Avaya Solution & Interoperability Test Lab

Application Notes for NICE Uptivity with Avaya Aura® Application Enablement Services and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes cover the interoperability compliance testing of the NICE Uptivity recording solution with Avaya Aura® Communication Manager and Avaya Aura® Application Enablement Services (AES).

In the compliance testing, NICE Uptivity used various registration features from the Avaya Aura® Application Enablement Services Device, Media, and Call Control interface to capture media associated with monitored agent stations for call recording.

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

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

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1. Introduction

These Application Notes describe the configuration steps for the Uptivity recording solution from NICE to interoperate with the Avaya solution consisting of Avaya Aura® Communication Manager R7.1.3, Avaya Aura® Session Manager R7.1.3, and Avaya Aura® Application Enablement Services R7.1.3. Uptivity uses Communication Manager's Multiple Registration and/or Single Step Conference (SSC) 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**.

The NICE Uptivity system interfaces with Avaya Aura® Communication Manager and Avaya Aura® Application Enablement Services, using the Telephony Service API (TSAPI) to obtain call event information and the Device, Media & Call Control (DMCC) API to obtain audio via various Registration methods.

DMCC allows software vendors to create soft phones on a recording server, and use them to monitor and record Avaya phones. This is purely a software solution and does not require telephony boards or any wiring beyond a typical network infrastructure.

The compliance testing focused on the monitoring and recording performed by NICE Uptivity of calls placed to and/or from digital, H.323, and SIP telephones, H.323 and SIP softphones, agents, hunt groups and Vector Directory Numbers (VDNs) supported by Communication Manager. NICE Uptivity uses:

- The TSAPI interface of AES to monitor extensions and hunt groups to obtain call and login events.
- The DMCC interface of AES to register the recorder with Communication Manager in order to record devices.

2. General Test Approach and Test Results

The feature test cases were performed manually. Upon start of the NICE Uptivity application, the application established a DMCC Stream with Application Enablement Services to register the recorder as a Main or (In)Dependent IP Endpoint for each of the virtual or target stations on Communication Manager, and to receive Third Party call events via a TSAPI Stream.

Each call was handled manually at the agent station with generation of unique audio content for recording. Necessary agent actions such as hold and reconnect were performed from the Desk Phone or Softphone to test various call scenarios.

The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet connection to NICE Uptivity and Application Enablement Services.

The verification of tests included use of status screens, logs for proper message exchanges and use of the NICE Uptivity web interfaces for proper logging and playback of call recordings.

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 interfaces between Avaya systems and NICE Uptivity utilized enabled capabilities of TLS for Application links between Application Enablement Services and CM, and streams between Application Enablement Services and NICE Uptivity. However, SRTP is not currently supported with the NICE Uptivity solution so all media sessions between the Gateways and recorder utilized Advanced Encryption Standard (AES) encrypted media.

This test was conducted in a lab environment simulating a basic customer network environment. The testing focused on the standards-based interface between the Avaya solution and the NICE Uptivity solution.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing.

The feature testing focused on verifying the following on NICE Uptivity:

- Handling of call events.
- Use of DMCC registration services to register the IP softphones.
- Use of TSAPI and\or DMCC monitoring services and media control events to obtain the media and call events from the phones.
- Proper recording, logging, and playback of calls for scenarios involving hold, reconnect, conference, transfer.

Serviceability testing focused on verifying the ability of NICE Uptivity to recover from adverse conditions, such as disconnecting and reconnecting the Ethernet connection to NICE Uptivity and Application Enablement Services.

2.2. Test Results

All Test cases were executed and verified. The only observations were the current lack of support for SRTP media streams although as stated, AES encryption is supported. Also, on consultative calls (transfers and conference), the internal station to station meta data is tagged to the second leg of the call, not the original external caller data. Blind transfer and conference tag original caller data to both legs of the call.

2.3. Support

Technical support on NICE Uptivity can be obtained through the following:

- 1-888-922-5526
- Or on the web at https://help.incontact.com/

3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**.

The agent station extensions used in the compliance testing were 30001 through 30006. Virtual extensions 33001-33009 were also available for testing.

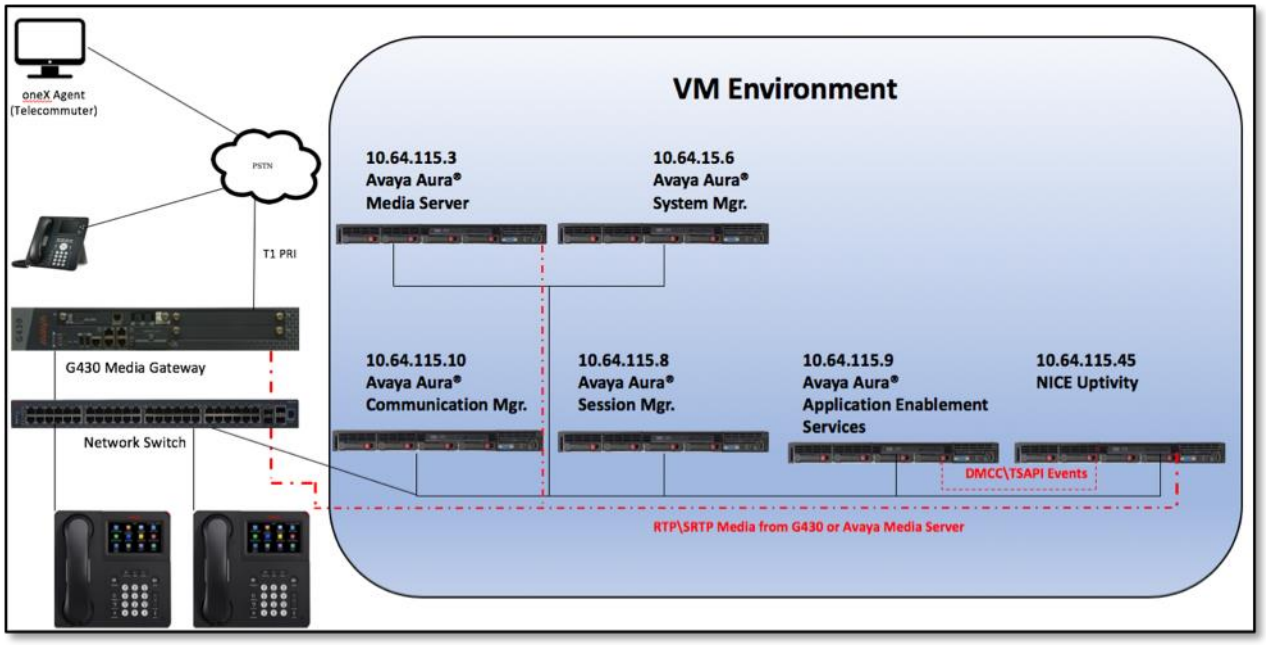


Figure 1: Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on VMWare ESXi 6.0	R7.1.3 (Feature Pack 3) (7.1.3.0.0.532.0-24515)
Avaya Aura® Application Enablement Services running on VMWare ESXi 6.0	R7.1.3 (7.1.3.0.1.7-0)
Avaya G430 Media Gateway	38.20.1/1
Avaya Aura® Media Server running on VMWare ESXi 6.0	7.8.0.333
Avaya 6408D Digital Station	N/A
Avaya 9670G	3.280A (H.323)
Avaya 9641G	7.1.1.09 (SIP)
Avaya 9611G	6.6506 (H.323)
Avaya 9630G	2.6.17 (SIP)
Avaya oneX® Agent	2.5.60129.0 (H.323)
NICE Uptivity on Microsoft Windows (running on VMWare) Avaya DMCC .NET	17.3.0605.652 2012R2 Standard ESXi 6.0 6.3.3

5. Configure Avaya Aura® Communication Manager

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

- Verify Feature and License for the integration
- Administer Communication Manager System Features
- Administer IP Services for Application Enablement Services
- Administer Computer Telephony Integration (CTI) Link
- Verify Recorded Extensions & Add Virtual Stations

All configuration changes in this section for Communication Manager are performed through the System Access Terminal (SAT) interface. For more details on configuring Communication Manager, refer to the Avaya product documentation in **Section 10**.

The test environment consisted of a mix of phones. PRI trunks connect the test systems to the PSTN enabling calls and EC500 interactions with external devices. These Application Notes do not cover the full environment, much of it is standard implementation. Rather, these notes focus on the parts that impact the integration with the tested application.

Recording can be performed via DMCC using one of three methods, depending on the target station types. Understanding this will help with perspective in this document. Bold indicates the methods used in the tested solution.

Target Endpoint Type	Multi-Registration	Service Observe	Single Step Conference
SIP	No	Yes (Main Mode)	**Yes (Main Mode)**
SIP Dual-Reg	**Yes (Independent Mode)**	Yes (Main Mode)	Yes (Main Mode)
H.323	**Yes (Dependent Mode)**	Yes (Main Mode)	Yes (Main Mode)
Digital	**Yes (Dependent Mode)**	Yes (Main Mode)	Yes (Main Mode)
Analog	N/A	Yes (Main Mode)	Yes (Main Mode)

The NICE Uptivity application uses Dependent (Multiple Registration) to record Digital and H.323 endpoints, Independent Mode (Multiple Registration) to record SIP Dual-Registration devices, and Single Step conference virtual extensions using Main mode to record any target that cannot use Multiple Registration.

5.1. Verify Feature and License for the integration

For recording solutions, the following license are required on Communication Manager:

- Recorders that use Single Step Conference or Service Observation (i.e. Registering using the MAIN option), will use a virtual extension to join the recorder to calls. Each recording port using these methods will consume a **Station** license when administered.

```
display system-parameters customer-options                    Page   1 of 12
                              OPTIONAL FEATURES

     G3 Version: V17                              Software Package: Enterprise
       Location: 2                                 System ID (SID): 1
       Platform: 28                                Module ID (MID): 1

                                                          USED
                                 Platform Maximum Ports: 6400  94
                                       Maximum Stations: 2400  16
                                Maximum XMOBILE Stations: 2400  0
                    Maximum Off-PBX Telephones - EC500: 9600  1
                    Maximum Off-PBX Telephones -   OPS: 9600  3
                    Maximum Off-PBX Telephones - PBFMC: 9600  0
                    Maximum Off-PBX Telephones - PVFMC: 9600  0
                    Maximum Off-PBX Telephones - SCCAN: 0     0
                         Maximum Survivable Processors: 313   0
```

- Recorders using the Multiple Registration (ie. Registering using the DEPENDENT or INDEPENDENT option) do not require additional station license. All methods will consume a **Concurrently Registered IP Station** license:

```
display system-parameters customer-options                      Page   2 of 12
                               OPTIONAL FEATURES

IP PORT CAPACITIES                                             USED
                     Maximum Administered H.323 Trunks: 4000  0
          Maximum Concurrently Registered IP Stations: 2400  11
            Maximum Administered Remote Office Trunks: 4000  0
Maximum Concurrently Registered Remote Office Stations: 2400  0
             Maximum Concurrently Registered IP eCons: 68    0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                       Maximum Video Capable Stations: 2400  0
                   Maximum Video Capable IP Softphones: 2400  0
                       Maximum Administered SIP Trunks: 4000  55
  Maximum Administered Ad-hoc Video Conferencing Ports: 4000  0
   Maximum Number of DS1 Boards with Echo Cancellation: 80    0
```

- In previous versions of Communication Manager, the IP_API_A (DMCC) may have been enforced on Communication Manager, and\or Application Enablement. With version 7 of Communication Manager, this RTU is completely controlled by Application Enablement Services (DMCC_DMC).
- Customers who purchase Application Enablement will have ASAI capabilities enabled on the Communication Manager. These include **ASAI Link Core Capabilities** and\or **Computer Telephony Adjunct Links** (enabled when TSAPI Basic RTU are purchased):

```
display system-parameters customer-options                      Page   4 of 12
                               OPTIONAL FEATURES

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

5.2. Administer Communication Manager System Features

If UCID is desired, make the following changes using an appropriate Node ID based on the customer requirements.

```
change system-parameters features                                Page   5 of 19
                        FEATURE-RELATED SYSTEM PARAMETERS

SYSTEM PRINTER PARAMETERS
  Endpoint:               Lines Per Page: 60

SYSTEM-WIDE PARAMETERS
                                    Switch Name: SIL Denver
           Emergency Extension Forwarding (min): 10
         Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? n
                              COR to Use for DPT: station
               EC500 Routing in Survivable Mode: dpt-then-ec500
MALICIOUS CALL TRACE PARAMETERS
              Apply MCT Warning Tone? n   MCT Voice Recorder Trunk Group:
      Delay Sending RELease (seconds): 0
SEND ALL CALLS OPTIONS
     Send All Calls Applies to: station    Auto Inspect on Send All Calls? n
             Preserve previous AUX Work button states after deactivation? n
UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? y    UCID Network Node ID: 1
```

```
change system-parameters features                                Page  13 of 19
                        FEATURE-RELATED SYSTEM PARAMETERS
 CALL CENTER MISCELLANEOUS
           Callr-info Display Timer (sec): 10
                         Clear Callr-info: next-call
        Allow Ringer-off with Auto-Answer? n

    Reporting for PC Non-Predictive Calls? n

            Agent/Caller Disconnect Tones? n
          Interruptible Aux Notification Timer (sec): 3
             Zip Tone Burst for Callmaster Endpoints: double

  ASAI
                    Copy ASAI UUI During Conference/Transfer? n
               Call Classification After Answer Supervision? n
                                          Send UCID to ASAI? y
                 For ASAI Send DTMF Tone to Call Originator? y
          Send Connect Event to ASAI For Announcement Answer? n
  Prefer H.323 Over SIP For Dual-Reg Station 3PCC Make Call? y
```

5.3. Administer IP Services for Application Enablement Services

Use the **change ip-services** command to Enable IP-Services for Application Enablement Services:

```
change ip-services                                          Page   1 of   3

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

On page 3, add the **hostname** for the Application Enablement Services server, and a **password** that will be entered in the AES setup in the next section.

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

   Server ID    AE Services       Password          Enabled    Status
                   Server
      1:       sildvaes          *                   y        in use
```

5.4. Administer Computer Telephony Integration (CTI) Link

Add a CTI-Link with **ADJ-IP** link Type, the name is not critical:

```
add cti-link 1                                              Page   1 of   3
                                  CTI LINK
 CTI Link: 1
Extension: 30000
     Type: ADJ-IP
                                                                    COR: 1
     Name: SILDVAES
```

5.5. Verify Recorded Extensions & Add Virtual Stations

For recording solutions using MAIN registration type (Single Step Conference or Service Observe), virtual extensions are administered for each recording port. In this test environment, stations 33000 – 33009 were previously built as:

- **Type** = 9630
- **Security Code** = eg: 123456 (this will be required when setting up the recorder)
- **IP Softphone** = y
- **COR** = 1 (note, only relevant for Service Observe methods)

```
change station 33000                                        Page   1 of   5
                                    STATION

Extension: 33000                        Lock Messages? n                BCC: 0
     Type: 9630                       Security Code: *                 TN: 1
     Port: S00002                   Coverage Path 1: ____              COR: 1
     Name: DMCC1                    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: 33000
           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
```

- All other settings may be left at defaults.

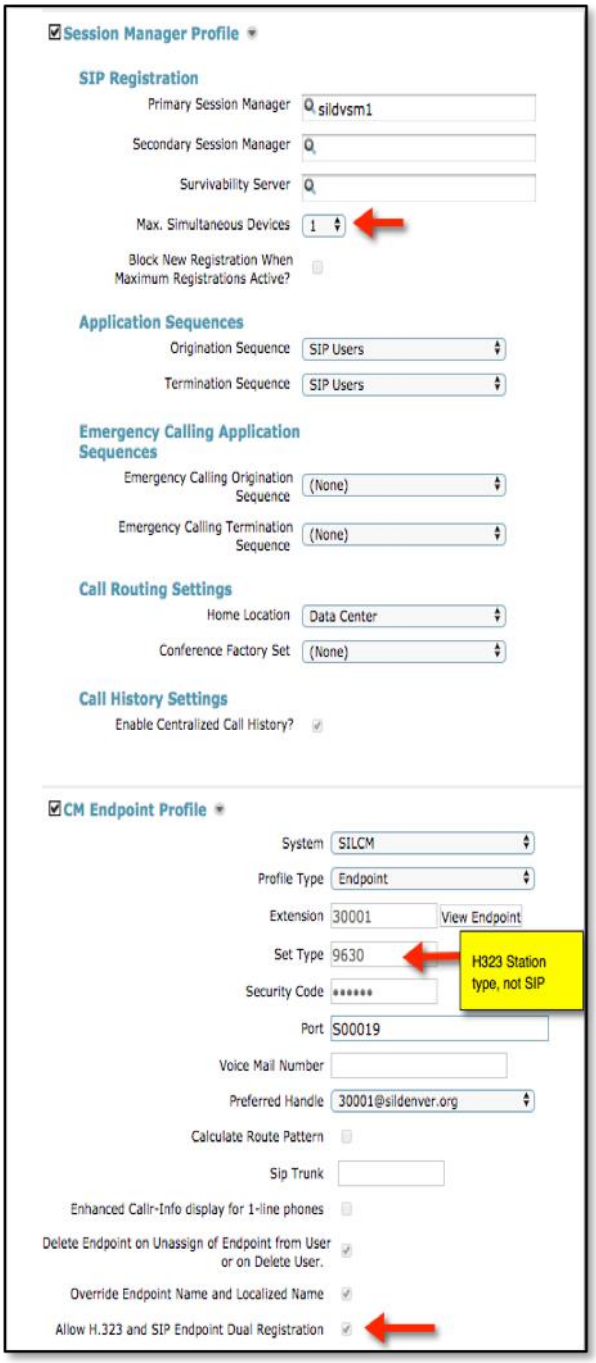
For Multiple Registration methods, the agent extensions must be administered as follows:

- **Security Code** = eg: 123456 (this will be required when setting up the recorder)
- **IP Softphone** = y

Agent Stations that will be recorded using Service Observation or Single Step Conference do not require IP Softphone to be enabled.

For SIP endpoints to be able to be recorded using Multiple Registration, the SIP user profile must be associated with an H.323 station (Dual-Registration). Else, these endpoints can be recorded using Service Observe or Single Step Conference.

The relevant settings in the System Manager User Profile for a Dual-Registration user are shown below:



Additionally, for Dual-registered devices, the station mapping must be manually entered to enable the SIP device to receive the media for calls to that station:

```
change off-pbx-telephone station-mapping 30001                Page   1 of   3
                  STATIONS WITH OFF-PBX TELEPHONE INTEGRATION

 Station       Application Dial   CC  Phone Number    Trunk       Config  Dual
 Extension                 Prefix                    Selection   Set     Mode
 30001           OPS           -     30001            aar        1
```

Call Center and routing administration tasks in Communication Manager were minimal, and not covered in these notes.

To ensure the recorder received media matching its requirements, the IP Address of the Application Enablement Services server was associated with network-region 2, which used ip-codec-set 2 as shown below. Shuffling (IP-IP Direct Audio) was disabled for this network region.

```
display ip-network-map                                        Page   1 of  63
                            IP ADDRESS MAPPING

                                                  Subnet Network     Emergency
 IP Address                                       Bits   Region VLAN Location Ext
 ------------------------------------------------ ------ ------ ---- -------------
 FROM: 10.64.115.9                                /      2      n
   TO: 10.64.115.9
 FROM: 10.64.115.33                               /      1      n
   TO: 10.64.115.255
```

```
change ip-network-region 2                                    Page   1 of  20
                            IP NETWORK REGION
  Region: 2       NR Group: 2
Location: 1     Authoritative Domain: sildenver.org
    Name: recorder              Stub Network Region: n
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 2              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                       IP Audio Hairpinning? n
   UDP Port Max: 3329
```

```
change ip-codec-set 2                                         Page   1 of   2

                         IP MEDIA PARAMETERS
    Codec Set: 2

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n          2       20
 2:


     Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: aes
 3: none
```

6. Configure Avaya Aura® Application Enablement Services

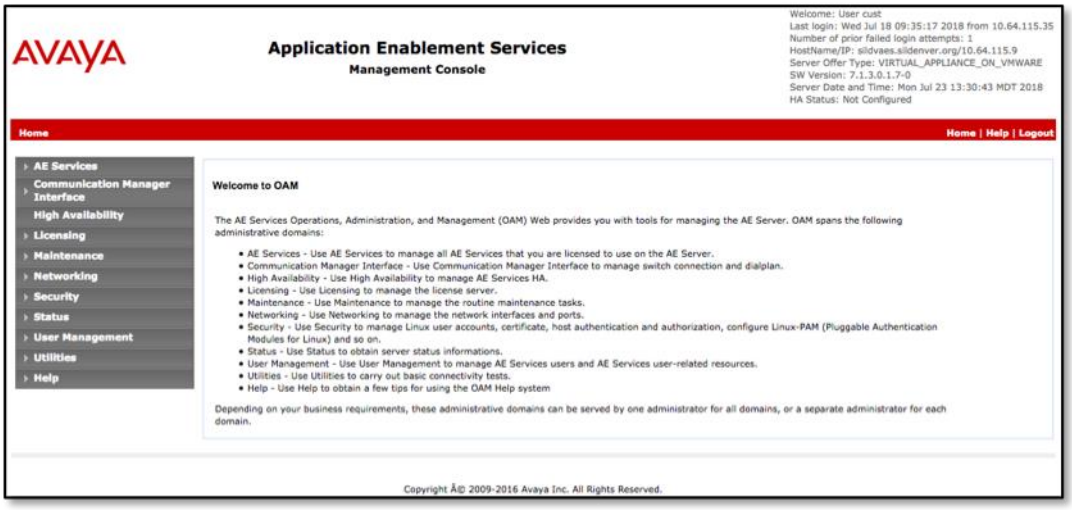
All administration of Application Enablement Services is performed via a web browser. Enter https://<ip-addr> in the URL field of a web browser where <ip-addr> is the IP address of the Application Enablement Services server. After a login step, the **Welcome to OAM** page is displayed. All navigation is performed by clicking links in the Navigation Panel on the left side of the screen, context panels will then appear on the right side of the screen.

Note that this server was updated during the testing to version 7.1.3.0.1.7-0, some screenshots display the previous version.

All connections were secure, meaning the rootCA from System Manager was installed on Communication Manager, Application Enablement Services, and the NICE Uptivity server. Identity certificates were generated in System Manager for the Avaya Aura components. By installing the rootCA on the NICE Uptivity server, secure DMCC links and media were possible using a Shared Key methodology. For more secure needs, a Mutual Authentication methodology is supported but was not tested.

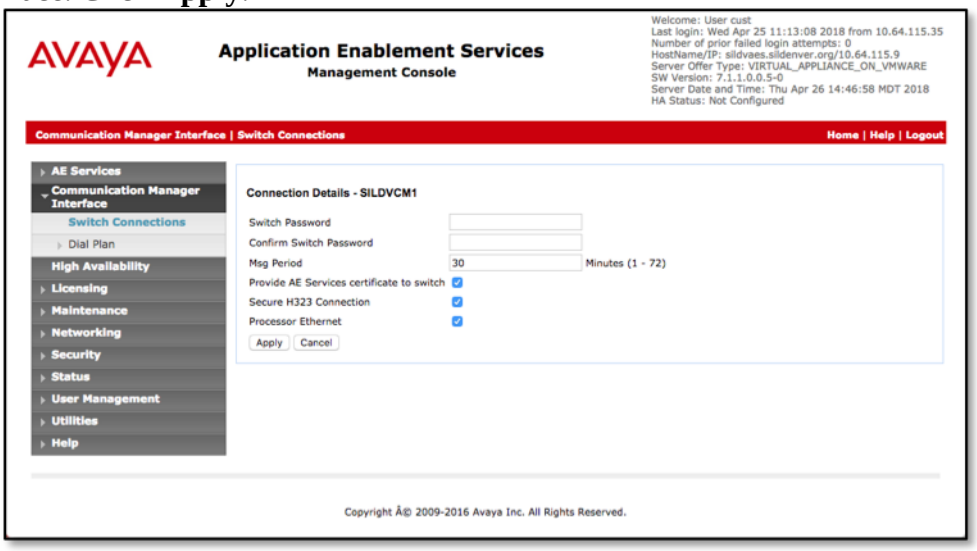
The procedures fall into the following areas:

- Configure Communication Manager Switch Connections
- Add TSAPI Links
- Note the TLink Information
- Configure a CTI User for NICE Uptivity
- Enable Unrestricted Access for the NICE Uptivity User
- Confirm TSAPI and DMCC Licenses
- Restart TSAPI Service

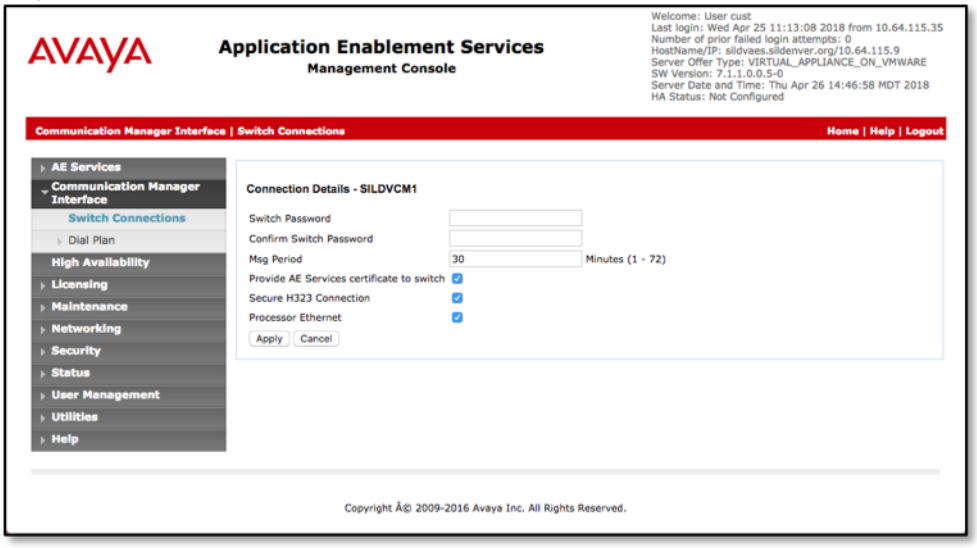


6.1. Configure Communication Manager Switch Connections

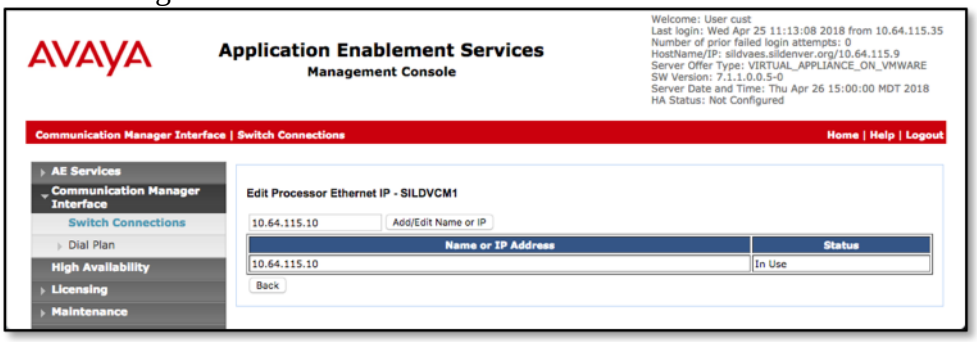
Navigate to the **Communication Manager Interface > Switch Connections** page and enter a name for the new switch connection (e.g. **SILDVCM1**) and click the **Add Connection** button (not shown). The **Connection Details** screen is shown. Enter the **Switch Password** configured in **Section 5.3** and check the **Secure H323 Connection** and **Processor Ethernet** box if using the **procr** interface. Click **Apply**.



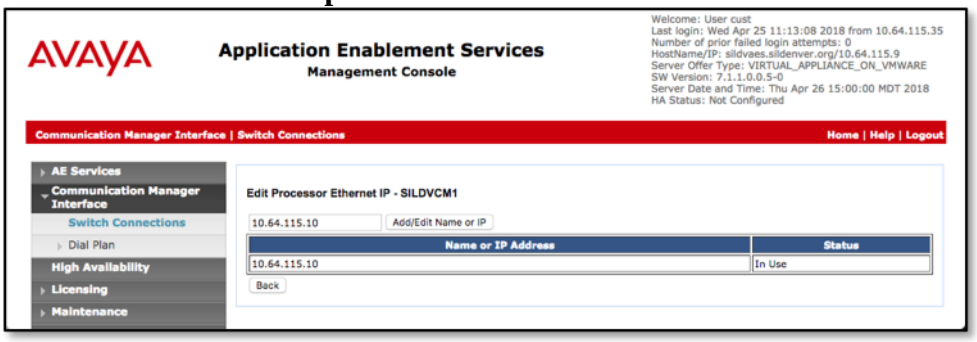
Once applied, the **Switch Connections** list will confirm the addition of the connection.



Click on the **Edit PE/CLAN IPs** button and enter the IP Address for the PROCR of Communication Manager:



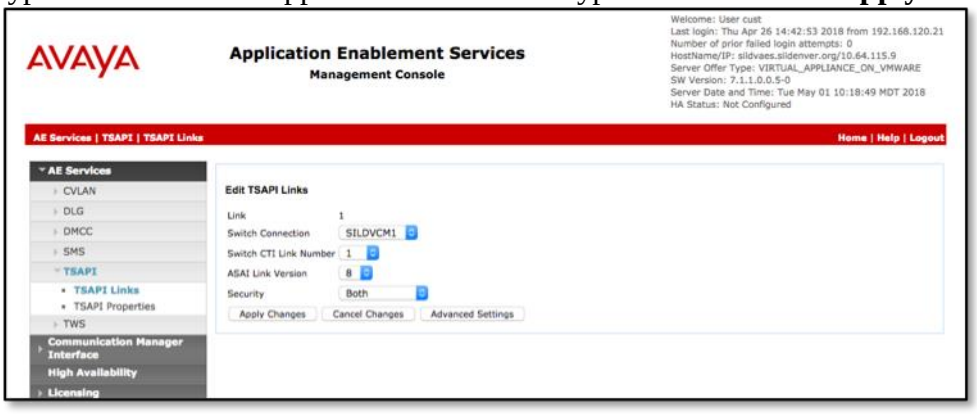
Repeat for the **Edit H.323 Gatekeeper**:



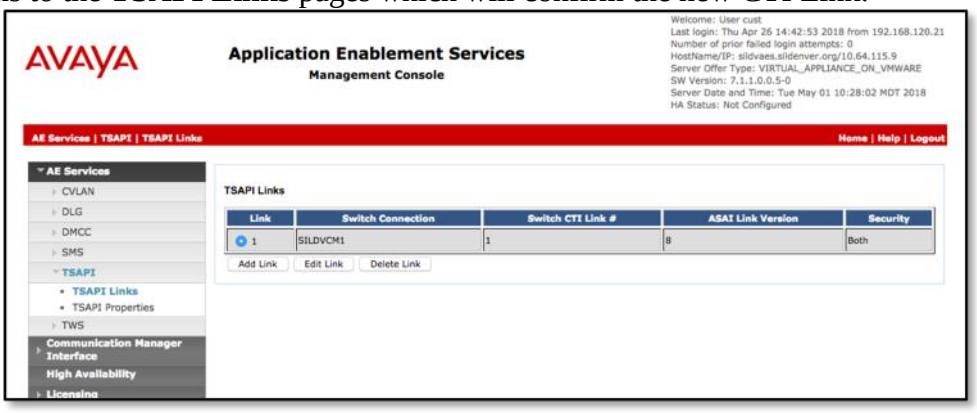
6.2. Configure TSAPI Links

Navigate to **AE Services > TSAPI > TSAPI Links** and click **Add Link** (not shown).

Select the **Switch Connection** created in **Section 6.1** in the drop-down menu (SILDVCM1), choose the **Switch CTI Link Number** that matches the link created in **Section 5.4** above. Choose an **ASAI Link Version**, 8 is generally recommended. For **Security**, choose either **Both** or **Encrypted**. Both will permit applications not capable of using secure streams to connect, while Encrypted will forces all applications to use Encrypted steams. Click **Apply Changes**.



This returns to the **TSAPI Links** pages which will confirm the new CTI Link:



6.3. Note the TLink Information

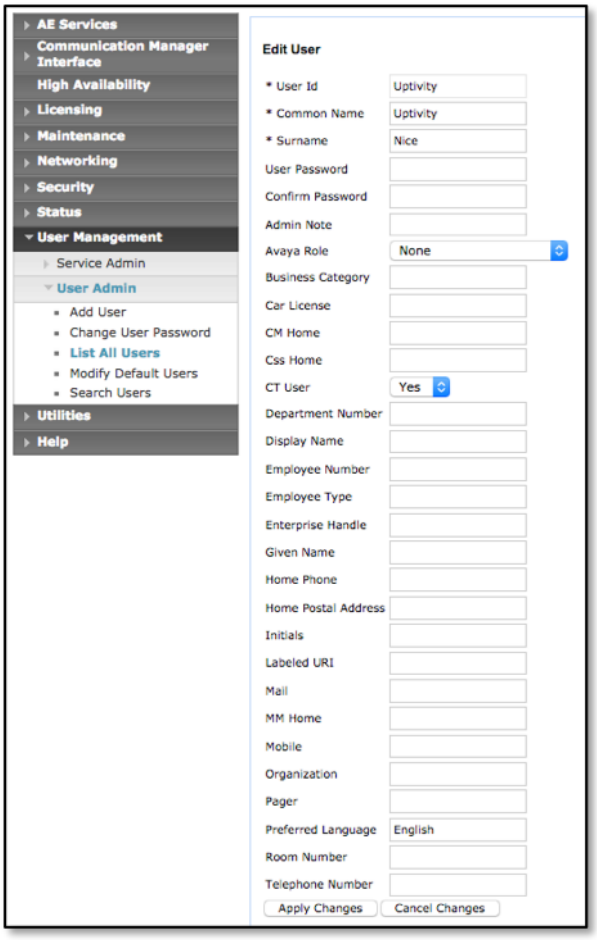
From the **TSAPI Links** page, click **Edit Link**, then **Advanced Settings** (not shown) and take note of the Tlinks Configured.

If **Both** was selected for Security in **Section 6.2** above, two Tlinks will appear with the format AVAYA#SwitchLinkName#CSTA#AESHostName. The link with CSTA-S is the secure link that will be used when configuring the NICE Uptivity application in **Section 7**.

6.4. Configure a CTI User for NICE Uptivity

Navigate to **User Management > User Admin > Add User**. Enter an appropriate **User Id, Common Name, Surname,** and **User Password**. Select **Yes** from the **CT User** dropdown list.

Click **Apply** at the bottom of the pages to save the entries.

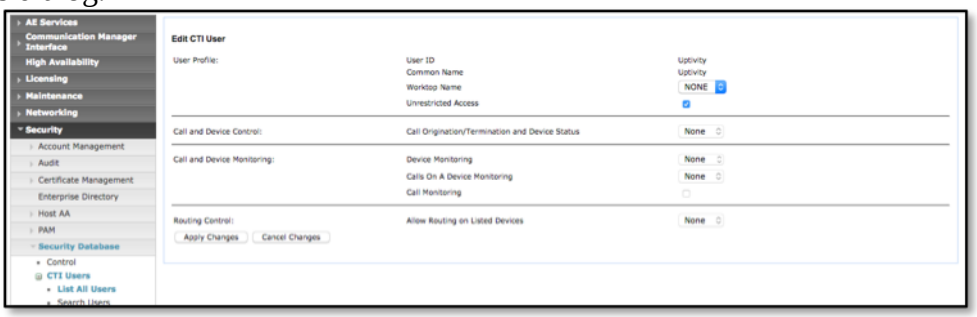


6.5. Enable Unrestricted Access for the NICE Uptivity User

If the Security Database (SDB) is enabled on Application Enablement Services, set the NICE Uptivity user account to Unrestricted Access to enable any device (station, ACD extension, DMCC virtual station) to be used implicitly. This step avoids the need to duplicate administration.

Navigate to **Security → Security Database → CTI Users → List All Users** and select the **NICE Uptivity** user and click **Edit** (not shown).

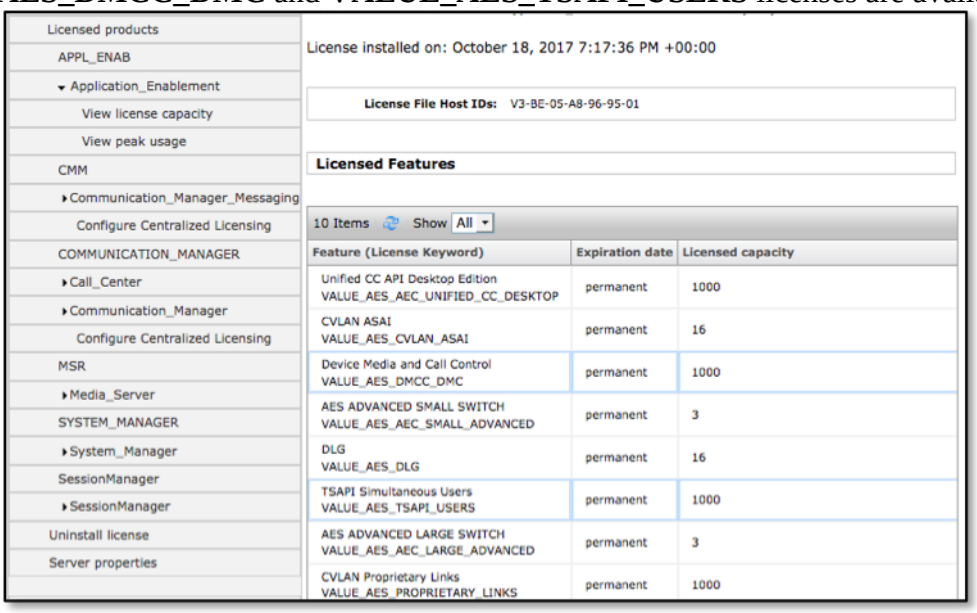
On the **Edit CTI User** panel, check the **Unrestricted Access** box and click the **Apply Changes** button. Click **Apply** when asked to confirm the change on the **Apply Changes to CTI User Properties** dialog.



6.6. Confirm TSAPI and DMCC Licenses

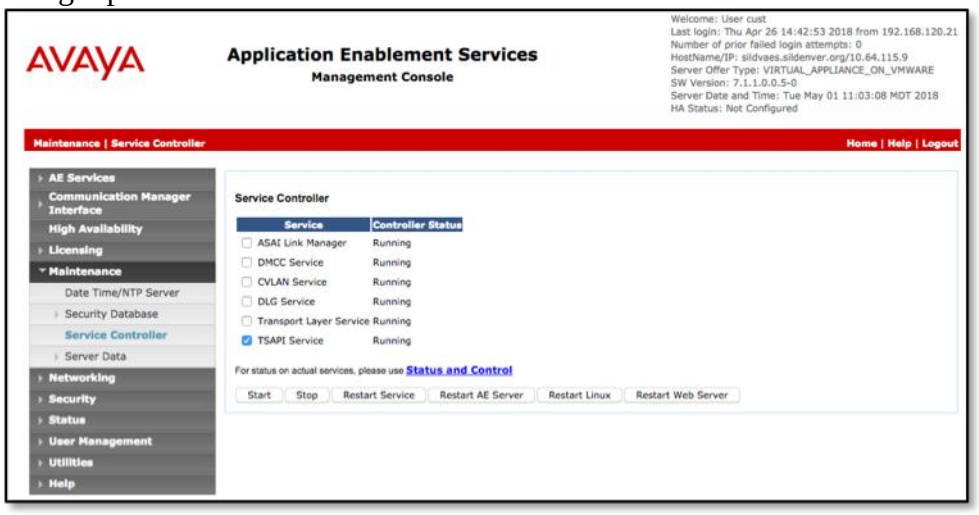
NICE Uptivity uses a DMCC (**VALUE_AES_DMCC_DMC**) license for each recording port. Additionally, a TSAPI Basic (**VALUE_AES_TSAPI_USERS**) license is used for each agent station being monitored, as well as each hunt group being monitored. Additionally, recorder ports that will use Single Step Conference or Service Observation will require a TSAPI license to add these ports to calls.

With version 7 and later, WebLM is typically installed and configured on Avaya Aura® System Manager. A **Web License Manager** login window is displayed. Enter proper credentials to log in. Click **Licensed products → APPL_ENAB → Application_Enablement** from the left pane. The Application Enablement Services license is displayed in the right pane. Ensure enough **VALUE_AES_DMCC_DMC** and **VALUE_AES_TSAPI_USERS** licenses are available.



6.7. Restart TSAPI Service

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

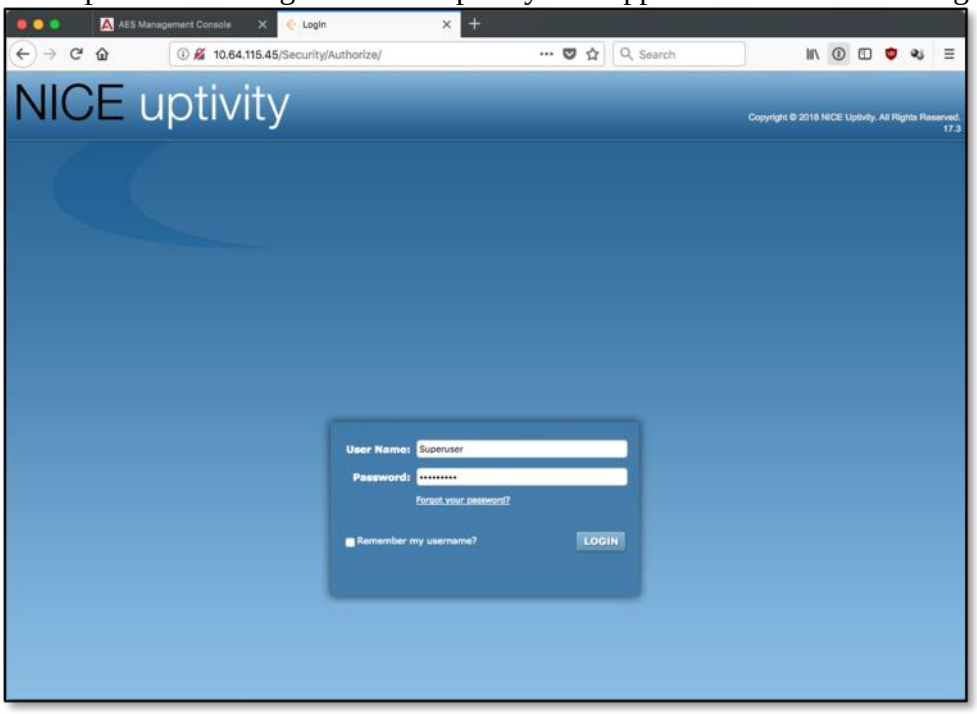


7. Configure NICE Uptivity

The initial configuration of the NICE Uptivity server is typically performed by NICE technicians or authorized installers. These Application Notes will only cover the steps necessary to configure the solution to interoperate with Communication Manager and Application Enablement Services.

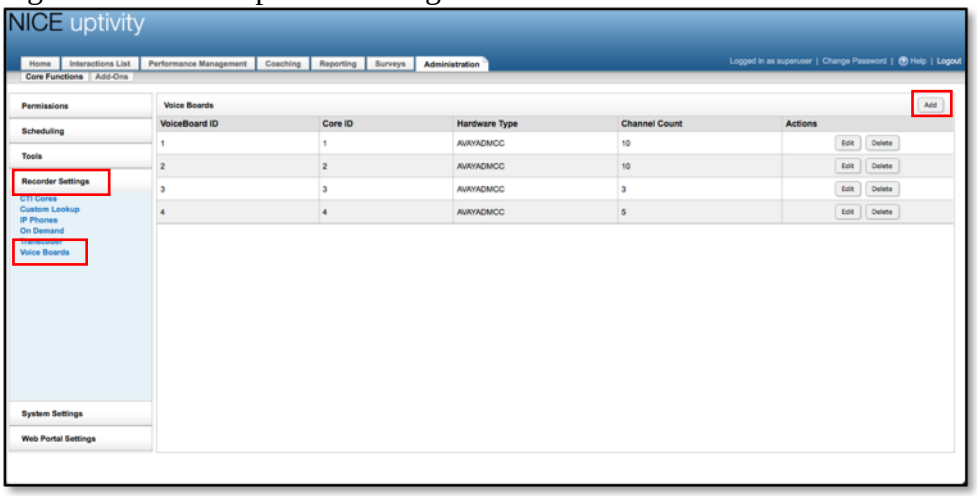
The Uptivity application is the next generation solution incorporating elements from previous products Call Copy and inContact, some UI elements use the legacy naming conventions.

Configuration is performed using the NICE Uptivity web application on the recording server.

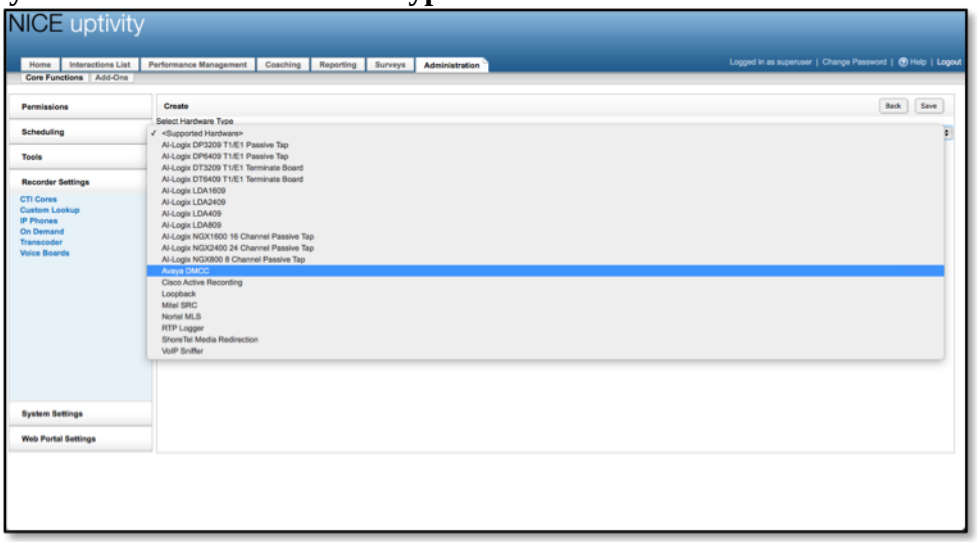


7.1. Create Voice Board

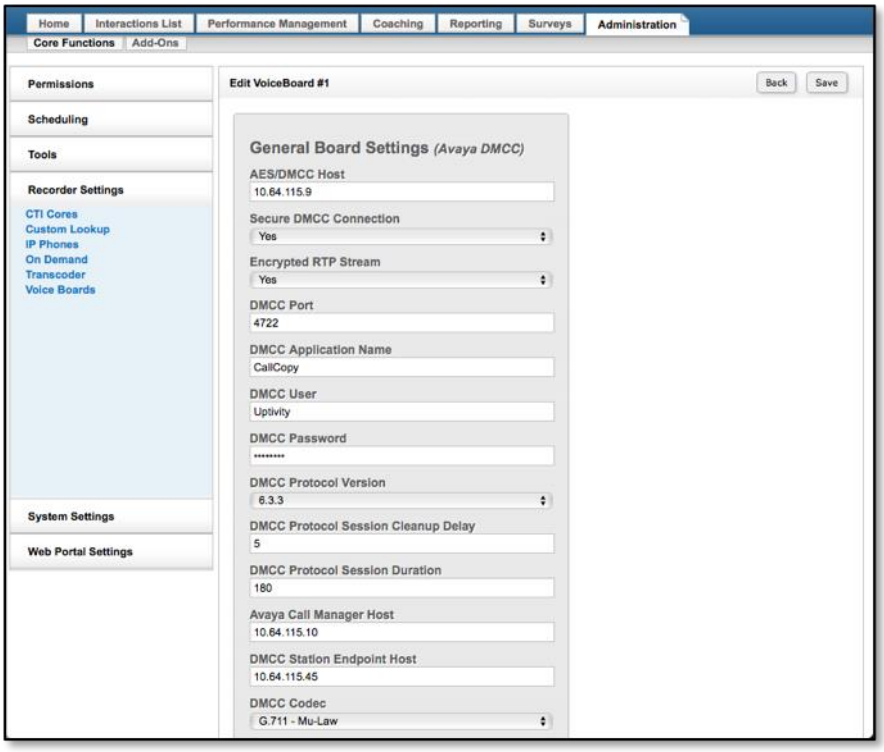
Click on the **Administration** tab and select **Recorder Settings > Voice Boards** in the left window, click on **Add** at the top right of the main window. This screenshot displays the four boards configured for the compliance testing.



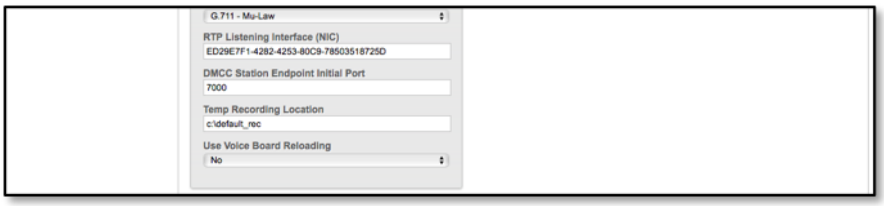
Select **Avaya DMCC** as the **Hardware Type**:



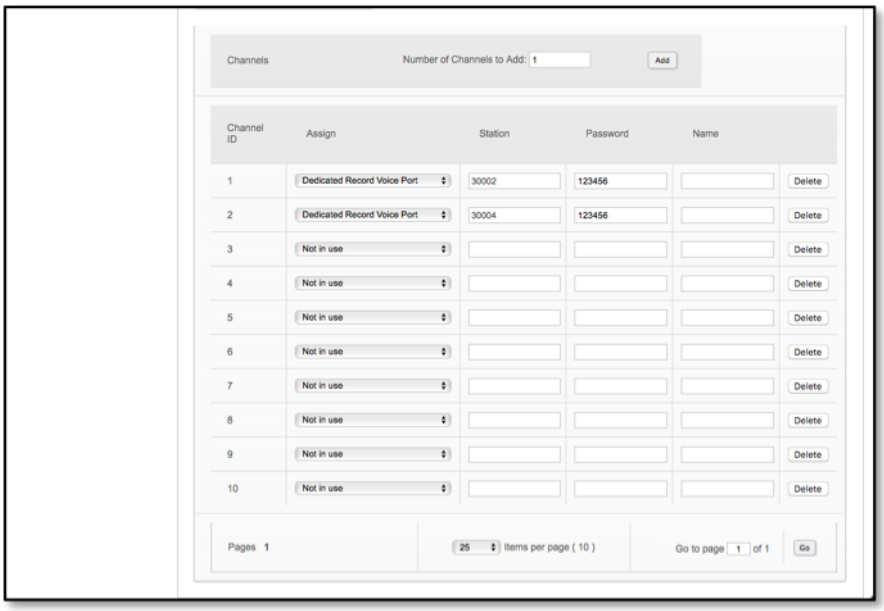
Enter the **AES/DMCC Host** information which will be the IP Address of the AES. The DMCC Connection can be secure or unsecure, for compliance testing **Secure DMCC Connection** was set to **Yes**. The **DMCC Port** for a secure connection is set to the DMCC default **4722**. Choose a suitable **DMCC Application Name** and enter the **DMCC User** and **DMCC Password** as per Section 6.4. Set the **DMCC Protocol Version** to **6.3.3**. Enter the IP Address of the Communication Manager PROCR in the **Avaya Call Manager Host** field, and the IP Address of the Uptivity Server in the **DMCC Station Endpoint Host** field. Finally, set the **DMCC Codec** to **G.711 – Mu-Law**.



Scroll the page to complete the remaining settings. The value for the RTP Listening Interface (NIC) can be obtained from either Wireshark or from the program cc_interfaceBrowser.exe as shown in Appendix A.

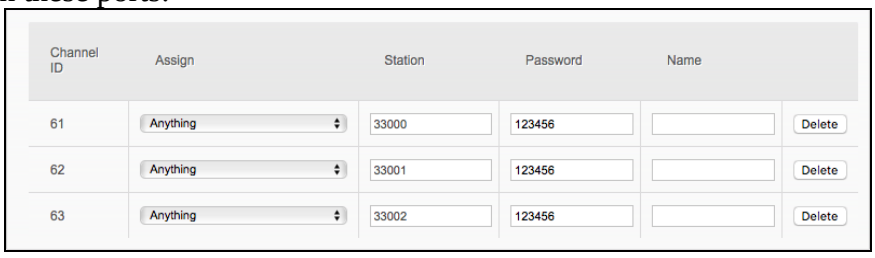


Further down on the page, enter the Number of Channels to Add, this will determine the number of simultaneous recording ports that will be used by this voice board. For this board, 2 were chosen allowing each port to be associated with a different Communication Manager extension. This Voice Board was configured to record agent extensions 30002 and 30004 using a **Dedicated Record Voice Port**. When all information is entered, scroll to the top of the page and click **Save**.



Repeat the above steps for as many Voice Boards as will be needed for the site. In the compliance test, 4 were configured. The second board was configured identical to the above using a **Dedicated Record Voice Port** for agent station 30001 (the Dual-Registered SIP\H.323 station).

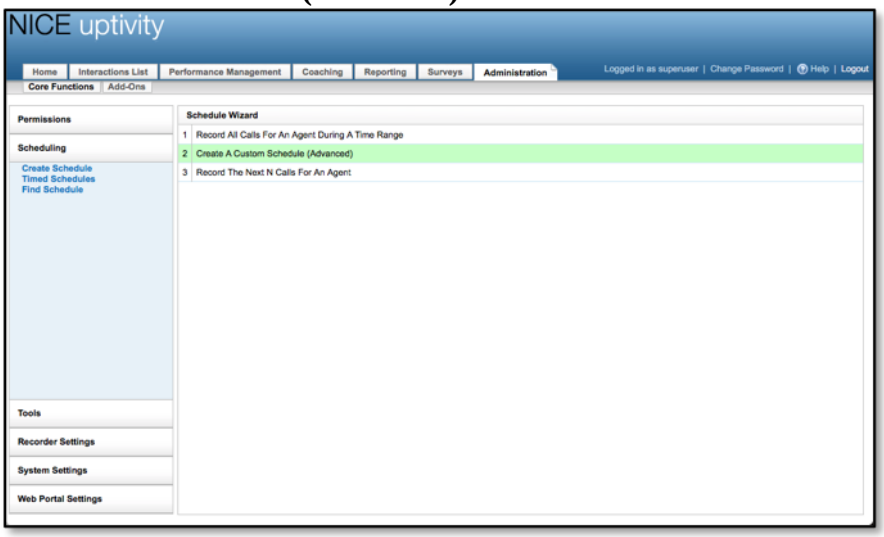
The third board was configured for Single Step Conference using the following Channels, these are virtual extensions in Communication Manager designated for use by the recording application to be registered using the MAIN registration type. The Channel assignment was set to **Anything** on these ports:



The fourth board was identical, but used only one channel for Single Step Conference to record the Digital set using virtual extension 33003.

7.2. Create Schedule

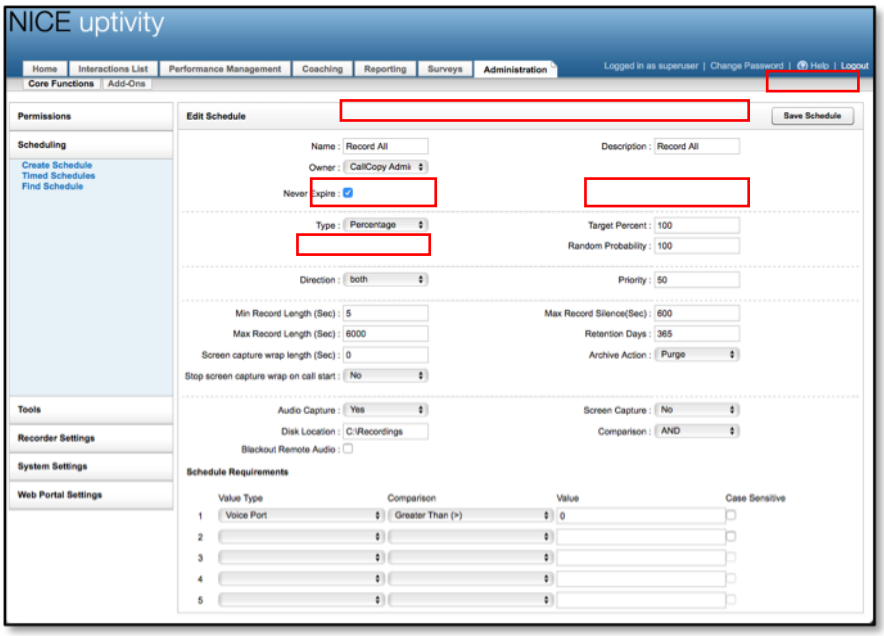
Remain in the **Administration** tab and select **Scheduling > Create Schedule** in the left window, click on **Create a Custom Schedule (Advanced)** in the main window.



Enter a suitable name for the schedule. For compliance testing the following were set,

- **Type** Set to **Percentage**
- **Target Percent** Set to **100**
- **Direction** Set to **Both**
- **Schedule Requirements** **Voice Port Greater Than 0**

The other values can be left as default. Click on **Save Schedule** to save this.



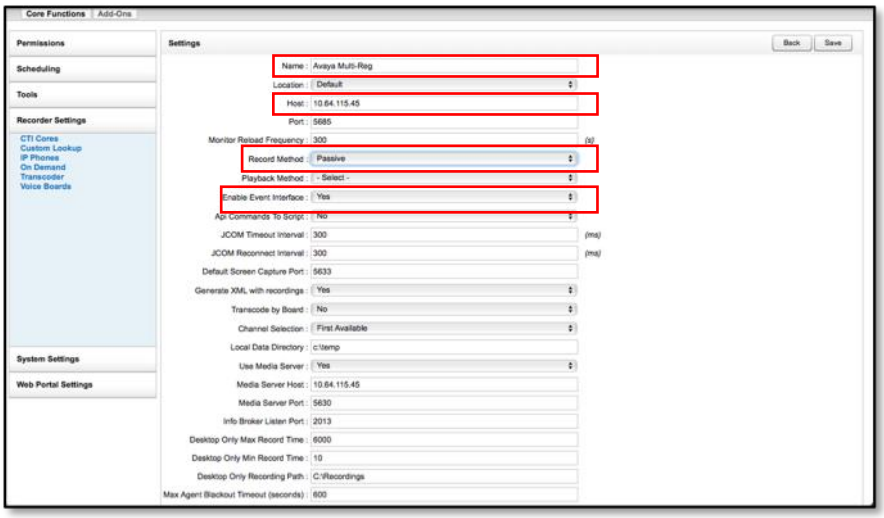
7.3. Create CTI Cores

Remain in the **Administration** tab and select **Recorder Settings > CTI Cores** in the left window, click on **Add Core** at the top right of the main window. One CTI Core was created for each of the four Voice Boards created in **Section 7.1**.

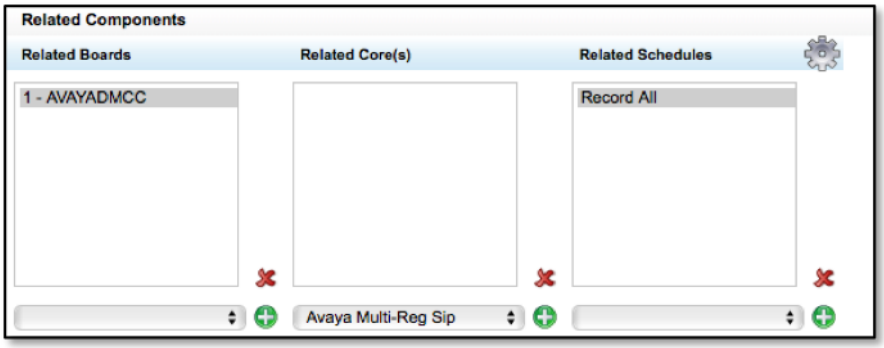
The first Core was used for Multi-Registration to record the H.323 stations 30002/30004. Enter a suitable name for the Core. For compliance testing the following were set,

- **Host** Set to **IP Address of the Uptivity Server**
- **Record Method** Set to **Passive**
- **Enable Event Interface** Set to **Yes**

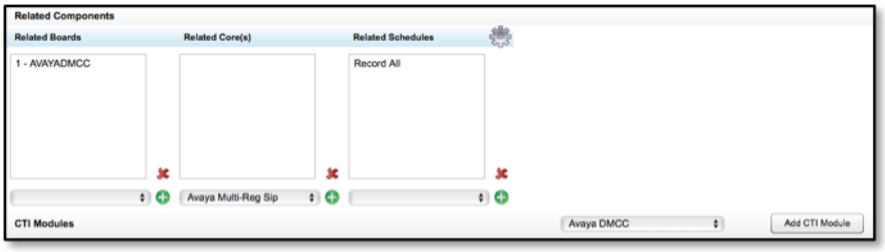
Note that **Single Step Conference** recording method was specified in the Cores configured to record the Digital and SIP devices, otherwise, all settings were the same.



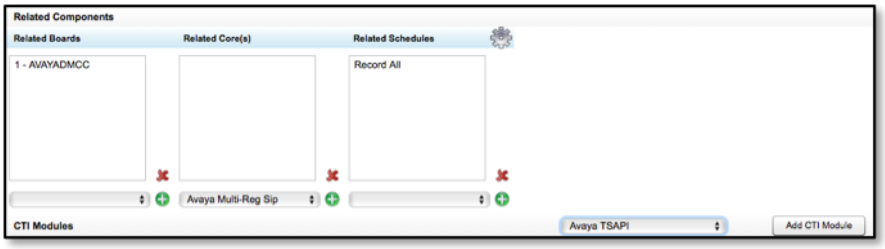
Voice Board 1 was associated with this Core, as was the **Record All** Schedule:



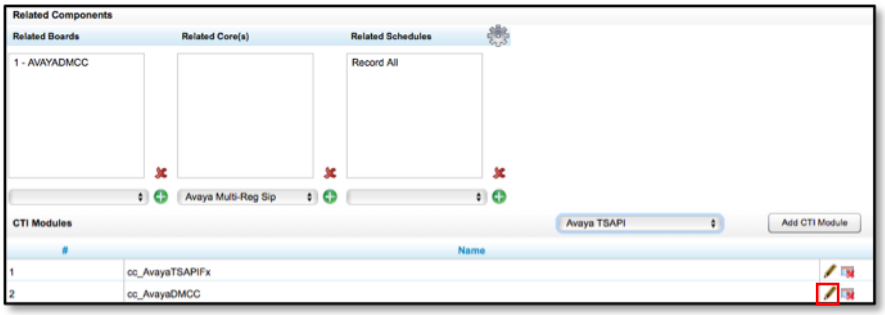
With the Board and Schedule selected, click on the drop-down menu highlighted and select **Avaya DMCC**, with this selected click on **Add CTI Module**.



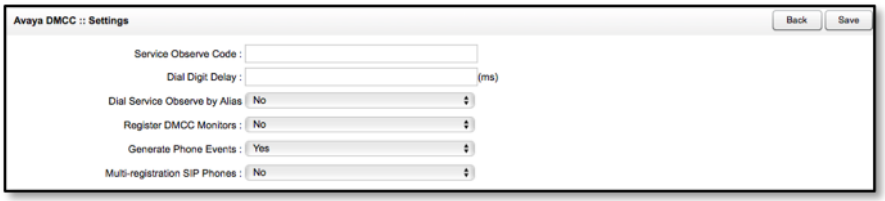
From the drop-down menu select **Avaya TSAPI** and click on **Add CTI Module**.



Each new CTI Module needs to be edited, click on the edit icon as shown below and edit each CTI core starting with **cc_AvayaDMCC** as shown below.

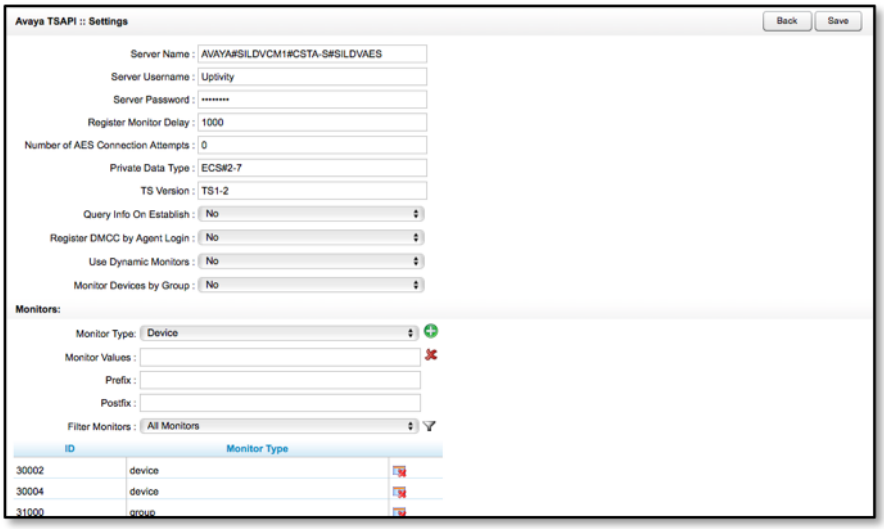


The settings below show the DMCC configuration used during compliance testing.



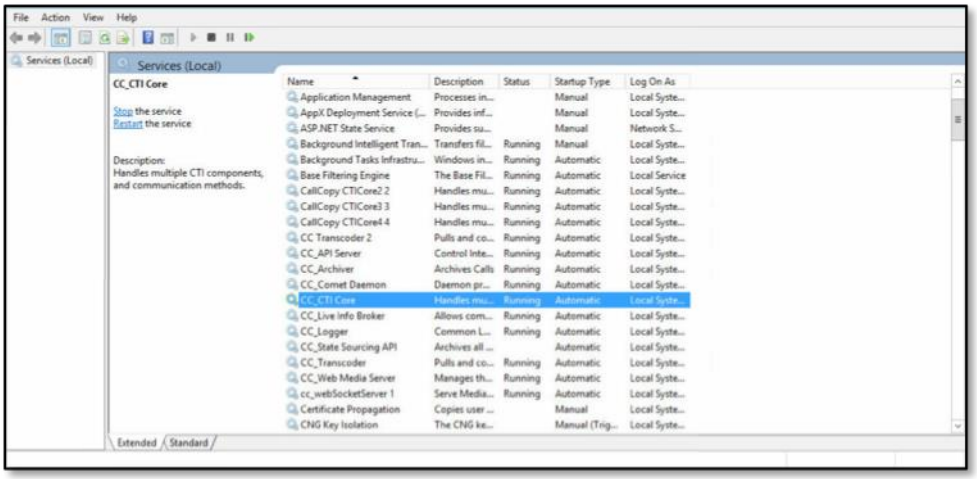
Similarly, edit the **cc_AvayaTSAPIFx** module.

Enter the TLINK information (found from **Section 6.3**) into the Server Name box. Enter the CTI user and password as per **Section 6.4** for the Server Username and Server Password. The other information can be left as default. Scroll down to Monitors, each device that needs to be monitored can be added here, as shown below these are Agent and Hunt Group extensions 30002, 30004, 31000. Click **Save** when all entries are complete.



Repeat these steps for the remaining 3 Cores, associating them with the remaining Voice Boards and entering the TSAPI devices to be monitored for each Core.

At this point, the four CTI Cores can be started from the Windows Services control panel.



8. Verification Steps

8.1. Verify Communication Manager Status

From a Communication Manager SAT session, the **list registered-ip-stations** command will show an IP_API_A registration with the Application Enablement server address for Multi-Registration (eg. 30001, 30002, 30004) and Virtual Extensions (eg. 33000) registered to the Uptivity application.

```
list registered-ip-stations                                          Page   1

                          REGISTERED IP STATIONS

Station Ext   Set Type/ Prod ID/       Station IP Address/
or Orig Port  Net Rgn   Release    Skt Gatekeeper IP Address
------------ --------- ---------- --- ---------------------------------------
30001         9630      IP_API_A   tls 10.64.115.9
              2         3.2040         10.64.115.10
30002         9611      IP_Phone   tls 10.64.115.36
              1         6.6506         10.64.115.10
30002         9611      IP_API_A   tls 10.64.115.9
              2         3.2040         10.64.115.10
30004         9650      IP_Phone   tcp 10.64.115.31
              1         3.280A         10.64.115.10
30004         9650      IP_API_A   tls 10.64.115.9
              2         3.2040         10.64.115.10
30007         4621      IP_Phone   tcp 10.64.115.37
              1         2.9020         10.64.115.10
33000         9630      IP_API_A   tls 10.64.115.9
              2         3.2040         10.64.115.10
33001         9630      IP_API_A   tls 10.64.115.9
              2         3.2040         10.64.115.10
33002         9630      IP_API_A   tls 10.64.115.9
              2         3.2040         10.64.115.10
33003         9630      IP_API_A   tls 10.64.115.9
              2         3.2040         10.64.115.10
```

The **list monitored-station** command will show stations with TSAPI monitors.

```
list monitored-station

                             MONITORED STATION

   Associations:     1       2       3       4       5       6       7       8
                  CTI     CTI     CTI     CTI     CTI     CTI     CTI     CTI
Station Ext       Lnk CRV Lnk CRV Lnk CRV Lnk CRV Lnk CRV Lnk CRV Lnk CRV Lnk CRV
---------------- ------- ------- ------- ------- ------- ------- ------- -------
30001             1  0003
30002             1  0008
30003             1  0016
30004             1  0001
30005             1  0002
30006             1  0015
```

With an active call, the **status station** command can demonstrate the media properties of a call being recorded. The recorder is 10.64.115.45 connected to the G430 with g711u and aes encrypted media, the station is connected to the Avaya Media Server (AMS) with g729a and srtp encrypted in the example below:

```
status station 30002                                          Page   7 of   9
                    SRC PORT TO DEST PORT TALKPATH
src port: S00005
S00005:TX:10.64.115.36:2328/g729a/20ms/1-srtp-aescm128-hmac80   - station 30002
AMS1:RX:10.64.115.3:6142/g729a/20ms/1-srtp-aescm128-hmac80:TX:cnfID:0  - AMS
AMS1:RX:cnfID:0:TX:10.64.115.3:6144/g729/20ms/1-srtp-aescm128-hmac80   - AMS
001V012:RX:10.64.115.2:2050/g729/20ms/1-srtp-aescm128-hmac80:TX:ctxID:233 - G430
001V011:RX:ctxID:233:TX:10.64.115.2:2056/g711u/20ms/aes
S00003:RX:10.64.115.45:7000/g711u/20ms/aes      - recorder
```

The following captures the audio channels involved in the connections with AMS to station 30002 using G.729a, and G430 to the application via AES using G.711MU:

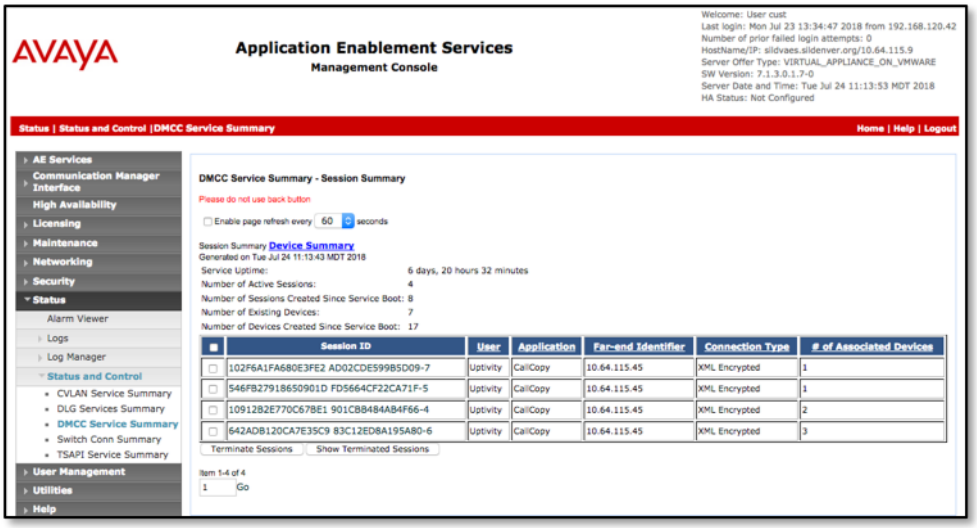
```
status station 30002                                          Page   5 of   9

                         AUDIO CHANNEL Port: S00005
G.729A        Switch-End Audio Location: AMS1
            IP Address                              Port  Node Name       Rgn
Other-End: 10.64.115.3                              6142  sildvams        1
  Set-End: 10.64.115.36                             2328                  1
Audio Connection Type: ip-tdm

                         AUDIO CHANNEL Shared Port: S00003
G.711MU       Switch-End Audio Location: MG1
            IP Address                              Port  Node Name       Rgn
Other-End: 10.64.115.2                              2056                  1
  Set-End: 10.64.115.9                              7000                  2
Audio Connection Type: ip-tdm
```

The following captures the Call Control associations for the devices registered to station 30002:

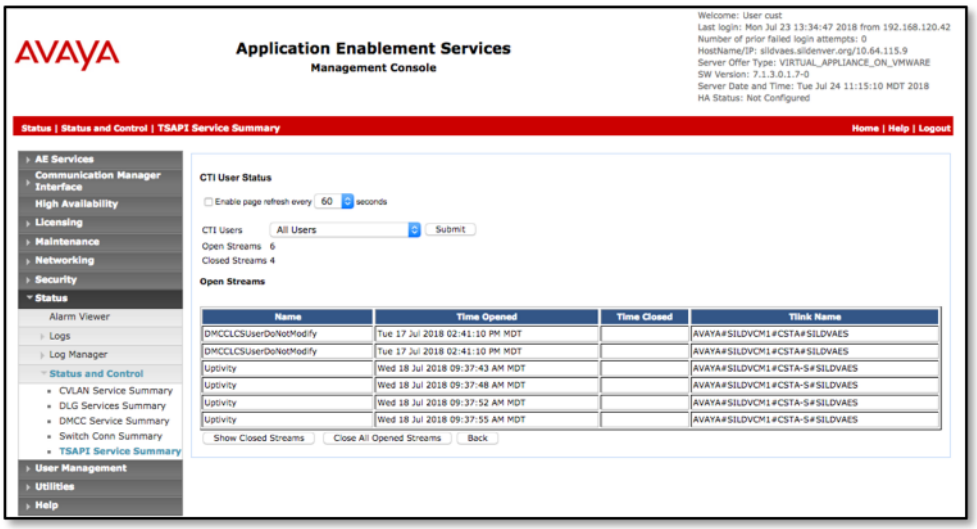
```
 status station 30002                                          Page   4 of   9
                            CALL CONTROL SIGNALING
Port: S00005         Switch-End IP Signaling Loc: PROCR    H.245 Port:
            IP Address                              Port  Node Name       Rgn
Switch-End: 10.64.115.10                            1300  procr           1
Reg Set End:10.64.115.36                            47482                 1
Alt Set End:not applicable
H.245 Near:
 H.245 Set:
Shared Port: S00003  Switch-End IP Signaling Loc: PROCR
            IP Address                              Port  Node Name       Rgn
Switch-End:10.64.115.10                             1300  procr           1
   Reg Set End:10.64.115.9                          27341                 2
```

8.2. Verify Application Enablement Services Status

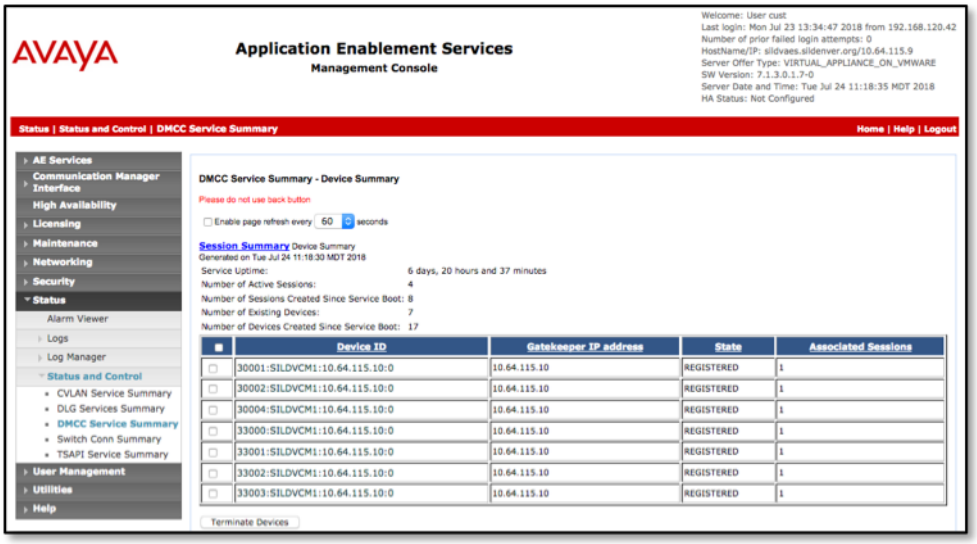
From Application Enablement Services, the **Status > DMCC Service Summary > Session Summary** will reflect the Secure DMCC sessions the recorder Voice Boards have established.



The **Status > TSAPI Service Summary > User Status** should display one stream for each of the CTI Cores:

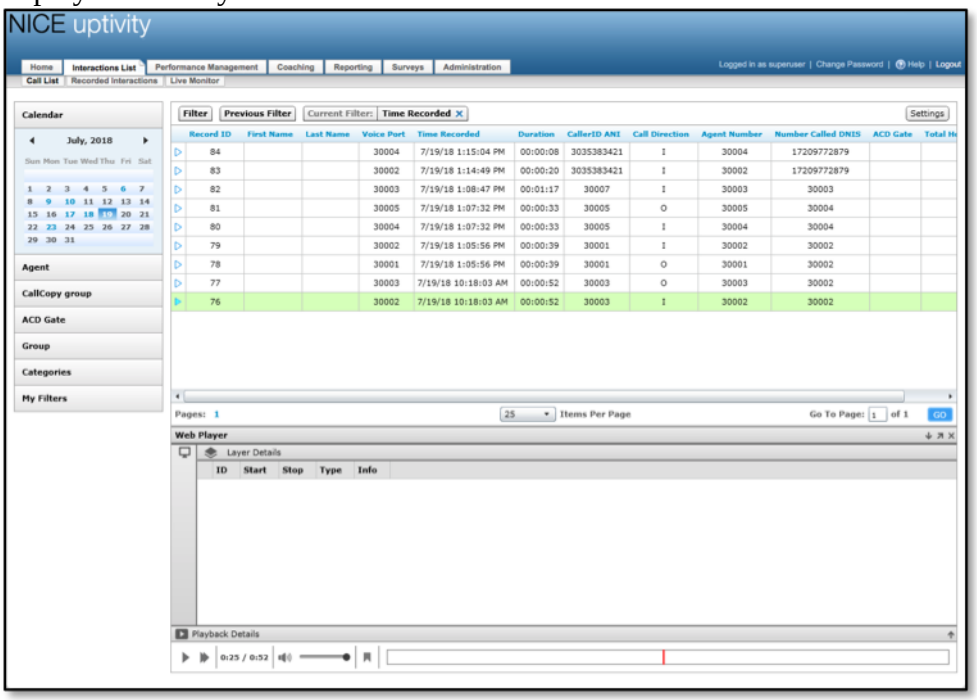


The **Status > DMCC Service Summary > Device Summary** will reflect the registrations that the recorder has established.



8.3. Verify Recording and Playback

Using a browser, access the NICE Uptivity user interface and navigate to the **Interactions List** tab and click on the date you wish to search for calls. Note there are more refined searches that can be performed, the following will display all recordings the logged in user has rights to view based on permissions administered elsewhere in the web application. In most browsers, the wav form will display in the Playback Details section of the screen.



9. Conclusion

These Application Notes describe the procedures for configuring NICE Uptivity to monitor and record calls placed to and from agents and phones on Avaya Aura® Communication Manager. In the configuration described in these Application Notes, NICE Uptivity uses the Device and Media Control Services of Avaya Aura® Application Enablement Services to perform recording. During compliance testing, NICE Uptivity successfully recorded calls placed to and from agents and stations.

Refer to **Section 2.2** for details regarding secure media limitations.

10. Additional References

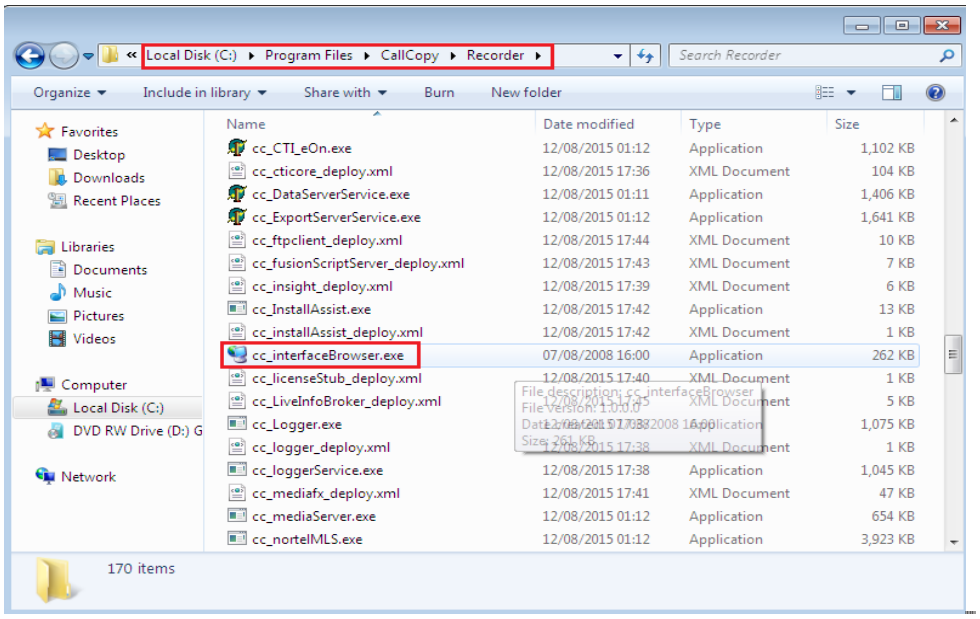
Product documentation for Avaya products may be found at http://support.avaya.com.

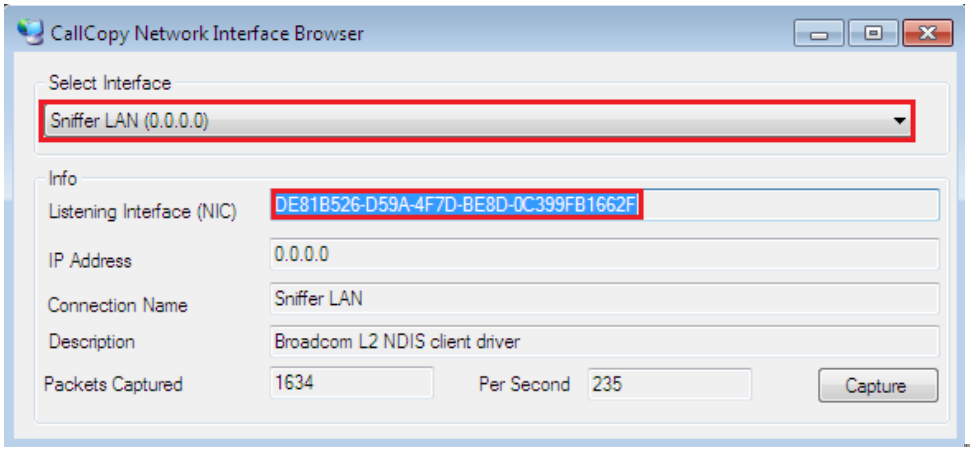
- *Administering Avaya Aura® Communication Manager*, Release 7.1.1, Issue 2, August 2017
- *Avaya Aura® Application Enablement Services Administration and Maintenance Guide*, Release 7.1.1, Issue 3, September 2017
- *Avaya Aura® Application Enablement Services Device, Media and Call Control .NET API Programmers Guide Release 7.1.1*, Issue 1, Document Number 02-602658

Appendix A

Open the application called cc_interfaceBrowser.exe, this should be located in Program Files →CallCopy→Recorder folder.

Select the correct Network Interface which is used to capture the RTP from the data switch. The Listening Interface should then be populated and this is used for the setup of the NICE Uptivity Call Recording server in **Section 7**.





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.