

Avaya Solution & Interoperability Test Lab

Application Notes for Numonix Recite Interaction Recording Solution with Avaya Aura® Application Enablement Services and Avaya Aura® Communication Manager - Issue 1.0

Abstract

These Application Notes outline how to configure the Numonix Recite recording solution to interoperate with Avaya Aura[®] Communication Manager and Avaya Aura[®] Application Enablement Services (AES).

In the compliance testing, Numonix Recite used the Multiple Registration feature 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.

1. Introduction

Numonix's Recite solution is an interaction recording suite designed to boost business success. Recite delivers insights into employee-customer interactions by capturing content from multiple media modalities, including voice, video, screen and chat/IM.

The Numonix Recite 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 the Multiple Registrations method.

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

Recite uses:

- The TSAPI interface of AES (via DMCC) to monitor extensions to obtain call events.
- The DMCC interface of AES to register the recorder as an additional registered endpoint with Communication Manager in order to record devices.

Serviceability tests were also conducted to assess the reliability of the Recite solution.

2. General Test Approach and Test Results

The feature test cases were performed manually. Upon start of the Recite application, the application automatically established a DMCC stream with AES to register the recorder as a Dependent additional IP Endpoint for each of the target stations on Communication Manager, and to receive third party call events via TSAPI through the DMCC stream.

Each call was handled manually at the agent 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 Recite and Avaya components.

The verification of tests included use of logs for proper message exchanges and use of the Recite 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 interface between Avaya systems and Numonix Recite utilized enabled capabilities of TLS for Application links between Application Enablement and CM, and streams between Application Enablement and Recite. Further, SRTP was used to encrypt all media streams.

This test was conducted in a lab environment simulating a basic customer enterprise network environment. The testing focused on the standards-based interface between the Avaya solution and the Numonix solution. The results of testing are therefore considered to be applicable to either a premise-based deployment or to a hosted or cloud deployment where some elements of the Numonix solution may reside beyond the boundaries of the enterprise network, or at a different physical location from the Avaya components.

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

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

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

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing.

The feature testing focused on verifying the following on Recite:

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

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

2.2. Test Results

All test cases were executed and verified. The following observations were made from the compliance testing.

Held calls appear as two calls in the playback query, each with meta data identifying the
parties on the call. A reviewer would know these were related by the context of the audio
recorded.

2.3. Support

Technical support on Numonix Recite can be obtained through the following:

- Email: support@numonixrecording.com
- Phone: +1-561-952-2600 opt 2.
- Web: https://numonixrecording.com/contact-support/

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.

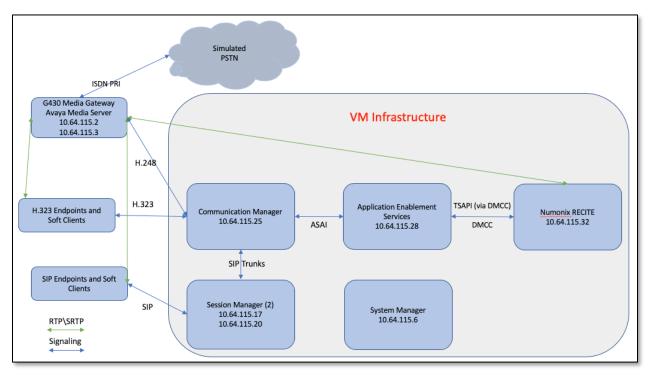


Figure 1: Numonix Recite Compliance Testing Configuration

4. Equipment and Software Validated

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

Equipment/Software	Version			
Avaya Aura® Communication Manager running on	R018x.00.0.822.0			
virtualized environment				
Avaya Aura® Application Enablement Services running	8.0.1.0.3.5-0			
on virtualized environment				
Avaya Aura® Session Manager running on virtualized	8.0.0.0.800035			
environment				
Avaya Aura® System Manager	8.0.0.0.098174			
Avaya Aura® Media Server				
Avaya G450 Media Gateway	FW 38.20.1/1			
Avaya 96x1 Series IP Telephone				
• 96x1 (H.323)	6_6_5_06-080917			
• 96x1 (SIP)	7.0.1			
Avaya 1416 Digital Telephones	FW 1			
Desktop PC running Avaya One-X® Agent	2.5.60129.0 (H.323)			
Numonix RECITE	3.0.4.2			
Windows 2012R2 Standard Server	2012 R2			
VMWare (Host)	ESXi 6.0			
DMCC .NET SDK	6.3.3.14			

5. Configure Avaya Aura® Communication Manager

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

- Confirm Licensing
- 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 (see **Section 4** for details on these). 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.

The Numonix Recite application uses Dependent (Multiple Registration) to record Digital and H.323 endpoints, Independent Mode (Multiple Registration) to record SIP devices.

5.1. Confirm Licensing

The Multiple Registration DMCC recording method does not require additional "Virtual Extensions" built for the recorder to register and use to join calls like Single Step Conference and Service Observe recording methods do.

 Recorders using the Multiple Registration (i.e., 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
                                                                   2 of 12
                                                             Page
                              OPTIONAL FEATURES
IP PORT CAPACITIES
                   Maximum Administered H.323 Trunks: 4000 0
         Maximum Concurrently Registered IP Stations: 2400 11
           Maximum Administered Remote Office Trunks: 4000
Maximum Concurrently Registered Remote Office Stations: 2400 0
        Maximum Concurrently Registered IP eCons: 68
 Max Concur Registered Unauthenticated H.323 Stations: 100
                      Maximum Video Capable Stations: 2400
            Maximum Video Capable JP Softphones: 2400 0
                    Maximum Administered SIP Trunks: 4000 55
 Maximum Administered Ad-hoc Video Conferencing Ports: 4000
  Maximum Number of DS1 Boards with Echo Cancellation: 80
```

- 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 Services 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
Abbreviated Dialing Emmanced Dict.

Access Security Gateway (ASG)? y

Analog Trunk Incoming Call ID? y

A/D Grp/Sys List Dialing Start at 01? y
                                                      Authorization Codes? y
                                                                      CAS Branch? n
                                                                        CAS Main? n
Answer Supervision by Call Classifier? y
                                                              Change COR by FAC? n
                                    ARS? y Computer Telephony Adjunct Links? y
                  ARS/AAR Partitioning? y
                                              Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y
                                                                    DCS (Basic)? y
                                                              DCS Call Coverage? y
          ASAI Link Core Capabilities? 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. Note, these changes are outlined in Bold.

```
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?
```

5.3. Administer IP Services for Avaya Aura® Application Enablement Services

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

change ip-services						Page	1 of	3
			IP	SERVICES				
Service	Enabled	Local		Local	Remote	Remote		
Type		Node		Port	Node	Port		
AESVCS	y p	rocr		<u>8765</u>				

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

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

CTI LINK

CTI Link: 1

Extension: 30000

Type: ADJ-IP

Name: SILDVAES
```

5.5. Verify Recorded Extensions & Add Virtual Stations

For recording solutions using Dependent or Independent mode registration type, agent extensions are administered as follows to allow the recording port to register.

- **Type** = 9630
- Security Code = eg: 123456 (this will be required when setting up the recorder)
- **IP Softphone** = y

```
change station 30001
                                                                Page 1 of 5
                                     STATION
                                       Lock Messages? n
Security Code: *
                                                                      BCC: 0
Extension: 30001
                   Type: 9630
     Port: S00002
    Name: Agent1_
                      Hunt-to Station:
STATION OPTIONS
                                           Time of Day Lock Table:
             Loss Group: 19 Personalized Ringing Pattern: 1
      Speakerphone: 2-way Mute Button Enabled? Y
Display Language: english Button Modules: 0
vable GK Node Name:
Survivable COR: internal Media Complex Ext:
vivable Trunk Dest? Y
Survivable GK Node Name:
   Survivable Trunk Dest? y
                                              IP Video Softphone? <u>n</u>
                              Short/Prefixed Registration Allowed: default
                                              Customizable Labels? y
```

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

```
change ip-network-region 2

IP NETWORK REGION

Region: 2 NR Group: 2
Location: 1 Authoritative Domain: sildenver.org

Name: recorder

MEDIA PARAMETERS

Codec Set: 2

UDP Port Min: 2048

UDP Port Max: 3329

Page 1 of 20

IP NETWORK REGION

Sildenver.org

Stub Network Region: n

Intra-region IP-IP Direct Audio: no

Inter-region IP-IP Direct Audio: no

IP Audio Hairpinning? n
```

IP Codec used for compliance testing.

```
change ip-codec-set 2

IP MEDIA PARAMETERS

Codec Set: 2

Audio Silence Frames Packet
Codec Suppression Per Pkt Size(ms)

1: G.711MU n 2 2 20

2:

Media Encryption Encrypted SRTCP: enforce-unenc-srtcp

1: 1-srtp-aescm128-hmac80
2: aes none
```

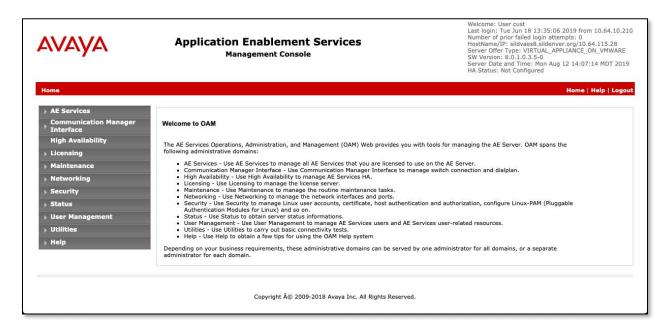
6. Configure Avaya Aura® Application Enablement Services

All administration of the AES 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 AES 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.

All connections were secure, meaning the root CA from System Manager was installed on Communication Manager, AES, and the Numonix Recite server. Identity certificates were generated in System Manager for the Avaya Aura components. By installing the root CA on the Recite 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.

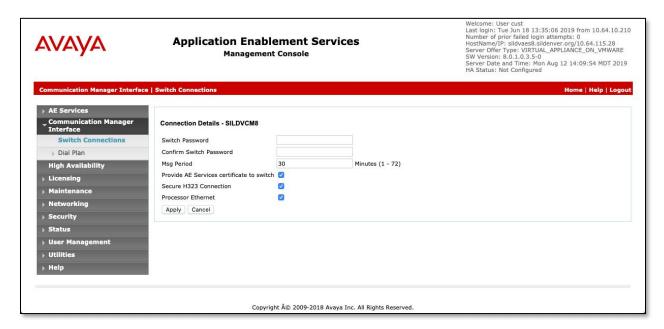
The procedures fall into the following areas:

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

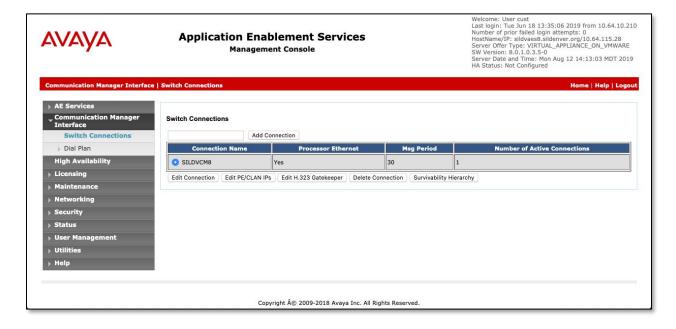


6.1. Configure Communication Manager Switch Connections

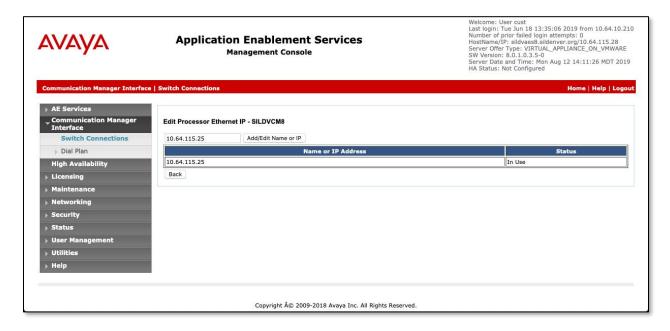
Navigate to the Communication Manager Interface → Switch Connections page and enter a name for the new switch connection (e.g. SILDVCM8) and click the Add Connection button (not shown). The Connection Details screen is shown. Enter the Switch Password configured in Section 5.3 and check the Secure H323 Connection and Processor Ethernet box if using the procr interface. Click Apply.



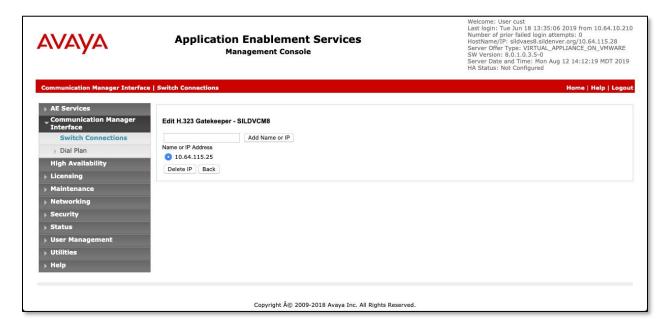
Once applied, the **Switch Connections** list will confirm the addition of the connection.



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



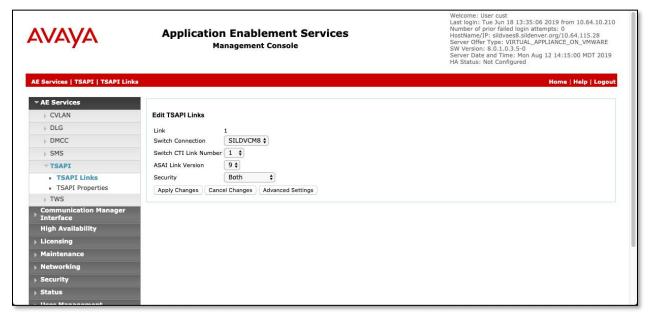
Repeat for the **Edit H.323 Gatekeeper**:



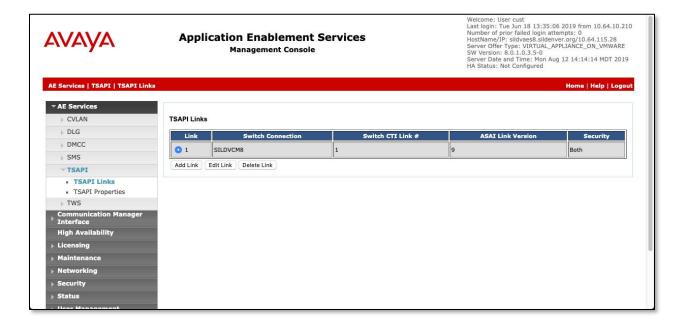
6.2. Configure TSAPI Links

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 (SILDVCM8), choose the **Switch CTI Link Number** that matches the link created in **Section 5.4** above. Choose an **ASAI Link Version**, **9** 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**.



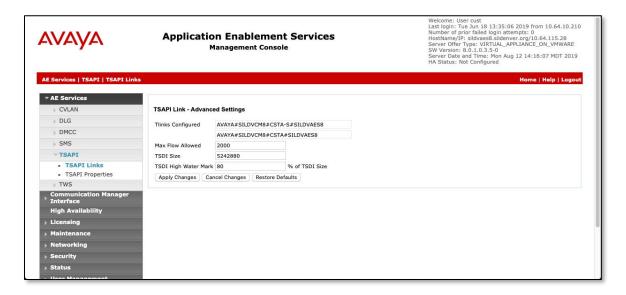
This returns to the **TSAPI Links** pages which will confirm the new CTI Link:



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.

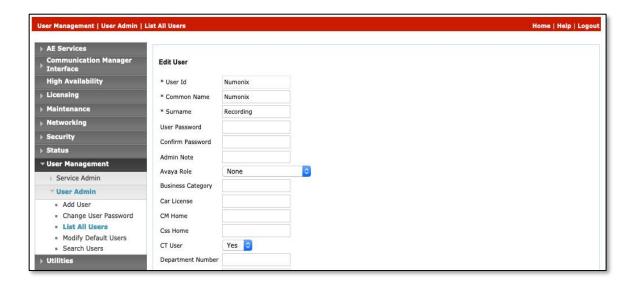
If **Both** was selected for Security in **Section 6.2** above, two Tlinks will appear with the format AVAYA#SwitchLinkName#CSTA#AESHostName. The link with CSTA-S is the secure link that will be used when configuring the Recite application in **Section 7**.



6.4. Configure a CTI User for Recite

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.

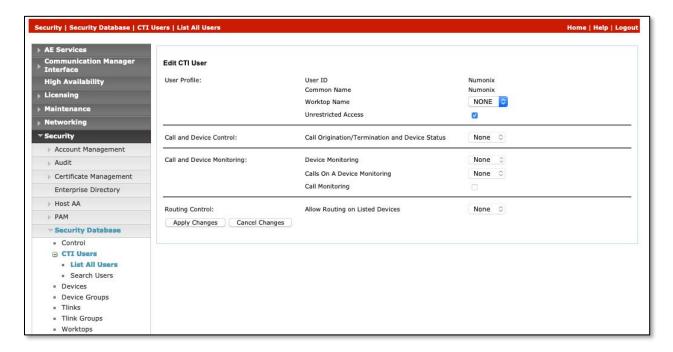


6.5. Enable Unrestricted Access for the Recite User

If the Security Database (SDB) is enabled on Application Enablement Services, set the Recite 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.

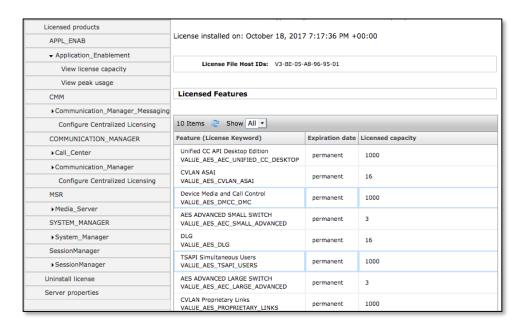
Navigate to Security → Security Database → CTI Users → List All Users and select the Numonix user and click Edit (not shown).

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.



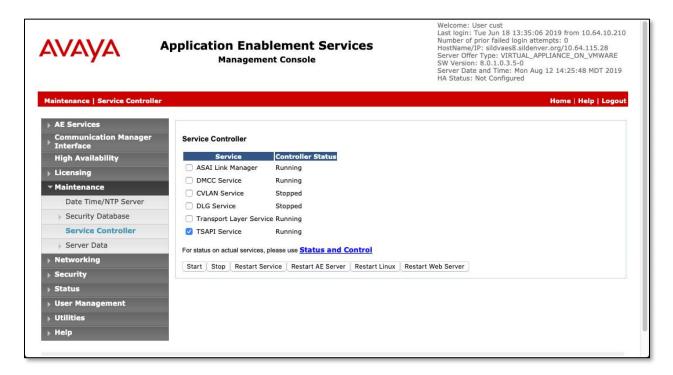
6.6. Confirm TSAPI and DMCC Licenses

Recite 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.



6.7. Restart TSAPI Service

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

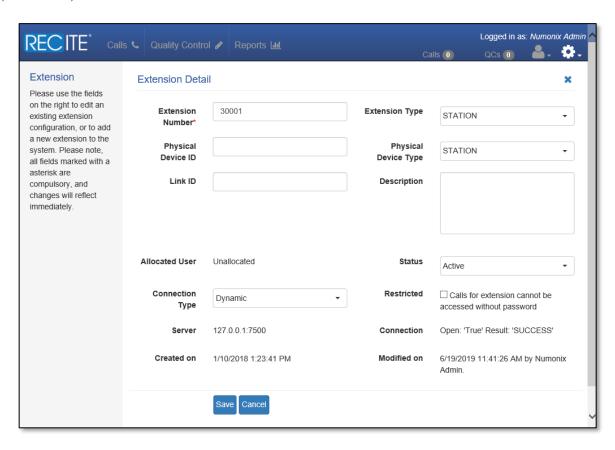


7. Configure Numonix Recite

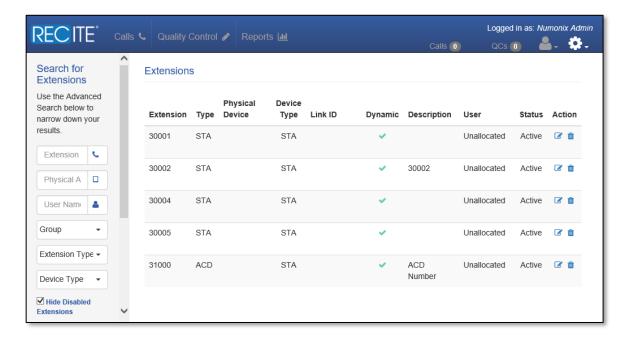
Numonix installers will perform most of the initial configuration of the system, some of the steps to enable connectivity to Communication Manager and AES are described below. Note that most of the configuration settings are contained in the *CTILinkService.app.config* XML file on the Recite server. Most other tasks are performed using a web browser.

7.1. Configure Extensions, ACD Queues and Agents

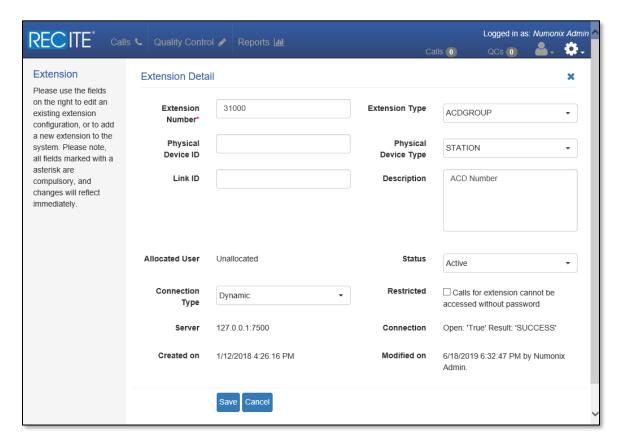
Click on the gear icon in the upper right of the web interface, choose Extensions, then click Add (not shown) to create stations to be recorded.



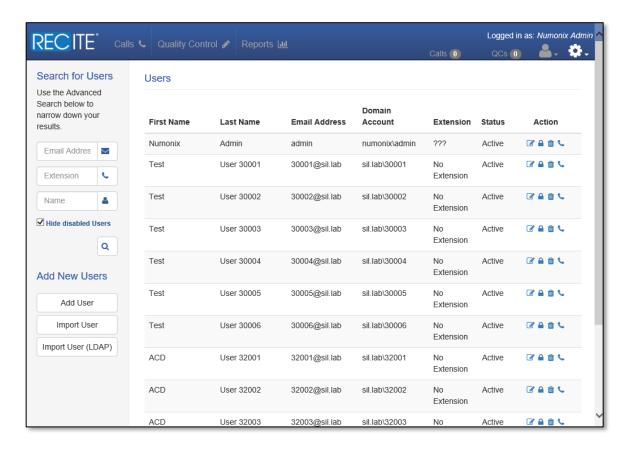
Repeat for all stations and confirm in the list shown below.

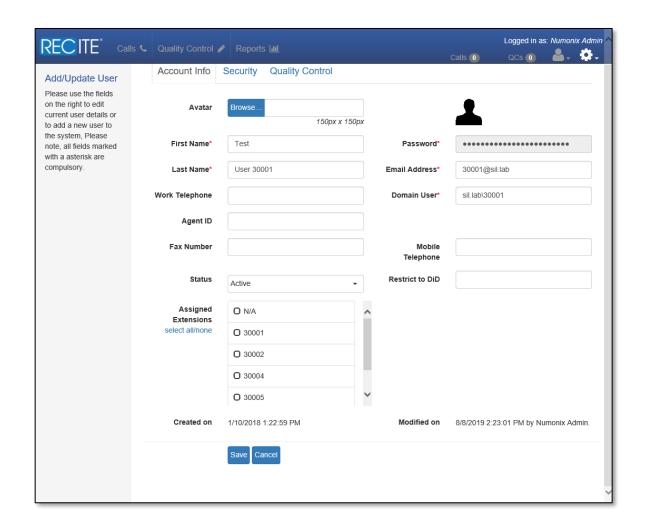


ACD Queues are configured as extensions, with an **ACDGROUP** Type:

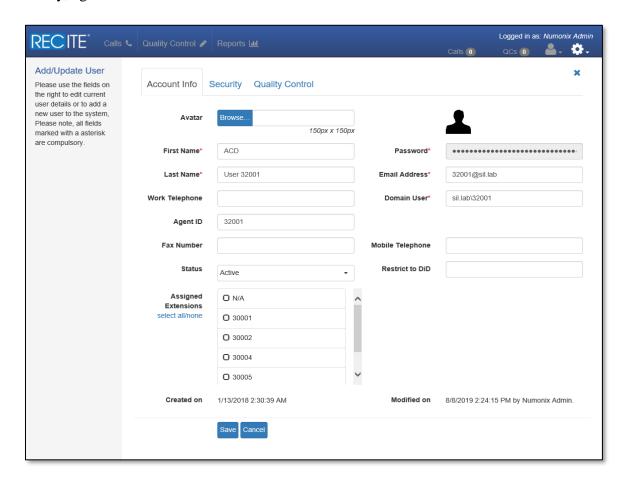


Users are created similarly and can be configured to allow playback of calls for specific extensions.





Similarly, agents are created:



7.2. Configure Recite for Avaya Aura® Application Enablement Services Device Registrations

In the web interface, create a new "Registration" Group containing all targeted users.

Note: Confirm all Users' extensions are associated with their profile before adding the User to the Registration Group.

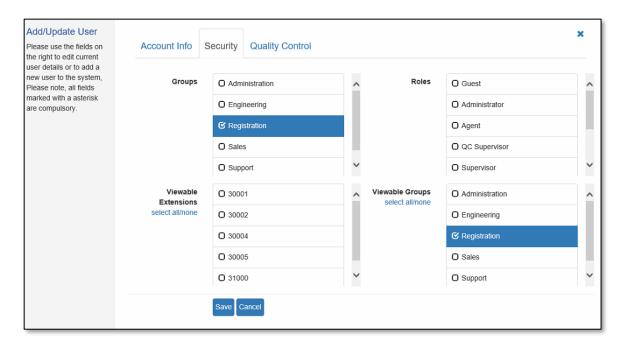


Configure this Registration Group in CTILink under "RegistrationGroupName" in the file *CTILinkService.app.config* on the Recite server. This XML file is where all of the connection settings are configured as well.

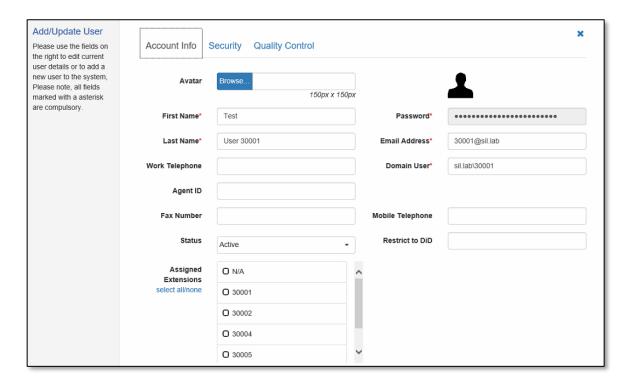
```
maxconcurrentconmandoperations
   <CommandOperationTimeout Value="5" />
   <CommandOperationRetryCount Value="2" />
 </AsvncCommandProvider>
 <DomainRequired Value="true" />
 <RemotePartyHeaderName Value="" />
 <NormalizePhoneNumber Value="true" />
 <EnableNATMapping Value="false" />
 <ExtensionRecordingOptions Value="Any" />
 <ApplicationInstanceID Value="" />
 <Priority Value="1" />
 <MessageCredentialType Value="Windows" />
 <EnableTargetActivationTimer Value="true" />
 <EnableAutoStopReplacedCalls Value="false" />
 <CertificateName Value="Avaya" />
 <CMSwitchName Value="SILDVCM8" />
 <CMSwitchAddress Value="10.64.115.25" />
 <UseSSL Value="true" />
 <AppName Value="Recite" />
 <VersionNumeric Value="8.0.1" />
 <Version Value="http://www.ecma-international.org/standards/ecma-323/csta/ed3/privC" />
 <DeviceInstance Value="2" />
 <DependencyMode Value="1" />
 <EnableRegistration Value="true" />
 <EnableNATAutoMapping Value="false" />
 <ExcludePayloadTypes Value="" />
 <SupportedCodecs Value="g711U,g711A,g729A,g726A,g723" />
  <SupportedEncryptionModes Value="srtp-aescm128-hmac80, none" />
 <RegistrationGroupName Value="Registration" />
 <EnableMetaDataMap Value="false" />
 <ExcludeAudioPayloadTypes Value="G7221/32000:bitrate=48000,SIREN14/16000:bitrate=48000" />
 <ExcludePayloadFormatDetails Value="true" />
 <EnableDeviceSnapshot Value="false" />
 <ResetTargetOnActivationTimeout Value="false" />
 <TargetHeartbeatTimeout Value="3900" />
 <ScreenRecordGroupName Value="ScreenRecord" />
</Link001>
```

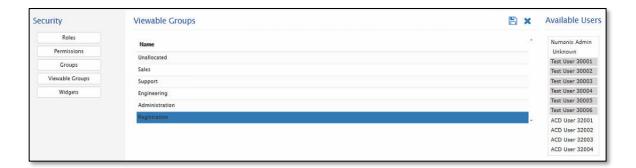
7.3. Set Agent passwords

Create a new "Registration User" and make this user a "Group Viewer" of the Registrations Group in Step 1 and member of the Registrations groups.



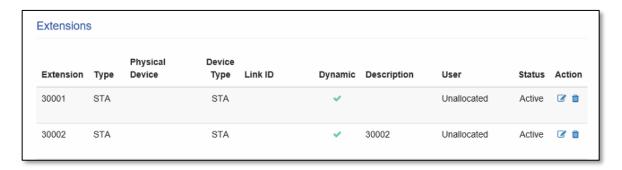
The password for the Registration User must match the registration password for every extension.



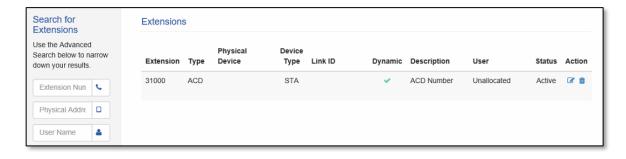


7.4. Configure Recite for Free Seating

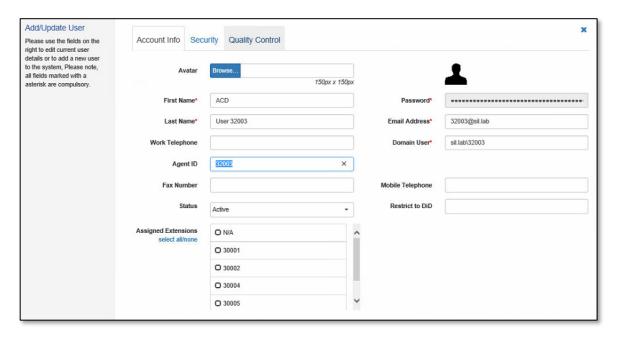
Base Device Extensions should be entered as STA type devices.



Each skill group number should be entered as an ACD type device.



The PBX AgentID/AgentPIN number must be entered in the associated User form in the **Agent ID** input box.



With the above configured, the user with correlating agent ID in their profile will be associated with the extension.

If the Avaya environment is not running the Avaya EAS (Expert Agent Selection) then the above config will be invalid for the site and free seating will not be supported. In this case the base extensions will be loaded as STA devices and assigned directly to the user.

The property setting "EnableDeviceSnapshot" is a Link level property and should be set to "false" when Avaya EAS is not enabled, this is set in the file *CTILinkService.app.config*.

```
<CommandOperationTimeout Value="5" />
 <CommandOperationRetryCount Value="2" />
</AsyncCommandProvider>
<DomainRequired Value="true" />
<RemotePartyHeaderName Value="" />
<NormalizePhoneNumber Value="true" />
<EnableNATMapping Value="false" />
<ExtensionRecordingOptions Value="Any" />
<ApplicationInstanceID Value="" />
<Priority Value="1" />
<MessageCredentialType Value="Windows" />
<EnableTargetActivationTimer Value="true" />
<EnableAutoStopReplacedCalls Value="false" />
<CertificateName Value="Avaya" />
<CMSwitchName Value="SILDVCM8" />
<CMSwitchAddress Value="10.64.115.25" />
<UseSSL Value="true" />
<AppName Value="Recite" />
<VersionNumeric Value="8.0.1" />
<Version Value="http://www.ecma-international.org/standards/ecma-323/csta/ed3/privC" />
<DeviceInstance Value="2" />
<DependencyMode Value="1" />
<EnableRegistration Value="true" />
<EnableNATAutoMapping Value="false" />
<ExcludePayloadTypes Value="" />
<SupportedCodecs Value="g711U,g711A,g729A,g726A,g723" />
<SupportedEncryptionModes Value="srtp-aescm128-hmac80, none" />
<RegistrationGroupName Value="Registration" />
<EnableMetaDataMap Value="false" />
<ExcludeAudioPayloadTypes Value="G7221/32000:bitrate=48000,SIREN14/16000:bitrate=48000" />
<ExcludePayloadFormatDetails Value="true" />
<EnableDeviceSnapshot Value="false" />
<ResetTargetOnActivationTimeout Value="false" />
<TargetHeartbeatTimeout Value="3900" />
<ScreenRecordGroupName Value="ScreenRecord" />
```

8. Verification Steps

This section describes various methods for verifying aspects of the solution.

8.1. Communication Manager

On Communication Manager, use the **status station** command to verify the application has registered to the agent's stations. The first screenshot is a SIP_CC device.

```
status station 30001
                                           Page 6 of 7
               IP ENDPOINT DATA
        Port: S00011
Product ID-Release:
                                 H.245 Tunneled? does not apply
Registration Status: unregistered
                                       MAC Address:
Native NAT Address:
ALG NAT WAN Address:
        Shared Port: S00012
                                Shd Signal Status: connected
Product ID-Release: IP_API_A 3.2040
                                        H.245 Tunneled? does not apply
Registration Status: registered-authenticated MAC Address: unavailable
Authentication Type: DES-56-plus
                                      Dependency Mode: independent
Native NAT Address: not applicable
ALG NAT WAN Address: not applicable
```

With an active call on a H.323 station, the IP station (10.64.115.36) is registered to Communication Manager (10.64.115.25), and the shared port connection is with AES (10.64.115.28):

```
status station 30002
                                          Page 4 of 10
                 CALL CONTROL SIGNALING
                Switch-End IP Signaling Loc: PROCR
Port: S00005
                                                   H.245 Port:
                         Port Node Name
      IP Address
                                                    Rgn
Switch-End: 10.64.115.25
                                      61441 procr
                                                       1
Reg Set End:10.64.115.36
                                       1720
Alt Set End:not applicable
H.245 Near:
H.245 Set:
Shared Port: S00003 Switch-End IP Signaling Loc: PROCR
                                 Port Node Name
      IP Address
                                                    Rgn
Switch-End:10.64.115.25
                                      61441 procr
 Reg Set End:10.64.115.28
                                       23097 sildvaes8
```

The audio connections show the station connected with the G430, and AES:

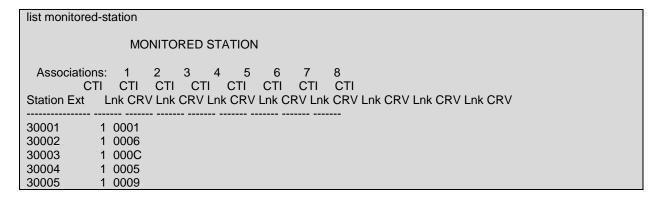
```
status station 30002
                                          Page 5 of 10
              AUDIO CHANNEL Port: S00005
G.729A
           Switch-End Audio Location: MG1
      IP Address
                                Port Node Name
                                     2058 sildvmg1
Other-End: 10.64.115.2
Set-End: 10.64.115.36
                                     2982
Audio Connection Type: ip-tdm
              AUDIO CHANNEL Shared Port: S00003
           Switch-End Audio Location: MG1
      IP Address
                                 Port Node Name
                                                     Rgn
Other-End: 10.64.115.2
                                      2056 sildvmg1
Set-End: 10.64.115.28
                                      20000 sildvaes8
Audio Connection Type: ip-tdm
```

```
status station 30002
                                           Page 6 of 7
               IP ENDPOINT DATA
        Port: S00005
Product ID-Release: IP_Phone 6.6506
                                         H.245 Tunneled? does not apply
Registration Status: registered-authenticated MAC Address: 50:cd:22:b2:62:42
Authentication Type: DES-56-plus
                                      Dependency Mode: main
Native NAT Address: not applicable
ALG NAT WAN Address: not applicable
        Shared Port: S00003
                                 Shd Signal Status: connected
Product ID-Release: IP_API_A 3.2040 H.245 Tunneled? does not apply
Registration Status: registered-authenticated MAC Address: unavailable
Authentication Type: DES-56-plus
                                      Dependency Mode: independent
Native NAT Address: not applicable
ALG NAT WAN Address: not applicable
```

With an active call, use **status station** on H.323 and DCP stations, or status trunk port x for SIP stations to view audio connection status. In the below screenshot, this verifies the SRTP media connection between the Recite recorder (10.64.115.30) and the gateway.

```
Page 8 of 10
status station 30002
            SRC PORT TO DEST PORT TALKPATH
src port: S00005
S00005:TX:10.64.115.36:2982/g729a/20ms/1-srtp-aescm128-hmac80
001V011:RX:10.64.115.2:2058/g729/20ms/1-srtp-aescm128-hmac80:TX:ctxID:275
001V011:RX:ctxID:275:TX:10.64.115.2:2056/g729/20ms/1-srtp-aescm128-hmac80
S00003:RX:10.64.115.32:20000/g729a/20ms/1-srtp-aescm128-hmac80
status station 30002
                                         Page 9 of 10
            SRC PORT TO DEST PORT TALKPATH
src port: S00005
S00005:TX:10.64.115.36:2982/g729a/20ms/1-srtp-aescm128-hmac80
001V011:RX:10.64.115.2:2058/g729/20ms/1-srtp-aescm128-hmac80:TX:ctxID:275
001V011:RX:ctxID:275:TX:10.64.115.2:2060/g729/20ms/1-srtp-aescm128-hmac80
T00001:RX:10.64.115.30:5004/g729/20ms/1-srtp-aescm128-hmac80
```

Use the **list monitored-station** command to verify ASAI associations to provide TSAPI events and control to the application:

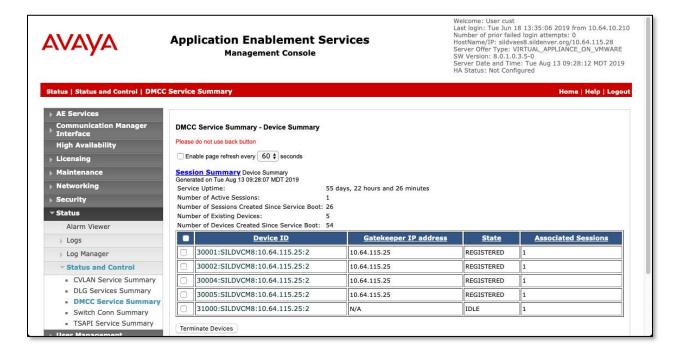


8.2. Avaya Aura® Application Enablement Services

On the AES, navigate to **Status > Status and Control > DMCC Service Summary** to verify Recite is connected using an encrypted session:



Click on the **Device Summary** link to view the registrations:

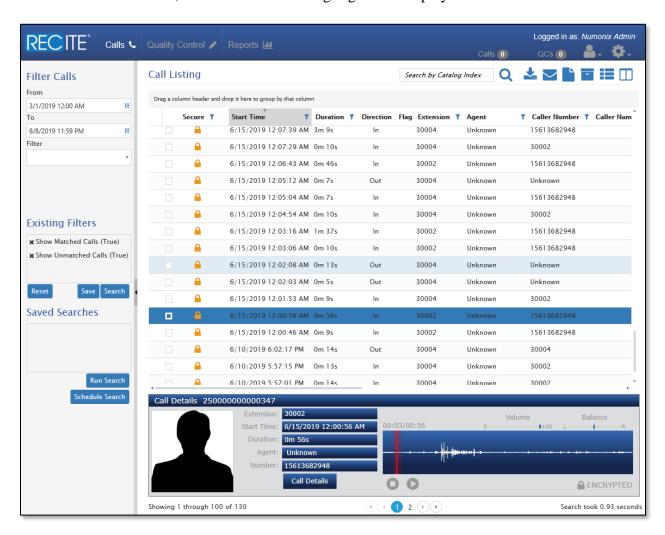


8.3. Numonix Recite

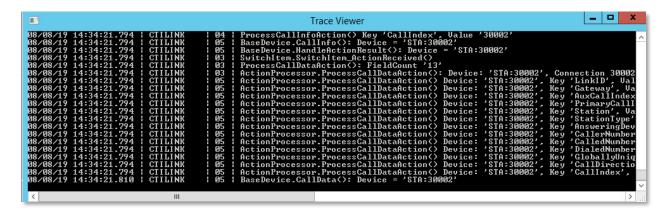
On the Recite server, the following applet will show the current connected status of the recorder when lights are green (when a channel is actively recording, it will turn red):



In the web user interface, search for calls and highlight them replay an interaction:



Details about event data will appear in the **Trace Viewer** applet on the Recite server as shown below, this can be useful to see events in real-time:



9. Conclusion

These Application Notes describe the configuration steps required for Numonix Recite to successfully interoperate with Avaya Aura[®] Communication Manager and Avaya Aura[®] Application Enablement Services to record audio calls. Recite passed all compliance test cases successfully, please refer to **Section 2.2** for results and any observations.

10. Additional References

This section references the product documentation relevant to these Application Notes. Product documentation for Avaya products may be found at http://support.avaya.com.

Avaya:

- 1. Administering Avaya Aura® Communication Manager, Release 8.0.x Issue 4, May 2019
- 2. Administering Avaya Aura® Application Enablement Services, Release 8.0.x Issue 3, August 2019

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