

Avaya Solution & Interoperability Test Lab

Application Notes for configuring Liquid Assure from Liquid Voice to interoperate with Avaya Aura® Communication Manager R7.0 and Avaya Aura® Application Enablement Services R7.0 using DMCC Multi-Registration to record calls - Issue 1.0

Abstract

These Application Notes describe the configuration steps for Liquid Voice Liquid Assure to interoperate with the Avaya solution consisting of an Avaya Aura® Communication Manager R7.0, an Avaya Aura® Session Manager R7.0, and Avaya Aura® Application Enablement Services R7.0 using Multi-Registration.

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

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

1. Introduction

These Application Notes describe the configuration steps for the Liquid Assure R7.1 from Liquid Voice to interoperate with the Avaya solution consisting of an Avaya Aura® Communication Manager R7.0, an Avaya Aura® Session Manager R7.0, and Avaya Aura® Application Enablement Services R7.0. Liquid Assure uses Communication Manager's Multiple Registrations 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**.

DMCC works by allowing software vendors to create soft phones on a recording server, and use them to monitor and record Avaya phonesets. This is purely a software solution and does not require telephony boards or any wiring beyond a typical network infrastructure. The DMCC API associated with the AES server monitors the digital and VoIP extensions. The application uses the AE Services DMCC service to register itself as a recording device at the target extension. When the target extension joins a call, the application automatically receives the call's aggregated RTP media stream via the recording device and records the call.

Liquid Assure is a modular based call-recorder with an easy-to-use web based interface. The modular design allows the system to be scaled to any number of extensions and sites. The web interface provides tools for searching and retrieving recording, forwarding exporting and annotating recordings, centralized system security authorization and auditing, and system status monitoring. The base solution can be amended with additional add-ons including Avaya powered speech analytics, screen-recordings and quality management. The same recording components are also available as an SME targeted product called Liquid Recording. Liquid Recoding has a reduced feature-set and is limited to recording 60 concurrent calls.

2. General Test Approach and Test Results

The interoperability compliance testing evaluated the ability of the Liquid Assure to carry out call recording in a variety of scenarios using DMCC Multi-Registration with AES and Communication Manager. A range of Avaya endpoints were used in the compliance testing all of which are listed in **Section 4**.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

2.1. Interoperability Compliance Testing

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

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

2.2. Test Results

All functionality and serviceability test cases were completed successfully, except for the following feature test which had an issue as follows.

1. **Call Park**. The un-parked call is not being recorded. It appears that there are no events being sent for un-parking a call by Communication Manager. Modification Report [**CM-9860**] has been raised with the Communication Manager support team. A fix for this issue will be implemented for release 7.1 of Communication Manager.

2.3. Support

Technical support can be obtained for Liquid Assure from:

Website http://www.liquidvoice.com
 Telephone +44 (0) 113 200 2020
 Email support@liquidvoice.com

3. Reference Configuration

The configuration in **Figure 1** was used to compliance test Liquid Assure with the Avaya solution using DMCC Multi-Registration to record calls. The Liquid Voice server is setup for DMCC Multi-Registration mode and connects to the AES.

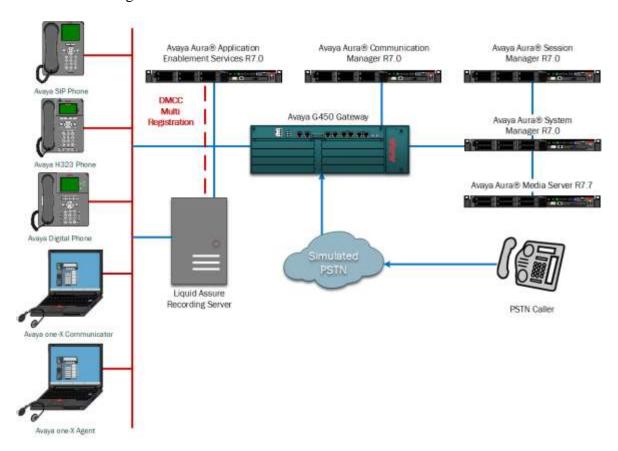


Figure 1: Connection of Liquid Assure R7.1 from Liquid Voice with Avaya Aura® Communication Manager R7.0, Avaya Aura® Session Manager R7.0 and Avaya Aura® Application Enablement Services R7.0

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® System Manager running on a virtual server	System Manager 7.0.0.1 Build No. – 7.0.0.0.16266-7.0.9.7001011 Software Update Revision No: 7.0.0.1.4212
Avaya Aura® Session Manager running on a virtual server	Session Manager R7.0 Build No. – 7.0.0.1.700102
Avaya Aura® Communication Manager running on a virtual server	R7.0 SP1 00.0.441.0-22684
Avaya Aura® Application Enablement Services running on a virtual server	R7.0 Build No – 7.0.0.0.1.13
Avaya G450 Gateway	37.19.0 /1
Avaya 9608 H323 Deskphone	96x1 H323 Release 6.6.028
Avaya 9641 SIP Deskphone	96x1 SIP Release 7.0.0.39
Avaya 9630 SIP Deskphone	R2.6.13.1
Avaya one-X® Communicator H.323	R6.2.4.07-FP4
Avaya one-X® Agent	R 2.5.50022.0
Avaya 9408 Digital Deskphone	FW Version 2
Avaya DECT Handsets	3725 DH4 (R3.3.11) 3720 DH3 (R3.3.11)
Liquid Voice, Liquid Assure - Liquid Assure Standalone Server - Liquid Recording Service	V7.1 Interface V7.0.0

5. Configure Avaya Aura® Communication Manager

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

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

5.1. Verify System Features

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

```
display system-parameters customer-options
                                                             Page
                                                                   3 of 11
                              OPTIONAL FEATURES
   Abbreviated Dialing Enhanced List? y
                                               Audible Message Waiting? y
      Access Security Gateway (ASG)? n
                                               Authorization Codes? y
      Analog Trunk Incoming Call ID? y
                                                            CAS Branch? n
A/D Grp/Sys List Dialing Start at 01? y
                                                              CAS Main? n
Answer Supervision by Call Classifier? y
                                                     Change COR by FAC? n
                               ARS? y Computer Telephony Adjunct Links? y
               ARS/AAR Partitioning? y Cvg Of Calls Redirected Off-net? y
         ARS/AAR Dialing without FAC? y
                                                           DCS (Basic)? y
                                                   DCS Call Coverage? y
         ASAI Link Core Capabilities? n
         ASAI Link Plus Capabilities? n
                                                     DCS with Rerouting? y
     Async. Transfer Mode (ATM) PNC? n
 Async. Transfer Mode (ATM) Trunking? n Digital Loss Plan Modification? y
             ATM WAN Spare Processor? n
                                                              DS1 MSP? y
                               ATMS? y
                                                  DS1 Echo Cancellation? y
                 Attendant Vectoring? y
```

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

Display the procr IP address by using the command **display node-names ip** and noting the IP address for the **procr** and AES (**aes70vmpg**).

display node-names	ip			Page	1 of	2
		IP NODE	NAMES			
Name	IP Address					
SM100	10.10.40.12					
aes70vmpg	10.10.40.16					
default	0.0.0.0					
G450	10.10.40.15					
procr	10.10.40.13					

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

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

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

change ip-s	1 of	4					
Service Type AESVCS	Enabled Y	Local Node procr	IP SERVICES Local Port 8765	Remote Node	Remote Port		

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

- AE Services Server: Name obtained from the AES server, in this case aes70vmpg.
- **Password:** Enter a password to be administered on the AES server.
- Enabled: Set to y.

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

change ip-serv	rices			Page	4 of	4
	AE					
Server ID	AE Services Server	Password	Enabled	Status		
1: 2: 3:	aes70vmpg	*****	У	idle		

5.4. Configure CTI Link for TSAPI Service

Add a CTI link using the **add cti-link n** command, where n is the n is the cti-link number as shown in the example below this is 1. Enter an available extension number in the **Extension** field. Enter **ADJ-IP** in the **Type** field, and a descriptive name in the **Name** field. Default values may be used in the remaining fields.

add cti-li	nk 1		Page	1 of	3
	CT	I LINK			
CTI Link:	1				
Extension:	7999				
Type:	ADJ-IP				
				COR:	l
Name:	aes70vmpg				

5.5. Configure H323 Stations for Multi-Registration

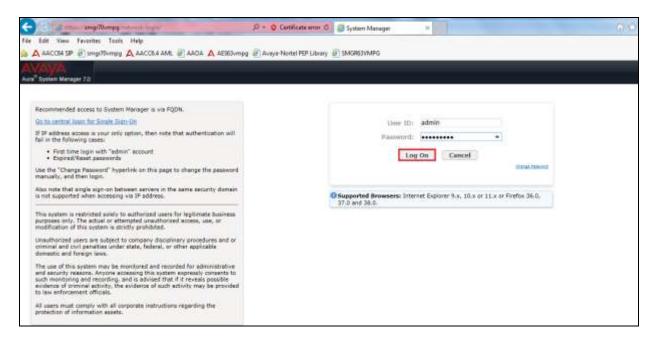
All endpoints that are to be monitored by Liquid Voice will need to have IP Softphone set to Y. IP Softphone must be enabled in order for Multi-Registration to work. Type **change station x** where x is the extension number of the station to be monitored also note this extension number for configuration required in **Section 8.1.** Note the **Security Code** and ensure that **IP SoftPhone** is set to y.

change station x			Page	1 of	6
		STATION			
Extension: x		Tools Magaagaaaaaaaa	D.C	:C: 0	
		Lock Messages? n			
Type: 9608		Security Code: 1234		i: 1	
Port: S00101		Coverage Path 1:	CC	R: 1	
Name: Extension		Coverage Path 2:	CC	S: 1	
		Hunt-to Station:			
STATION OPTIONS					
		Time of Day Lock Table:			
Loss Group:	1 9	Personalized Ringing Pattern:	1		
Hood Group.	10	Message Lamp Ext:	_		
Charlennhana	2				
Speakerphone:	-	Mute Button Enabled?	У		
Display Language:	english				
Survivable GK Node Name:					
Survivable COR:	internal	Media Complex Ext:			
Survivable Trunk Dest?	У	IP SoftPhone?	У		
		IP Video Softphone?	n		
	Short/	Prefixed Registration Allowed:			

5.6. Configure SIP Stations for Multi-Registration

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

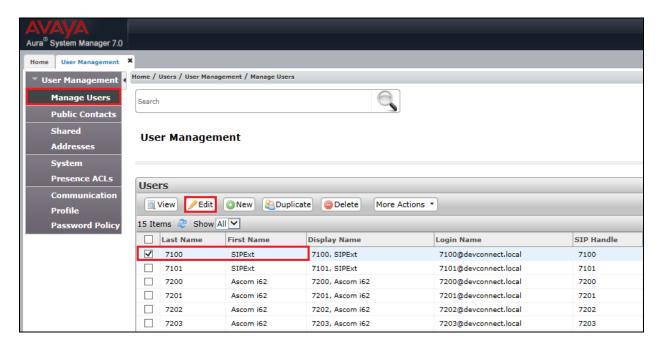
Note: The following shows changes a SIP extension and assumes that the SIP extension has been programmed correctly and is fully functioning.



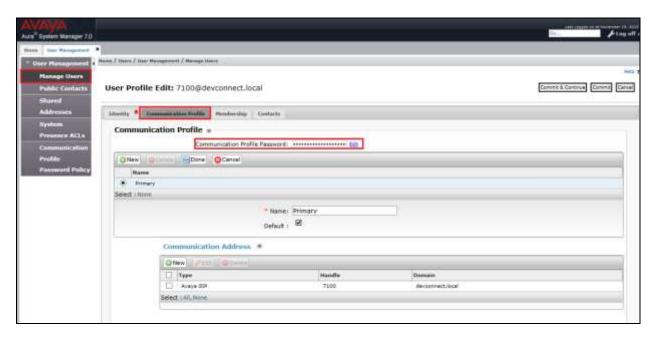
From the home page click on **User Management** highlighted below.



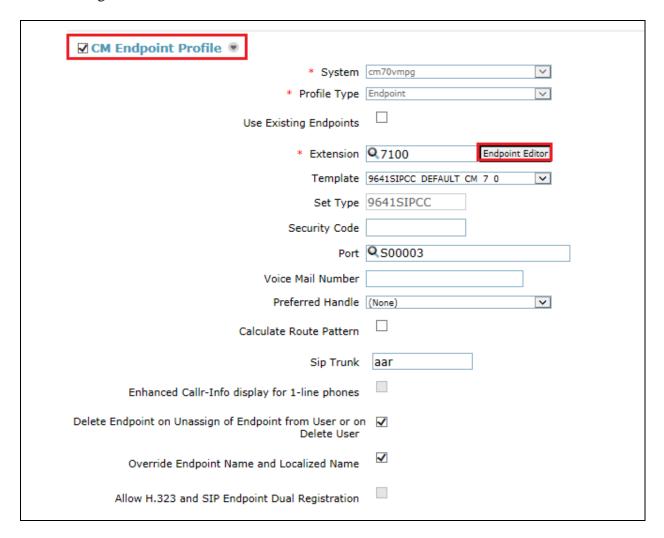
Click on Manager Users in the left window. Select the station to be edited and click on Edit.



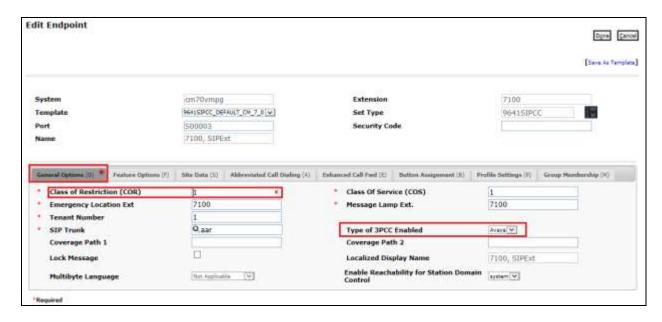
Click on the **Communication Profile** tab. Ensure that the **Communication Profile Password** is known and if not click on edit to change it.



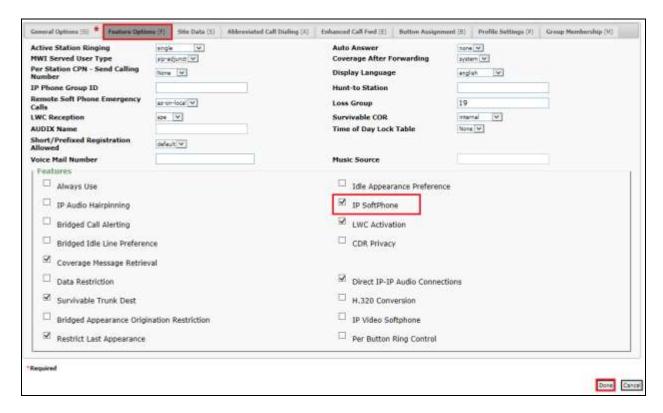
From the same page scroll down to **CM Endpoint Profile** click on **Endpoint Editor** to make further changes.



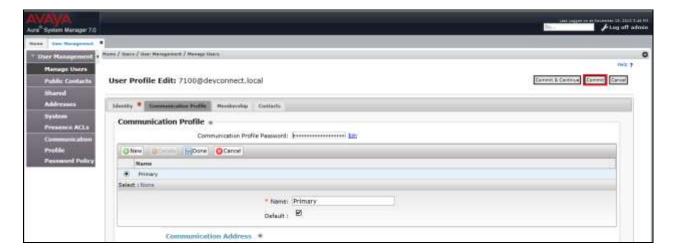
In the **General Options** tab ensure that **Type of 3PCC Enabled** is set to **Avaya** as is shown below.



Click on the **Feature Options** tab and ensure that **IP Softphone** is ticked as shown. Click on **Done**, at the bottom of the screen, once this is set.



Click on **Commit** once this is done to save the changes.



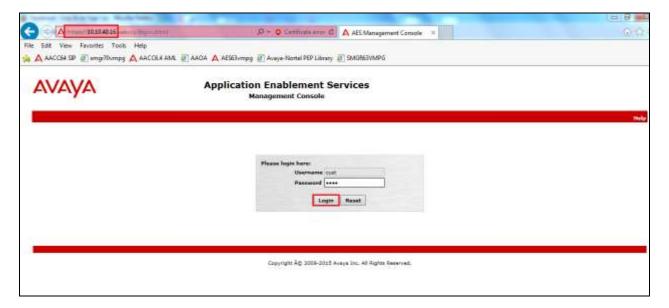
6. Configure Avaya Aura® Application Enablement Services

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

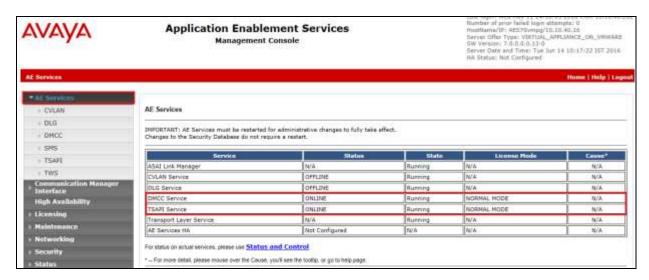
- Verify Licensing
- Create Switch Connection
- Administer TSAPI link
- Identify Tlinks
- Enable TSAPI and DMCC Ports
- Create CTI User
- Set Up Security Database on AES
- Associate Devices with CTI User

6.1. Verify Licensing

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



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

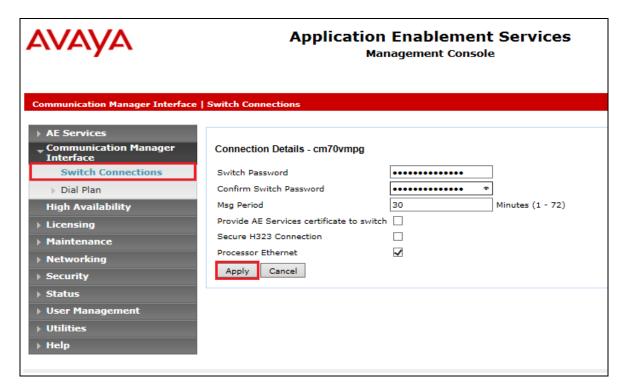


6.2. Create Switch Connection

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



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



From the **Switch Connections** screen, select the radio button for the recently added switch connection and select the **Edit PE/CLAN IPs** button (not shown, see screen at the bottom of the previous page. In the resulting screen, enter the IP address of the procr as shown in **Section 5.2** that will be used for the AES connection and select the **Add/Edit Name or IP** button.



6.3. Administer TSAPI link

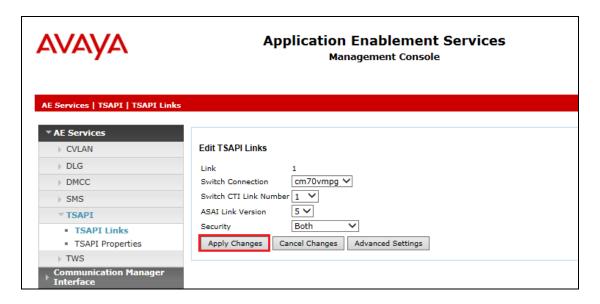
From the Application Enablement Services Management Console, select **AE Services** → **TSAPI** → **TSAPI Links**. Select **Add Link** button as shown in the screen below.



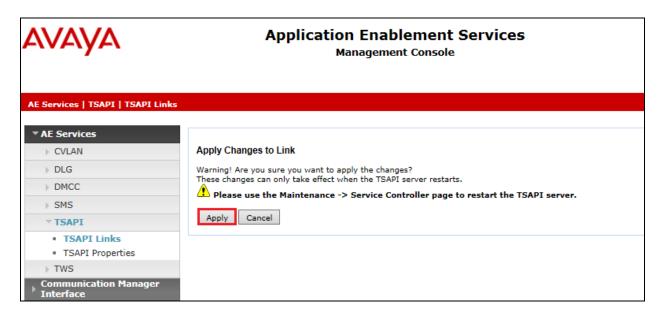
On the **Add TSAPI Links** screen (or the **Edit TSAPI Links** screen to edit a previously configured TSAPI Link as shown below), enter the following values:

- **Link:** Use the drop-down list to select an unused link number.
- **Switch Connection:** Choose the switch connection **cm70vmpg**, which has already been configured in **Section 6.2** from the drop-down list.
- **Switch CTI Link Number:** Corresponding CTI link number configured in **Section 5.4** which is **1**.
- **ASAI Link Version:** This can be left at the default value of 5.
- **Security:** This can be left at the default value of **both**.

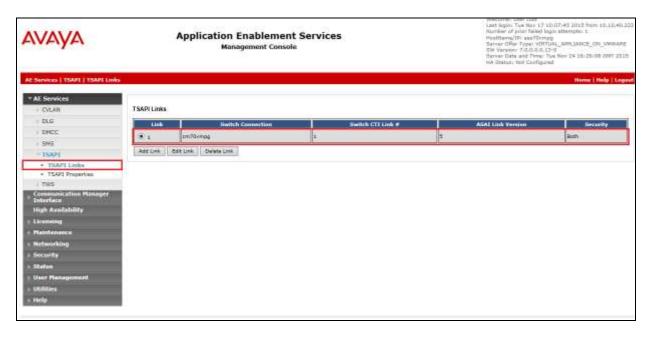
Once completed, select Apply Changes.



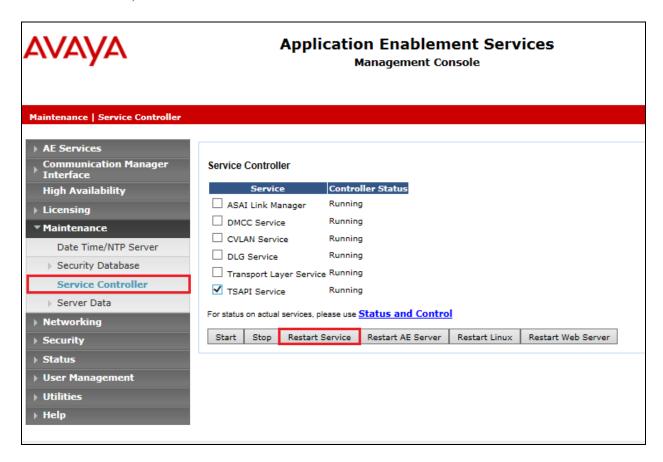
Another screen appears for confirmation of the changes made. Choose **Apply**.



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

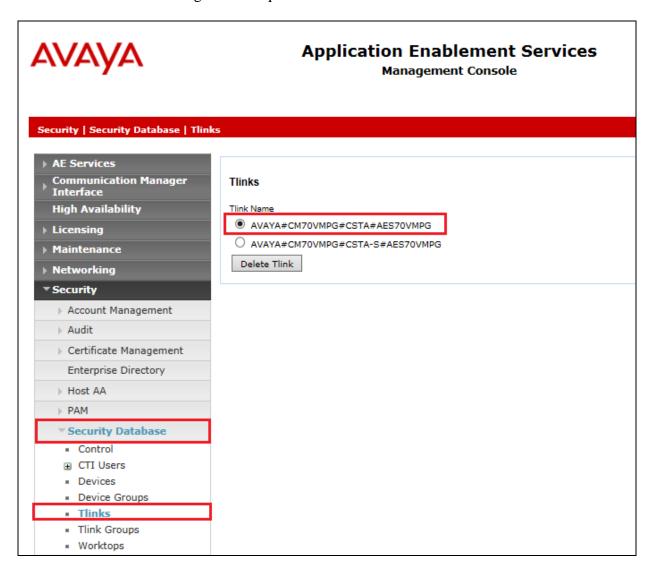


The TSAPI Service must be restarted to effect the changes made in this section. From the Management Console menu, navigate to **Maintenance** → **Service Controller**. On the Service Controller screen, tick the **TSAPI Service** and select **Restart Service**.



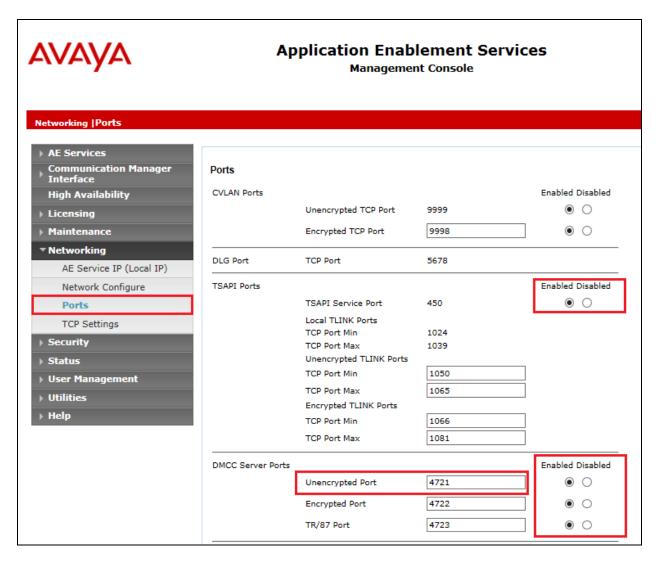
6.4. Identify Tlinks

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



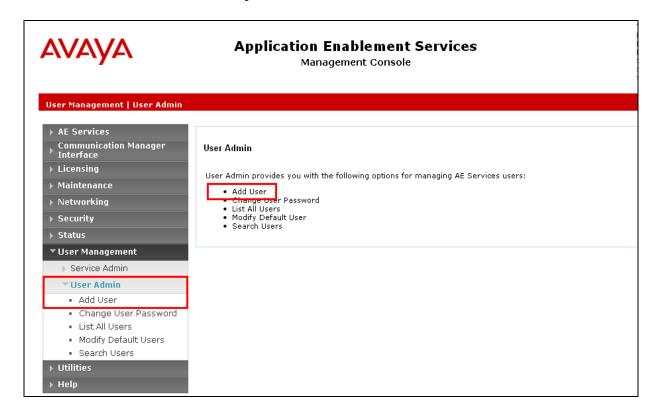
6.5. Enable TSAPI and DMCC Ports

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



6.6. Create CTI User

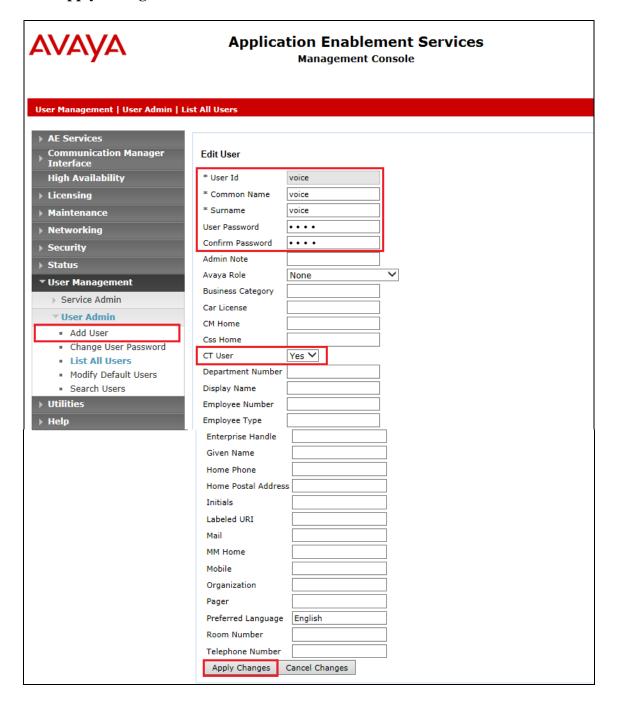
A User ID and password needs to be configured for the Liquid Assure to communicate with the Application Enablement Services server. Navigate to the **User Management** → **User Admin** screen then choose the **Add User** option.



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

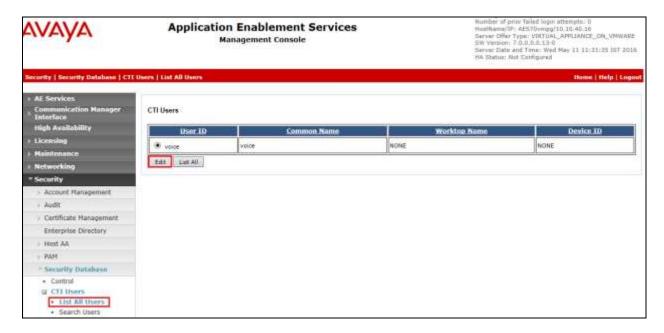
- User Id This will be used by the Liquid Assure setup in Section 7.1.
- Common Name and Surname Descriptive names need to be entered.
- User Password and Confirm Password This will be used with Liquid Assure setup in Section 7.1.
- **CT User -** Select **Yes** from the drop-down menu.

Click on **Apply Changes** at the bottom of the screen.

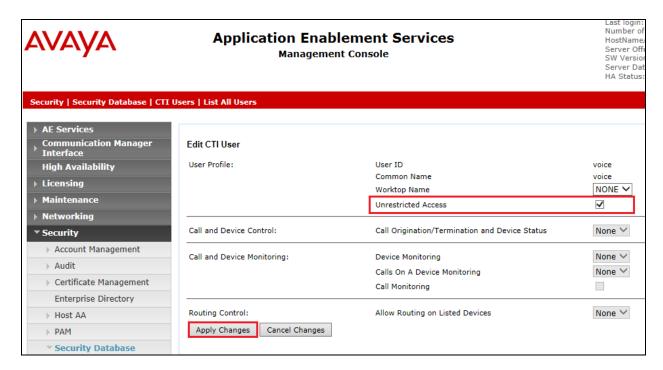


6.7. Associate Devices with CTI User

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



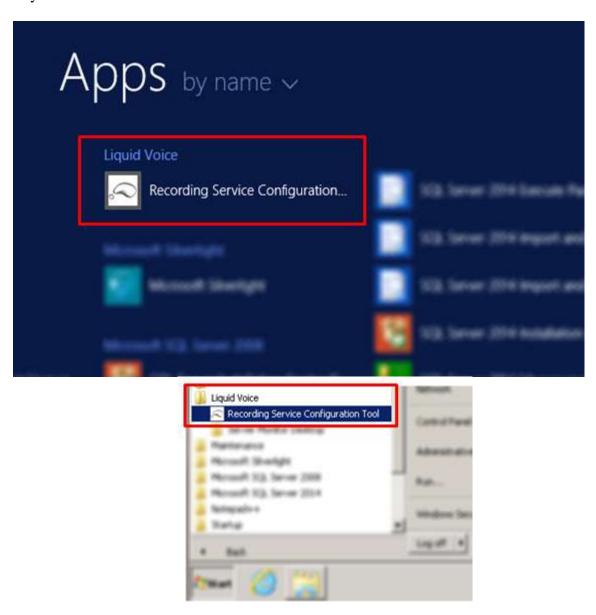
In the main window ensure that **Unrestricted Access** is ticked. Once this is done click on **Apply Changes**.



7. Configure Liquid Assure

The installation of Liquid Assure Recording is typically carried out by a Liquid Voice certified engineer and is outside the scope of these Application Notes. For information on the installation of Liquid Assure contact Liquid Voice as per the information provided in **Section 2.3**.

The following sections will outline the process involved in connecting the Liquid Assure to the Avaya solution. All configuration of the Liquid Assure for connection with the AES is performed from an application on the Liquid Assure server called **Recording Service Configuration Tool**, open this application as shown below depending on the Operating System of your server.



7.1. Configure connection to AES

With the Recording Service Configuration Tool opened, select the **DMCC** tab as shown, here the following information is added.

• **TLink String** This is the Tlink information from **Section 6.4**.

• **User/Password** This is the username and password configured in **Section 6.6**.

• **AES IP Address** This is the IP address of the AES server.

• CM/CLAN IP This is the IP address of Communication Manager as per Section

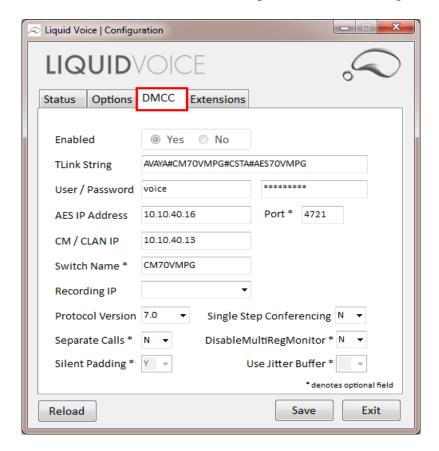
5.2 where the procr address is displayed.

• **Recording IP** This is the IP address of the Liquid Assure server.

Protocol Version Set to 7.0.
Conferencing Set to N.

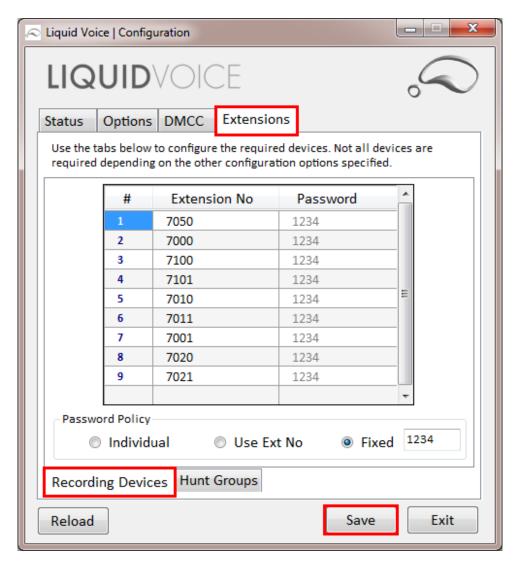
• **Separate Calls** Set to **Y/N**** (see below for further explaination).

The remaining fields were left as default. **Save** can be pressed before moving to another tab.



** Setting can be Y or N with Y separating calls for hold/transfer/conference. This setting will change the number of recordings present for calls that were held, transferred and conferenced. Separate Calls was set to both Y and N for compliance testing and works equally well for both. Whilst this setting is configurable it is advisable to consult Liquid Voice if you are considering changing this setting.

Click on the **Extensions** tab, all the Avaya phone sets that are to be "monitored" or recorded are added to the tab. If there are passwords on the phone sets and in this case there was of **1234** which was the case for all phone sets then a **Fixed** password can be applied as shown below. Entering each **Extension No** and clicking on Save at the bottom of the screen will result in the password being populated automatically for each extension.



This concludes the setup of the Liquid Assure Server for DMCC Multi-Registration recording.

8. Verification Steps

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

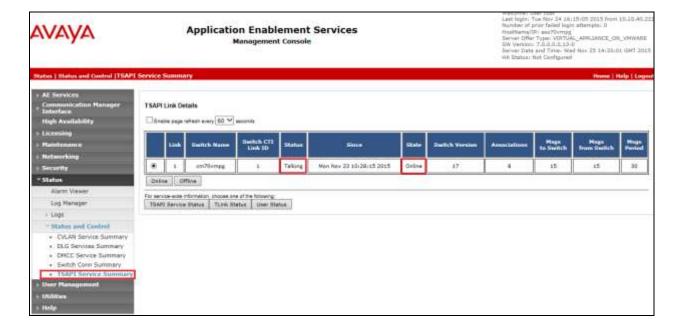
8.1. Verify Avaya Aura® Communication Manager CTI Service State

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

statu	s aesvcs ct	i-link				
			AE SERVICES	CTI LINK STATUS		
CTI Link	Version	Mnt Busy	AE Services Server	Service State	Msgs Sent	Msgs Rcvd
1	5	no	aes70vmpg	established	18	18

8.2. Verify TSAPI Link

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



8.3. Verify DMCC link on AES

Verify the status of the DMCC link by selecting Status \rightarrow Status and Control \rightarrow DMCC Service Summary to display the DMCC Service Summary – Session Summary screen. The screen below shows that the user **voice** is connected from the IP address **10.10.40.59** which is the Liquid Assure server.



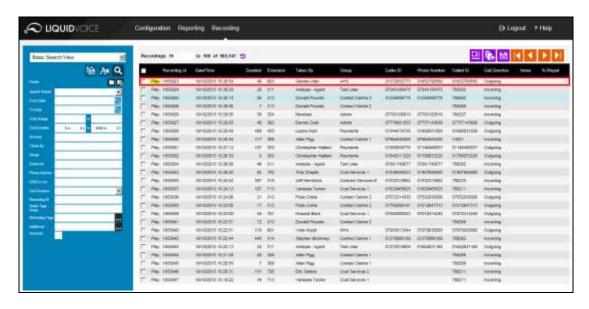
8.4. Verify calls are being recorded

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

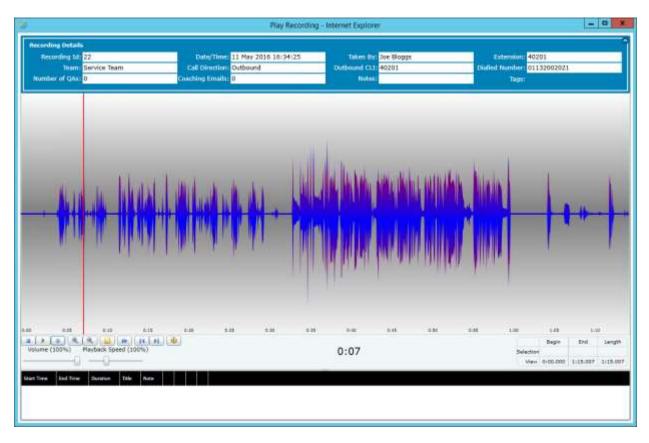
Open a browser session to the Liquid Assure server as is shown below. If Liquid Assure is configured for single sign-on the list of recordings will automatically appear, else enter appropriate login credentials and click on **Login**.



Once logged in the following window is automatically displayed. By default the system shows the most recent recordings first. There is a search panel on the left that can be used to filter the recording list by an abundance of different criteria. Click **Play** on the desired recording and this will open the built-in audio player and play back the call.



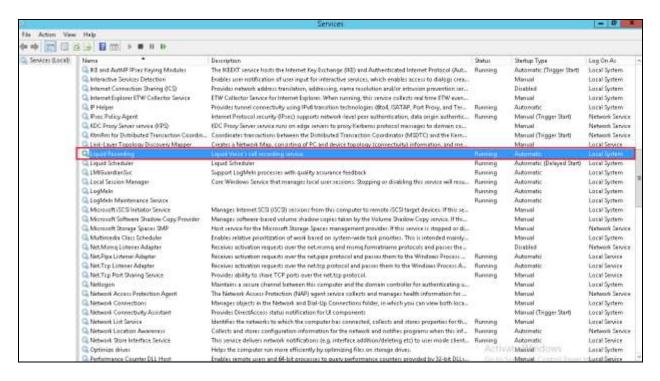
The player is opened and the recording is presented for playback. Click on the **Play/Pause** icons at the bottom of the screen to control and to play back the recording. Double clicking on the waveform will skip forward and backwards during playback as required.



8.5. Verify Liquid Voice Services

If these recordings are not present or cannot be played back the **Liquid Recording** service may not be running or may need to be restarted. The Liquid Assure server can be logged into and checked to ensure that the **Liquid Recording** service is running.

A restart of the Recording service is required following certain configuration changes whilst others such as adding and removing Recording Devices can be changed ad-hoc. There may be a number of installed services associated with Liquid Voice, for the purposes of this document only the Recording service requires a restart.



9. Conclusion

These Application Notes describe the configuration steps required for Liquid Assure R7.1 from Liquid Voice to successfully interoperate with Avaya Aura® Communication Manager R7.0 using Avaya Aura® Application Enablement Services R7.0 to connect to using DMCC Multi-Registration to record calls. All feature functionality and serviceability test cases were completed successfully with some issues and observations noted in **Section 2.2**.

10. Additional References

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

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

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

Product documentation for Liquid Voice products may be found at:

Website http://www.liquidvoice.com
 Telephone +44 (0) 113 200 2020
 Email support@liquidvoice.com

Appendix

Avaya one-X® Agent Softphone

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

```
display station 7011
                                                                        Page
                                                                                1 of
                                          STATION
Extension: 2100
                                             Lock Messages? n
                                                                                BCC: 0
                                        Security Code: *
Coverage Path 1:
Coverage Path 2:
Hunt-to Station:
                                                                                 TN: 1
     Type: 9630
     Port: S00031
                                                                                 COR: 1
     Name: one-X Agent1
                                                                                 cos: 1
                                                                             Tests? y
STATION OPTIONS
               Location: Time of Day Lock Table:
Loss Group: 19 Personalized Ringing Pattern: 1
        Speakerphone: 2-way Mute Button Enabled? y
Display Language: english Button Modules: 0
able GK Node Name:
                                                      Message Lamp Ext: 7011
 Survivable GK Node Name:
         Survivable COR: internal
                                                    Media Complex Ext:
   Survivable Trunk Dest? y
                                                            IP SoftPhone? y
                                                     IP Video Softphone? n
                                  Short/Prefixed Registration Allowed: default
                                                    Customizable Labels? Y
```

```
display station 7011
                                                                         Page 2 of 5
                                         STATION
FEATURE OPTIONS
          LWC Reception: spe Auto Select Any Idle Appearance? n
          LWC Activation? y
                                                           Coverage Msg Retrieval? y
 LWC Log External Calls? n
                                                                       Auto Answer: none
             CDR Privacy? n
                                                                 Data Restriction? n
  CDR Privacy? n
Redirect Notification? y
                                                     Idle Appearance Preference? n
 Per Button Ring Control? n
Bridged Call Alerting? n
                                                  Bridged Idle Line Preference? n
                                                        Restrict Last Appearance? y
  Active Station Ringing: single
                                                                EMU Login Allowed? n
       H.320 Conversion? n Per Station CPN - Send Calling Number?

Service Link Mode: as-needed EC500 State: enabled Multimedia Mode: enhanced Audible Message Waiting?

Served Moor Type: Display Client Redirection?
                                                               EC500 State: enabled
                                                         Audible Message Waiting? n
    MWI Served User Type:
                                                     Display Client Redirection? n
                                                     Select Last Used Appearance? n
               AUDIX Name:
                                                       Coverage After Forwarding? s
                                                         Multimedia Early Answer? n
Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
  Emergency Location Ext: 7011 Always Use? n IP Audio Hairpinning? n
```

```
display station 7011
                                                                Page
                                                                      3 of 5
                                     STATION
            Conf/Trans on Primary Appearance? n
  Bridged Appearance Origination Restriction? n
              Call Appearance Display Format: disp-param-default
                           IP Phone Group ID:
Enhanced Callr-Info Display for 1-Line Phones? n
                             ENHANCED CALL FORWARDING
                                      Forwarded Destination
                                                                    Active
Unconditional For Internal Calls To: 1000
                                                                       n
                  External Calls To: 1000
                                                                        n
         Busy For Internal Calls To:
                                                                       n
                  External Calls To:
                                                                       n
     No Reply For Internal Calls To:
                                                                       n
                  External Calls To:
                                                                       n
           SAC/CF Override: n
```

```
display station 7011
                                                                  Page
                                                                         4 of
                                      STATION
 SITE DATA
       Room:
                                                          Headset? n
       Jack:
                                                          Speaker? n
      Cable:
                                                        Mounting: d
     Floor:
                                                      Cord Length: 0
   Building:
                                                       Set Color:
ABBREVIATED DIALING
    List1:
                               List2:
                                                          List3:
BUTTON ASSIGNMENTS
1: call-appr
                                          5: manual-in
                                                                Grp:
                                          6: after-call
7: aux-work RC:
 2: call-appr
                                                                 Grp:
 3: call-appr
                                                                 Grp:
 4: auto-in
                       Grp:
                                          8:
    voice-mail
```

Avaya 9608 H.323 Deskphone

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

```
display station 7000
                                                                 Page
                                                                      1 of 5
                                    STATION
                                      Lock Messages? n
Security Code: *
Coverage Path 1: 1
Extension: 7000
                                                                      BCC: 0
    Type: 9608
                                                                       TN: 1
                                                                     COR: 1
    Port: S00000
                                      Coverage Path 2:
    Name: Ext2000
                                      Hunt-to Station:
                                                                    Tests? y
STATION OPTIONS
                                          Time of Day Lock Table:
             Loss Group: 19 Personalized Ringing Pattern: 1
                                                Message Lamp Ext: 7000
       Speakerphone: 2-way
Display Language: english
                                             Mute Button Enabled? y
                                                  Button Modules: 0
 Survivable GK Node Name:
         Survivable COR: internal
                                              Media Complex Ext:
   Survivable Trunk Dest? v
                                                     IP SoftPhone? v
                                              IP Video Softphone? n
                              Short/Prefixed Registration Allowed: yes
                                             Customizable Labels? y
```

```
display station 7000
                                                                    Page 2 of 5
                                       STATION
FEATURE OPTIONS
          LWC Reception: spe
LWC Activation? y
External Calls? n
                                      Auto Select Any Idle Appearance? n
                                                      Coverage Msg Retrieval? y
 LWC Log External Calls? n
                                                                  Auto Answer: none
            CDR Privacy? n
                                                             Data Restriction? n
  Redirect Notification? y
                                                Idle Appearance Preference? n
 Per Button Ring Control? n
Bridged Call Alerting? n
                                              Bridged Idle Line Preference? n
  Bridged Call Alerting? n
                                                    Restrict Last Appearance? y
 Active Station Ringing: single
                                                           EMU Login Allowed? n
        H.320 Conversion? n Per Station CPN - Send Calling Number?
       H.320 Conversion: n

Service Link Mode: as-needed EC500 State: enabled Audible Message Waiting? n
                                             Audible Message Walting? n
Display Client Redirection? n
    MWI Served User Type: sip-adjunct
                                                 Select Last Used Appearance? n
                                                   Coverage After Forwarding? s
                                                     Multimedia Early Answer? n
Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
  Emergency Location Ext: 7000 Always Use? n IP Audio Hairpinning? n
```

```
display station 7000
                                                                  Page
                                                                         3 of 5
                                     STATION
             {\tt Conf/Trans} on Primary Appearance? n
   Bridged Appearance Origination Restriction? n
                                                    Offline Call Logging? y
         Require Mutual Authentication if TLS? n
               Call Appearance Display Format: disp-param-default
                           IP Phone Group ID:
Enhanced Callr-Info Display for 1-Line Phones? n
                              ENHANCED CALL FORWARDING
                                       Forwarded Destination
                                                                     Active
 Unconditional For Internal Calls To:
                                                                         n
                   External Calls To:
                                                                         n
          Busy For Internal Calls To:
                                                                         n
                  External Calls To:
                                                                         n
      No Reply For Internal Calls To:
                                                                         n
                   External Calls To:
            SAC/CF Override: n
```

display station 7000			Page	4 of	5
	STAT	ION			
SITE DATA					
Room:		Headset?	n		
Jack:		Speaker?	n		
Cable:		Mounting:	d		
Floor:		Cord Length:	0		
Building:		Set Color:			
ABBREVIATED DIALING					
List1:	List2:	List3:			
BUTTON ASSIGNMENTS					
1: call-appr		5: call-park			
2: call-appr		6∶			
3: call-appr		7:			
4: extnd-call		3∶			
voice-mail					

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