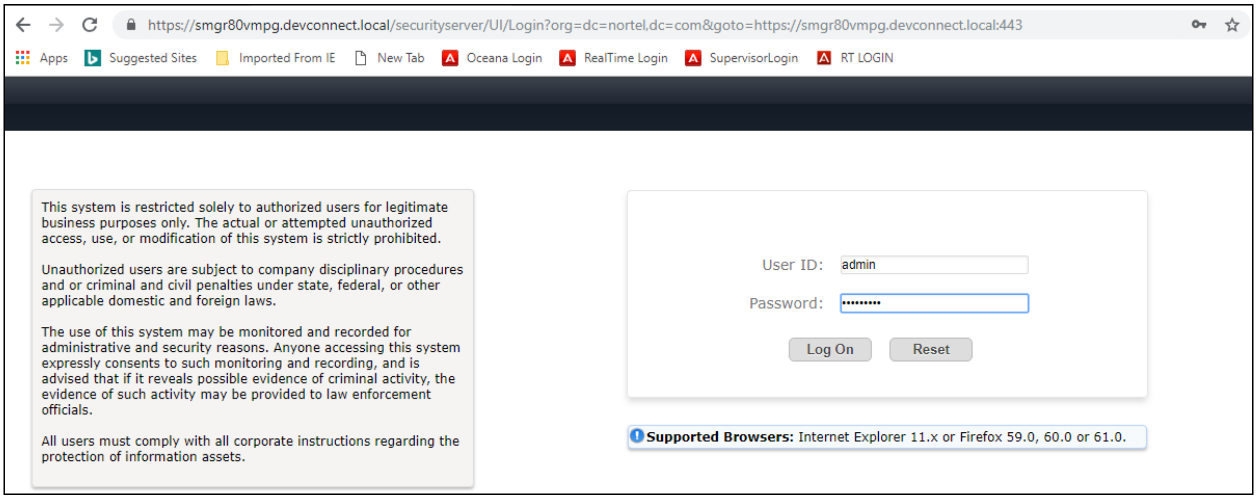
All endpoints that are to be monitored by NICE will need to have IP Softphone set to y. IP Softphone must be enabled in order for DMCC Service Observe and Single Step Conference to work. Type **change station x** where x is the extension number of the station to be monitored also note this extension number for configuration required in **Section 7.1.** Note the **Security Code** and ensure that **IP SoftPhone** is set to **y**.

```
change station x                                                    Page   1 of   6
                                     STATION

Extension: x                             Lock Messages? n               BCC: 0
     Type: 9608                        Security Code: 1234              TN: 1
     Port: S00101                      Coverage Path 1:                 COR: 1
     Name: Extension                   Coverage Path 2:                 COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 1591
            Speakerphone: 2-way                Mute Button Enabled? y
        Display Language: english
 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
```

5.7. Configure SIP Stations for Service Observation

Any SIP extension that is to be recorded requires some configuration changes to allow call recording using service observation. Changes of SIP phones on Communication Manager must be carried out from System Manager. Access the System Manager using a web browser by entering **http://<FQDN >/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager or **http://<IP Address >/SMGR**. Log in using appropriate credentials.

Note: The following shows changes a SIP extension and assumes that the SIP extension has been programmed correctly and is fully functioning.


https://smgr80vmpg.devconnect.local/securityserver/UI/Login?org=dc=nortel,dc=com&goto=https://smgr80vmpg.devconnect.local:443
Apps
Suggested Sites
Imported From IE
New Tab
Oceana Login
RealTime Login
SupervisorLogin
RT LOGIN
Log On
Reset

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

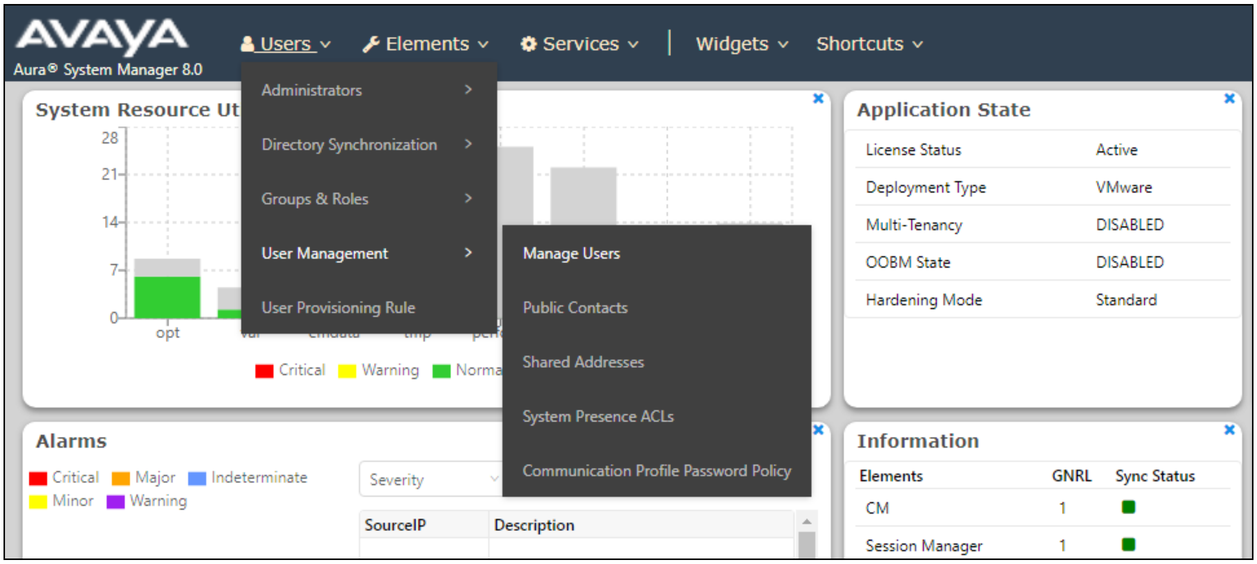
All users must comply with all corporate instructions regarding the protection of information assets.

User ID: admin

Password: •••••••••

Supported Browsers: Internet Explorer 11.x or Firefox 59.0, 60.0 or 61.0.

From the home page click on **Users → User Management → Manage Users** as highlighted below.



Application State

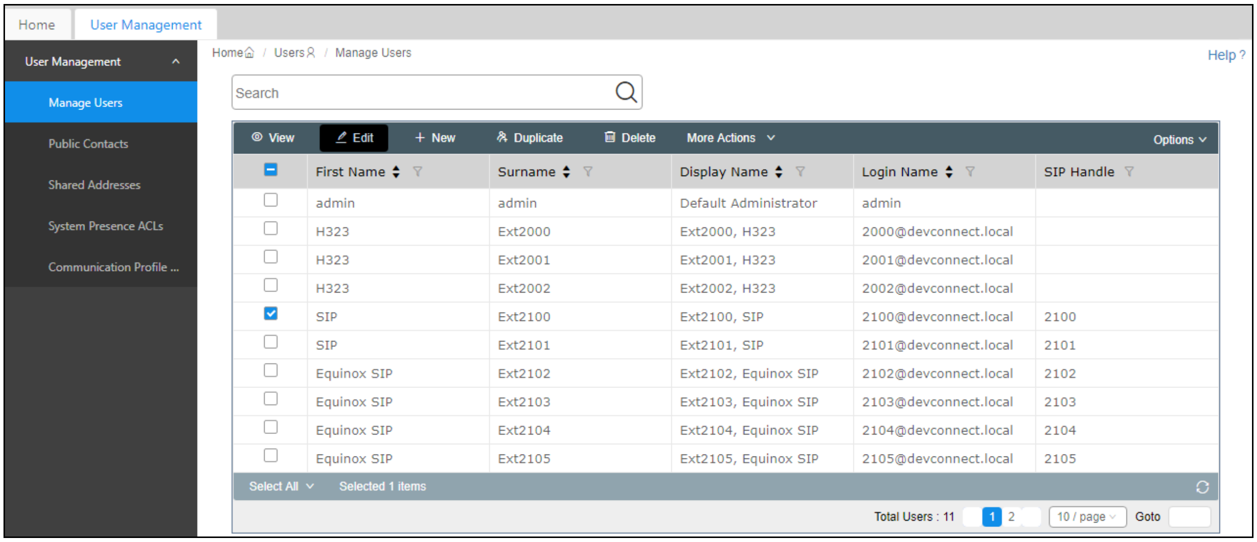
License Status	Active
Deployment Type	VMware
Multi-Tenancy	DISABLED
OOBM State	DISABLED
Hardening Mode	Standard

SourceIP	Description

Information

Elements	GNRL	Sync Status
CM	1	🟩
Session Manager	1	🟩

Select the station to be edited and click on **Edit**. The example below shows that SIP extension **2100** is selected.



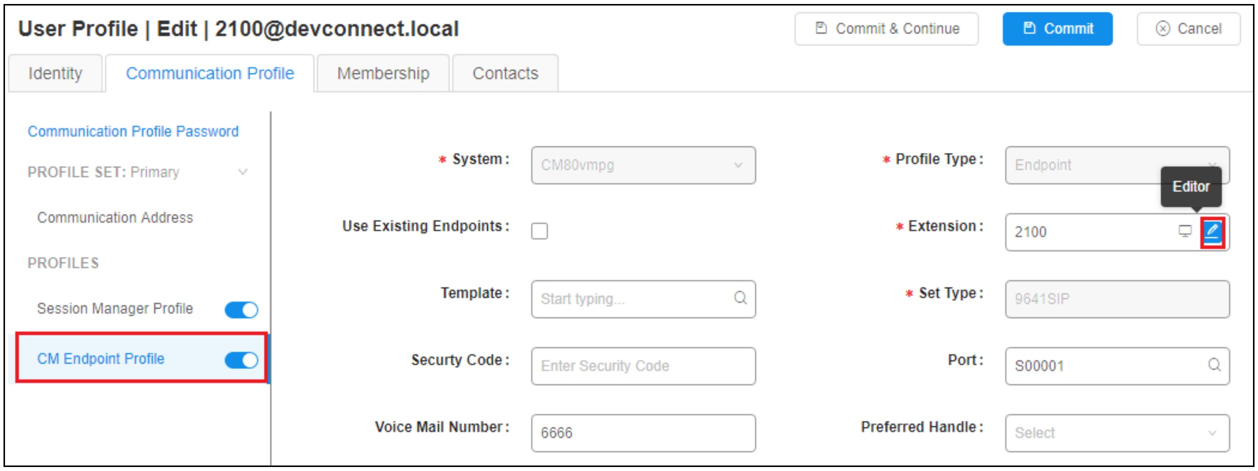
☐	First Name	Surname	Display Name	Login Name	SIP Handle
☐	admin	admin	Default Administrator	admin	
☐	H323	Ext2000	Ext2000, H323	2000@devconnect.local	
☐	H323	Ext2001	Ext2001, H323	2001@devconnect.local	
☐	H323	Ext2002	Ext2002, H323	2002@devconnect.local	
☑	SIP	Ext2100	Ext2100, SIP	2100@devconnect.local	2100
☐	SIP	Ext2101	Ext2101, SIP	2101@devconnect.local	2101
☐	Equinox SIP	Ext2102	Ext2102, Equinox SIP	2102@devconnect.local	2102
☐	Equinox SIP	Ext2103	Ext2103, Equinox SIP	2103@devconnect.local	2103
☐	Equinox SIP	Ext2104	Ext2104, Equinox SIP	2104@devconnect.local	2104
☐	Equinox SIP	Ext2105	Ext2105, Equinox SIP	2105@devconnect.local	2105

Select All Selected 1 items

Total Users : 11 1 2 10 / page Goto

To set the password for the SIP extension click on **Communication Profile Password** in the left window and set the password in the main window (not shown here).

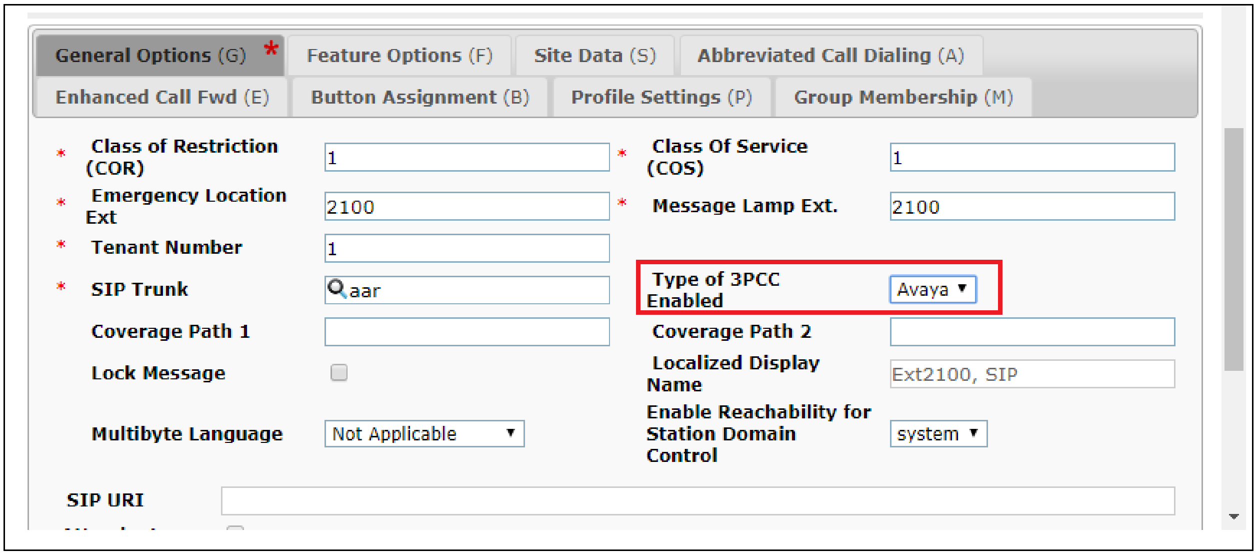
Click on the **CM Endpoint Profile** in the left window. Click on the **Editor** icon in the main window.



User Profile | Edit | 2100@devconnect.local

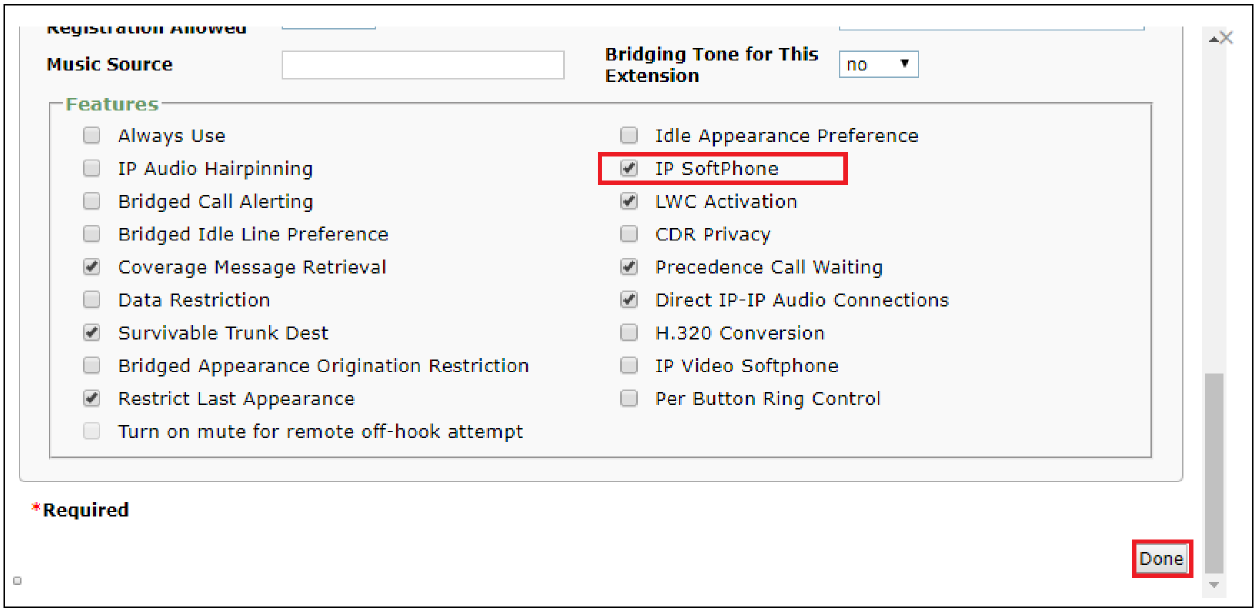
* System :	CM80vmpg	* Profile Type :	Endpoint
Use Existing Endpoints :	☐	* Extension :	2100
Template :	Start typing...	* Set Type :	9641SIP
Securty Code :	Enter Security Code	Port :	S00001
Voice Mail Number :	6666	Preferred Handle :	Select

Ensure that **Type of 3PCC Enabled** is set to **Avaya**. Click on the **Feature Options** tab after that. Ensure that both the **Class of Restriction (COR)** and the **Class of Service (COS)** are set correctly.



* Class of Restriction (COR)	1	* Class Of Service (COS)	1
* Emergency Location Ext	2100	* Message Lamp Ext.	2100
* Tenant Number	1		
* SIP Trunk	aar	Type of 3PCC Enabled	Avaya ▾
Coverage Path 1		Coverage Path 2	
Lock Message	□	Localized Display Name	Ext2100, SIP
Multibyte Language	Not Applicable ▾	Enable Reachability for Station Domain Control	system ▾
SIP URI			

Under Feature Options, scroll down and ensure that **IP Softphone** is ticked as shown. Click on **Done**.



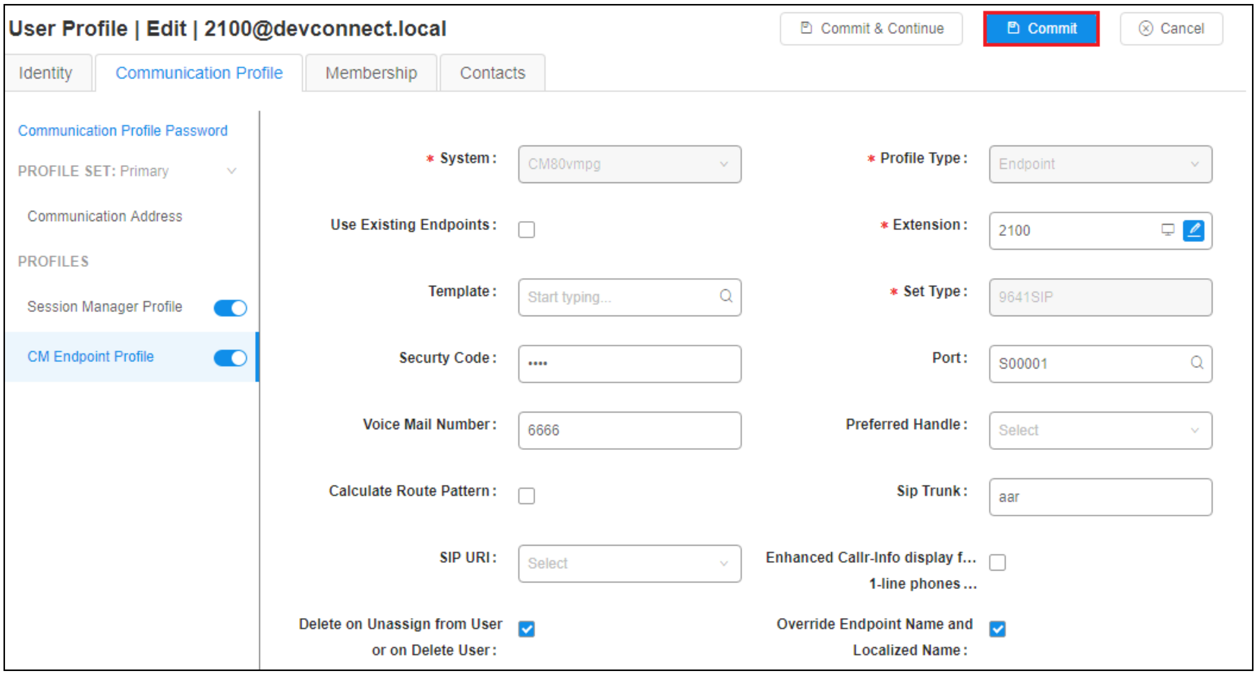
Registration Allowed			
Music Source		Bridging Tone for This Extension	no ▾

Features

- □ Always Use
- □ IP Audio Hairpinning
- □ Bridged Call Alerting
- □ Bridged Idle Line Preference
- ☑ Coverage Message Retrieval
- □ Data Restriction
- ☑ Survivable Trunk Dest
- □ Bridged Appearance Origination Restriction
- ☑ Restrict Last Appearance
- □ Turn on mute for remote off-hook attempt
- □ Idle Appearance Preference
- ☑ IP SoftPhone
- ☑ LWC Activation
- □ CDR Privacy
- ☑ Precedence Call Waiting
- ☑ Direct IP-IP Audio Connections
- □ H.320 Conversion
- □ IP Video Softphone
- □ Per Button Ring Control

*Required

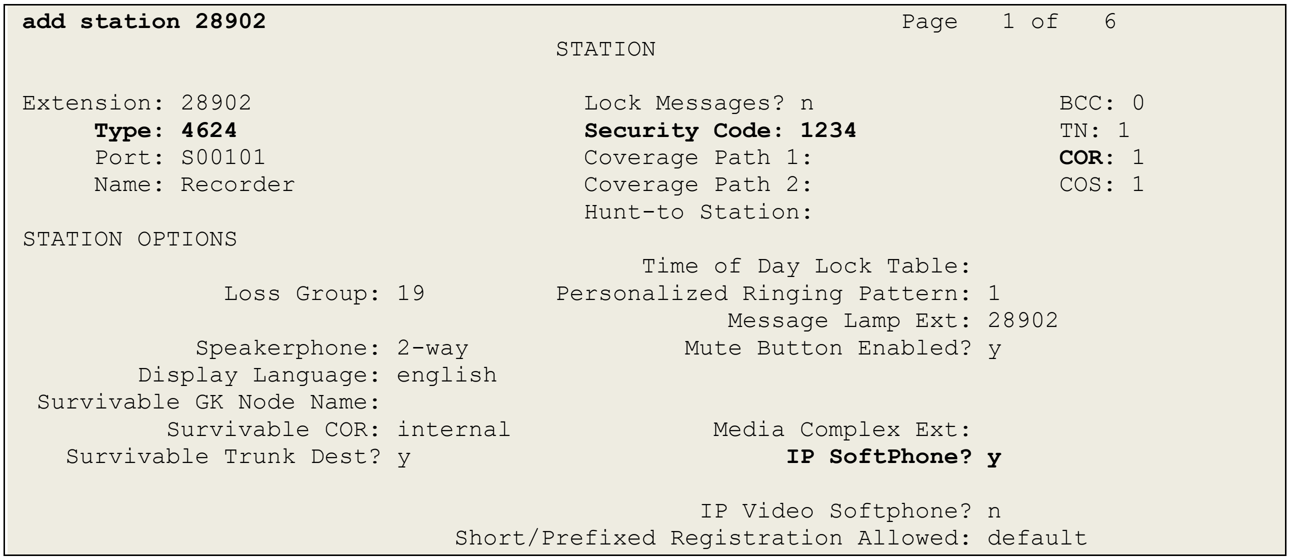
Click on **Commit**, as shown.



* System :	CM80vmpg	* Profile Type :	Endpoint
Use Existing Endpoints :	□	* Extension :	2100
Template :	Start typing...	* Set Type :	9641SIP
Securty Code :	••••	Port :	S00001
Voice Mail Number :	6666	Preferred Handle :	Select
Calculate Route Pattern :	□	Sip Trunk :	aar
SIP URI :	Select	Enhanced Callr-Info display f... 1-line phones ...	□
Delete on Unassign from User or on Delete User :	☑	Override Endpoint Name and Localized Name :	☑

5.8. Configure Virtual Stations for Single Step Conference and Service Observation

Add virtual stations to allow NICE Engage Platform record calls using Single Step Conference and Service Observe. Type **add station x** where x is the extension number of the station to be configured also note this extension number for configuration required in **Section 7.1**. Note the **Security Code** and ensure that **IP SoftPhone** is set to **y**. Note also the **COR** for the stations, this will be set to that configured in **Section 5.5**.



```
add station 28902                                            Page   1 of   6
                                     STATION

Extension: 28902                       Lock Messages? n                BCC: 0
     Type: 4624                        Security Code: 1234              TN: 1
     Port: S00101                      Coverage Path 1:                COR: 1
     Name: Recorder                    Coverage Path 2:                COS: 1
                                       Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 28902
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english
 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
```

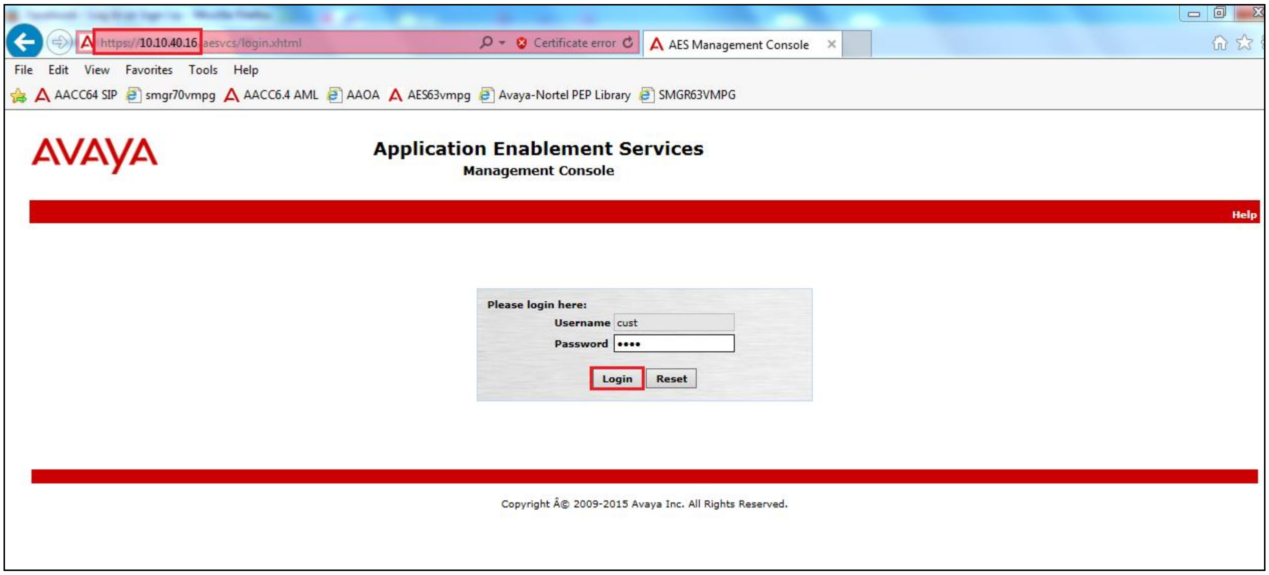
6. Configure Avaya Aura® Application Enablement Services

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

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.


https://10.10.40.16/aesvcs/login.xhtml
Certificate error
AES Management Console
File Edit View Favorites Tools Help
AACC64 SIP smgr70vmpg AACC6.4 AML AAOA AES63vmpg Avaya-Nortel PEP Library SMGR63VMPG
AVAYA
Help
Login
Reset

Application Enablement Services
Management Console

Please login here:
Username cust
Password ••••

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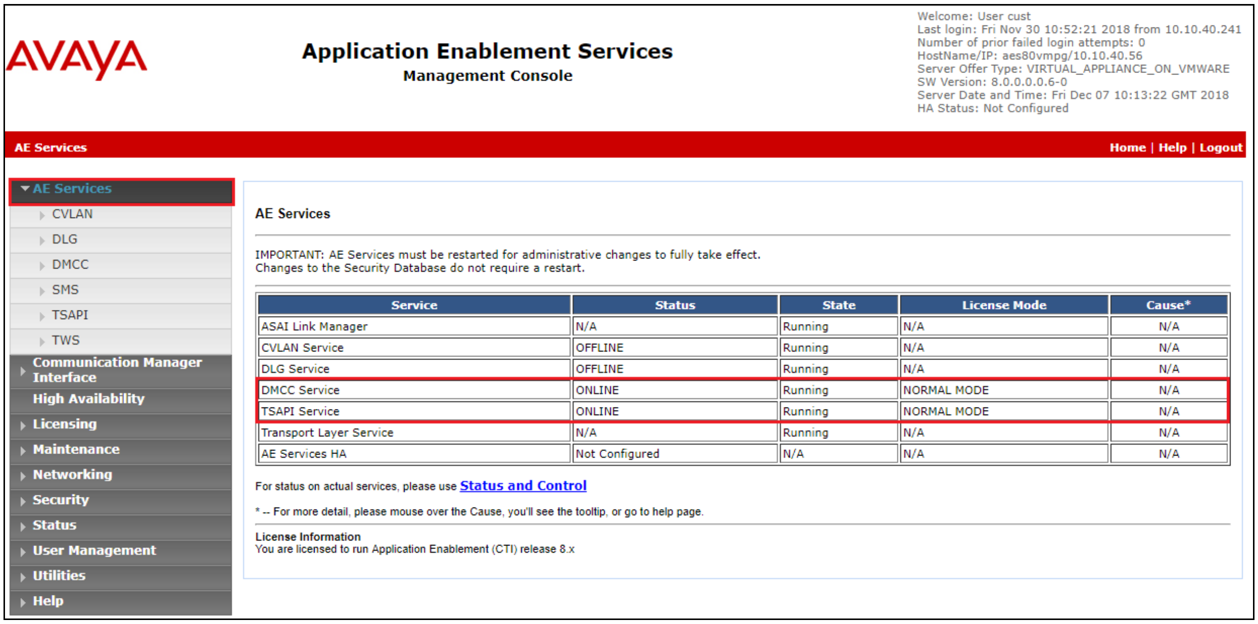
The Application Enablement Services Management Console appears displaying the **Welcome to OAM** screen (not shown). Select **AE Services** and verify that the **TSAPI Service** and **DMCC Service** are licensed by ensuring that the **License Mode** is showing **NORMAL MODE**. If not, contact an Avaya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



Welcome: User cust
Last login: Fri Nov 30 10:52:21 2018 from 10.10.40.241
Number of prior failed login attempts: 0
HostName/IP: aes80vmpg/10.10.40.56
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 8.0.0.0.0.6-0
Server Date and Time: Fri Dec 07 10:13:22 GMT 2018
HA Status: Not Configured

AE Services

IMPORTANT: AE Services must be restarted for administrative changes to fully take effect.
Changes to the Security Database do not require a restart.

Service	Status	State	License Mode	Cause*
ASAI Link Manager	N/A	Running	N/A	N/A
CVLAN Service	OFFLINE	Running	N/A	N/A
DLG Service	OFFLINE	Running	N/A	N/A
DMCC Service	ONLINE	Running	NORMAL MODE	N/A
TSAPI Service	ONLINE	Running	NORMAL MODE	N/A
Transport Layer Service	N/A	Running	N/A	N/A
AE Services HA	Not Configured	N/A	N/A	N/A

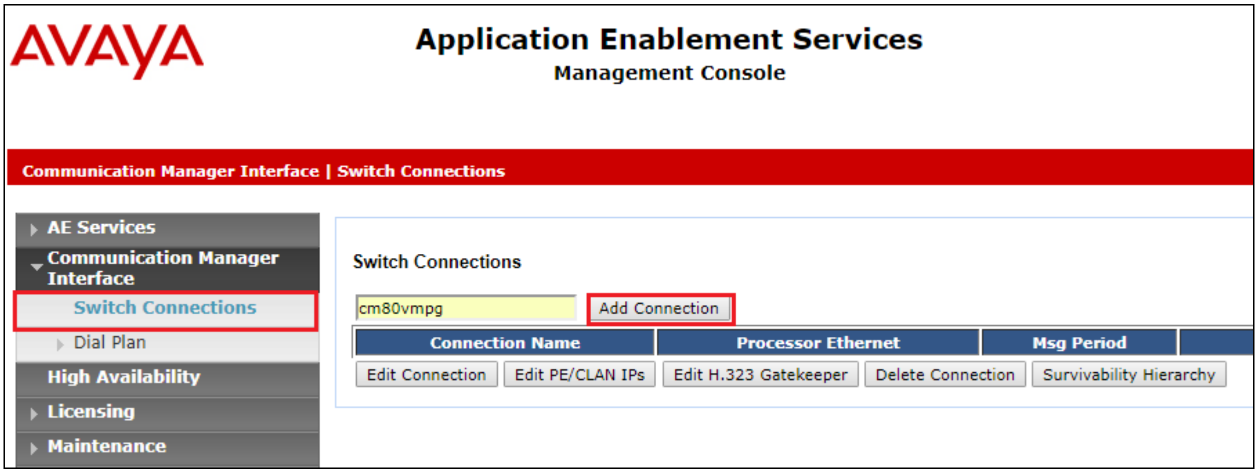
For status on actual services, please use Status and Control

* -- For more detail, please mouse over the Cause, you'll see the tooltip, or go to help page.

License Information
You are licensed to run Application Enablement (CTI) release 8.x

6.2. Create Switch Connection

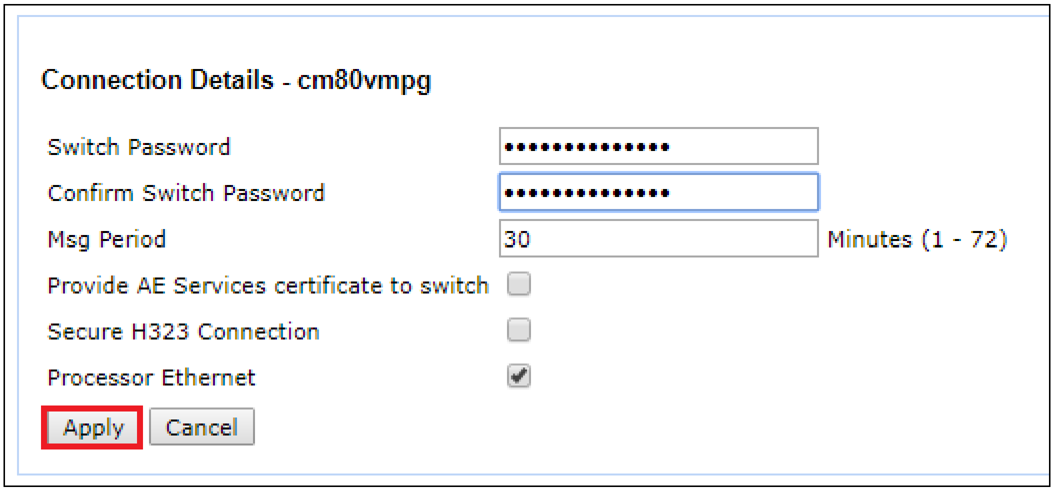
From the AES Management Console navigate to **Communication Manager Interface → Switch Connections** to set up a switch connection. Enter a name for the Switch Connection to be added and click the **Add Connection** button.



Switch Connections

cm80vmpg

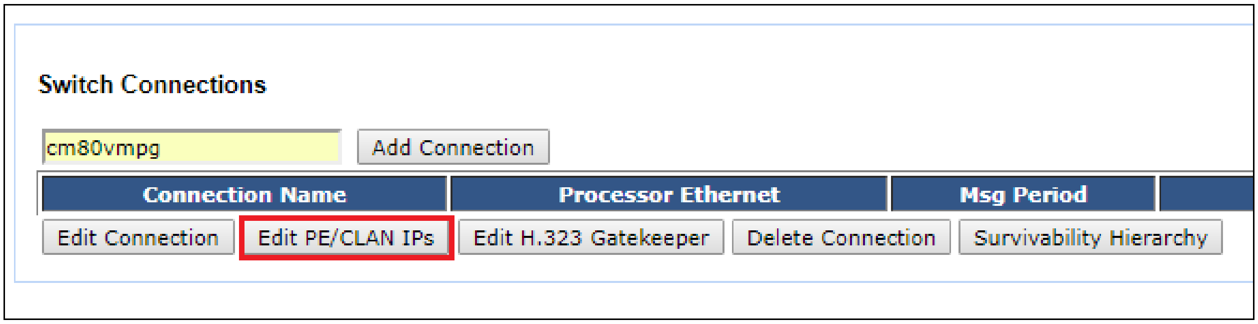
In the resulting screen enter the **Switch Password**; the Switch Password must be the same as that entered into Communication Manager AE Services Administration screen via the **change ip-services** command, described in **Section 5.3**. Default values may be accepted for the remaining fields. Click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



Connection Details - cm80vmpg

Switch Password	••••••••••••••
Confirm Switch Password	••••••••••••••
Msg Period	30 Minutes (1 - 72)
Provide AE Services certificate to switch	□
Secure H323 Connection	□
Processor Ethernet	☑

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

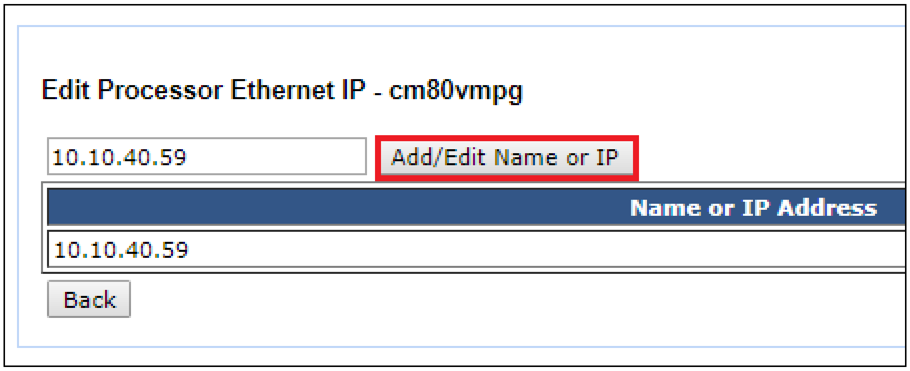


Switch Connections

cm80vmpg

Connection Name	Processor Ethernet	Msg Period

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.



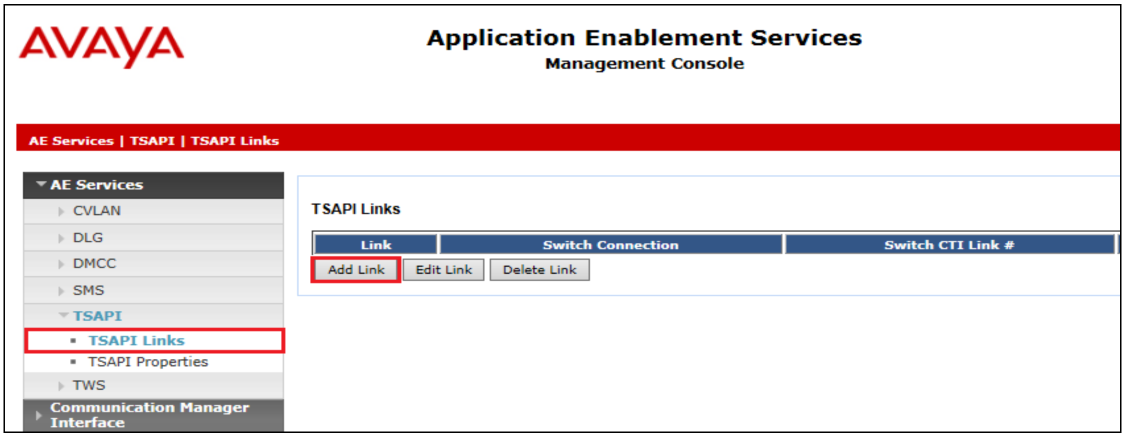
Edit Processor Ethernet IP - cm80vmpg

10.10.40.59

Name or IP Address
10.10.40.59

6.3. Administer TSAPI link

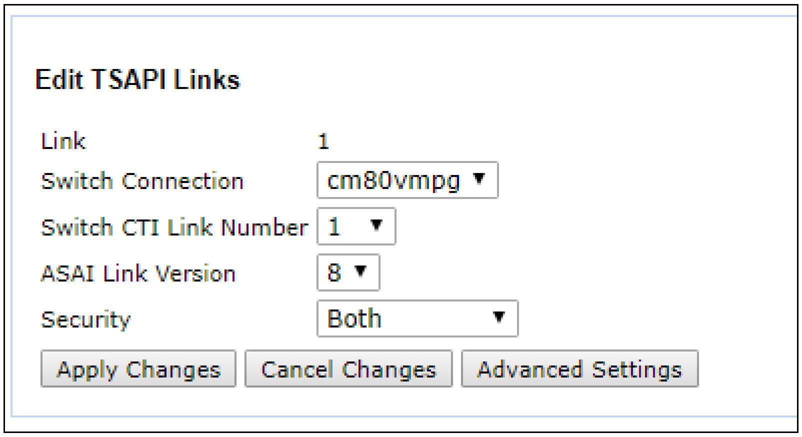
From the Application Enablement Services Management Console, select **AE Services → TSAPI → TSAPI Links**. Select **Add Link** button as shown in the screen below.



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:

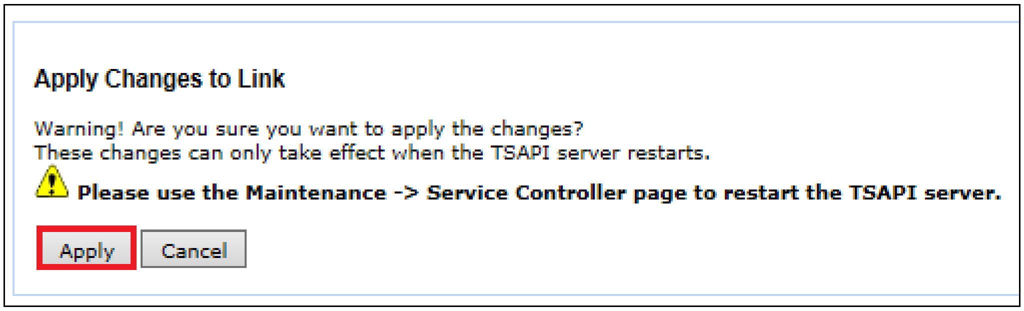
- **Link:** Use the drop-down list to select an unused link number.
- **Switch Connection:** Choose the switch connection **cm80vmpg**, which has already been configured in **Section 6.2** from the drop-down list.
- **Switch CTI Link Number:** Corresponding CTI link number configured in **Section 5.4** 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**.


Edit TSAPI Links

Link	1
Switch Connection	cm80vmpg ▼
Switch CTI Link Number	1 ▼
ASAI Link Version	8 ▼
Security	Both ▼

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

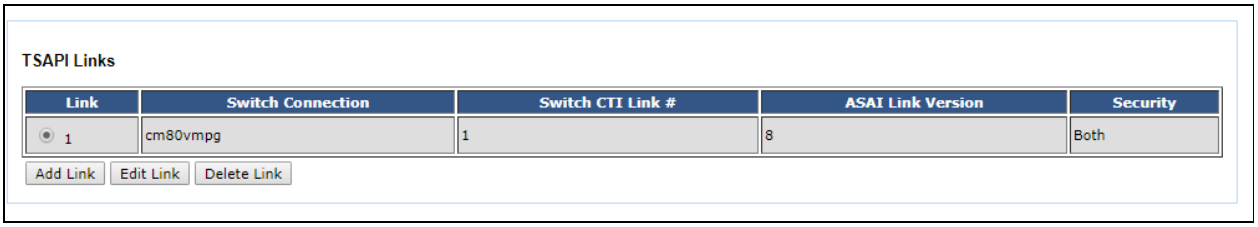


Apply Changes to Link

Warning! Are you sure you want to apply the changes?
These changes can only take effect when the TSAPI server restarts.

Please use the Maintenance -> Service Controller page to restart the TSAPI server.

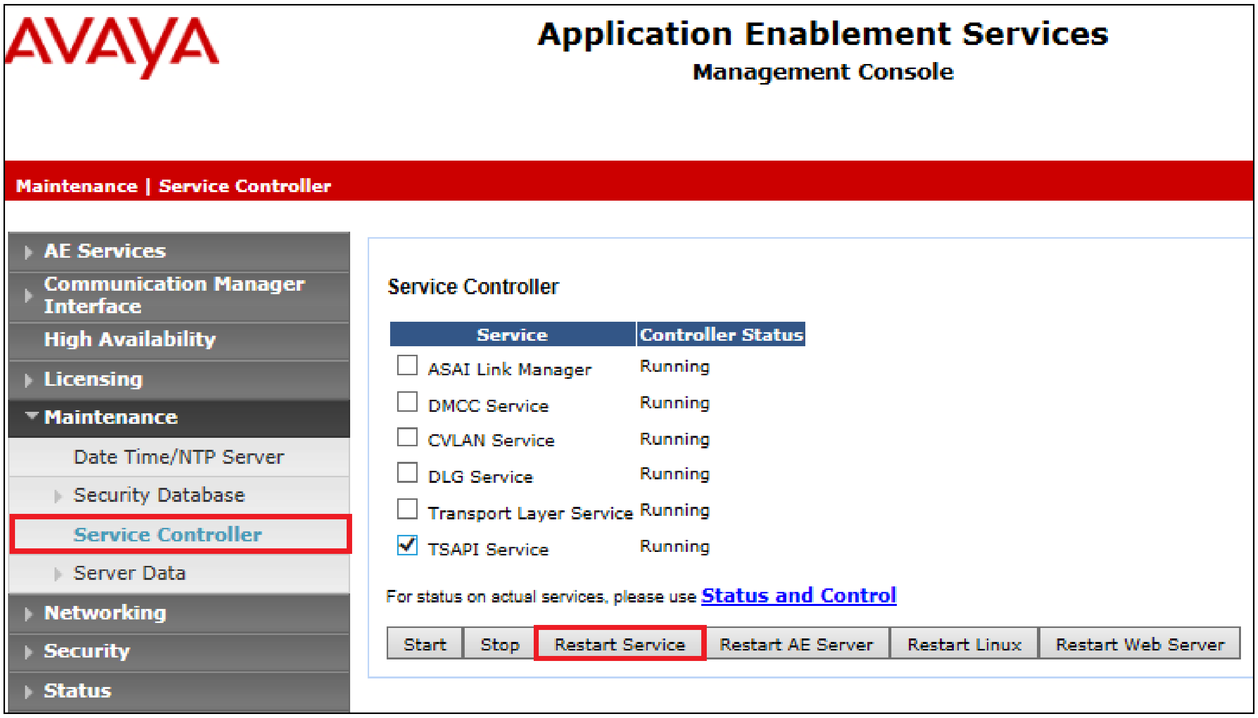
When the TSAPI Link is completed, it should resemble the screen below.



TSAPI Links

Link	Switch Connection	Switch CTI Link #	ASAI Link Version	Security
● 1	cm80vmpg	1	8	Both

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**.



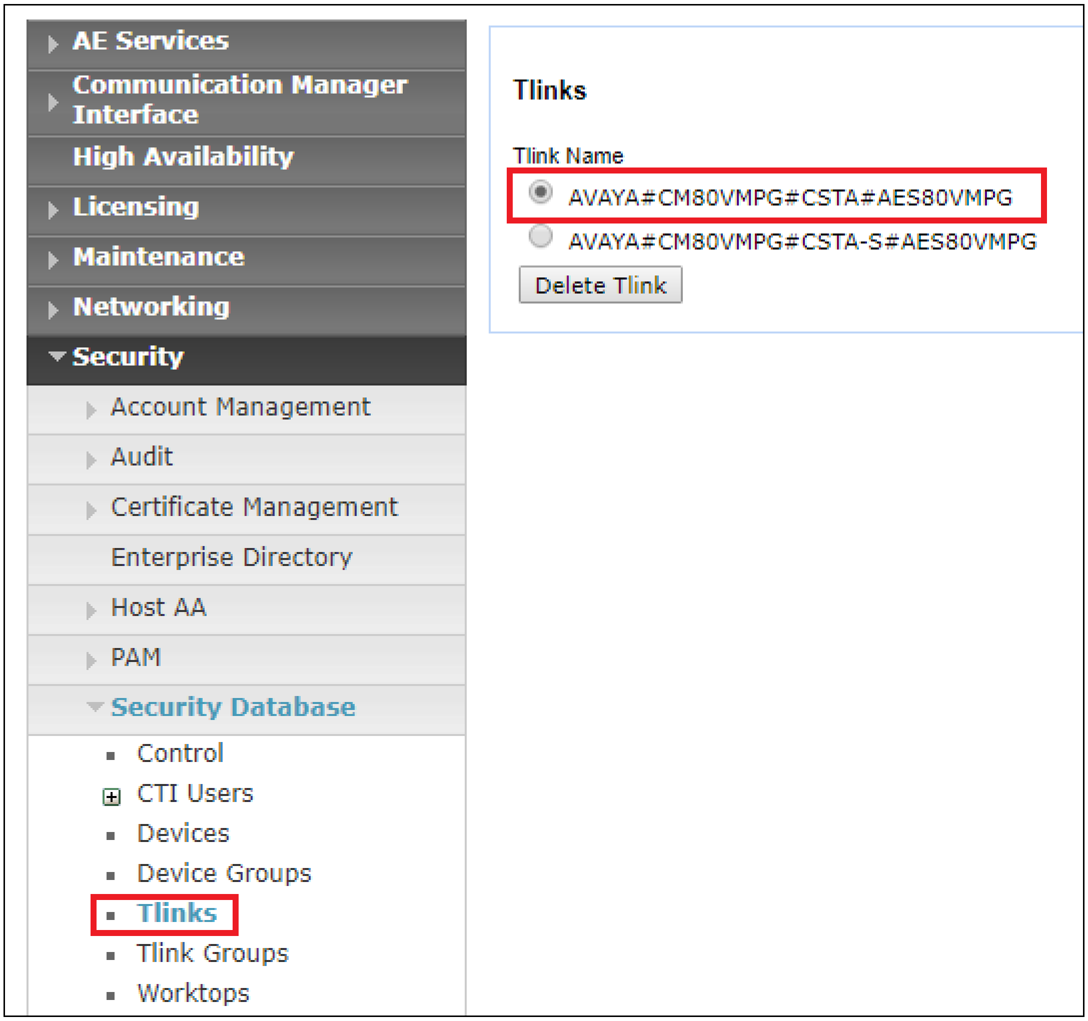
Service Controller

Service	Controller Status
□ ASAI Link Manager	Running
□ DMCC Service	Running
□ CVLAN Service	Running
□ DLG Service	Running
□ Transport Layer Service	Running
☑ TSAPI Service	Running

For status on actual services, please use **Status and Control**

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**.



Tlinks

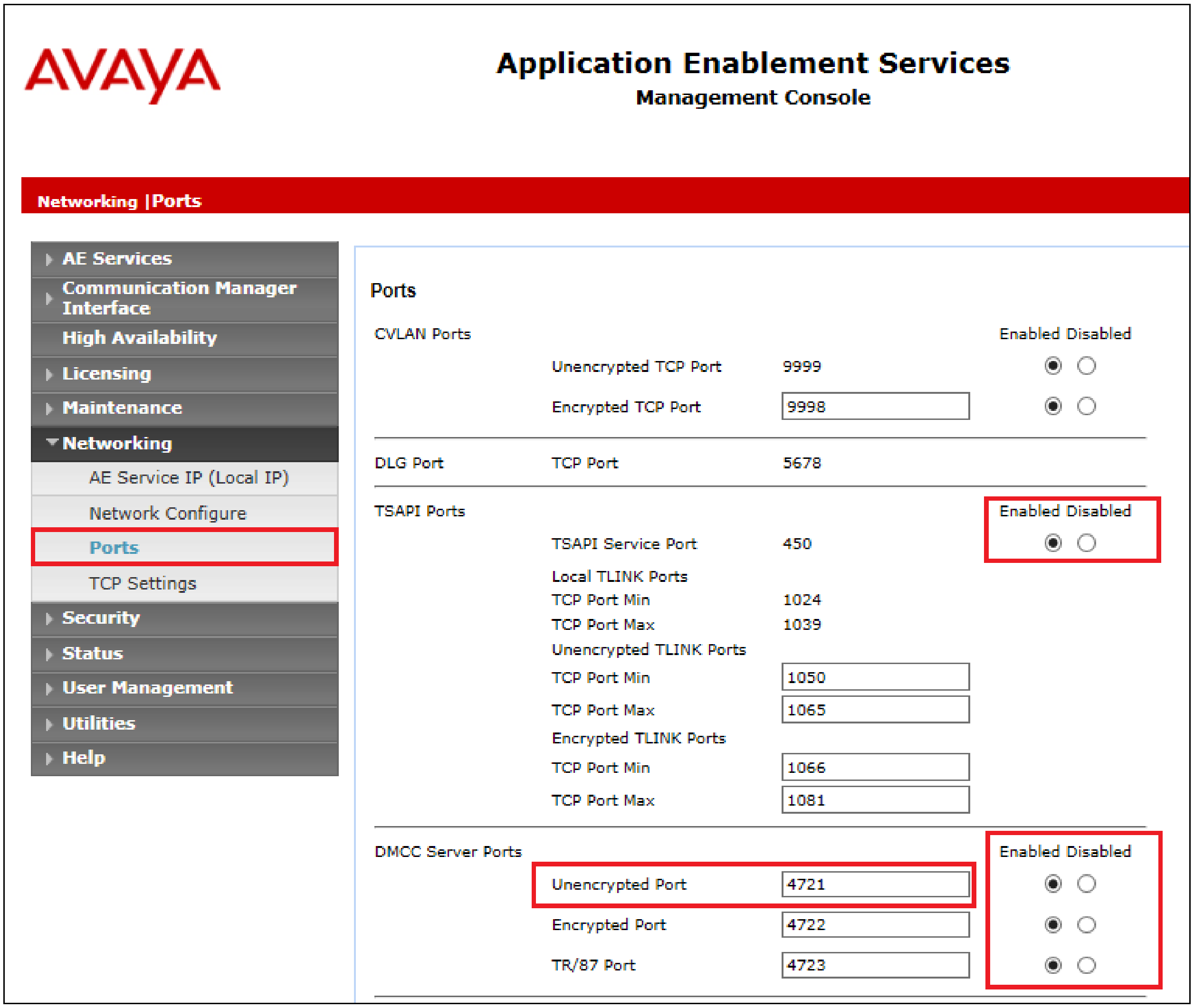
Tlink Name

● AVAYA#CM80VMPG#CSTA#AES80VMPG

○ AVAYA#CM80VMPG#CSTA-S#AES80VMPG

6.5. Configure Networking Ports

To ensure that TSAPI ports are enabled, navigate to **Networking → Ports**. Ensure that the TSAPI ports are set to **Enabled** as shown below. Ensure that the **DMCC Server Ports** are also **Enabled** and take note of the **Unencrypted Port 4721** which will be used later in **Section 7.1**.

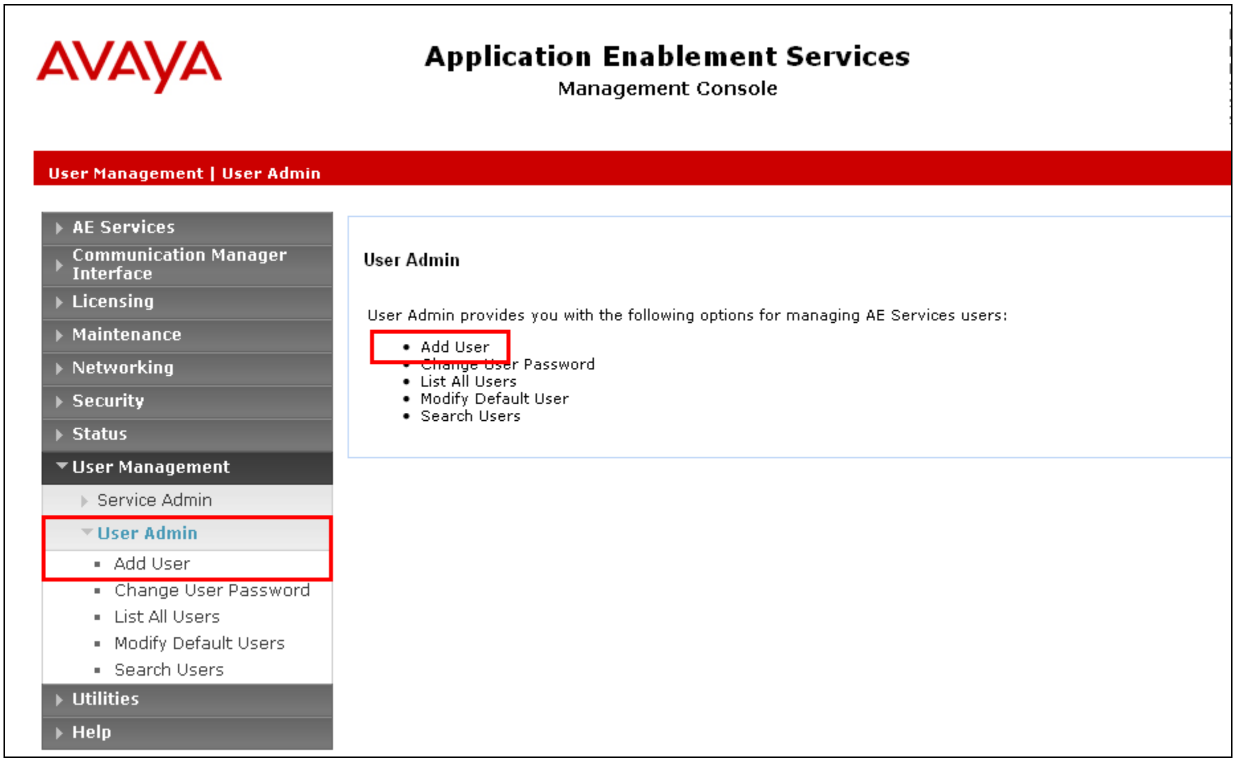


Ports

			Enabled	Disabled
CVLAN Ports				
	Unencrypted TCP Port	9999	●	○
	Encrypted TCP Port	9998	●	○
DLG Port	TCP Port	5678		
TSAPI Ports			Enabled	Disabled
	TSAPI Service Port	450	●	○
	Local TLINK Ports			
	TCP Port Min	1024		
	TCP Port Max	1039		
	Unencrypted TLINK Ports			
	TCP Port Min	1050		
	TCP Port Max	1065		
	Encrypted TLINK Ports			
	TCP Port Min	1066		
	TCP Port Max	1081		
DMCC Server Ports			Enabled	Disabled
	Unencrypted Port	4721	●	○
	Encrypted Port	4722	●	○
	TR/87 Port	4723	●	○

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.



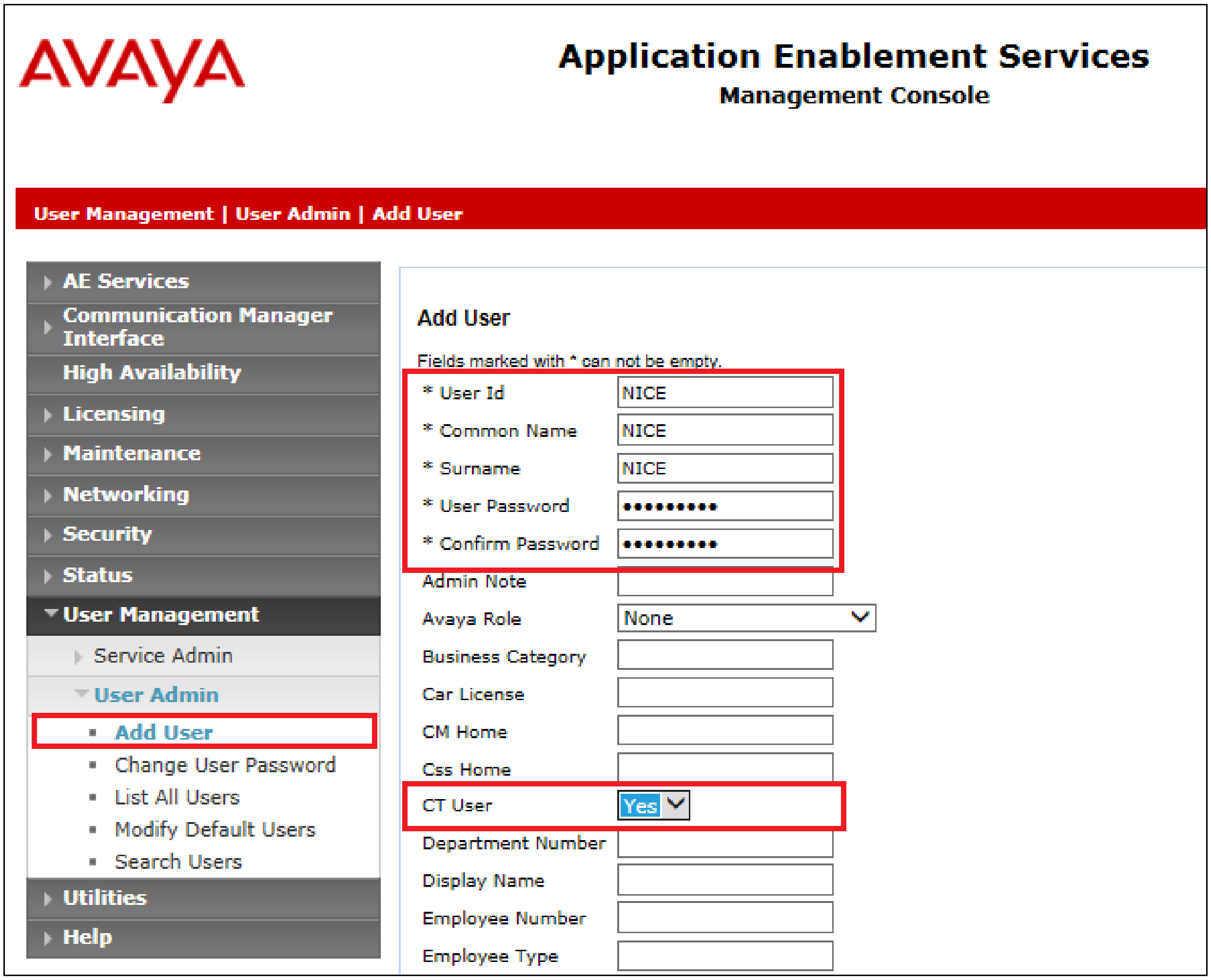
User Admin

User Admin provides you with the following options for managing AE Services users:

- Add User
- Change User Password
- List All Users
- Modify Default User
- Search Users

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.



Add User

Fields marked with * can not be empty.

* User Id	NICE
* Common Name	NICE
* Surname	NICE
* User Password	•••••••••
* Confirm Password	•••••••••
Admin Note	
Avaya Role	None
Business Category	
Car License	
CM Home	
Css Home	
CT User	Yes
Department Number	
Display Name	
Employee Number	
Employee Type	

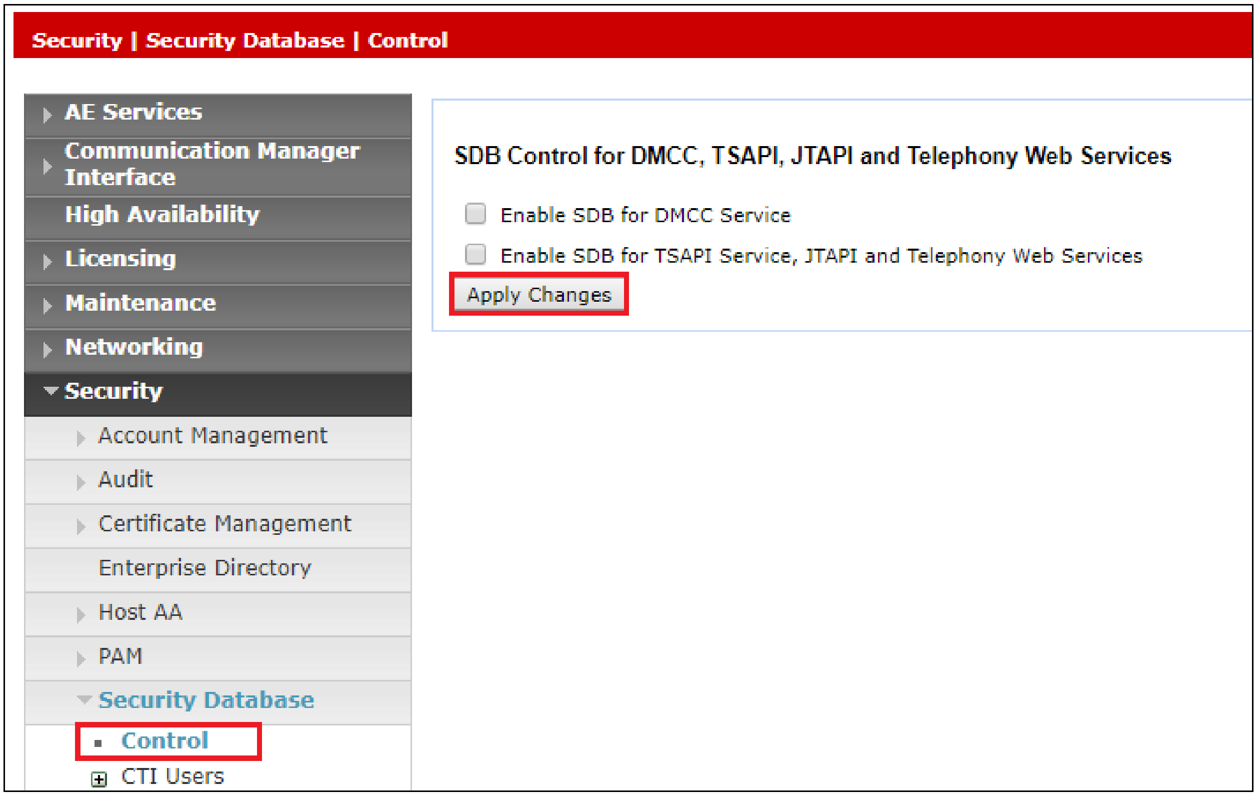
Scroll down and click on **Apply Changes** (not shown).

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.



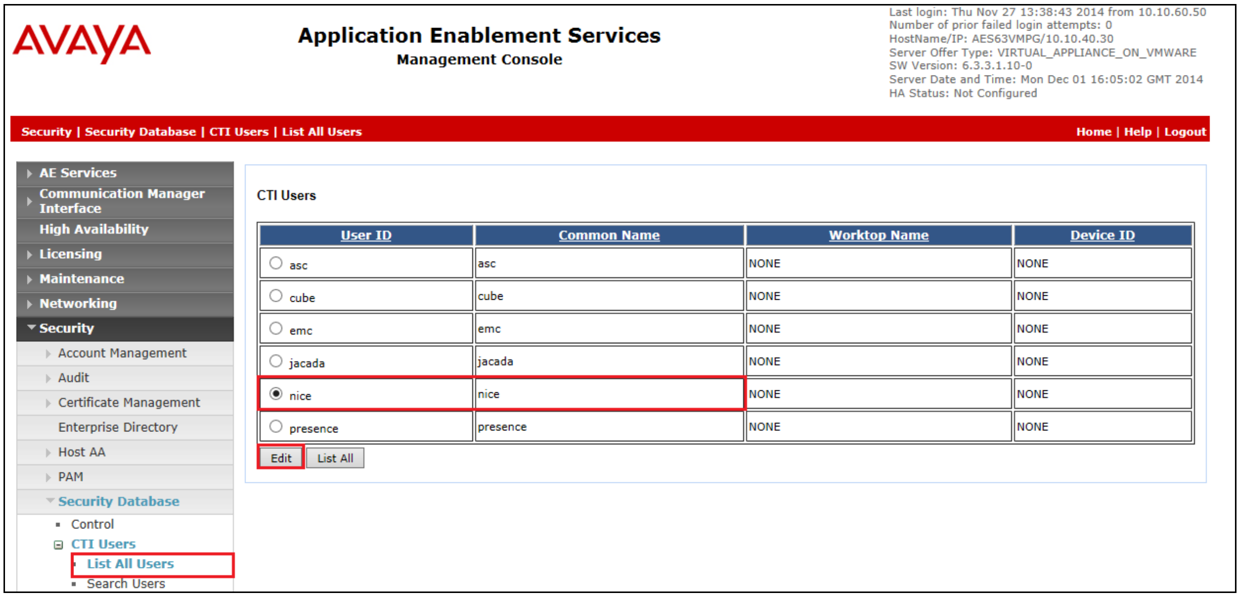
SDB Control for DMCC, TSAPI, JTAPI and Telephony Web Services

☐ Enable SDB for DMCC Service

☐ Enable SDB for TSAPI Service, JTAPI and Telephony Web Services

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**.

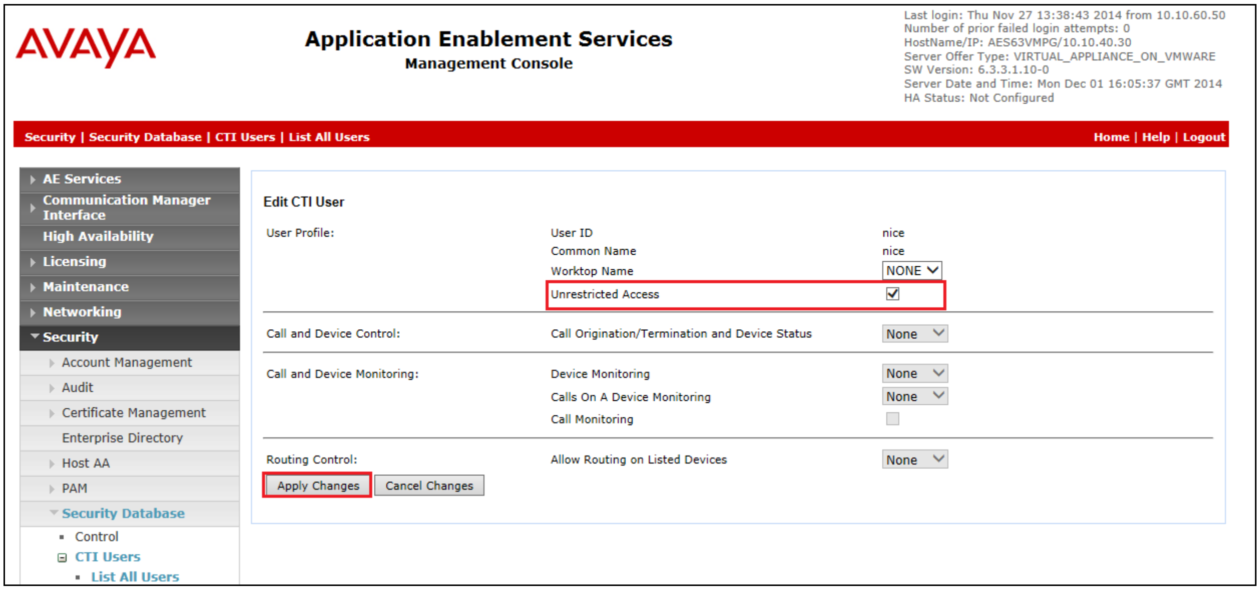


Last login: Thu Nov 27 13:38:43 2014 from 10.10.60.50
Number of prior failed login attempts: 0
HostName/IP: AES63VMPG/10.10.40.30
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 6.3.3.1.10-0
Server Date and Time: Mon Dec 01 16:05:02 GMT 2014
HA Status: Not Configured

CTI Users

User ID	Common Name	Worktop Name	Device ID
○ asc	asc	NONE	NONE
○ cube	cube	NONE	NONE
○ emc	emc	NONE	NONE
○ jacada	jacada	NONE	NONE
● nice	nice	NONE	NONE
○ presence	presence	NONE	NONE

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



Last login: Thu Nov 27 13:38:43 2014 from 10.10.60.50
Number of prior failed login attempts: 0
HostName/IP: AES63VMPG/10.10.40.30
Server Offer Type: VIRTUAL_APPLIANCE_ON_VMWARE
SW Version: 6.3.3.1.10-0
Server Date and Time: Mon Dec 01 16:05:37 GMT 2014
HA Status: Not Configured

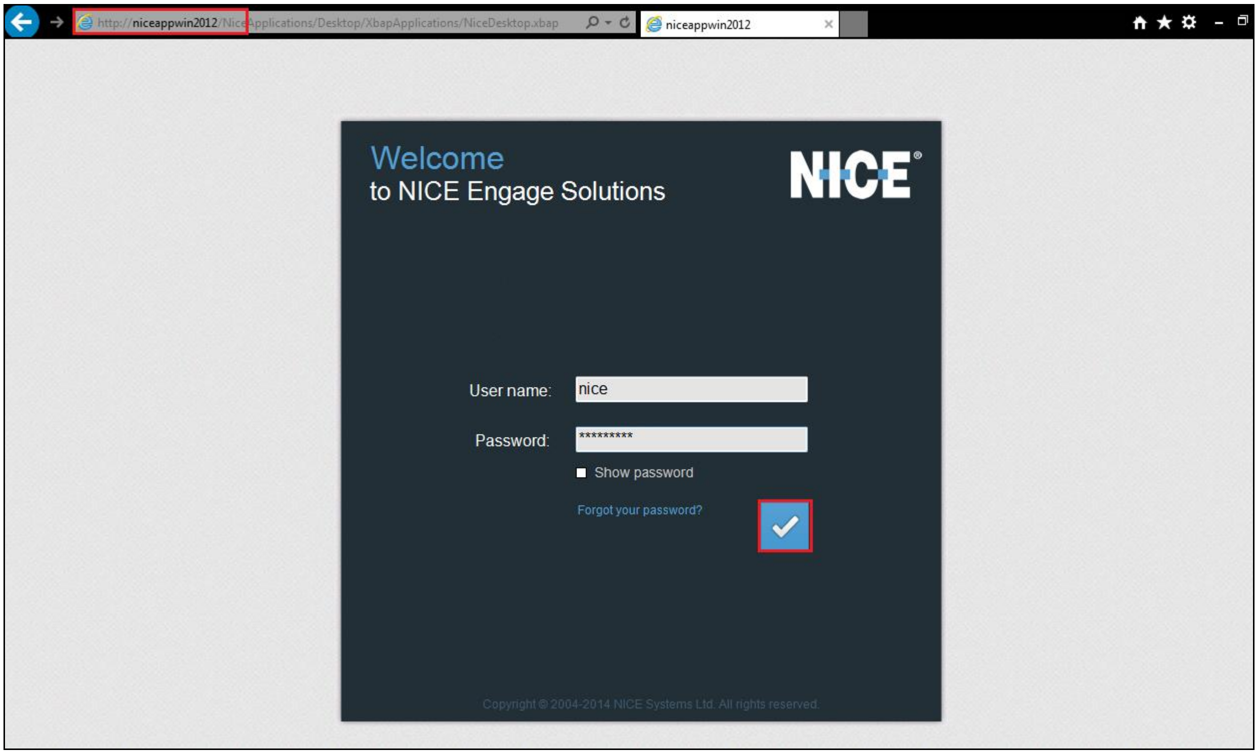
Edit CTI User

User Profile:	User ID	nice
	Common Name	nice
	Worktop Name	NONE
	Unrestricted Access	☑
Call and Device Control:	Call Origination/Termination and Device Status	None
Call and Device Monitoring:	Device Monitoring	None
	Calls On A Device Monitoring	None
	Call Monitoring	□
Routing Control:	Allow Routing on Listed Devices	None

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



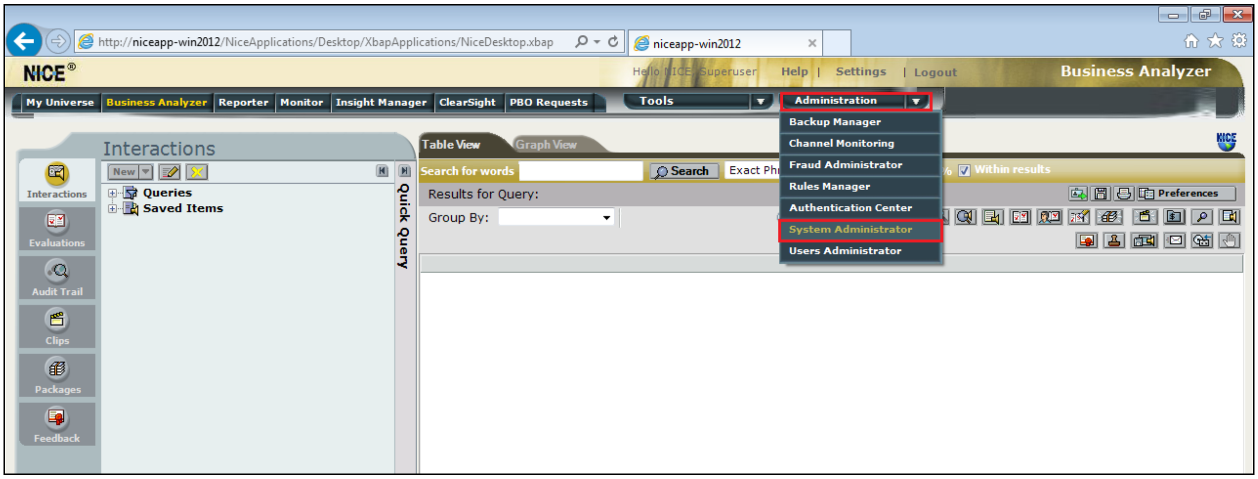
http://niceappwin2012/Nice/Applications/Desktop/XbapApplications/NiceDesktop.xbap

User name: nice

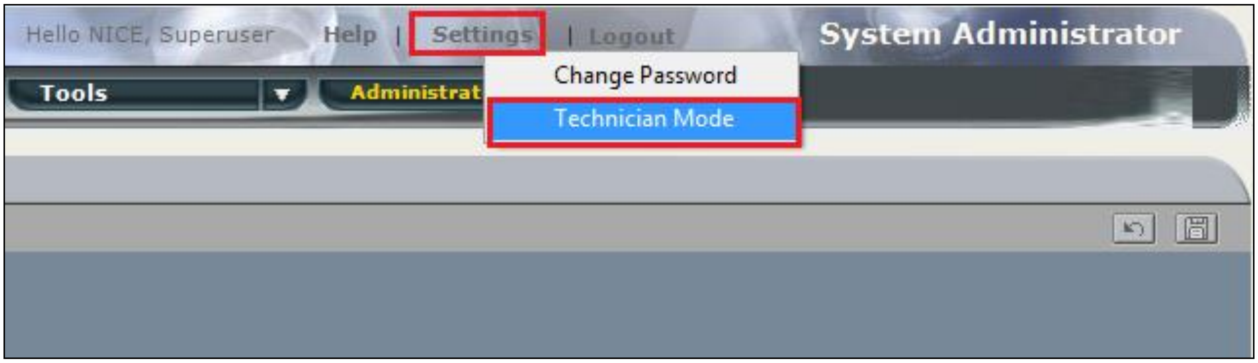
Password: *********

□ Show password
Forgot your password?
✓
Copyright © 2004-2014 NICE Systems Ltd. All rights reserved.

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

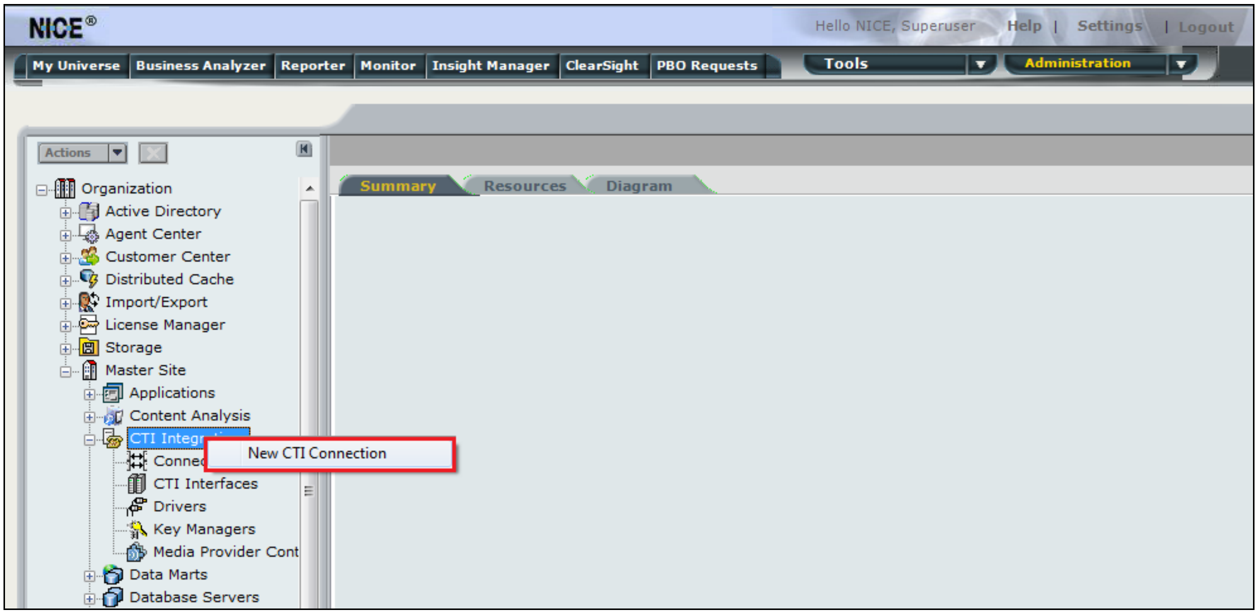


Before any changes can be made, switch to **Technician Mode** by clicking into **Settings** at the top of the screen as shown below.

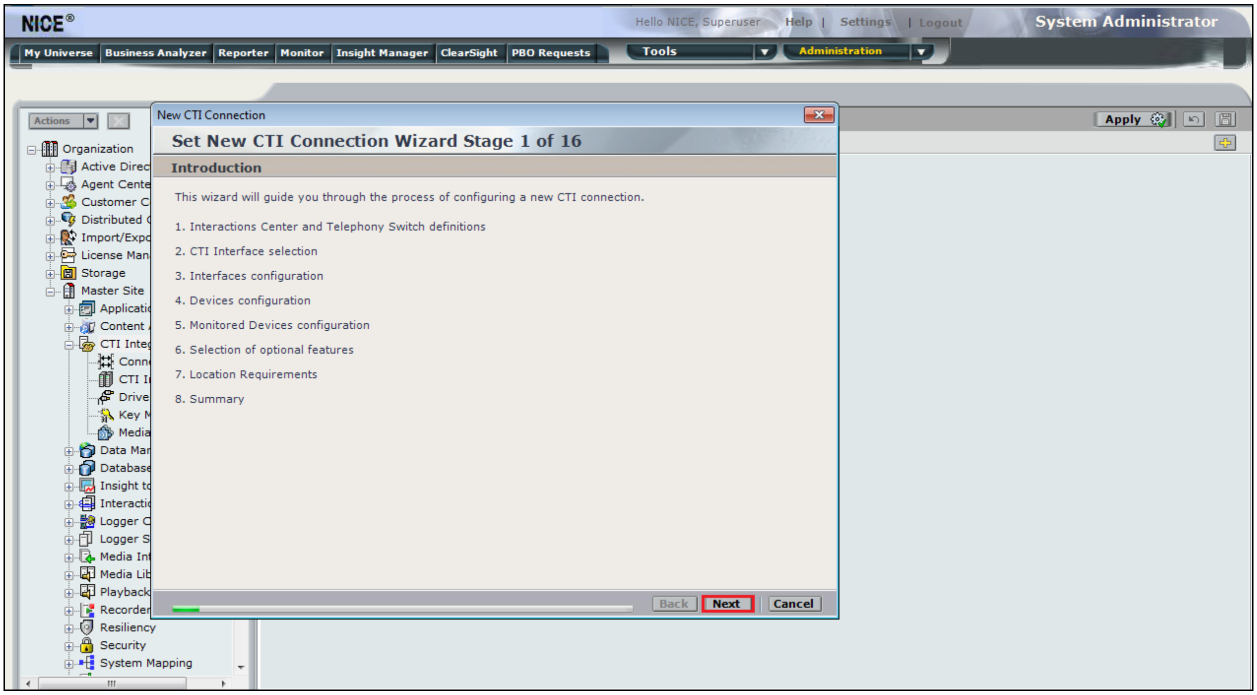


7.1. New CTI Connection

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 16 steps required to setup the connection to the AES for DMCC Service Observe and Single Step Conference type of call recording. Click on **Next** to continue.



Set New CTI Connection Wizard Stage 1 of 16

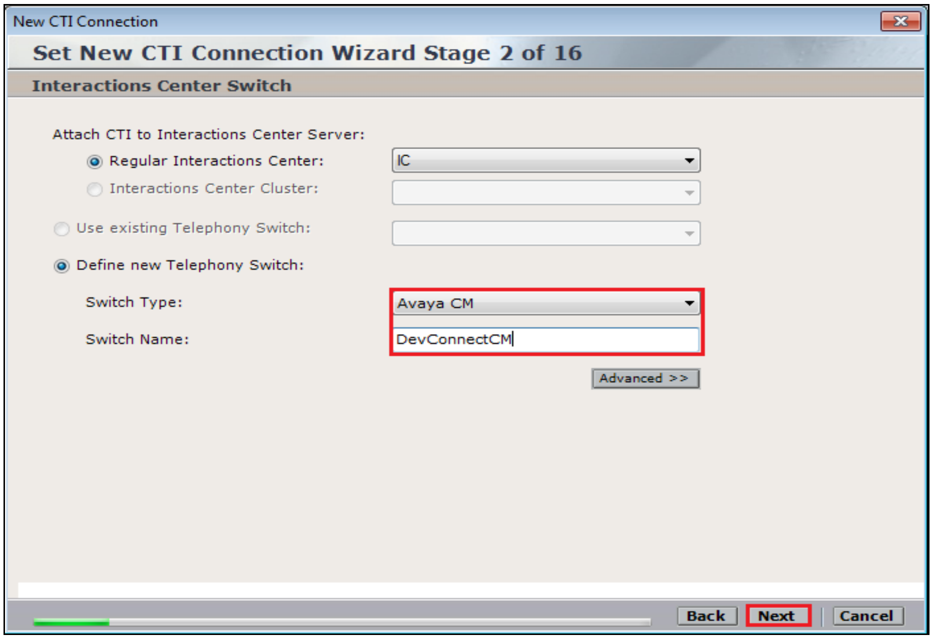
Introduction

This wizard will guide you through the process of configuring a new CTI connection.

1. Interactions Center and Telephony Switch definitions
2. CTI Interface selection
3. Interfaces configuration
4. Devices configuration
5. Monitored Devices configuration
6. Selection of optional features
7. Location Requirements
8. Summary

The value for **Regular Interactions Center** (**IC**) 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.


New CTI Connection

Set New CTI Connection Wizard Stage 2 of 16

Interactions Center Switch

Attach CTI to Interactions Center Server:

● Regular Interactions Center: IC

○ Interactions Center Cluster:

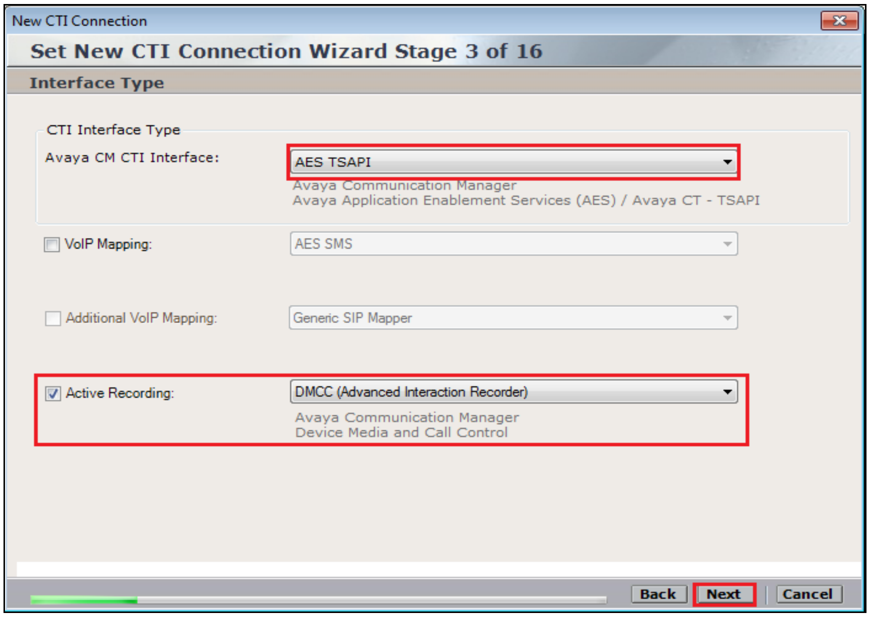
○ Use existing Telephony Switch:

● Define new Telephony Switch:

Switch Type: Avaya CM

Switch Name: DevConnectCM

Select **AES TSAPI** for the **Avaya CM CTI Interface**, ensure that **Active Recording** is ticked and select the **DMCC (Advanced integration Recorder)** from the dropdown menu. Click on **Next** to continue.


New CTI Connection

Set New CTI Connection Wizard Stage 3 of 16

Interface Type

CTI Interface Type

Avaya CM CTI Interface: AES TSAPI

Avaya Communication Manager
Avaya Application Enablement Services (AES) / Avaya CT - TSAPI

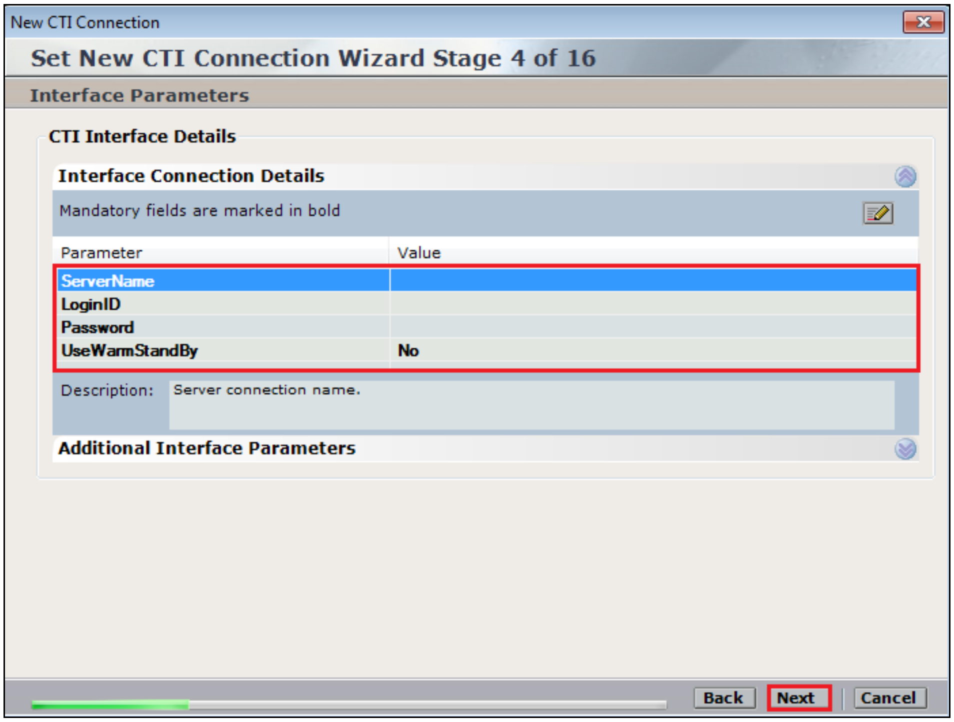
☐ VoIP Mapping: AES SMS

☐ Additional VoIP Mapping: Generic SIP Mapper

☑ Active Recording: DMCC (Advanced Interaction Recorder)

Avaya Communication Manager
Device Media and Call Control

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



New CTI Connection

Set New CTI Connection Wizard Stage 4 of 16

Interface Parameters

CTI Interface Details

Interface Connection Details

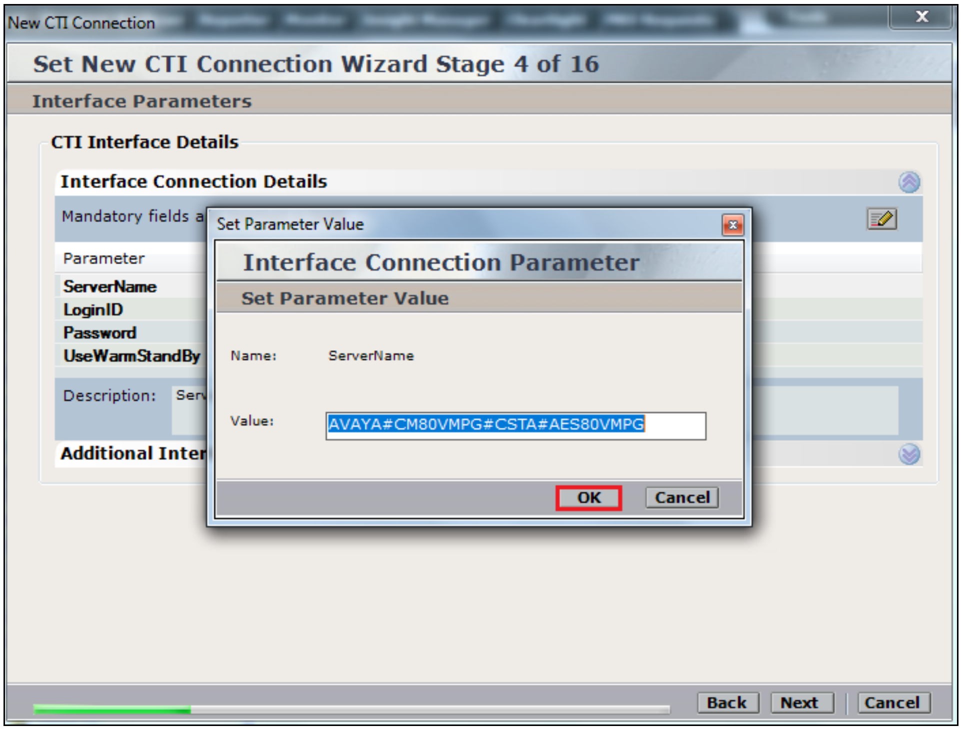
Mandatory fields are marked in bold

Parameter	Value
ServerName	
LoginID	
Password	
UseWarmStandBy	**No**

Description: Server connection name.

Additional Interface Parameters

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



Set Parameter Value

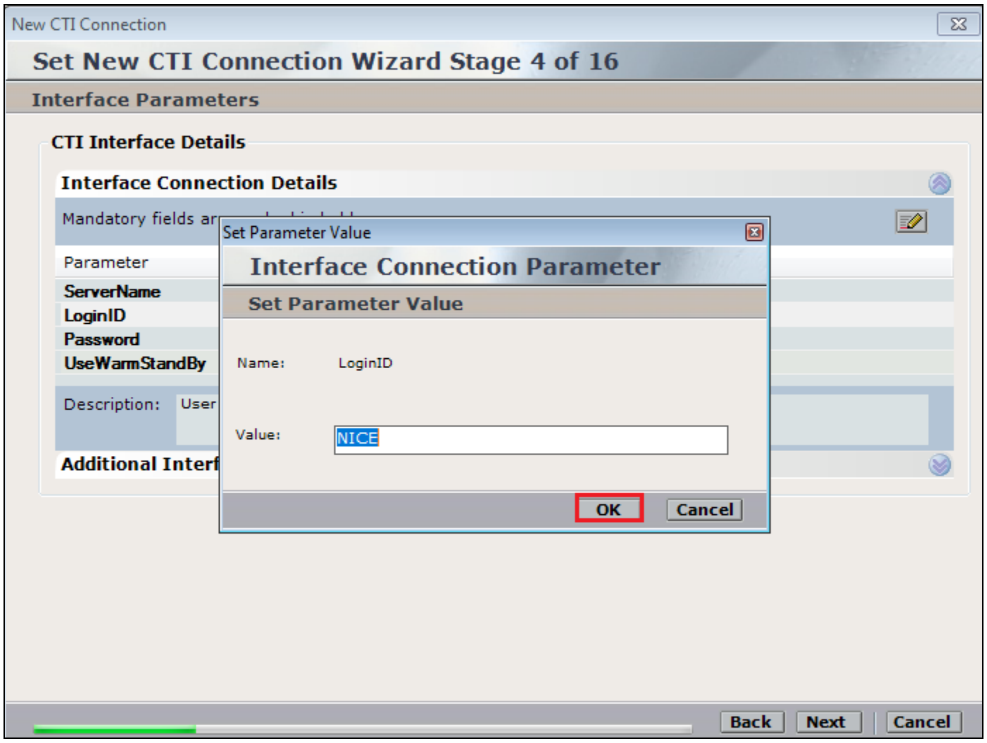
Interface Connection Parameter

Set Parameter Value

Name: ServerName

Value: AVAYA#CM80VMPG#CSTA#AES80VMPG

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



New CTI Connection

Set New CTI Connection Wizard Stage 4 of 16

Interface Parameters

CTI Interface Details

Interface Connection Details

Mandatory fields ar

Parameter
ServerName
LoginID
Password
UseWarmStandBy

Description: User

Additional Interf

Set Parameter Value

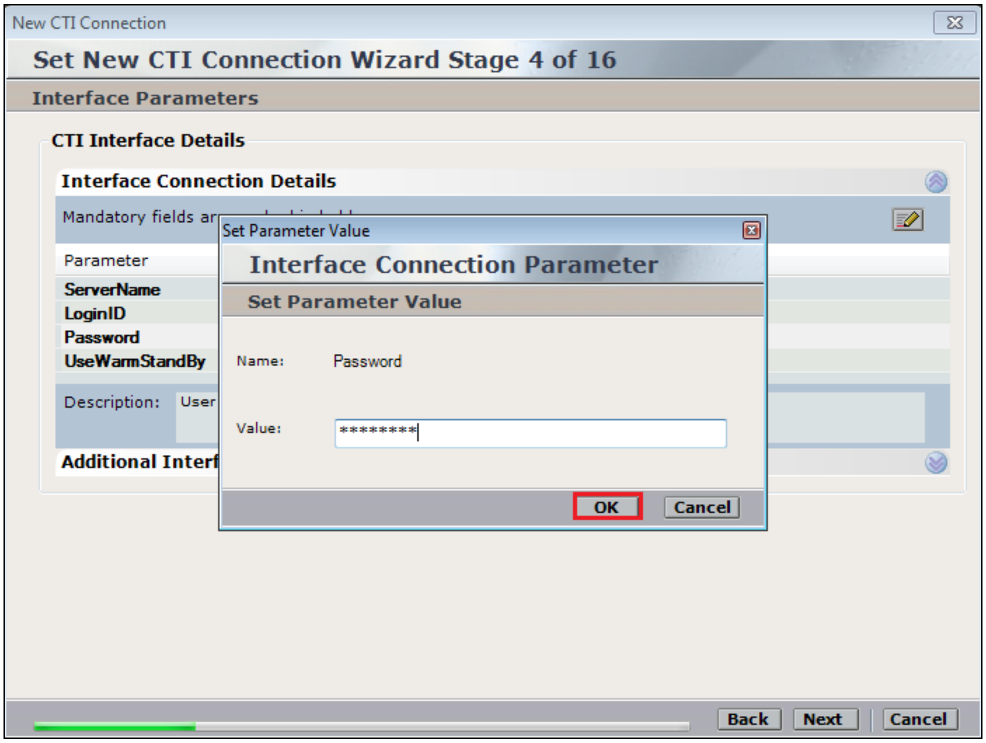
Interface Connection Parameter

Set Parameter Value

Name: LoginID

Value: NICE

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



New CTI Connection

Set New CTI Connection Wizard Stage 4 of 16

Interface Parameters

CTI Interface Details

Interface Connection Details

Mandatory fields ar

Parameter
ServerName
LoginID
Password
UseWarmStandBy

Description: User

Additional Interf

Set Parameter Value

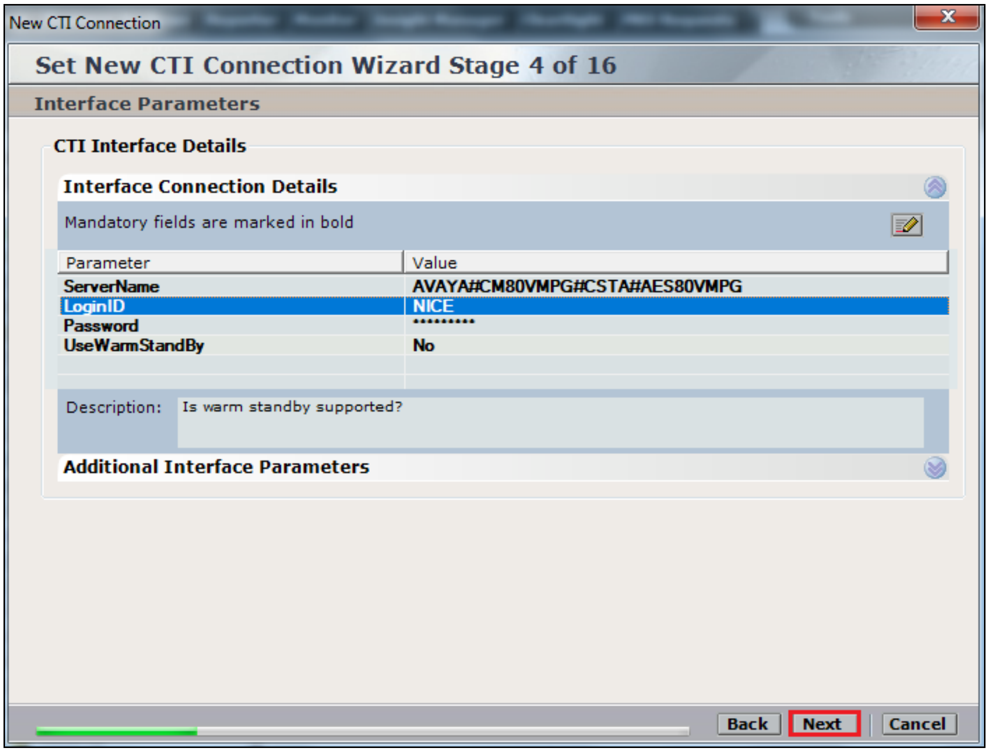
Interface Connection Parameter

Set Parameter Value

Name: Password

Value: *********

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



New CTI Connection

Set New CTI Connection Wizard Stage 4 of 16

Interface Parameters

CTI Interface Details

Interface Connection Details

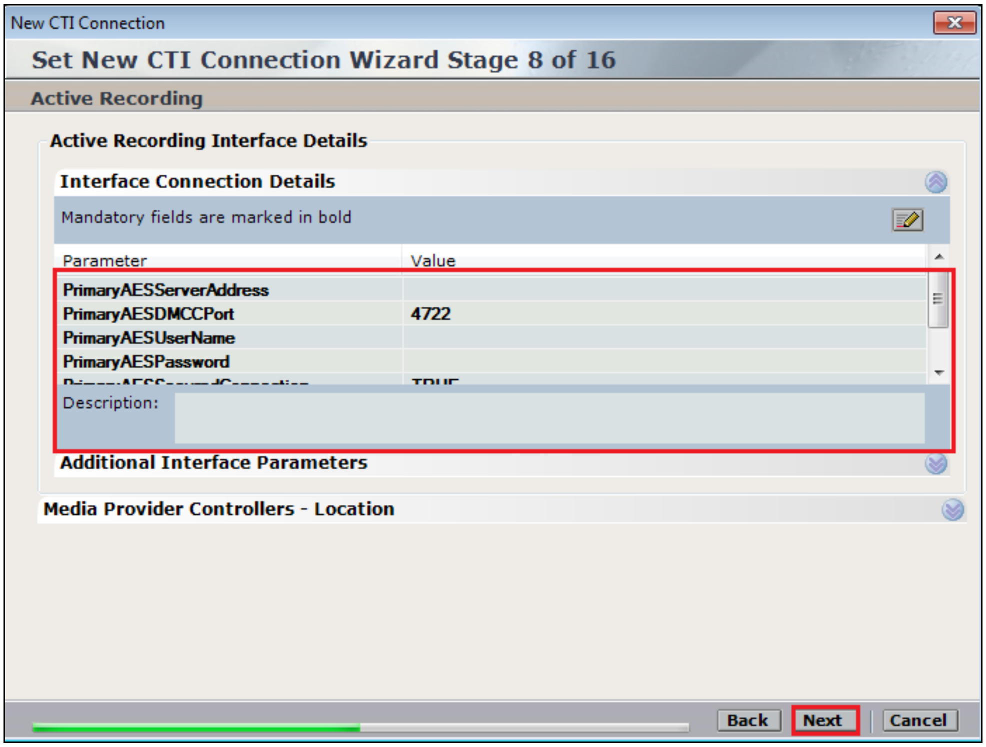
Mandatory fields are marked in bold

Parameter	Value
ServerName	**AVAYA#CM80VMPG#CSTA#AES80VMPG**
LoginID	**NICE**
Password	*********
UseWarmStandBy	**No**

Description: Is warm standby supported?

Additional Interface Parameters

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



New CTI Connection

Set New CTI Connection Wizard Stage 8 of 16

Active Recording

Active Recording Interface Details

Interface Connection Details

Mandatory fields are marked in bold

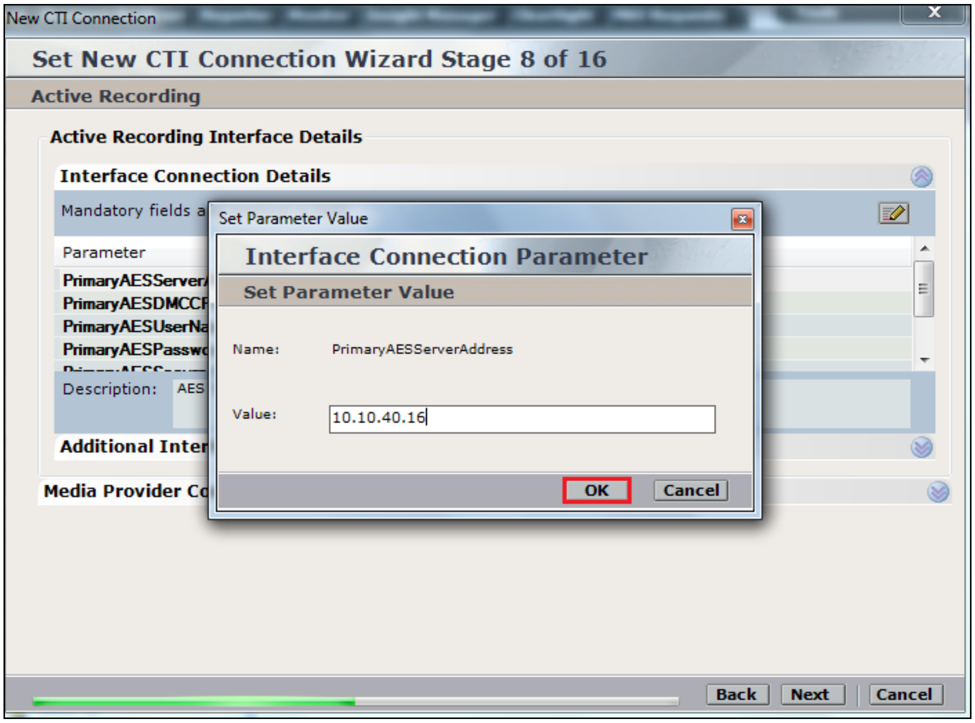
Parameter	Value
PrimaryAESServerAddress	
PrimaryAESDMCCPort	**4722**
PrimaryAESUserName	
PrimaryAESPassword	
PrimaryAESSecuredConnection	**TRUE**

Description:

Additional Interface Parameters

Media Provider Controllers - Location

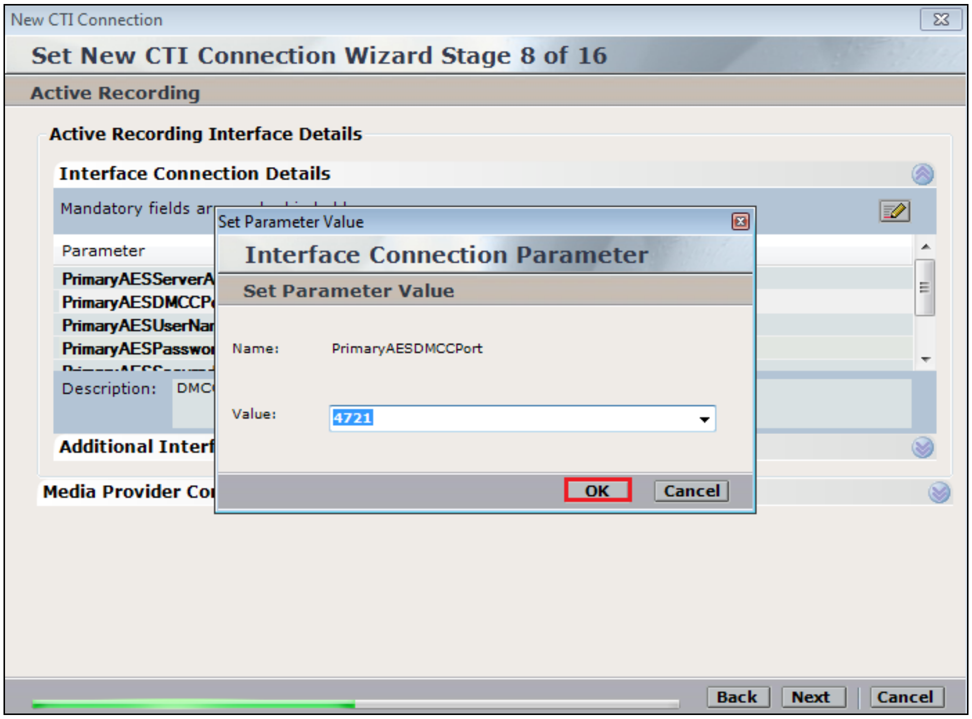
Enter the **Value** for the **AESServerAddress**, note this is the IP address of the AES server. Click on **OK**.



Name: PrimaryAESServerAddress

Value: 10.10.40.16

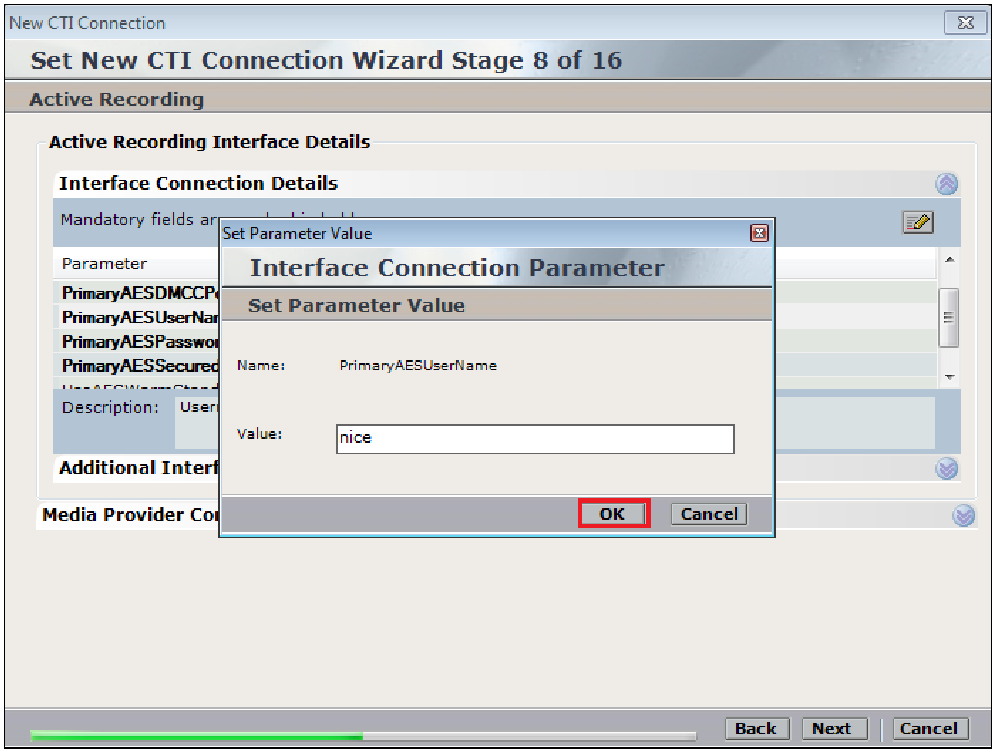
Enter the **Value** for the **AESDMCCPort**, note this will be the same port that was configured in **Section 6.5**. In this example the unencrypted port **4721** is entered.



Name: PrimaryAESDMCCPort

Value: 4721

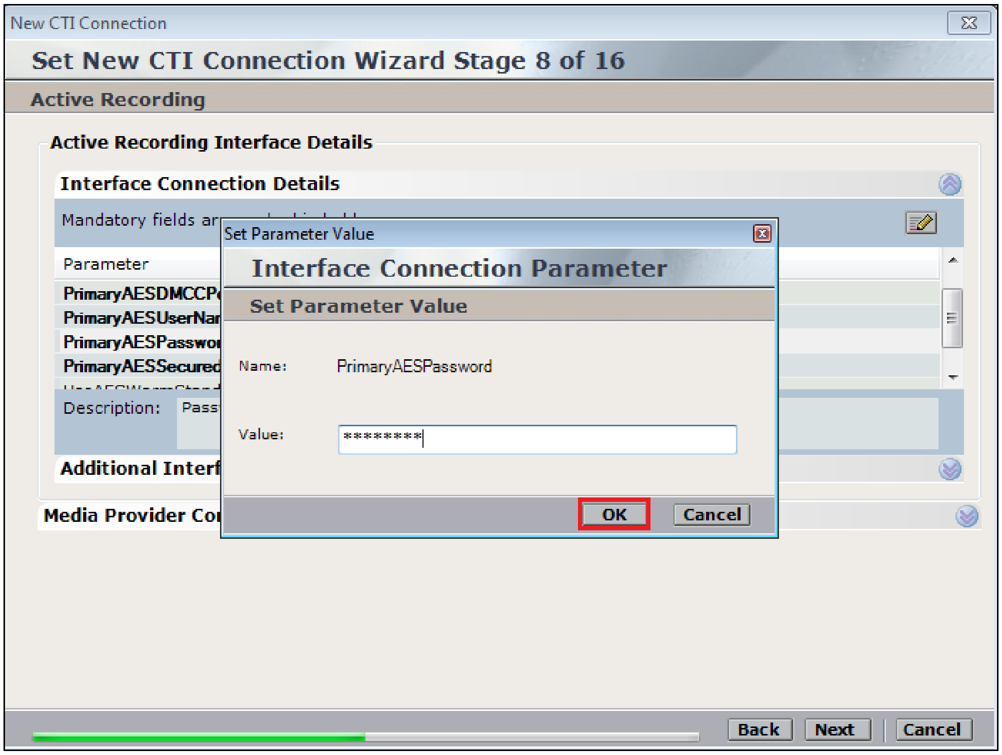
As before, enter the username that was created in **Section6.6** and click on **OK**.



Name: PrimaryAESUserName

Value: nice

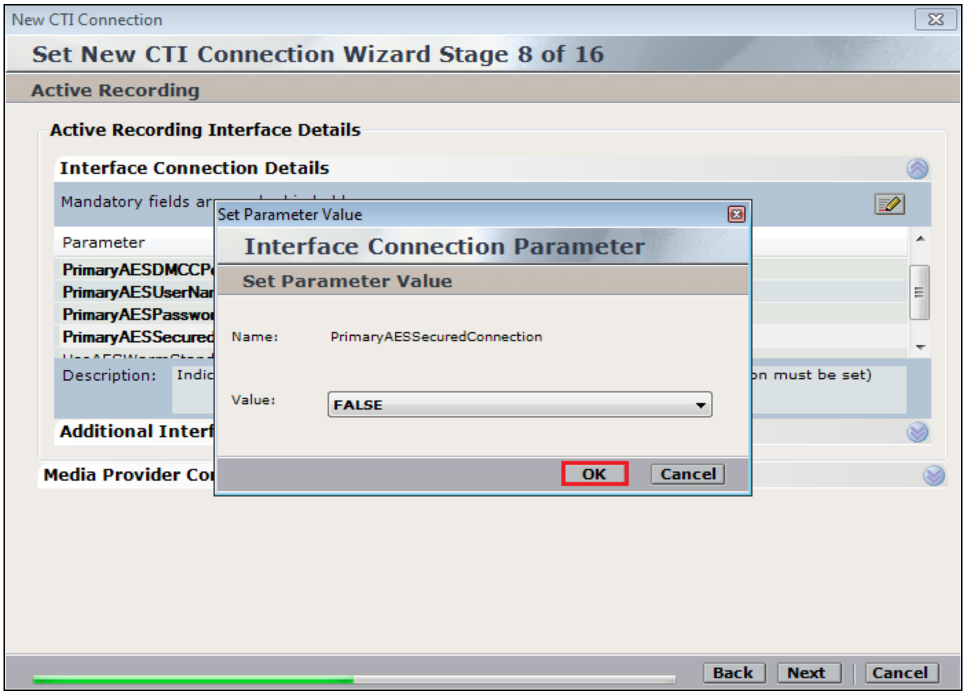
Enter the password that was created in **Section 6.6** and click on **OK**.



Name: PrimaryAESPassword

Value: *********

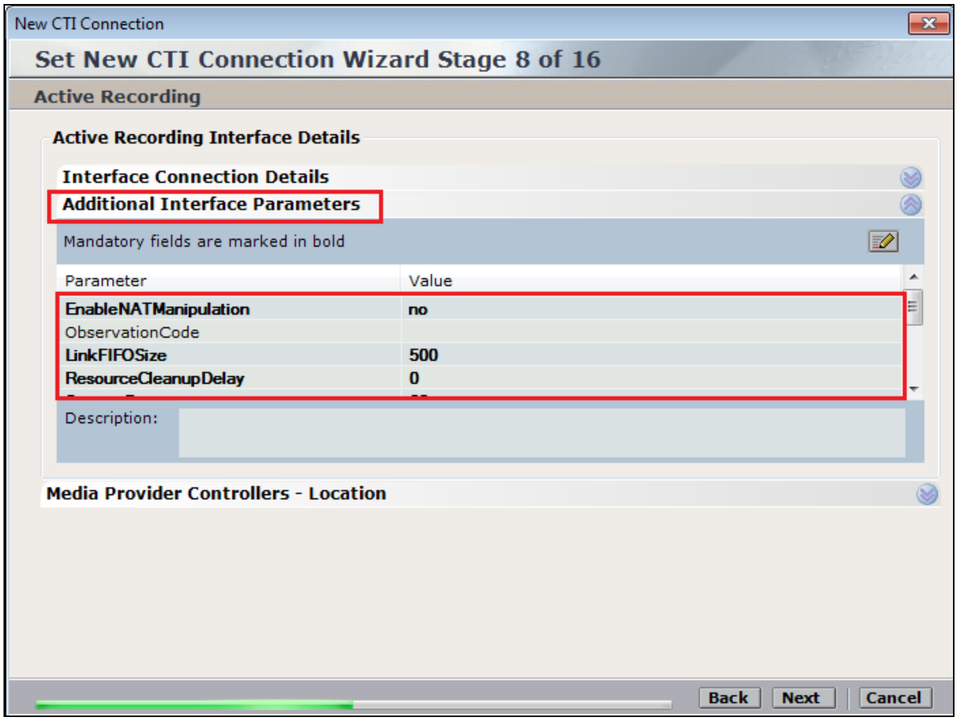
Because the unencrypted port was chosen, select **False** for the **PrimaryAESSecuredConnection**. Click on **OK** and then **Next** (not shown) to continue.



Name: PrimaryAESSecuredConnection

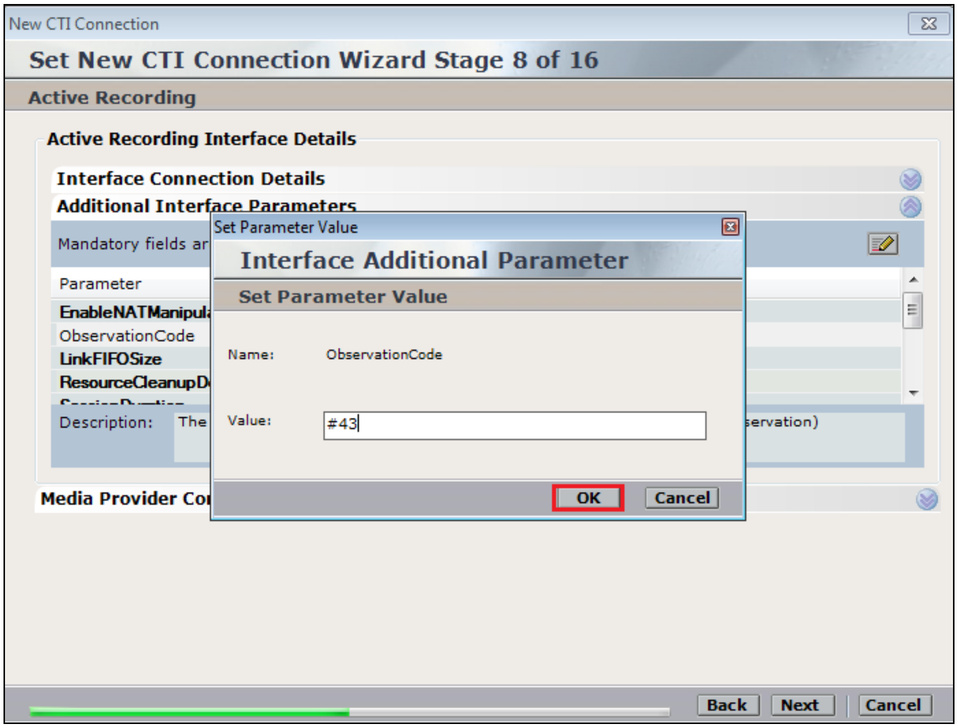
Value: FALSE

Click on **Additional Interface Parameters**, then to change the Service Observation Code double-click on **ObservationCode**.


New CTI Connection
Set New CTI Connection Wizard Stage 8 of 16
Active Recording
Active Recording Interface Details
Interface Connection Details
Additional Interface Parameters
Mandatory fields are marked in bold
Description:
Media Provider Controllers - Location
Back
Next
Cancel

Parameter	Value
EnableNATManipulation	**no**
ObservationCode	
LinkFIFOSize	**500**
ResourceCleanupDelay	**0**

Enter the **Value** that was created in **Section 5.5**. This was the Service Observing Listen Only Access Code **#43**. Click on **OK** to continue.



New CTI Connection

Set New CTI Connection Wizard Stage 8 of 16

Active Recording

Active Recording Interface Details

Interface Connection Details

Additional Interface Parameters

Set Parameter Value

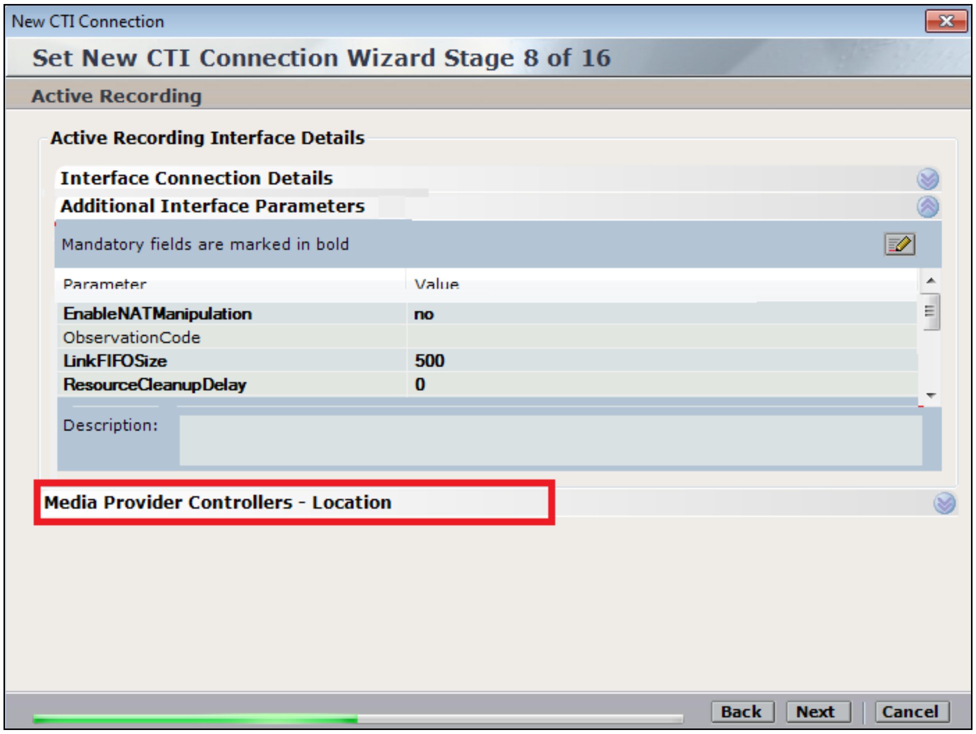
Interface Additional Parameter

Set Parameter Value

Name: ObservationCode

Value: #43

Click on **Media Provider Controllers – Location** to expand this.



New CTI Connection

Set New CTI Connection Wizard Stage 8 of 16

Active Recording

Active Recording Interface Details

Interface Connection Details

Additional Interface Parameters

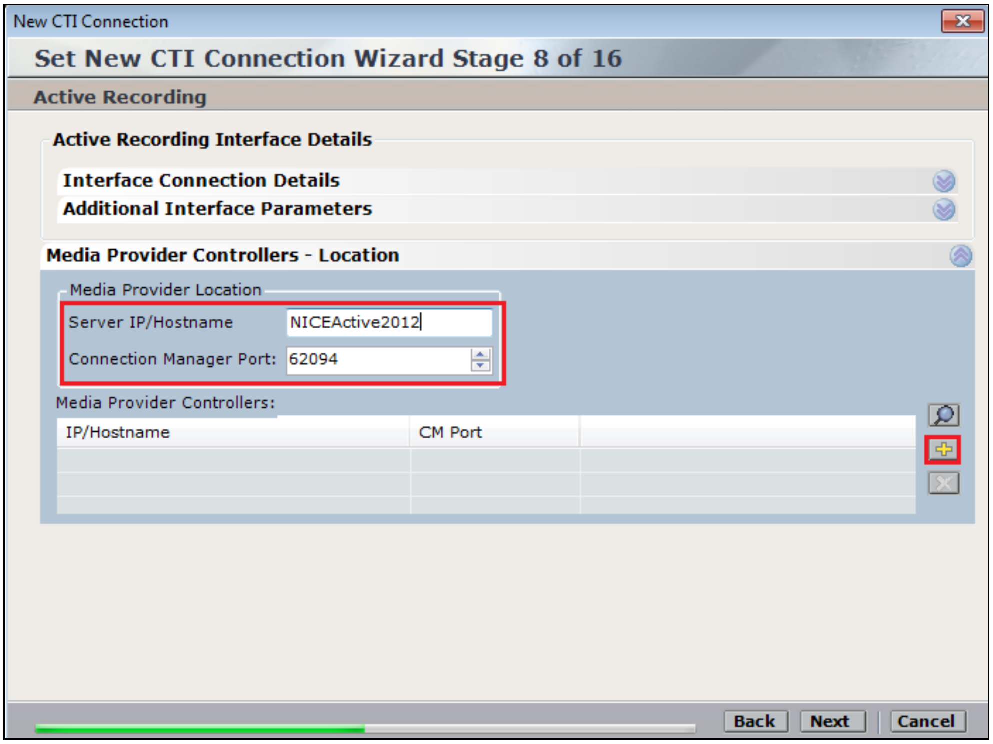
Mandatory fields are marked in bold

Parameter	Value
EnableNATManipulation	**no**
ObservationCode	
LinkFIFOSize	**500**
ResourceCleanupDelay	**0**

Description:

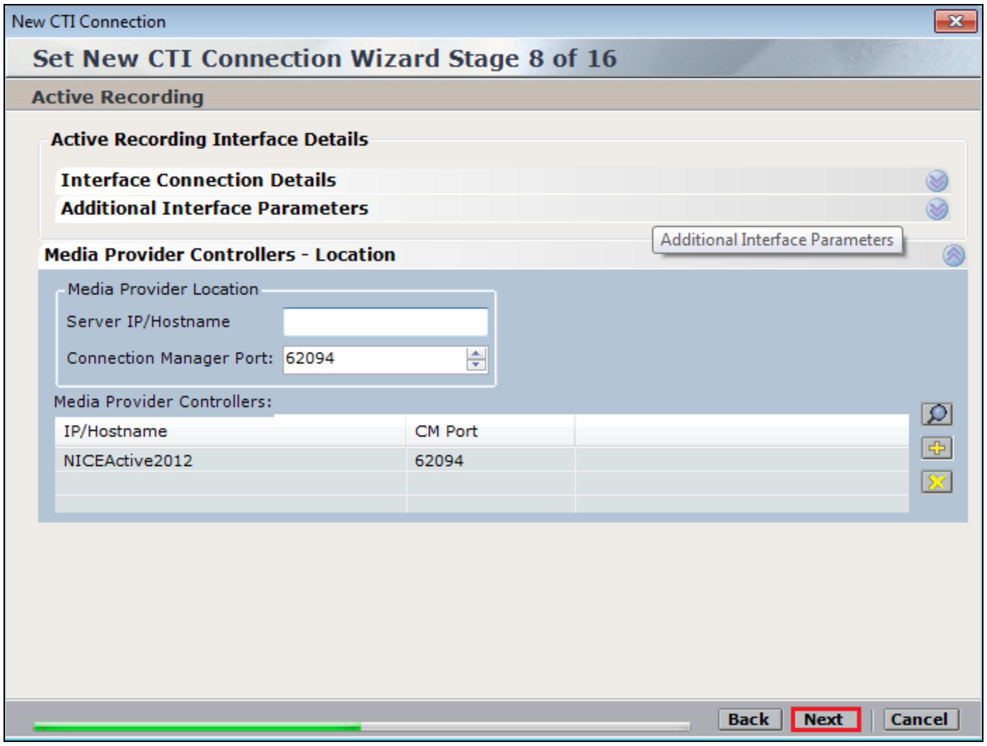
Media Provider Controllers - Location

Enter the **IP/Hostname** of the Nice Advaced Interactions Server, then click on the **+** icon to add this.


New CTI Connection
Set New CTI Connection Wizard Stage 8 of 16
Active Recording
Active Recording Interface Details
Interface Connection Details
Additional Interface Parameters
Media Provider Controllers - Location
Media Provider Location
Media Provider Controllers:
IP/Hostname
CM Port
Back
Next
Cancel

Server IP/Hostname: NICEActive2012
Connection Manager Port: 62094

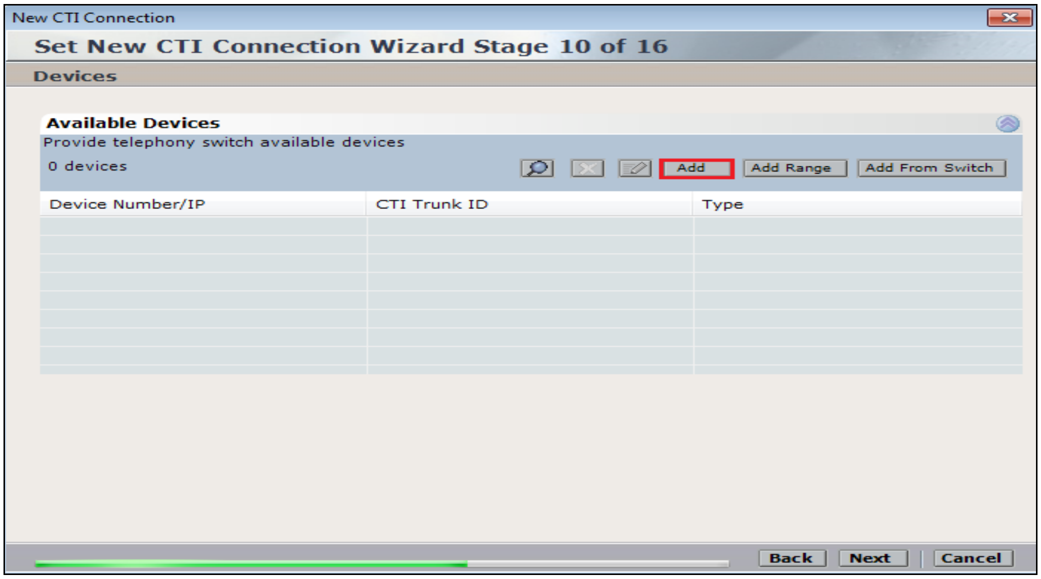
Click on **Next** to continue.


New CTI Connection
Set New CTI Connection Wizard Stage 8 of 16
Active Recording
Active Recording Interface Details
Interface Connection Details
Additional Interface Parameters
Additional Interface Parameters
Media Provider Controllers - Location
Media Provider Location
Media Provider Controllers:
Back
Next
Cancel

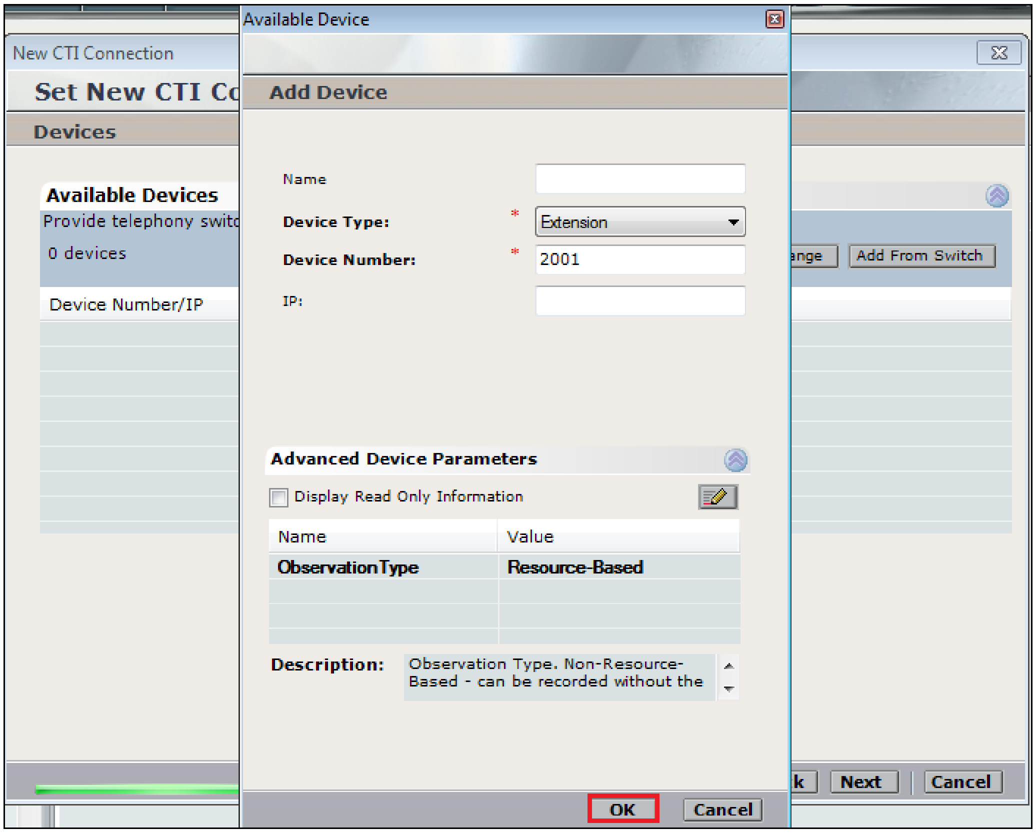
Server IP/Hostname:
Connection Manager Port: 62094

IP/Hostname	CM Port
NICEActive2012	62094

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



The **Device Type** should be **Extension** and insert the extension number of a phoneset that is to be recorded the example below showing extension **2001**. Expand **Advanced Device Parameters** and ensure that the **Value** for **Observation Type** is set to **Resourced-Based**. Click on **OK** to continue.



Name:

Device Type: * Extension

Device Number: * 2001

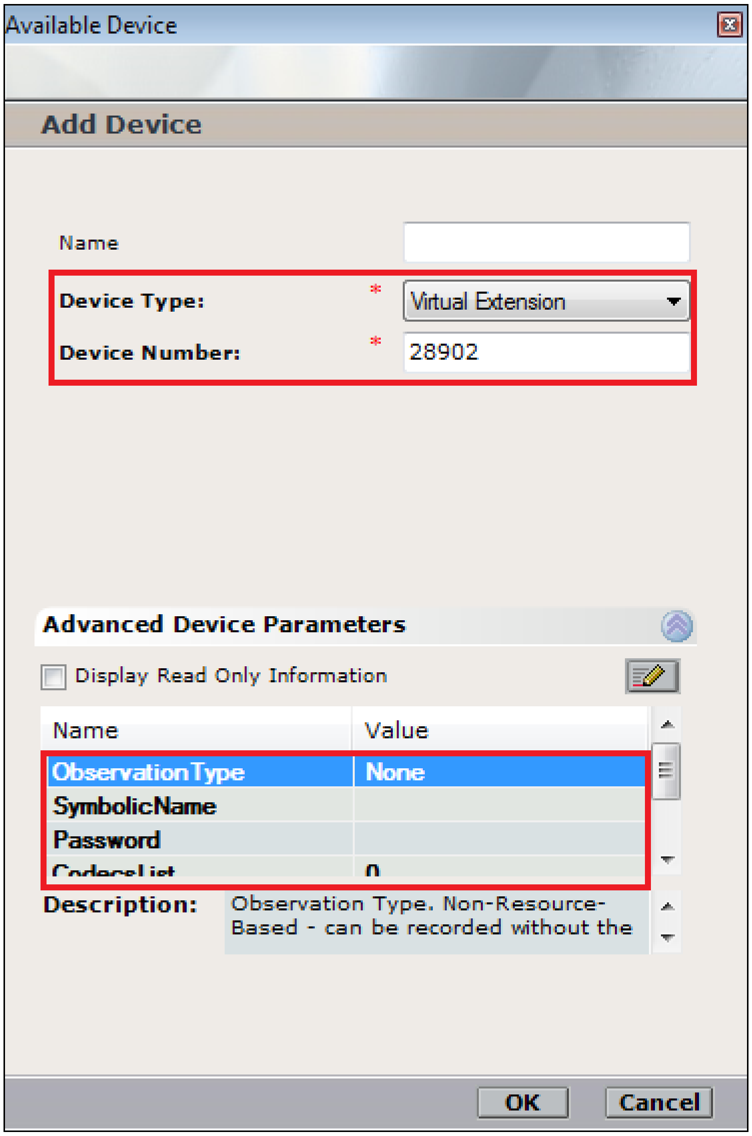
IP:

☐ Display Read Only Information

Name	Value
ObservationType	Resource-Based

Description: Observation Type. Non-Resource-Based - can be recorded without the

For Service Observe and Single Step Conference virtual extensions need to be added. These are the virtual extensions that were created in **Section 5.8**. Ensure that **Device Type** is set to **Virtual Extension** and add the correct extension for **Device Number**. Each of the **Parameters** highlighted at the bottom of the screen need to be configured and these are done by double-clicking on each parameter.



Available Device

Add Device

Name

Device Type: * Virtual Extension

Device Number: * 28902

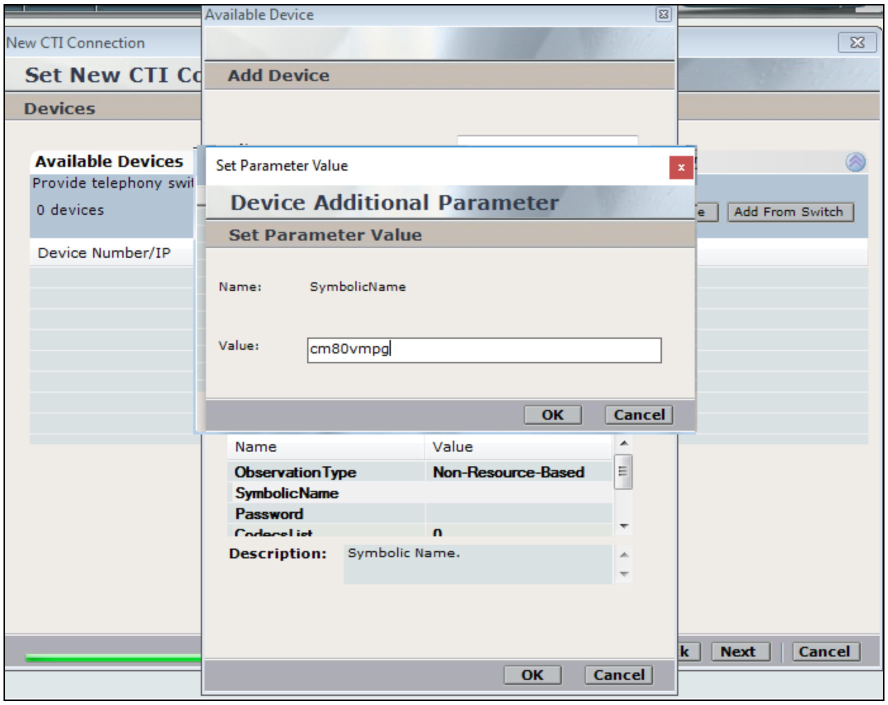
Advanced Device Parameters

□ Display Read Only Information

Name	Value
ObservationType	None
SymbolicName	
Password	
CodecsList	0

Description: Observation Type. Non-Resource-Based - can be recorded without the

Enter the correct **Value** for **SymbolicName**. Double-click on **SymbolicName** to set the value. This should be the same as the switch name entered in **Section 6.2**.



Set Parameter Value

Device Additional Parameter

Set Parameter Value

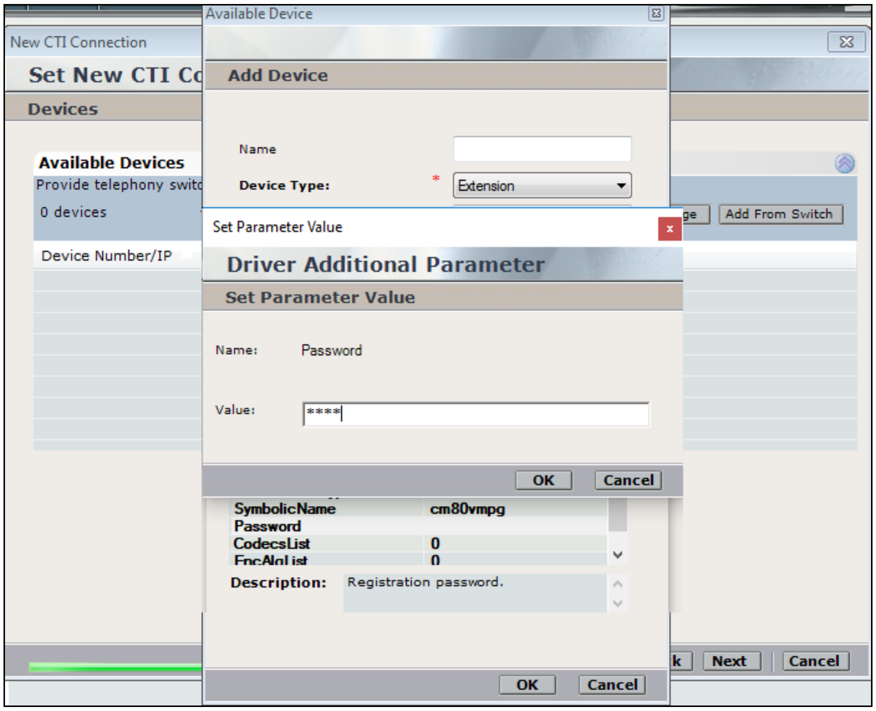
Name: SymbolicName

Value: cm80vmpg

Name	Value
ObservationType	Non-Resource-Based
SymbolicName	
Password	
CodecsList	0

Description: Symbolic Name.

Enter the correct **Password** and note this is the password for the extension that is being added here. This is the station password which was entered during the creation of the station. A printout of an extension can be found in **Section 5.6** of these Application Notes.



Name

Device Type: * Extension

Set Parameter Value

Driver Additional Parameter

Set Parameter Value

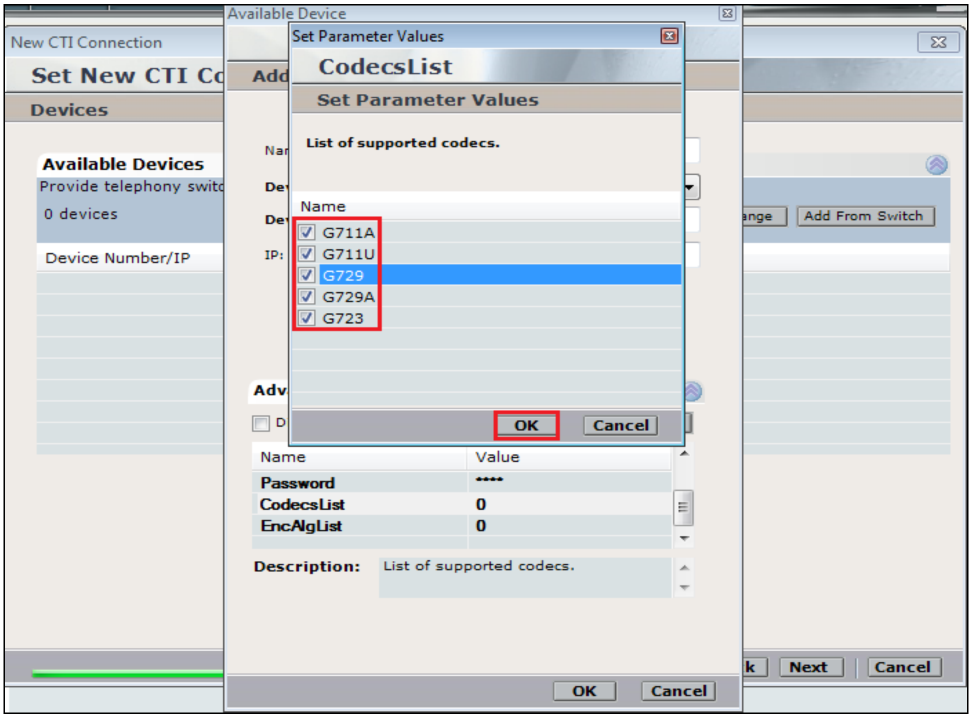
Name: Password

Value: ****

SymbolicName	cm80vmpg
Password	
CodecsList	0
EncAlgList	0

Description: Registration password.

Double-click on **CodecsList** and ensure that all the values are ticked as shown below. Click on **OK** to continue.



Set Parameter Values

CodecsList

Set Parameter Values

List of supported codecs.

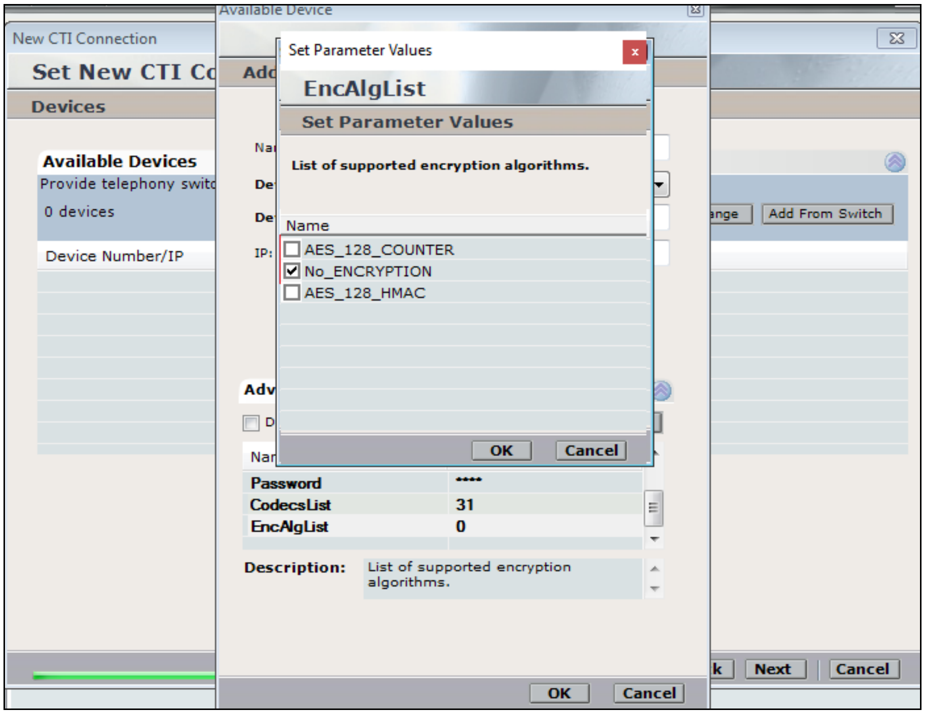
Name
☑ G711A
☑ G711U
☑ G729
☑ G729A
☑ G723

OK Cancel

Name	Value
Password	****
CodecsList	0
EncAlgList	0

Description: List of supported codecs.

Double-click on **EncAlgList** and seen as no SRTP was being recorded on this occasion **No_ENCRYPTION** was ticked. Click on **OK** to continue.



Set Parameter Values

EncAlgList

Set Parameter Values

List of supported encryption algorithms.

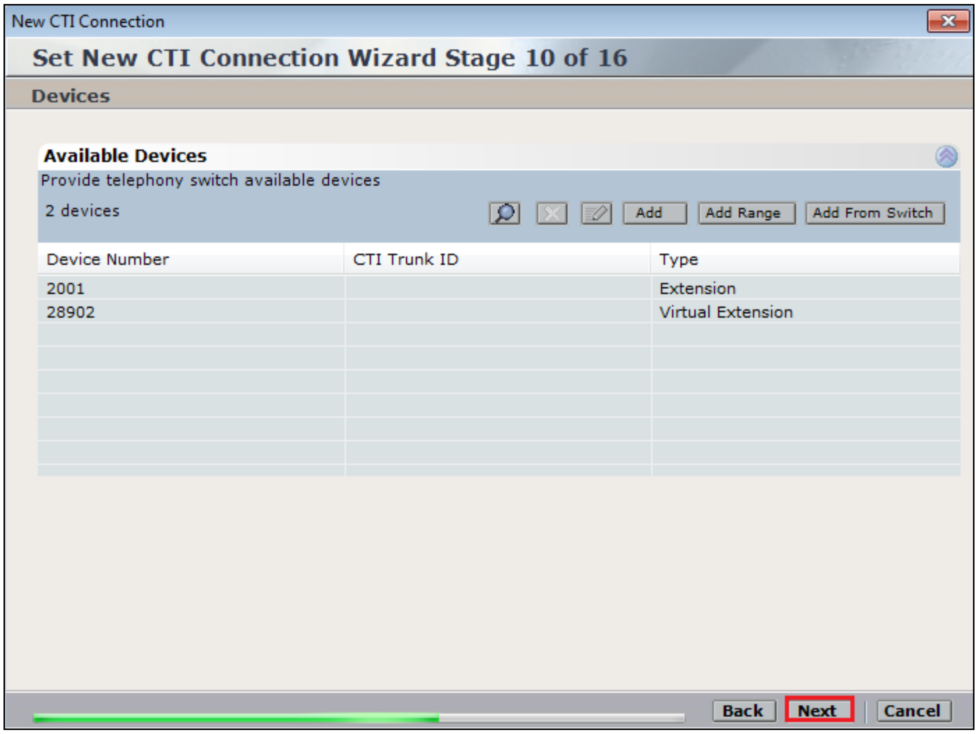
Name
□ AES_128_COUNTER
☑ No_ENCRYPTION
□ AES_128_HMAC

OK Cancel

Nar	
Password	****
CodecsList	31
EncAlgList	0

Description: List of supported encryption algorithms.

Click on **Next** to continue.



Set New CTI Connection Wizard Stage 10 of 16

Devices

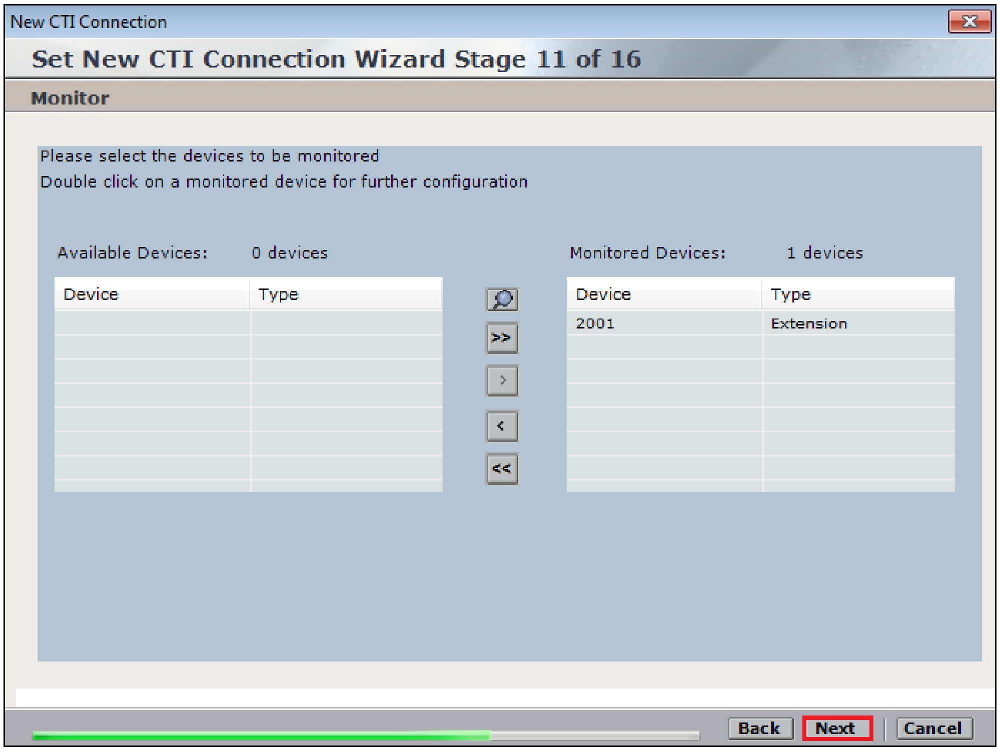
Available Devices

Provide telephony switch available devices

2 devices

Device Number	CTI Trunk ID	Type
2001		Extension
28902		Virtual Extension

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



Set New CTI Connection Wizard Stage 11 of 16

Monitor

Please select the devices to be monitored

Double click on a monitored device for further configuration

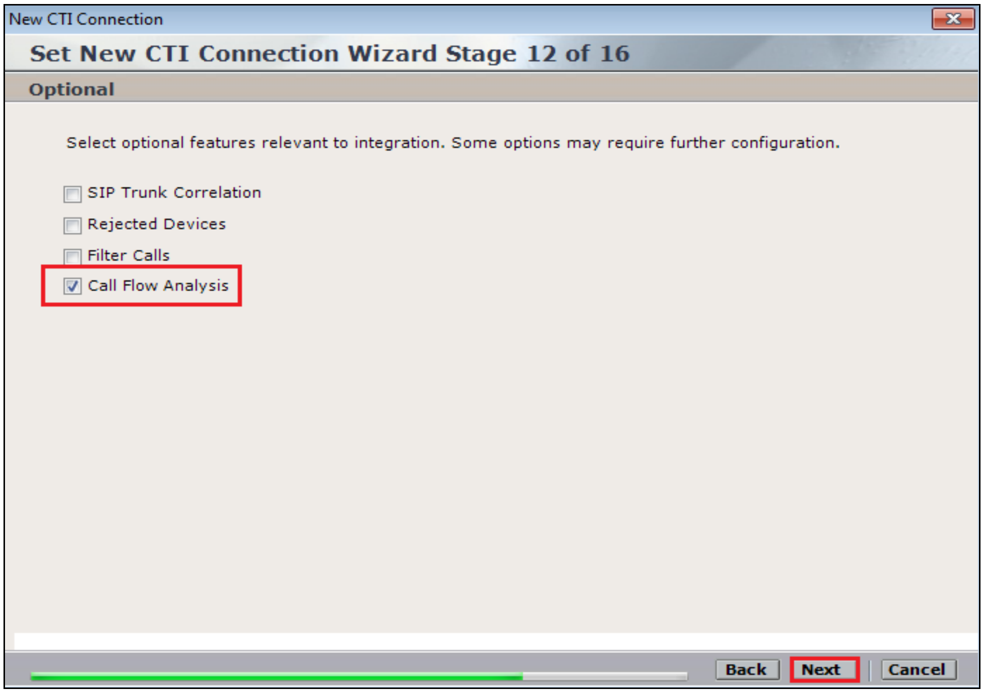
Available Devices: 0 devices

Device	Type

Monitored Devices: 1 devices

Device	Type
2001	Extension

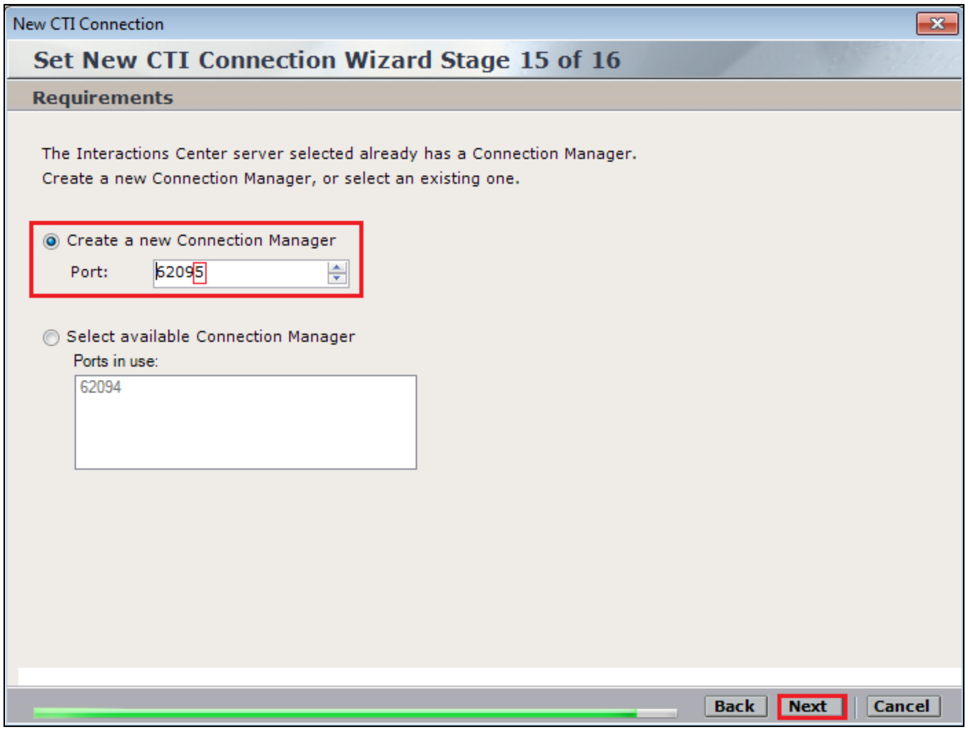
It is optional, but for better analysis tick on **Call Flow Analysis** and click on **Next** to continue.



Select optional features relevant to integration. Some options may require further configuration.

□ SIP Trunk Correlation
□ Rejected Devices
□ Filter Calls
☑ Call Flow Analysis

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

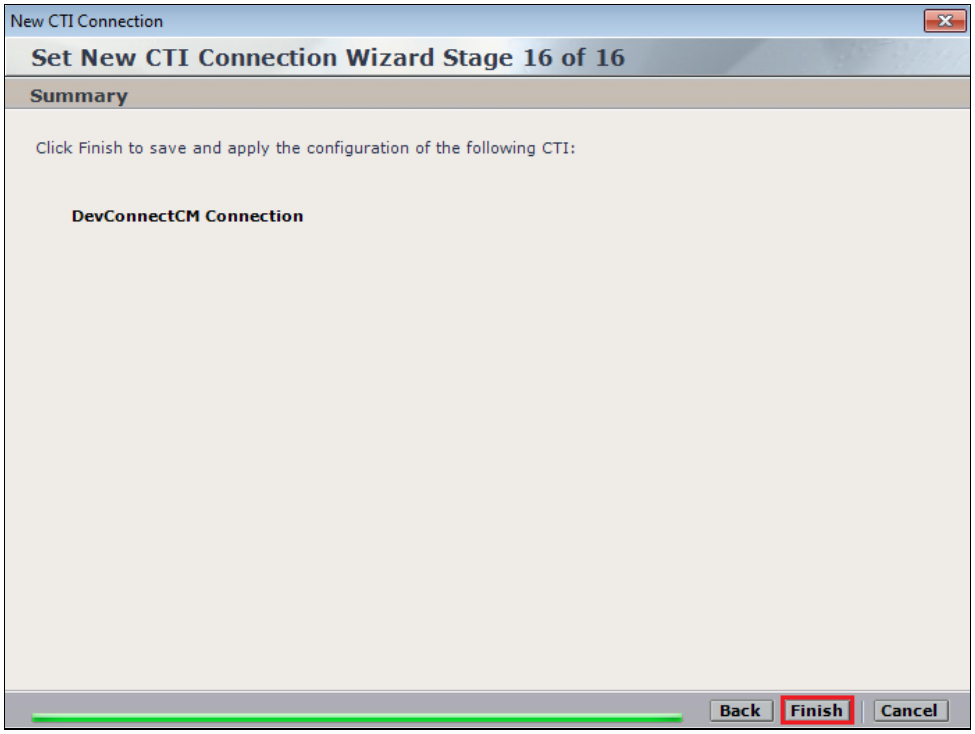


The Interactions Center server selected already has a Connection Manager.
Create a new Connection Manager, or select an existing one.

● Create a new Connection Manager
Port: 62095

○ Select available Connection Manager
Ports in use:
62094

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


New CTI Connection

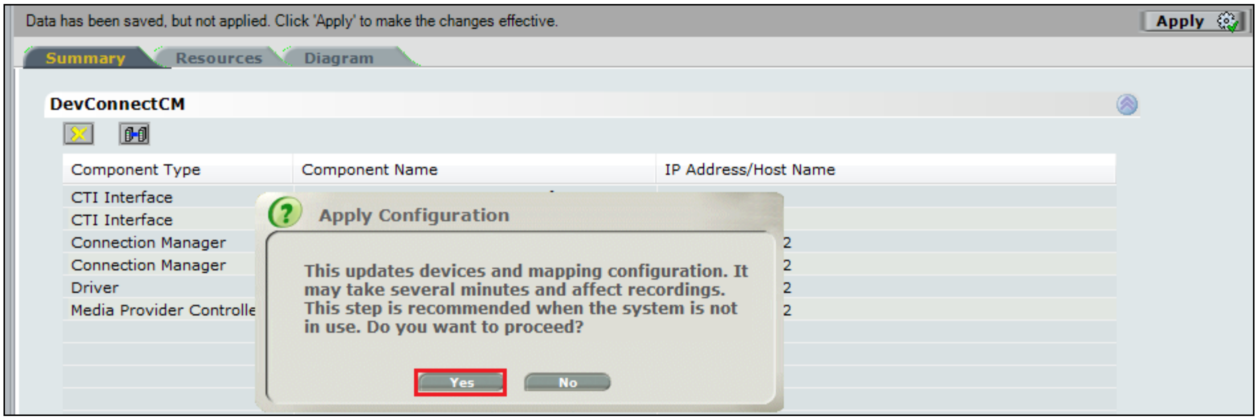
Set New CTI Connection Wizard Stage 16 of 16

Summary

Click Finish to save and apply the configuration of the following CTI:

DevConnectCM Connection

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


Data has been saved, but not applied. Click 'Apply' to make the changes effective.

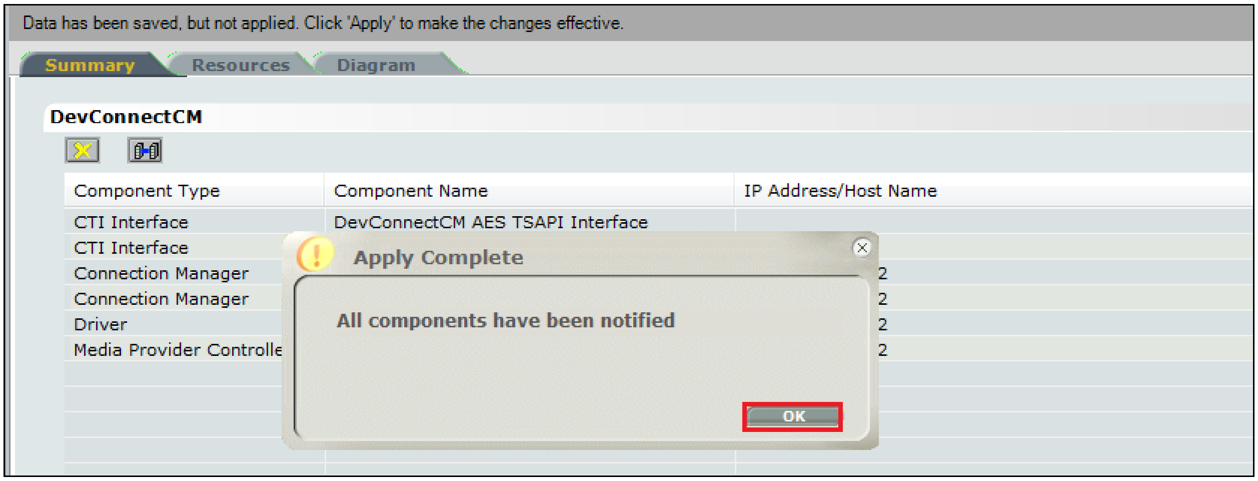
DevConnectCM

Component Type	Component Name	IP Address/Host Name
CTI Interface		
CTI Interface		
Connection Manager		2
Connection Manager		2
Driver		2
Media Provider Controlle		2

Apply Configuration

This updates devices and mapping configuration. It may take several minutes and affect recordings. This step is recommended when the system is not in use. Do you want to proceed?

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



Data has been saved, but not applied. Click 'Apply' to make the changes effective.

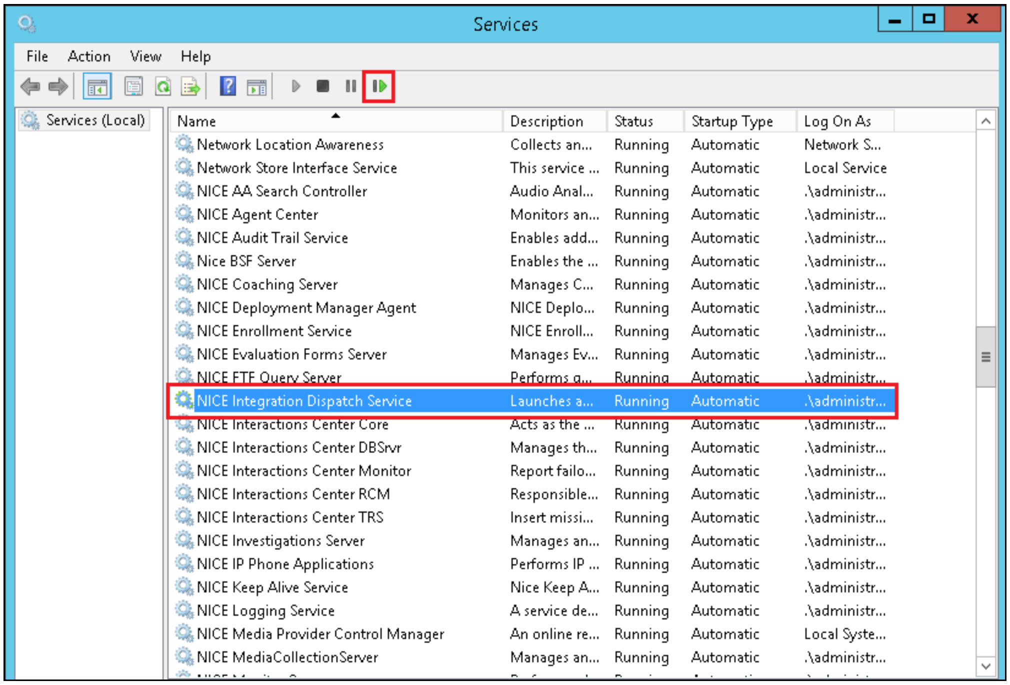
DevConnectCM

Component Type	Component Name	IP Address/Host Name
CTI Interface	DevConnectCM AES TSAPI Interface	
CTI Interface		
Connection Manager		2
Connection Manager		2
Driver		2
Media Provider Controlle		2

Apply Complete

All components have been notified

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

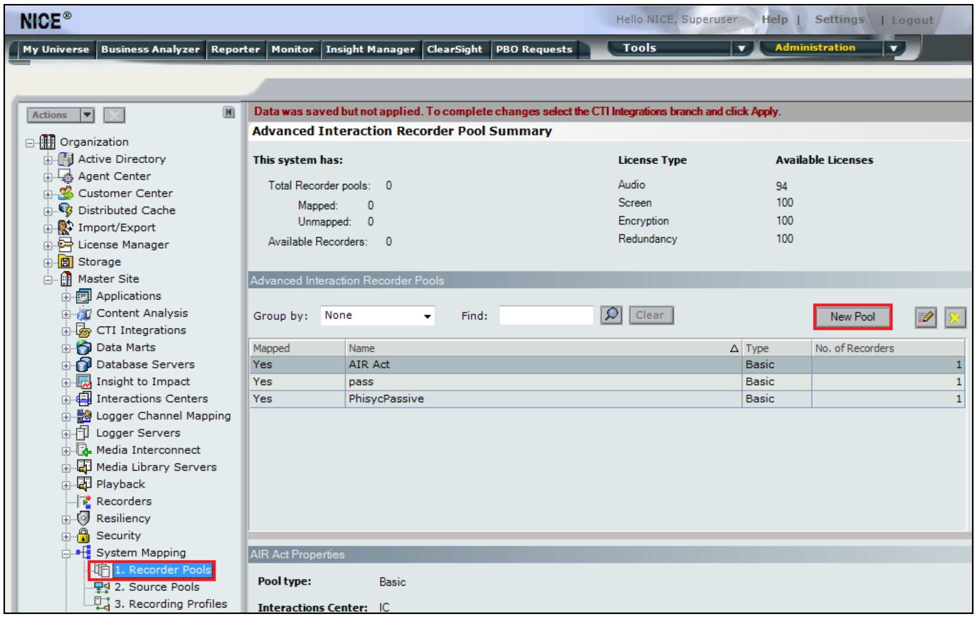


Services

Name	Description	Status	Startup Type	Log On As
Network Location Awareness	Collects an...	Running	Automatic	Network S...
Network Store Interface Service	This service ...	Running	Automatic	Local Service
NICE AA Search Controller	Audio Anal...	Running	Automatic	.\administr...
NICE Agent Center	Monitors an...	Running	Automatic	.\administr...
NICE Audit Trail Service	Enables add...	Running	Automatic	.\administr...
Nice BSF Server	Enables the ...	Running	Automatic	.\administr...
NICE Coaching Server	Manages C...	Running	Automatic	.\administr...
NICE Deployment Manager Agent	NICE Deplo...	Running	Automatic	.\administr...
NICE Enrollment Service	NICE Enroll...	Running	Automatic	.\administr...
NICE Evaluation Forms Server	Manages Ev...	Running	Automatic	.\administr...
NICE FTF Query Server	Performs q...	Running	Automatic	.\administr...
NICE Integration Dispatch Service	Launches a...	Running	Automatic	.\administr...
NICE Interactions Center Core	Acts as the ...	Running	Automatic	.\administr...
NICE Interactions Center DBSrvr	Manages th...	Running	Automatic	.\administr...
NICE Interactions Center Monitor	Report failo...	Running	Automatic	.\administr...
NICE Interactions Center RCM	Responsible...	Running	Automatic	.\administr...
NICE Interactions Center TRS	Insert missi...	Running	Automatic	.\administr...
NICE Investigations Server	Manages an...	Running	Automatic	.\administr...
NICE IP Phone Applications	Performs IP ...	Running	Automatic	.\administr...
NICE Keep Alive Service	Nice Keep A...	Running	Automatic	.\administr...
NICE Logging Service	A service de...	Running	Automatic	.\administr...
NICE Media Provider Control Manager	An online re...	Running	Automatic	Local Syste...
NICE MediaCollectionServer	Manages an...	Running	Automatic	.\administr...

7.2. System Mapping

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



Data was saved but not applied. To complete changes select the CTI Integrations branch and click Apply.

Advanced Interaction Recorder Pool Summary

This system has:

Total Recorder pools: 0
Mapped: 0
Unmapped: 0
Available Recorders: 0

License Type	Available Licenses
Audio	94
Screen	100
Encryption	100
Redundancy	100

Advanced Interaction Recorder Pools

Group by: None   Find:

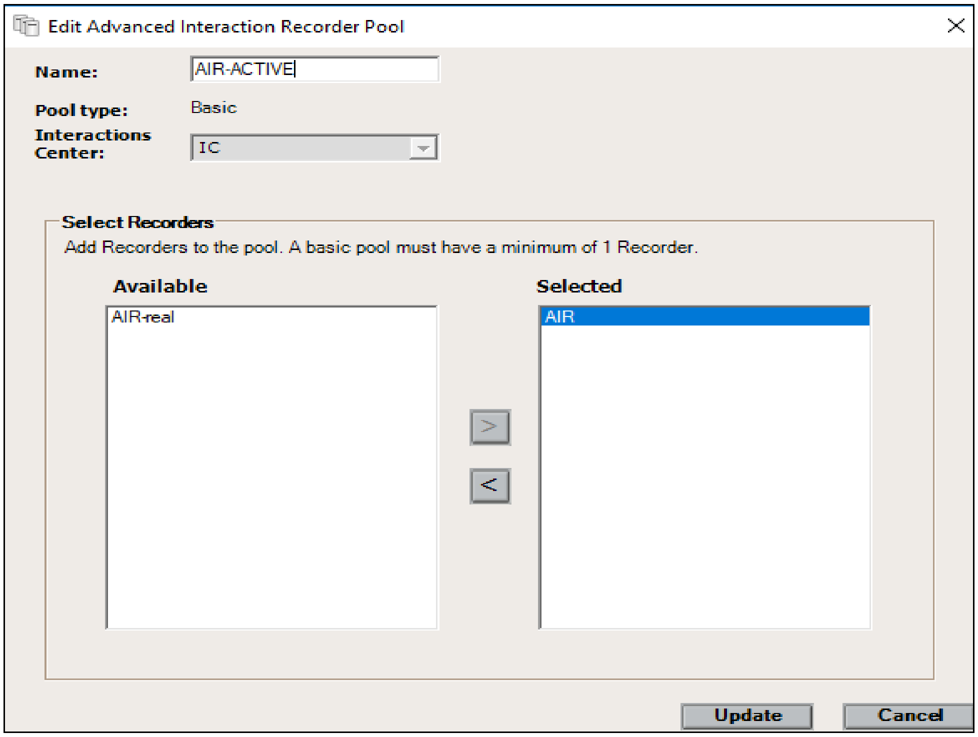
Mapped	Name	Type	No. of Recorders
Yes	AIR Act	Basic	1
Yes	pass	Basic	1
Yes	PhisycPassive	Basic	1

AIR Act Properties

Pool type: Basic

Interactions Center: IC

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



Edit Advanced Interaction Recorder Pool

Name: AIR-ACTIVE

Pool type: Basic

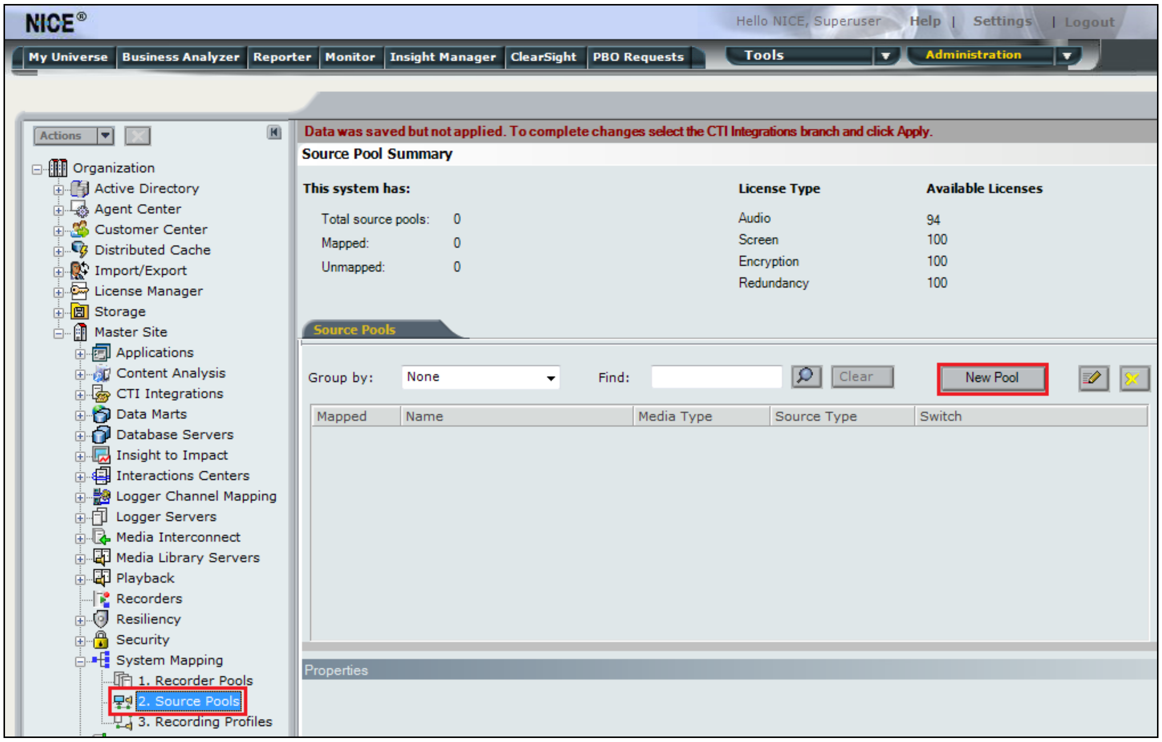
Interactions Center: IC

Select Recorders

Add Recorders to the pool. A basic pool must have a minimum of 1 Recorder.

Available	Selected
AIR-real	AIR

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



Data was saved but not applied. To complete changes select the CTI Integrations branch and click Apply.

Source Pool Summary

This system has:

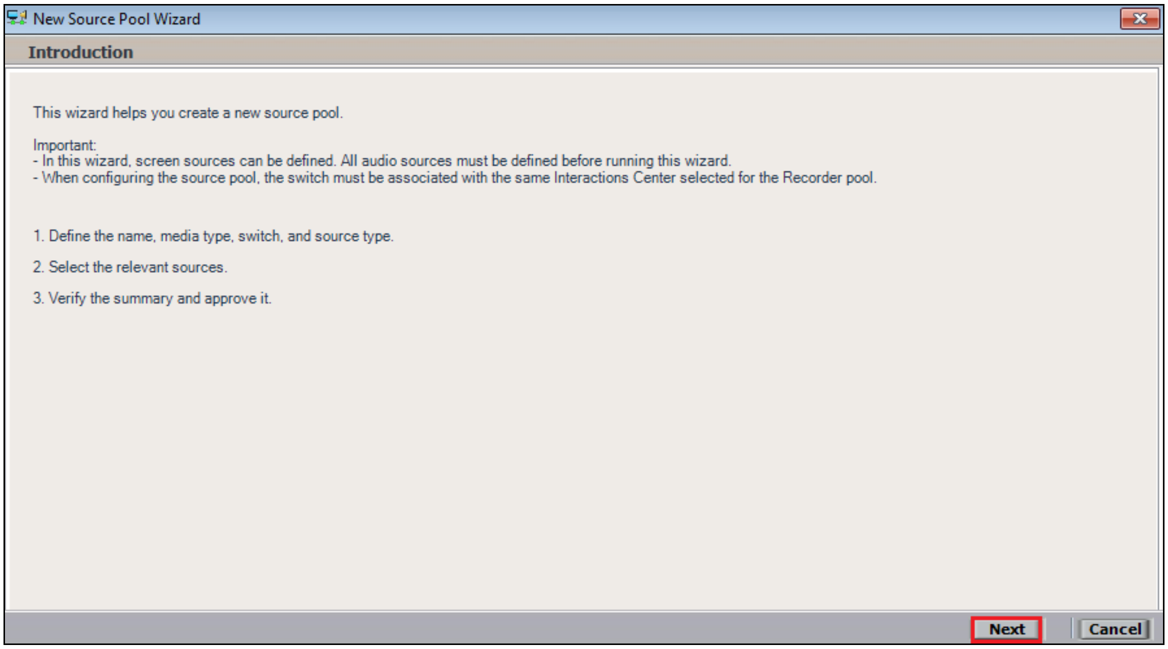
Total source pools:	0
Mapped:	0
Unmapped:	0

License Type	Available Licenses
Audio	94
Screen	100
Encryption	100
Redundancy	100

Group by: None

Mapped	Name	Media Type	Source Type	Switch

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



Introduction

This wizard helps you create a new source pool.

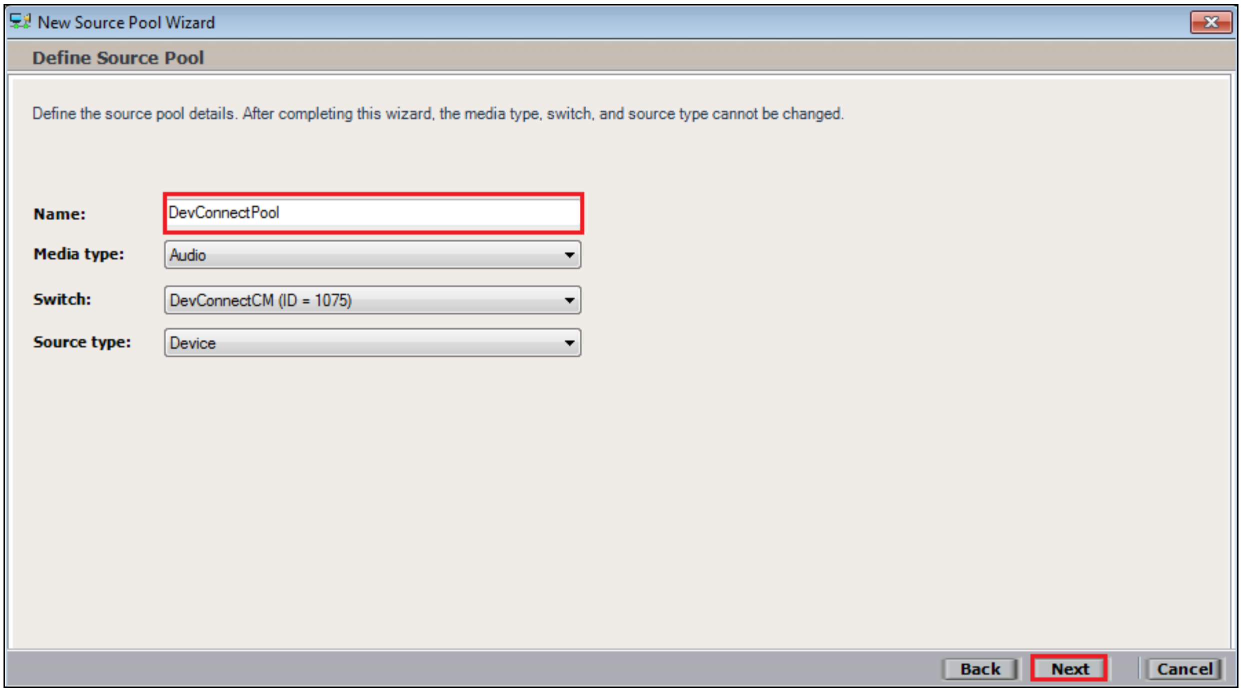
Important:
- In this wizard, screen sources can be defined. All audio sources must be defined before running this wizard.
- When configuring the source pool, the switch must be associated with the same Interactions Center selected for the Recorder pool.

1. Define the name, media type, switch, and source type.

2. Select the relevant sources.

3. Verify the summary and approve it.

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



Define the source pool details. After completing this wizard, the media type, switch, and source type cannot be changed.

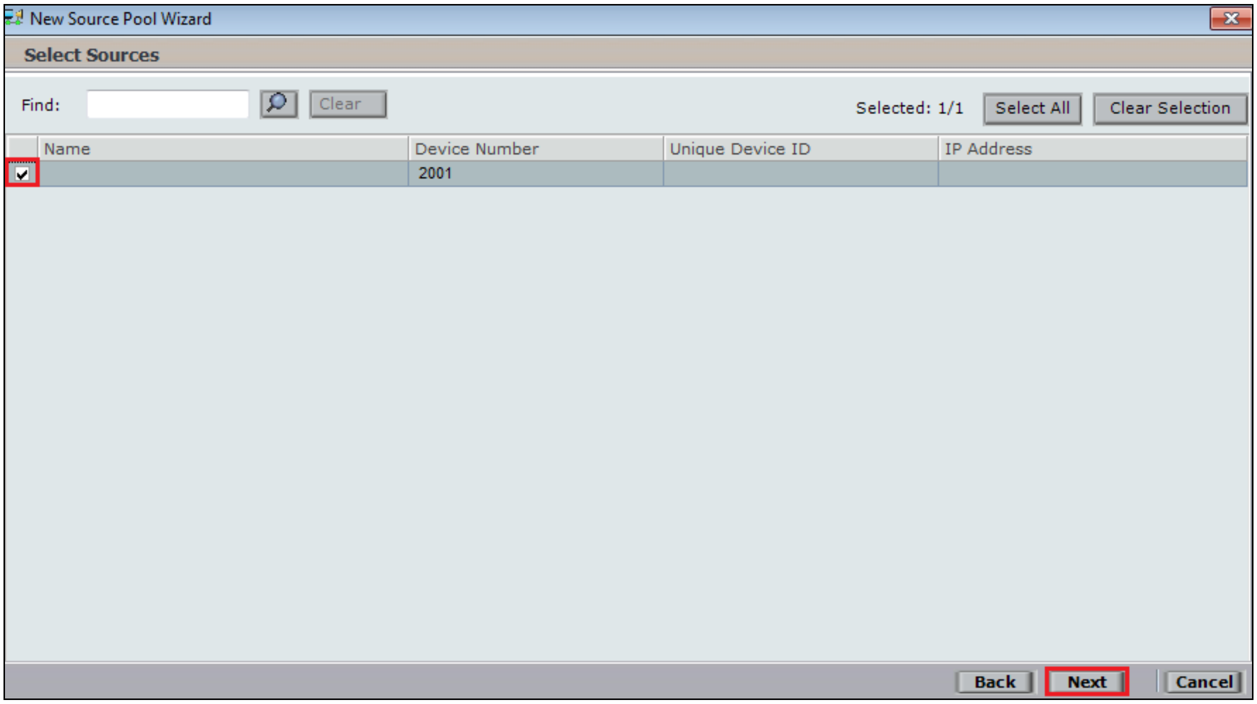
Name: DevConnectPool

Media type: Audio

Switch: DevConnectCM (ID = 1075)

Source type: Device

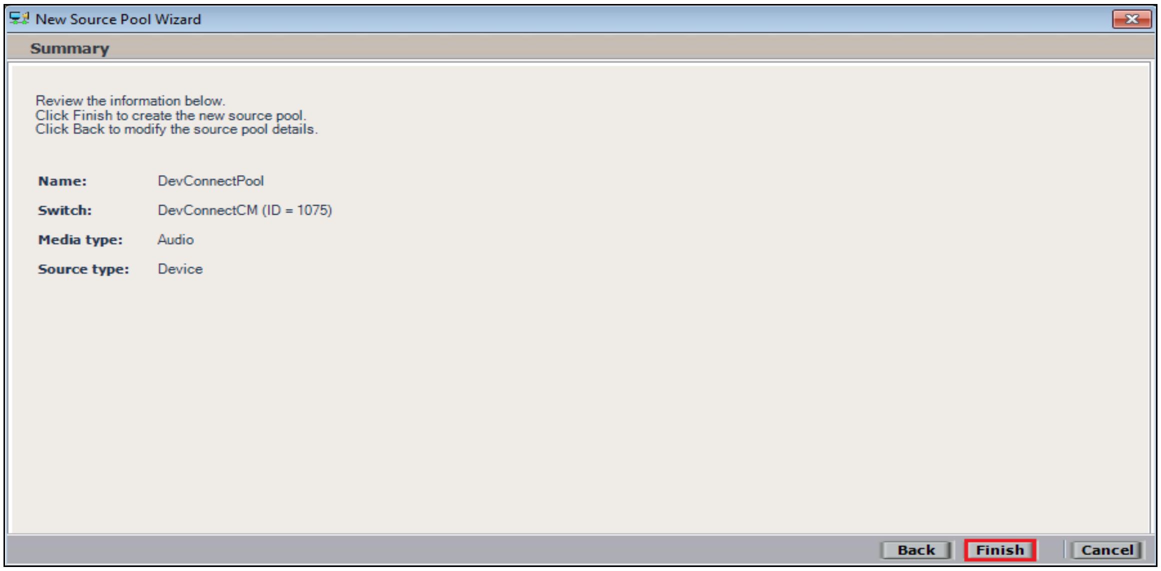
Select the extensions that were created in **Section 7.1**, note only one extension number is shown in the example below but this is not typical. Click on **Next** to continue.



Selected: 1/1

	Name	Device Number	Unique Device ID	IP Address
☑		2001		

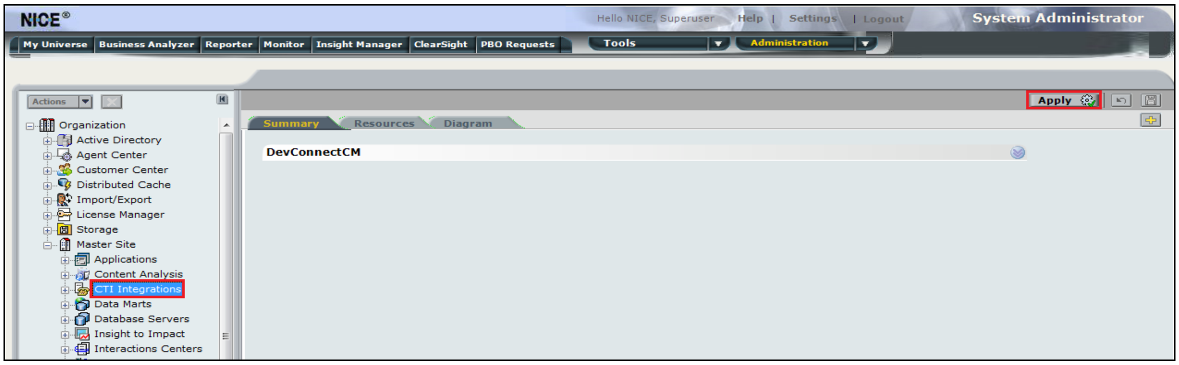
Click on **Finish** to complete the **New Source Pool Wizard**.



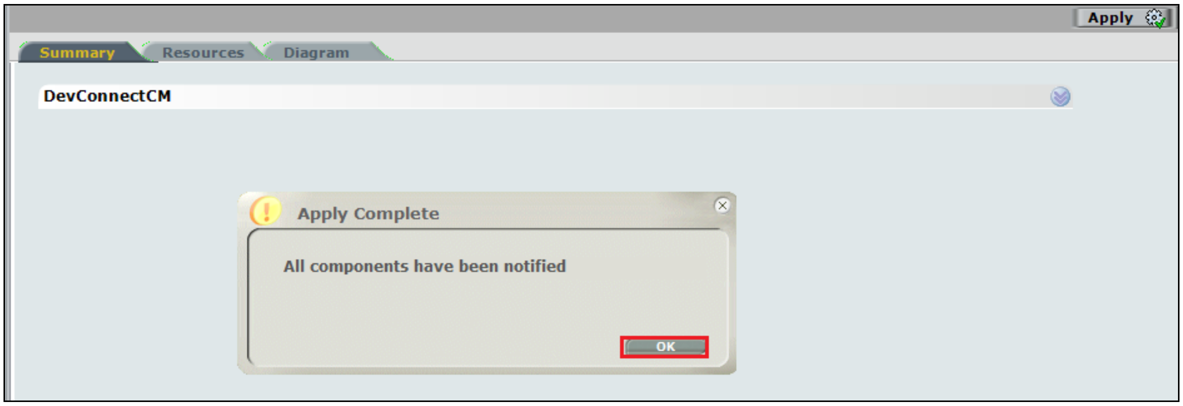
Review the information below.
Click Finish to create the new source pool.
Click Back to modify the source pool details.

Name: DevConnectPool
Switch: DevConnectCM (ID = 1075)
Media type: Audio
Source type: Device

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

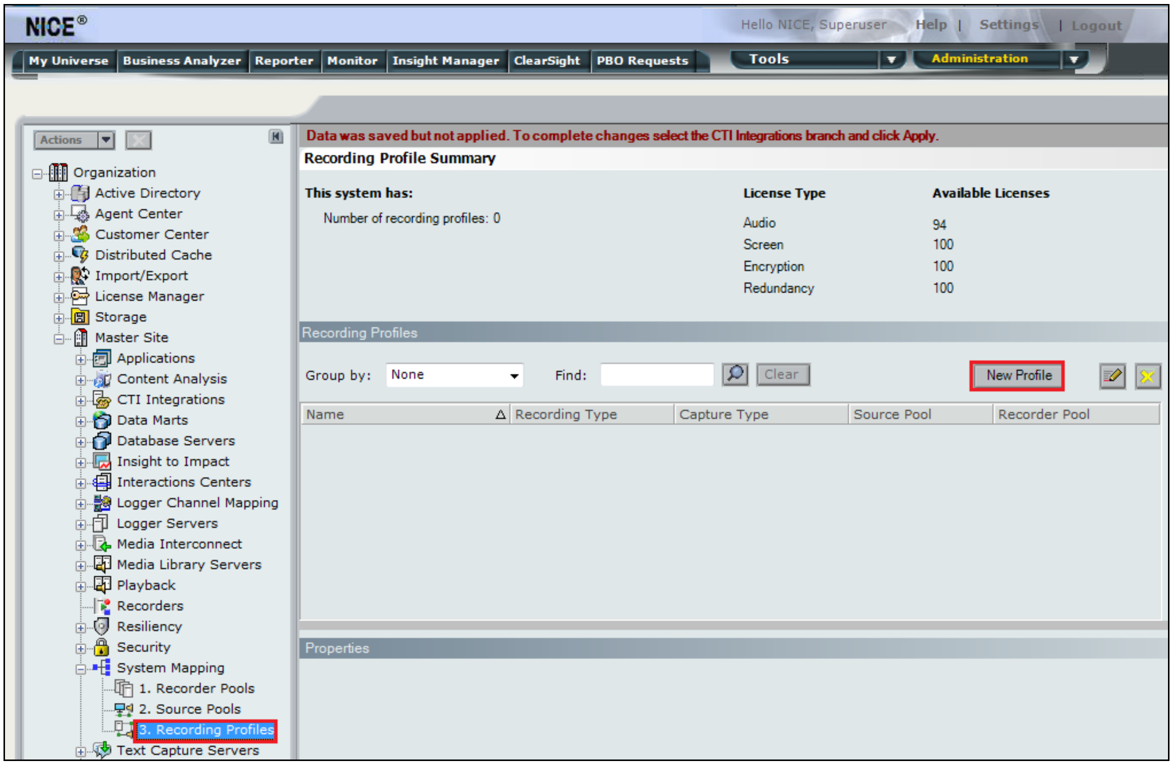


The following screen shows the changes were saved correctly. Click on **OK** to continue.



All components have been notified

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



Data was saved but not applied. To complete changes select the CTI Integrations branch and click Apply.

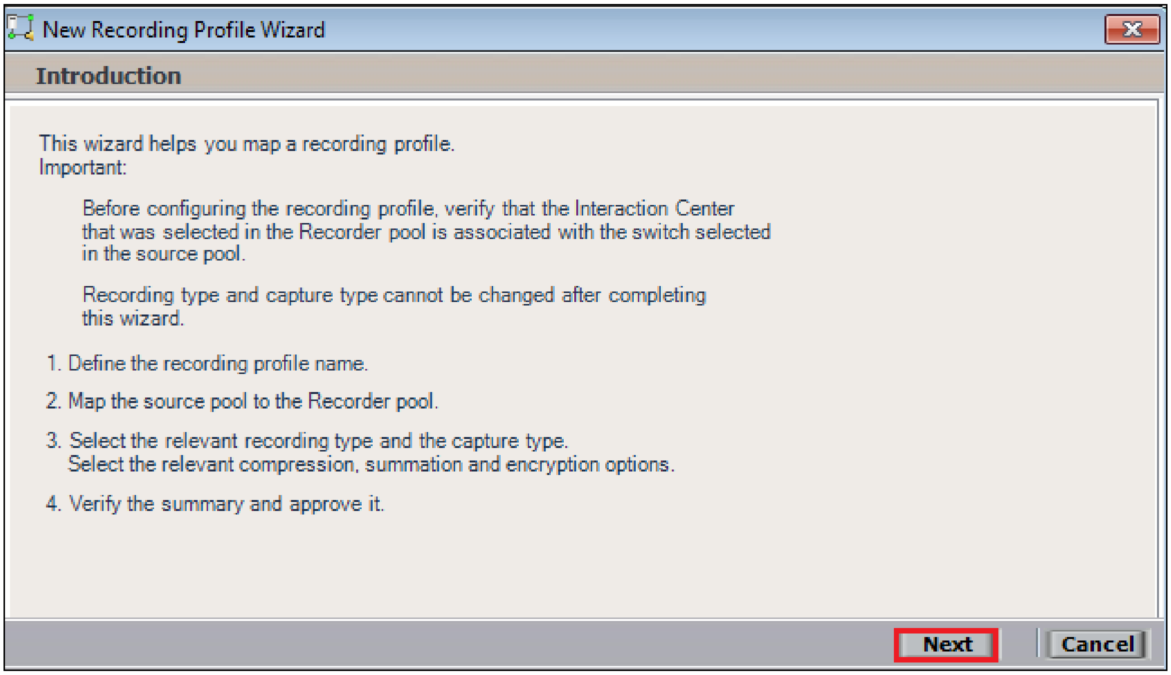
Recording Profile Summary

This system has:

Number of recording profiles: 0

License Type	Available Licenses
Audio	94
Screen	100
Encryption	100
Redundancy	100

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



Introduction

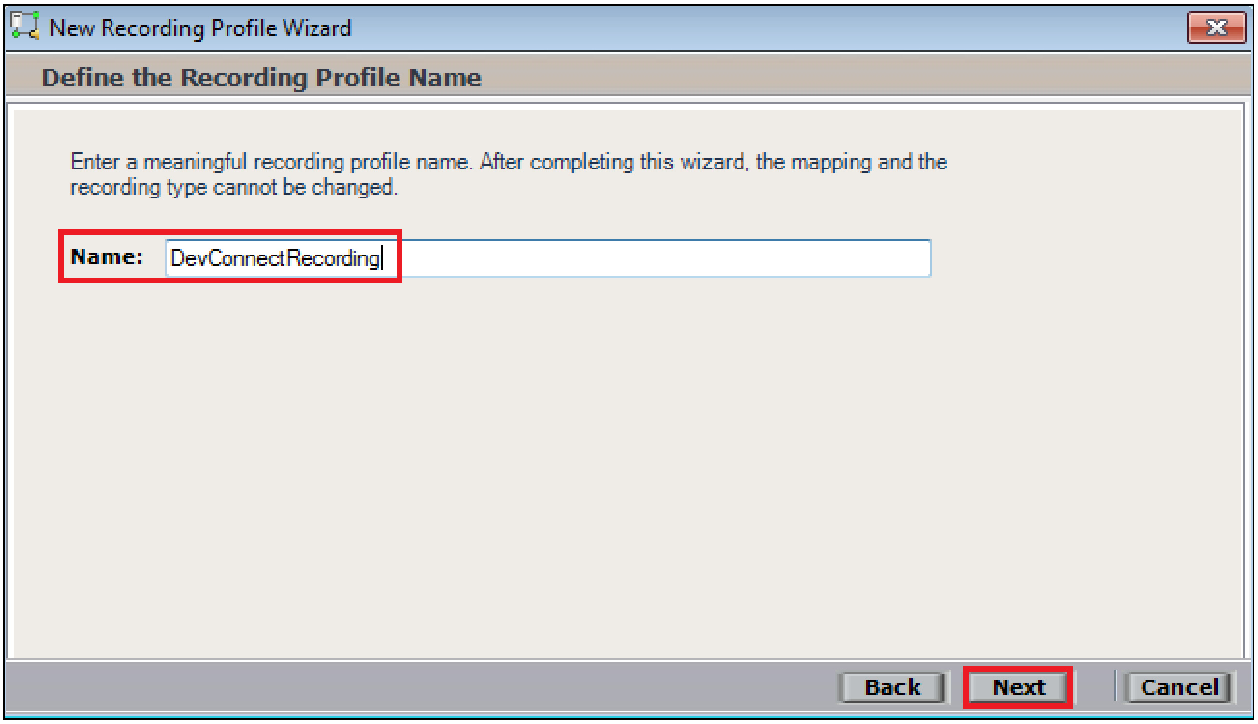
This wizard helps you map a recording profile.
Important:

Before configuring the recording profile, verify that the Interaction Center that was selected in the Recorder pool is associated with the switch selected in the source pool.

Recording type and capture type cannot be changed after completing this wizard.

1. Define the recording profile name.
2. Map the source pool to the Recorder pool.
3. Select the relevant recording type and the capture type.
   Select the relevant compression, summation and encryption options.
4. Verify the summary and approve it.

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

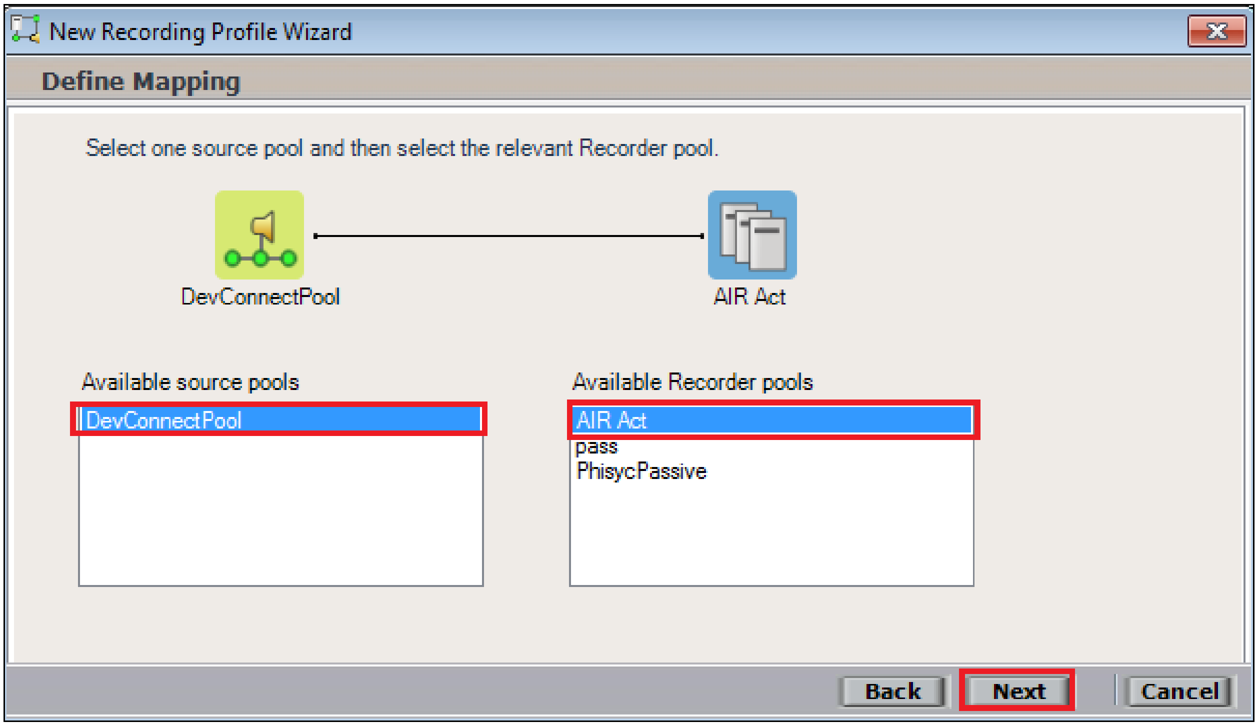


New Recording Profile Wizard
Define the Recording Profile Name
Back
Next
Cancel

Enter a meaningful recording profile name. After completing this wizard, the mapping and the recording type cannot be changed.

Name: DevConnectRecording

Select the correct **source pool** and **Recorder pool**, and then click **Next** to continue.

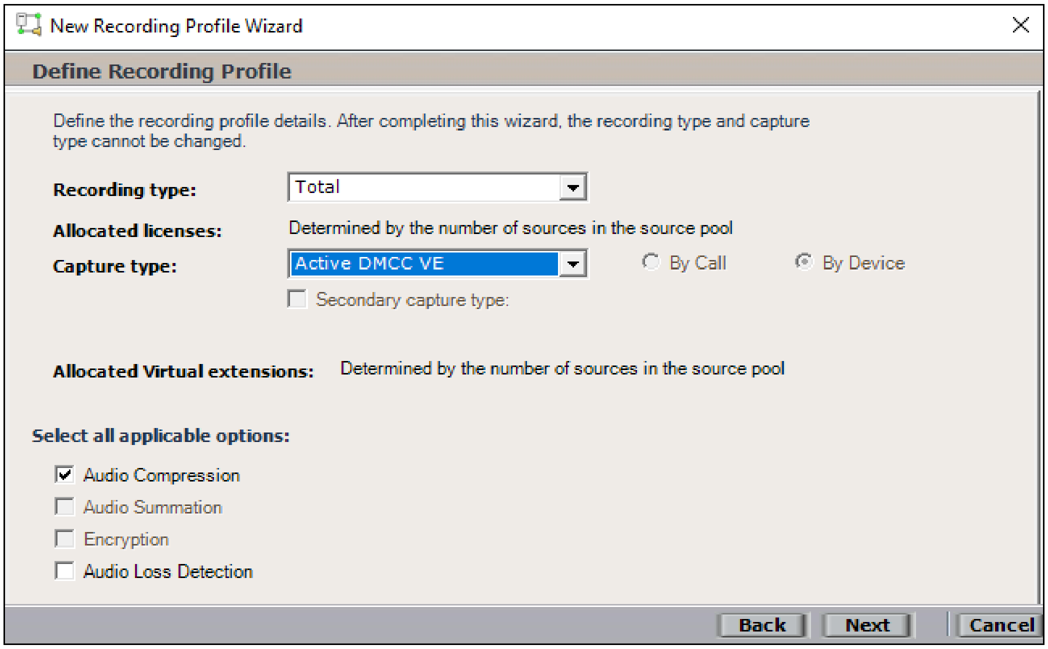


Select one source pool and then select the relevant Recorder pool.

Available source pools
DevConnectPool

Available Recorder pools
AIR Act
pass
PhisycPassive

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



New Recording Profile Wizard

Define Recording Profile

Define the recording profile details. After completing this wizard, the recording type and capture type cannot be changed.

Recording type: Total

Allocated licenses: Determined by the number of sources in the source pool

Capture type: Active DMCC VE ○ By Call ● By Device

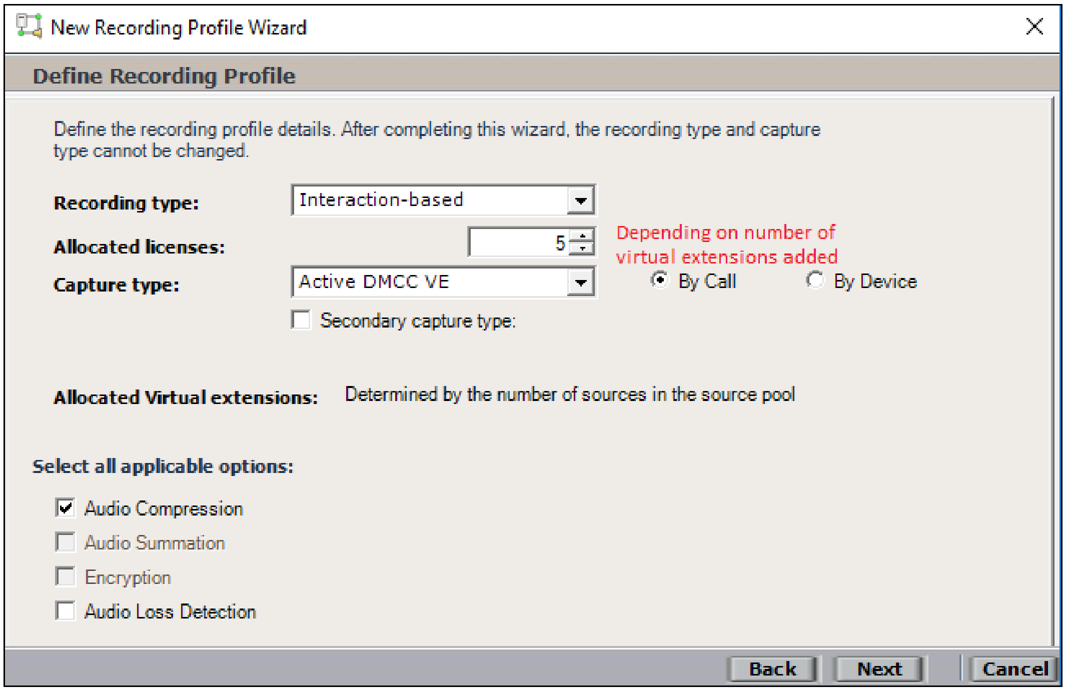
□ Secondary capture type:

Allocated Virtual extensions: Determined by the number of sources in the source pool

Select all applicable options:

☑ Audio Compression
□ Audio Summation
□ Encryption
□ Audio Loss Detection

Note: The only difference in the setup for Single Step Conference is with both the choice of **Recording type** which is set to **Interaction-based** and **Capture type** which will be **Active DMCC VE** and **By Call** as shown below.



New Recording Profile Wizard

Define Recording Profile

Define the recording profile details. After completing this wizard, the recording type and capture type cannot be changed.

Recording type: Interaction-based

Allocated licenses: 5 Depending on number of virtual extensions added

Capture type: Active DMCC VE ● By Call ○ By Device

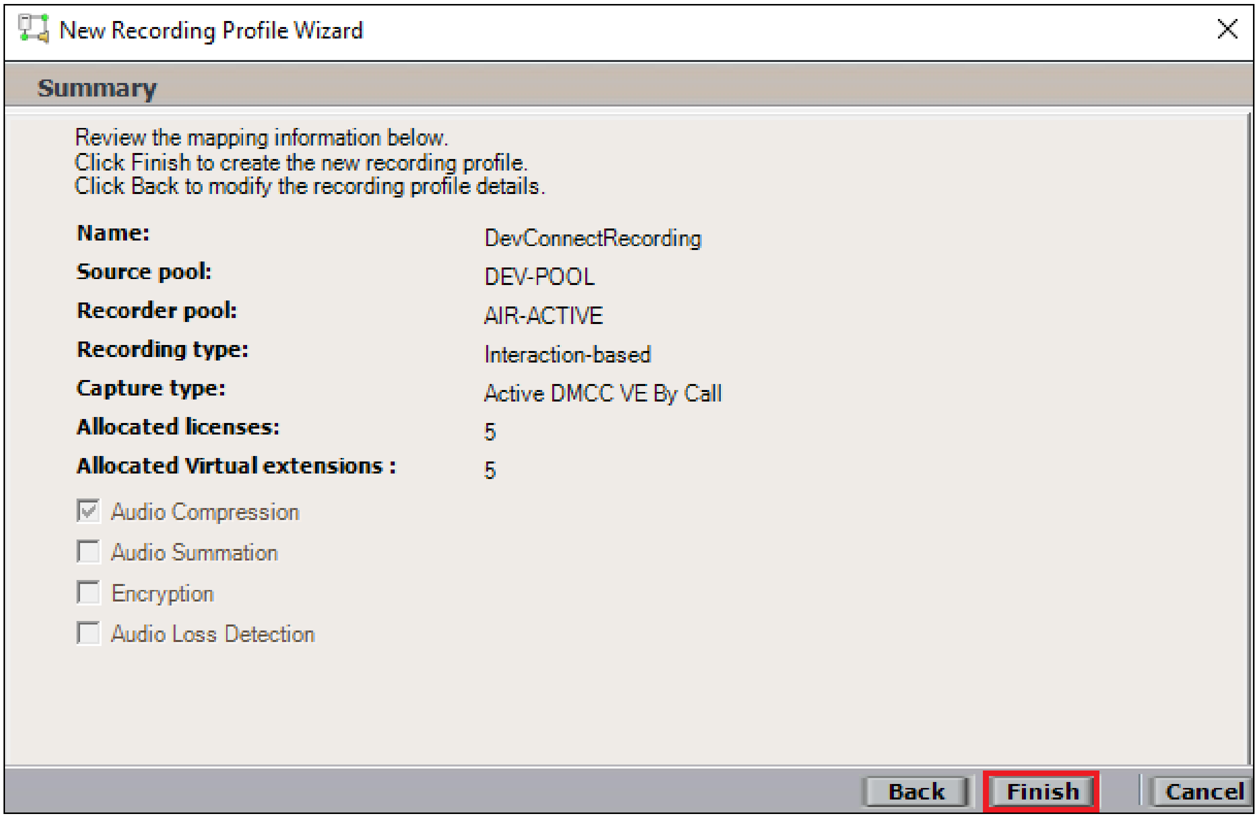
□ Secondary capture type:

Allocated Virtual extensions: Determined by the number of sources in the source pool

Select all applicable options:

☑ Audio Compression
□ Audio Summation
□ Encryption
□ Audio Loss Detection

Click on **Finish** to complete the **New Recording Profile Wizard**. The screen below shows that for Service Observe.



New Recording Profile Wizard

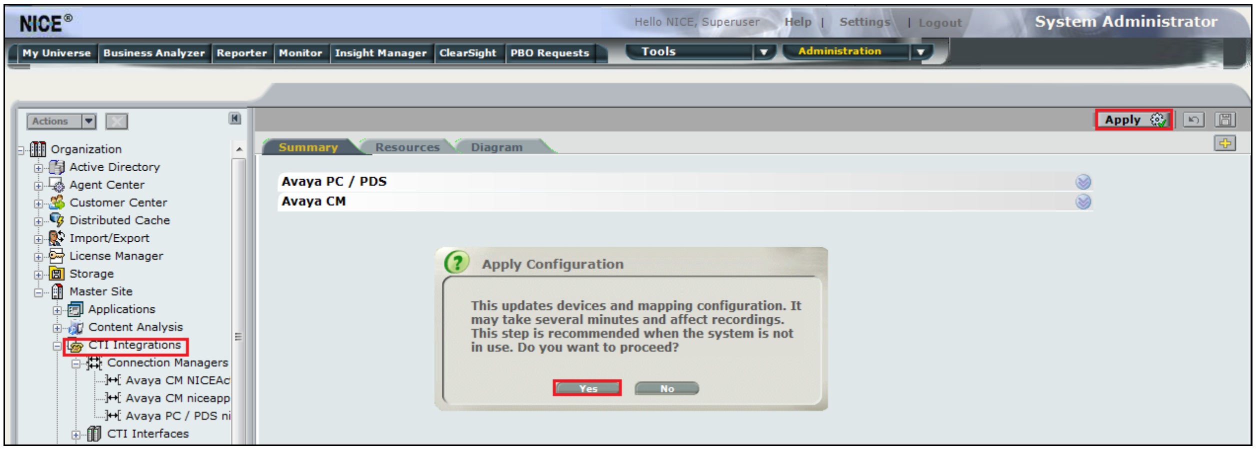
Summary

Review the mapping information below.
Click Finish to create the new recording profile.
Click Back to modify the recording profile details.

Name:	DevConnectRecording
Source pool:	DEV-POOL
Recorder pool:	AIR-ACTIVE
Recording type:	Interaction-based
Capture type:	Active DMCC VE By Call
Allocated licenses:	5
Allocated Virtual extensions :	5

☑ Audio Compression
☐ Audio Summation
☐ Encryption
☐ Audio Loss Detection

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



This updates devices and mapping configuration. It may take several minutes and affect recordings. This step is recommended when the system is not in use. Do you want to proceed?

This concludes the setup of the NICE Application Server for DMCC Service Observe and Single Step Conference 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.

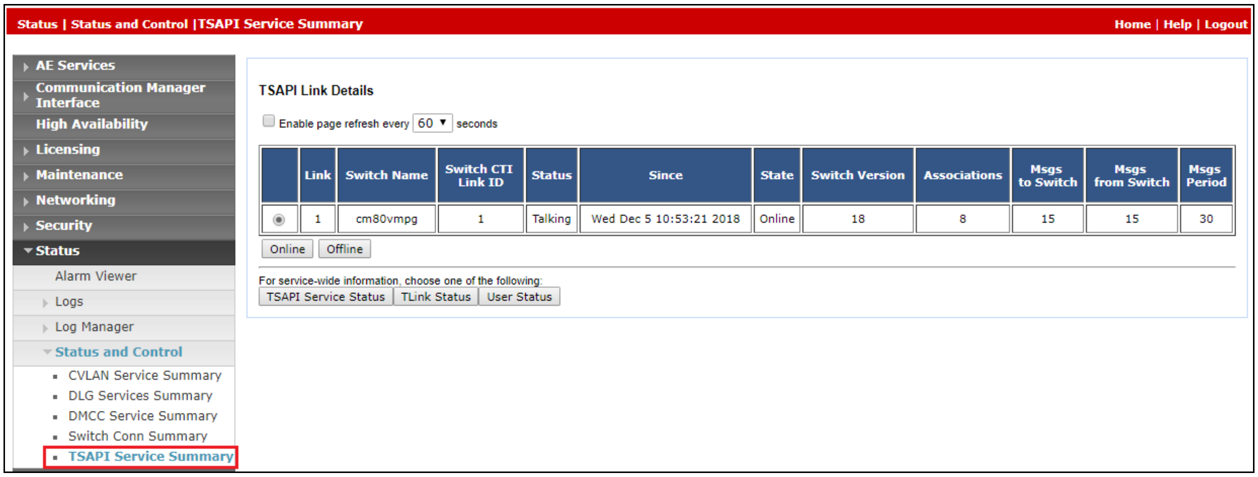
8.1. Verify Avaya Aura® Communication Manager CTI Service State

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**.

```
status aesvcs cti-link

                          AE SERVICES CTI LINK STATUS

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

1          8       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**.



TSAPI Link Details

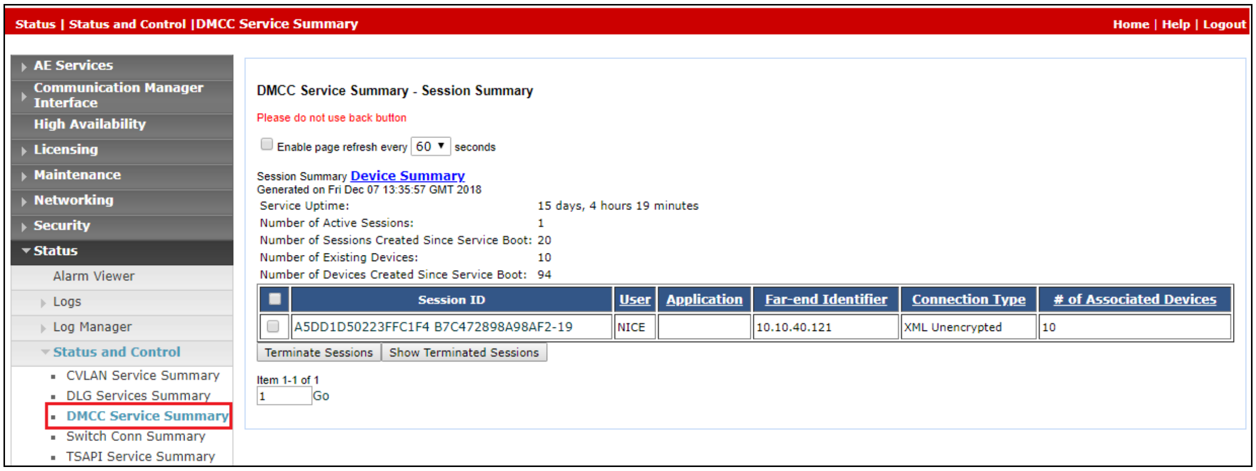
□ Enable page refresh every 60 ▾ seconds

	Link	Switch Name	Switch CTI Link ID	Status	Since	State	Switch Version	Associations	Msgs to Switch	Msgs from Switch	Msgs Period
●	1	cm80vmpg	1	Talking	Wed Dec 5 10:53:21 2018	Online	18	8	15	15	30

For service-wide information, choose one of the following:

8.3. Verify DMCC link on AES

Verify the status of the DMCC link by selecting **Status → Status and Control → DMCC Service Summary** to display the **DMCC Service Summary – Session Summary** screen. The screen below shows that the user **NICE** is connected from the IP address **10.10.40.121**, which is the NICE Application server.



DMCC Service Summary - Session Summary

Please do not use back button

□ Enable page refresh every 60 ▼ seconds

Session Summary Device Summary
Generated on Fri Dec 07 13:35:57 GMT 2018

Service Uptime:	15 days, 4 hours 19 minutes
Number of Active Sessions:	1
Number of Sessions Created Since Service Boot:	20
Number of Existing Devices:	10
Number of Devices Created Since Service Boot:	94

□	Session ID	User	Application	Far-end Identifier	Connection Type	# of Associated Devices
□	A5DD1D50223FFC1F4 B7C472898A98AF2-19	NICE		10.10.40.121	XML Unencrypted	10

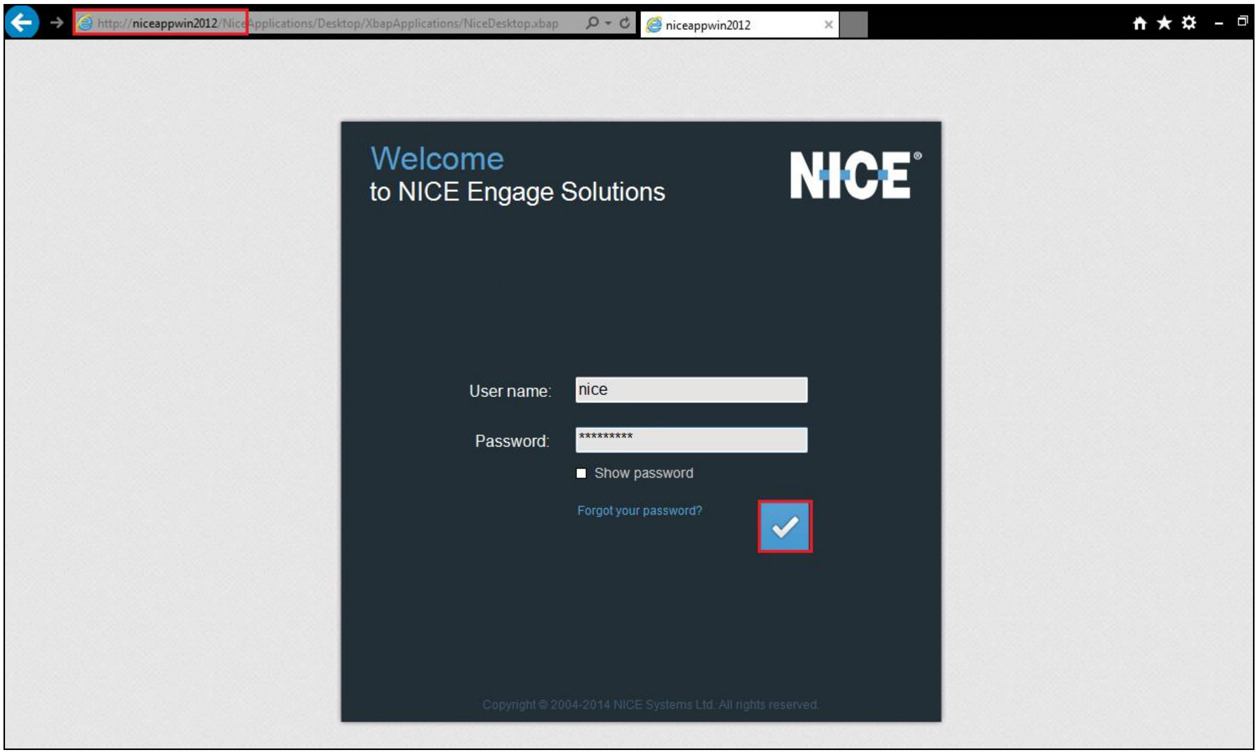
Terminate Sessions | Show Terminated Sessions

Item 1-1 of 1
1 Go

8.4. 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**.

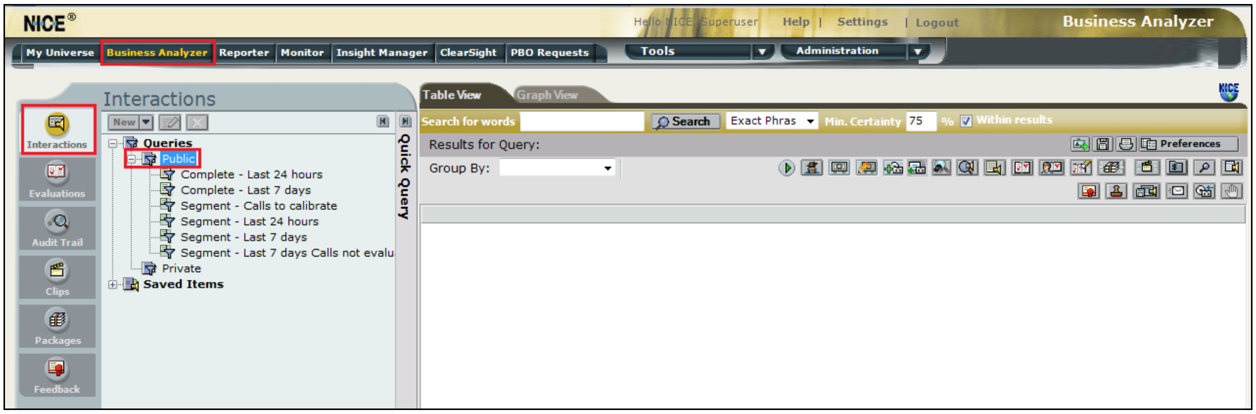

http://niceappwin2012/NiceApplications/Desktop/XbapApplications/NiceDesktop.xbap
niceappwin2012
Welcome
to NICE Engage Solutions
NICE
Forgot your password?
Copyright © 2004-2014 NICE Systems Ltd. All rights reserved.

User name: nice

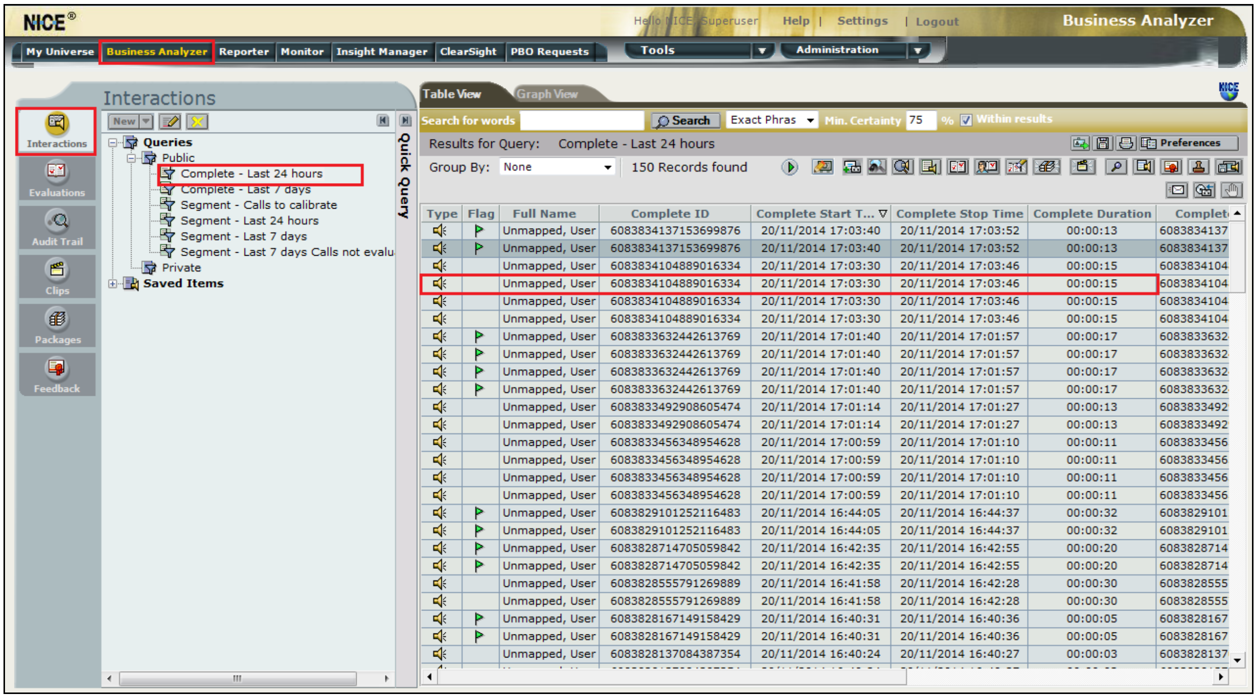
Password: *********

□ Show password

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

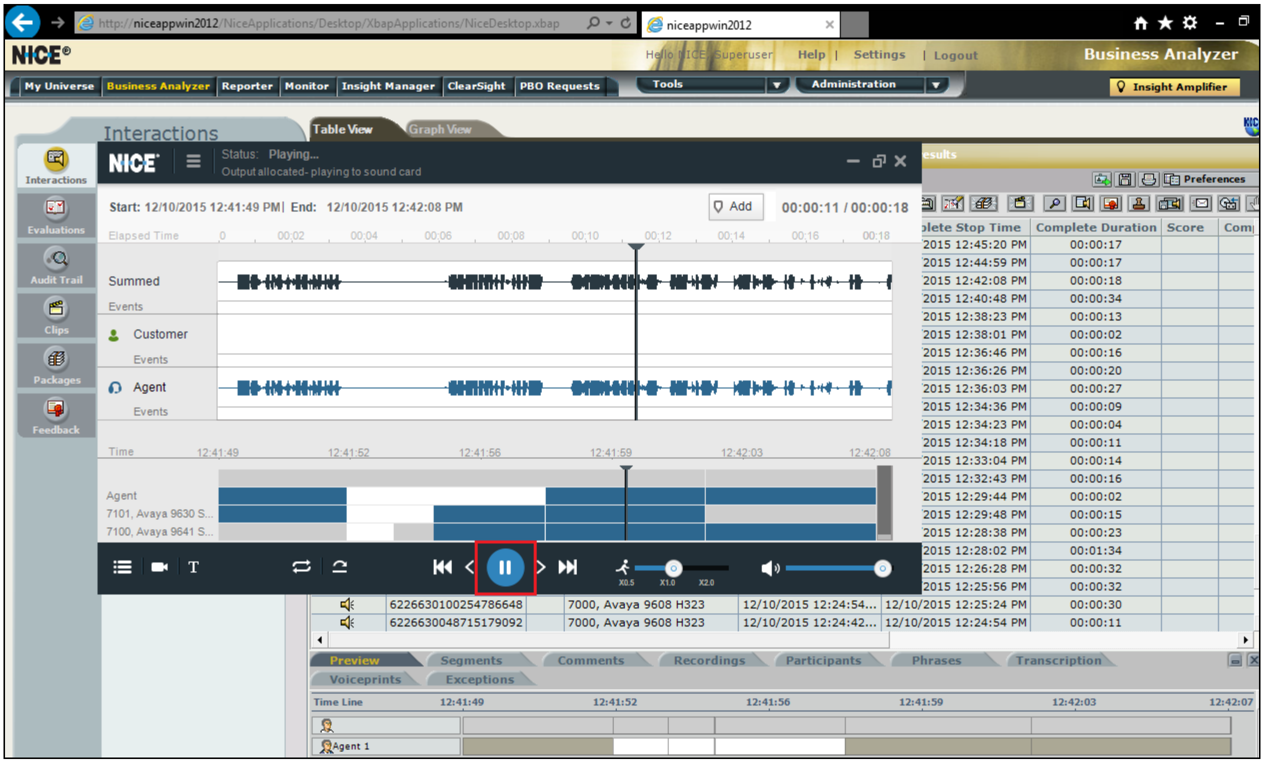


Results for Query: Complete - Last 24 hours

Group By: None 150 Records found

Type	Flag	Full Name	Complete ID	Complete Start T...	Complete Stop Time	Complete Duration	Complet
🔈	⚑	Unmapped, User	6083834137153699876	20/11/2014 17:03:40	20/11/2014 17:03:52	00:00:13	6083834137
🔈	⚑	Unmapped, User	6083834137153699876	20/11/2014 17:03:40	20/11/2014 17:03:52	00:00:13	6083834137
🔈		Unmapped, User	6083834104889016334	20/11/2014 17:03:30	20/11/2014 17:03:46	00:00:15	6083834104
🔈		Unmapped, User	6083834104889016334	20/11/2014 17:03:30	20/11/2014 17:03:46	00:00:15	6083834104
🔈		Unmapped, User	6083834104889016334	20/11/2014 17:03:30	20/11/2014 17:03:46	00:00:15	6083834104
🔈		Unmapped, User	6083834104889016334	20/11/2014 17:03:30	20/11/2014 17:03:46	00:00:15	6083834104
🔈	⚑	Unmapped, User	6083833632442613769	20/11/2014 17:01:40	20/11/2014 17:01:57	00:00:17	6083833632
🔈	⚑	Unmapped, User	6083833632442613769	20/11/2014 17:01:40	20/11/2014 17:01:57	00:00:17	6083833632
🔈	⚑	Unmapped, User	6083833632442613769	20/11/2014 17:01:40	20/11/2014 17:01:57	00:00:17	6083833632
🔈	⚑	Unmapped, User	6083833632442613769	20/11/2014 17:01:40	20/11/2014 17:01:57	00:00:17	6083833632
🔈		Unmapped, User	6083833492908605474	20/11/2014 17:01:14	20/11/2014 17:01:27	00:00:13	6083833492
🔈		Unmapped, User	6083833492908605474	20/11/2014 17:01:14	20/11/2014 17:01:27	00:00:13	6083833492
🔈		Unmapped, User	6083833456348954628	20/11/2014 17:00:59	20/11/2014 17:01:10	00:00:11	6083833456
🔈		Unmapped, User	6083833456348954628	20/11/2014 17:00:59	20/11/2014 17:01:10	00:00:11	6083833456
🔈		Unmapped, User	6083833456348954628	20/11/2014 17:00:59	20/11/2014 17:01:10	00:00:11	6083833456
🔈		Unmapped, User	6083833456348954628	20/11/2014 17:00:59	20/11/2014 17:01:10	00:00:11	6083833456
🔈	⚑	Unmapped, User	6083829101252116483	20/11/2014 16:44:05	20/11/2014 16:44:37	00:00:32	6083829101
🔈	⚑	Unmapped, User	6083829101252116483	20/11/2014 16:44:05	20/11/2014 16:44:37	00:00:32	6083829101
🔈	⚑	Unmapped, User	6083828714705059842	20/11/2014 16:42:35	20/11/2014 16:42:55	00:00:20	6083828714
🔈	⚑	Unmapped, User	6083828714705059842	20/11/2014 16:42:35	20/11/2014 16:42:55	00:00:20	6083828714
🔈		Unmapped, User	6083828555791269889	20/11/2014 16:41:58	20/11/2014 16:42:28	00:00:30	6083828555
🔈		Unmapped, User	6083828555791269889	20/11/2014 16:41:58	20/11/2014 16:42:28	00:00:30	6083828555
🔈	⚑	Unmapped, User	6083828167149158429	20/11/2014 16:40:31	20/11/2014 16:40:36	00:00:05	6083828167
🔈	⚑	Unmapped, User	6083828167149158429	20/11/2014 16:40:31	20/11/2014 16:40:36	00:00:05	6083828167
🔈		Unmapped, User	6083828137084387354	20/11/2014 16:40:24	20/11/2014 16:40:27	00:00:03	6083828137

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.



NICE

Status: Playing...
Output allocated- playing to sound card

Start: 12/10/2015 12:41:49 PM| End: 12/10/2015 12:42:08 PM

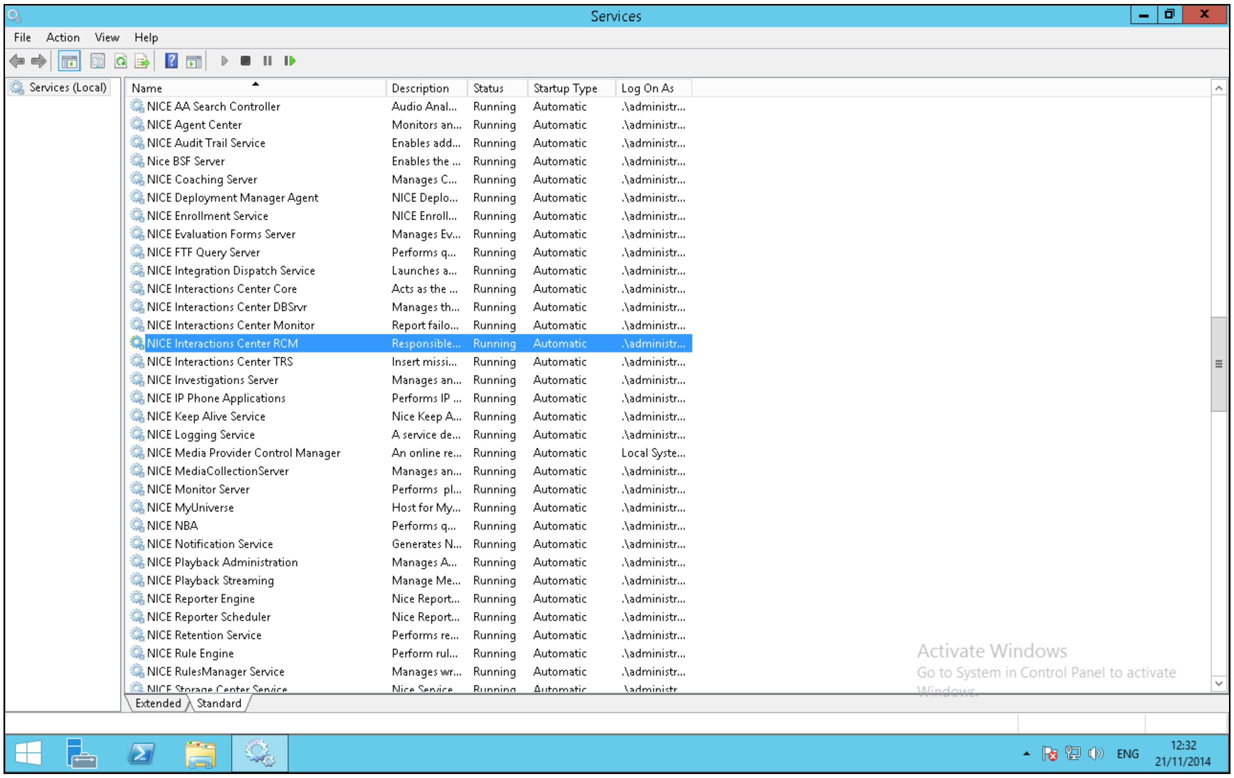
Add

00:00:11 / 00:00:18

				plete Stop Time	Complete Duration	Score	Com
				2015 12:45:20 PM	00:00:17		
				2015 12:44:59 PM	00:00:17		
				2015 12:42:08 PM	00:00:18		
				2015 12:40:48 PM	00:00:34		
				2015 12:38:23 PM	00:00:13		
				2015 12:38:01 PM	00:00:02		
				2015 12:36:46 PM	00:00:16		
				2015 12:36:26 PM	00:00:20		
				2015 12:36:03 PM	00:00:27		
				2015 12:34:36 PM	00:00:09		
				2015 12:34:23 PM	00:00:04		
				2015 12:34:18 PM	00:00:11		
				2015 12:33:04 PM	00:00:14		
				2015 12:32:43 PM	00:00:16		
				2015 12:29:44 PM	00:00:02		
				2015 12:29:48 PM	00:00:15		
				2015 12:28:38 PM	00:00:23		
				2015 12:28:02 PM	00:01:34		
				2015 12:26:28 PM	00:00:32		
				2015 12:25:56 PM	00:00:32		
🔈	6226630100254786648	7000, Avaya 9608 H323	12/10/2015 12:24:54...	12/10/2015 12:25:24 PM	00:00:30		
🔈	6226630048715179092	7000, Avaya 9608 H323	12/10/2015 12:24:42...	12/10/2015 12:24:54 PM	00:00:11		

8.5. 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 Advanced Interactions Server 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.



Name	Description	Status	Startup Type	Log On As
NICE AA Search Controller	Audio Anal...	Running	Automatic	.\administr...
NICE Agent Center	Monitors an...	Running	Automatic	.\administr...
NICE Audit Trail Service	Enables add...	Running	Automatic	.\administr...
Nice BSF Server	Enables the ...	Running	Automatic	.\administr...
NICE Coaching Server	Manages C...	Running	Automatic	.\administr...
NICE Deployment Manager Agent	NICE Deplo...	Running	Automatic	.\administr...
NICE Enrollment Service	NICE Enroll...	Running	Automatic	.\administr...
NICE Evaluation Forms Server	Manages Ev...	Running	Automatic	.\administr...
NICE FTF Query Server	Performs q...	Running	Automatic	.\administr...
NICE Integration Dispatch Service	Launches a...	Running	Automatic	.\administr...
NICE Interactions Center Core	Acts as the ...	Running	Automatic	.\administr...
NICE Interactions Center DBSrvr	Manages th...	Running	Automatic	.\administr...
NICE Interactions Center Monitor	Report failo...	Running	Automatic	.\administr...
NICE Interactions Center RCM	Responsible...	Running	Automatic	.\administr...
NICE Interactions Center TRS	Insert missi...	Running	Automatic	.\administr...
NICE Investigations Server	Manages an...	Running	Automatic	.\administr...
NICE IP Phone Applications	Performs IP ...	Running	Automatic	.\administr...
NICE Keep Alive Service	Nice Keep A...	Running	Automatic	.\administr...
NICE Logging Service	A service de...	Running	Automatic	.\administr...
NICE Media Provider Control Manager	An online re...	Running	Automatic	Local Syste...
NICE MediaCollectionServer	Manages an...	Running	Automatic	.\administr...
NICE Monitor Server	Performs pl...	Running	Automatic	.\administr...
NICE MyUniverse	Host for My...	Running	Automatic	.\administr...
NICE NBA	Performs q...	Running	Automatic	.\administr...
NICE Notification Service	Generates N...	Running	Automatic	.\administr...
NICE Playback Administration	Manages A...	Running	Automatic	.\administr...
NICE Playback Streaming	Manage Me...	Running	Automatic	.\administr...
NICE Reporter Engine	Nice Report...	Running	Automatic	.\administr...
NICE Reporter Scheduler	Nice Report...	Running	Automatic	.\administr...
NICE Retention Service	Performs re...	Running	Automatic	.\administr...
NICE Rule Engine	Perform rul...	Running	Automatic	.\administr...
NICE RulesManager Service	Manages wr...	Running	Automatic	.\administr...
NICE Storage Center Service	Nice Service ...	Running	Automatic	.\administr...

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 DMCC Service Observation and Single Step Conference to record calls. All feature functionality and serviceability test cases were completed successfully with some issues and 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 8.0

[4] *Avaya Aura® Session Manager Overview*, Doc # *03603323*

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

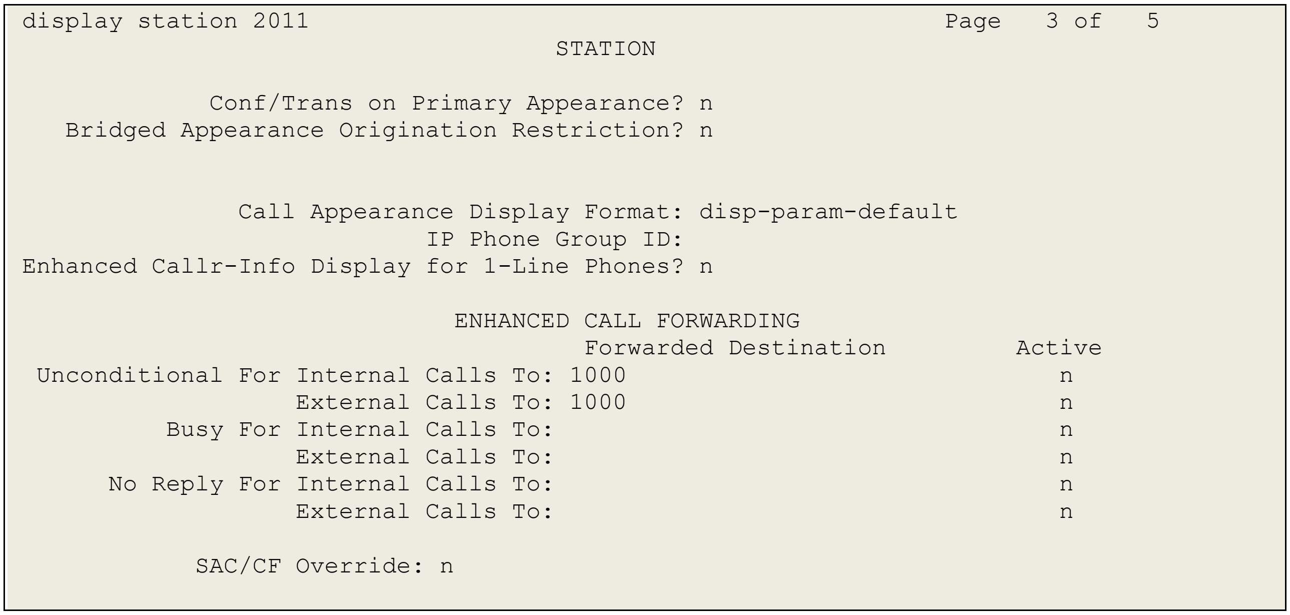
Appendix

Avaya one-X® Agent Softphone

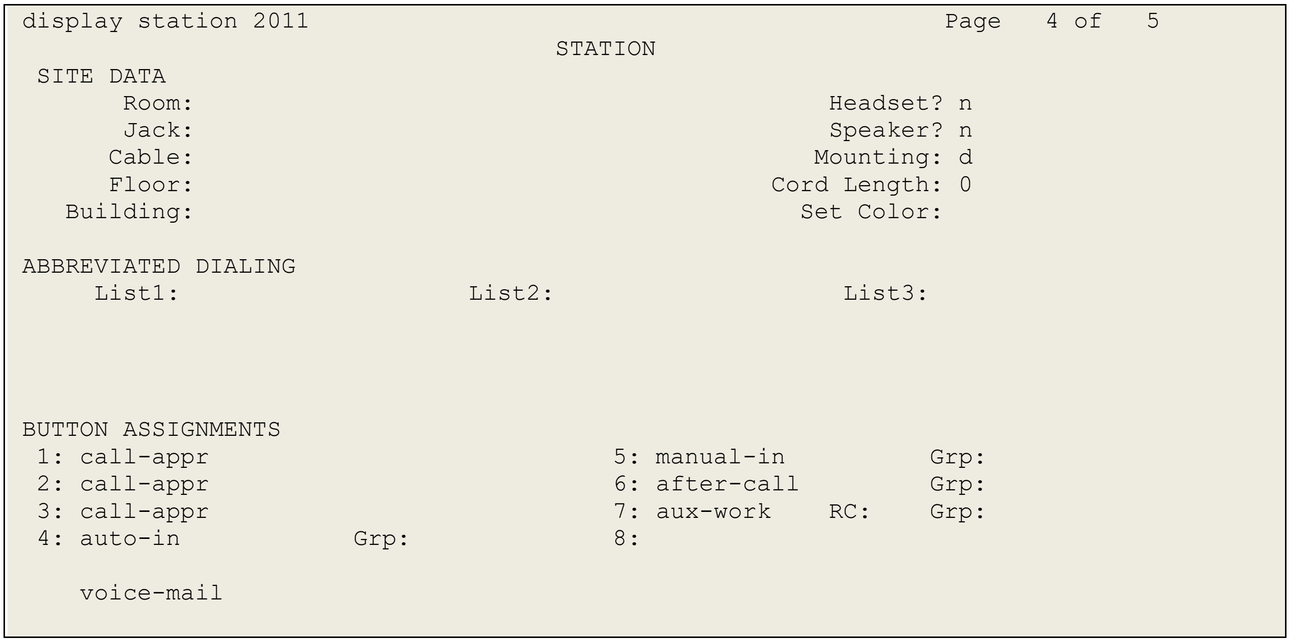
This is a printout of the Avaya one-X® Agent softphone used during compliance testing.

```
display station 2011                                            Page   1 of   5
                                     STATION

Extension: 2011                          Lock Messages? n               BCC: 0
     Type: 9630                          Security Code: *                TN: 1
     Port: S00031                      Coverage Path 1:                 COR: 1
     Name: one-X Agent1                Coverage Path 2:                 COS: 1
                                       Hunt-to Station:               Tests? y
STATION OPTIONS
                Location:                  Time of Day Lock Table:
              Loss Group: 19         Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 2011
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                  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: default

                                              Customizable Labels? Y
```

```
display station 2011                                            Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                              Data Restriction? n
   Redirect Notification? y                   Idle Appearance Preference? n
 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?
       Service Link Mode: as-needed                     EC500 State: enabled
         Multimedia Mode: enhanced                 Audible Message Waiting? n
    MWI Served User Type:                       Display Client Redirection? n
              AUDIX Name:                      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: 2011          Always Use? n IP Audio Hairpinning? n
```



```
display station 2011                                            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                              n
                   External Calls To: 1000                              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
```



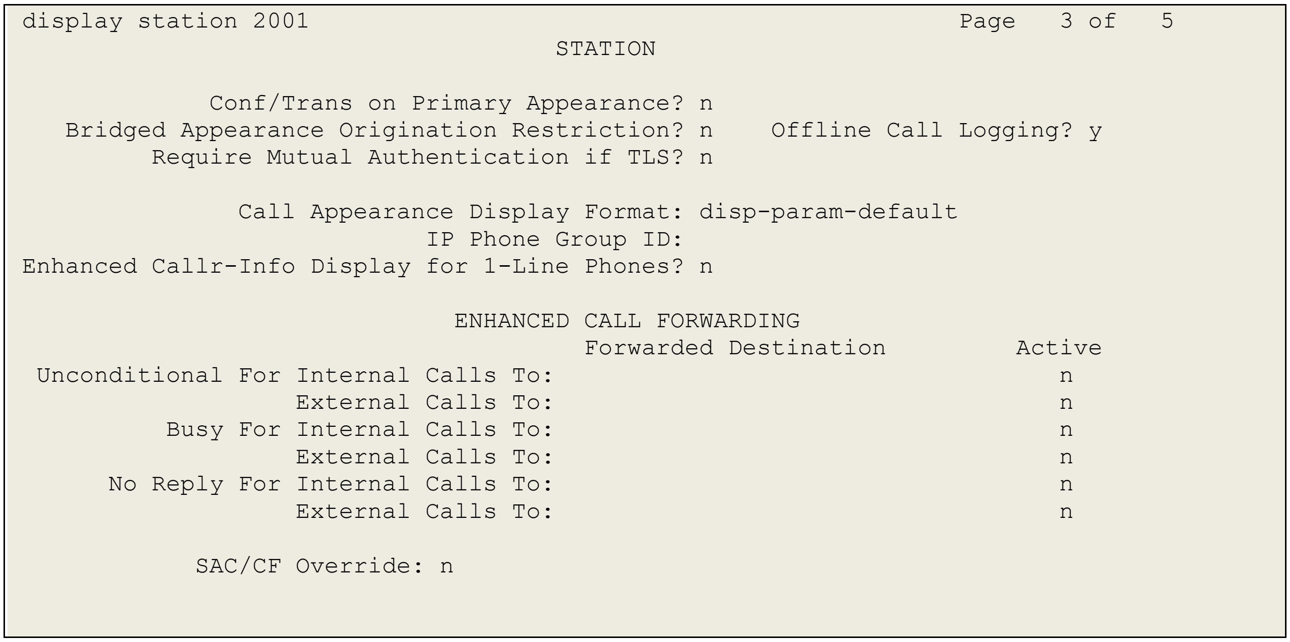
```
display station 2011                                            Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                            Headset? n
       Jack:                                            Speaker? n
      Cable:                                           Mounting: d
      Floor:                                        Cord Length: 0
   Building:                                          Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:




BUTTON ASSIGNMENTS
 1: call-appr                            5: manual-in         Grp:
 2: call-appr                            6: after-call        Grp:
 3: call-appr                            7: aux-work    RC:    Grp:
 4: auto-in            Grp:              8:

    voice-mail
```

Avaya 9608 H.323 Deskphone

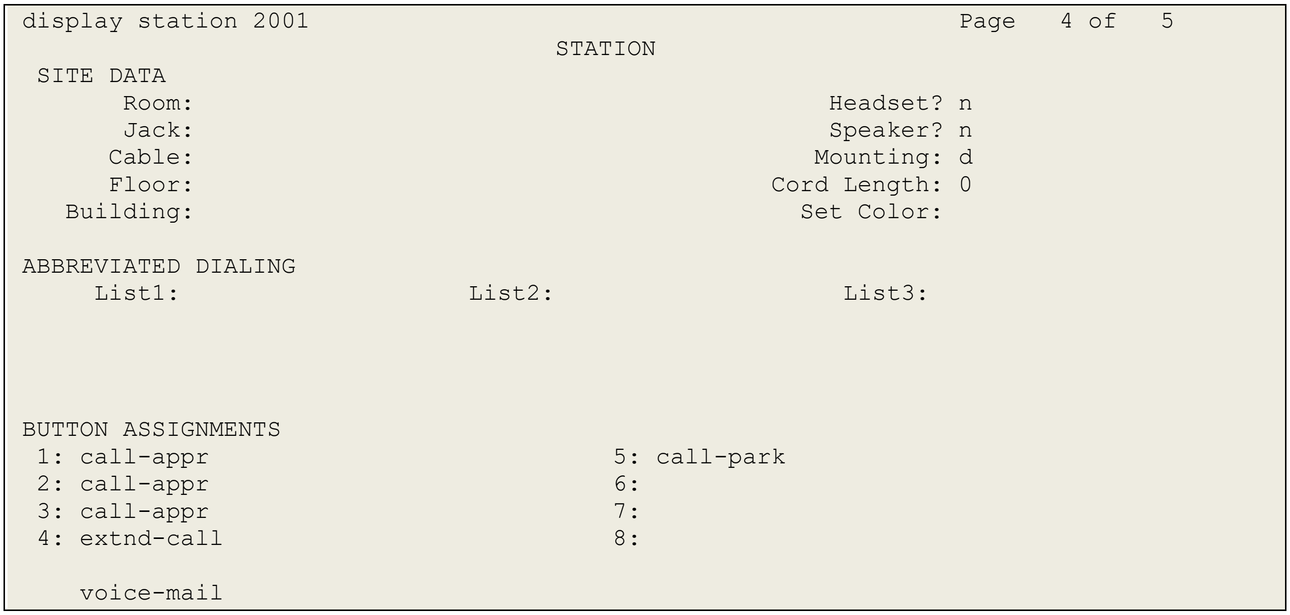
This is a printout of the Avaya 9608 H.323 deskphone used during compliance testing.

```
display station 2001                                             Page   1 of   5
                                     STATION

Extension: 2001                          Lock Messages? n               BCC: 0
     Type: 9608                          Security Code: *                TN: 1
     Port: S00000                      Coverage Path 1: 1               COR: 1
     Name: Ext2001                     Coverage Path 2:                 COS: 1
                                       Hunt-to Station:               Tests? y
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 2001
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                 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 2001                                             Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe              Auto Select Any Idle Appearance? n
          LWC Activation? y                         Coverage Msg Retrieval? y
  LWC Log External Calls? n                                    Auto Answer: none
             CDR Privacy? n                               Data Restriction? n
   Redirect Notification? y                     Idle Appearance Preference? n
 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?
       Service Link Mode: as-needed                        EC500 State: enabled
         Multimedia Mode: enhanced                 Audible Message Waiting? n
    MWI Served User Type: sip-adjunct           Display Client Redirection? n
                                               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: 2001         Always Use? n IP Audio Hairpinning? n
```



```
display station 2001                                             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:                                   n
                   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 2001                                             Page   4 of   5
                                     STATION
 SITE DATA
      Room:                                           Headset? n
      Jack:                                           Speaker? n
     Cable:                                          Mounting: d
     Floor:                                       Cord Length: 0
  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:

    voice-mail
```

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.