

Avaya Solution & Interoperability Test Lab

Application Notes for Liquid Voice Assure Interaction Recording with Avaya Aura® Contact Center and Avaya Aura® Communication Manager using port mirroring of Avaya Session Border Controller for Enterprise to record trunk calls – Issue 1.0

## Abstract

These Application Notes describe the configuration steps required for Liquid Voice Assure Interaction Recording V7.5 to interoperate with Avaya Aura® Communication Manager R8.1 and Avaya Aura® Contact Center R7.1 using 'port mirroring' of Avaya Session Border Controller for Enterprise R8.1.1 to record trunk calls.

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 required for Liquid Voice Assure Interaction Recording V7.5 to interoperate with Avaya Aura® Contact Center R7.1 and Avaya Aura® Communication Manager R8.1 using 'port mirroring' of Avaya Session Border Controller for Enterprise R8.1.1 to record trunk calls.

A PSTN is simulated using an Avaya Session Border Controller for Enterprise (ASBCE) connecting to another Avaya system. Calls are recorded by mirroring the external interface on the ASBCE. Recordings are started and stopped using SIP headers obtained from the mirrored port. Liquid Voice Assure Interaction Recording (Assure) connects to the Call Detail Recording (CDR) port on Communication Manager and to Communication Control Toolkit (CCT) Web Services to obtain both call and agent events to help provide the user with as much information as possible about each call recording.

On each of the agent's PCs the Liquid Voice PCI silencing client is installed, and this client has a connection in place with CCT to generate DTMF tones to the agent on call, pausing the recording when sensitive information is being exchanged.

This Liquid Voice system is fully integrated into a LAN (Local Area Network), and WAN (Wide Area Network) and includes easy-to-use Web based applications that works with .NET framework and used to retrieve telephone conversations from a comprehensive long-term calls database. The Liquid Voice applications suite contains tools for audio retrieval, centralized system security authorization, system control, and system status monitoring. Also included is a call parameters database, search tools, a wide variety of Recording-on-Demand capabilities, and comprehensive long-term call database for immediate retrieval.

# 2. General Test Approach and Test Results

The general test approach was to validate the ability of Assure to correctly and successfully record telephone calls in a call center environment. Contact Center agents were logged into various Avaya endpoints (outlined in **Section 4**) and by mirroring the external port of the SBCE all RTP was sent to a dedicated Network Interface Card (NIC) on the Assure server. Recordings were made using the SIP headers from the packets received by Assure from the SBCE mirrored port.

The connection to CCT web services and to Communication Manager CDR are used to obtain information on each recording such as CLID, DNIS, agent ID and skillset information. The recordings can be made with or without these connections to CCT and Communication Manager. Both of these connections can be used individually or as a hybrid solution depending on what information is required to be displayed to the user, see *Application Notes for Liquid Voice Assure Interaction Recording with Avaya Aura Communication Manager using port mirroring of Avaya Session Border Controller for Enterprise to record trunk calls.* 

The Liquid Voice PCI silencing client is used to "pause" the recording when taking sensitive information on the customer. The PCI silencing client has a connection to CCT which allows the

client to generate DTMF tones and send them down the line thus pausing the recording at that point. The PCI silencing client connects to the Pause Service for the configuration settings. The PCI silencing client uses the user's domain credentials on the client PC for Single Sign on to access the CCT service. When a pause request is generated, the PCI silencing client sends a request to the Avaya CCT service to generate DTMF tones on any of the user's terminals that are in a state that is capable of DTMF tone generation.

In total the Liquid Voice solution takes advantage of four separate connections to complete the full solution, that being:

- 1. CTI Call Detail connection to CCT to obtain agent events.
- 2. PCI silencing client to CCT to pause recordings.
- 3. CDR connection to Communication Manager to obtain call events.
- 4. Port Mirror connection to ASBCE to obtain the call recording itself.

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 Assure did not include use of any specific encryption features as requested by Liquid Voice.

## 2.1. Interoperability Compliance Testing

The interoperability compliance test included both feature functionality and serviceability testing. The feature functionality testing focused on interacting with the Assure platform in different call scenarios. The tests included:

- **Inbound calls** Test call recording for inbound calls to Communication Manager deskphones from the PSTN.
- **Outbound calls** Test call recording for outbound calls from Communication Manager deskphones to the PSTN.
- Hold/Transferred/Conference calls Test call recording for calls transferred to and in conference with PSTN callers.

- **Contact Center Agent calls** Call to CDN's answered by Agents, these calls include transfer and conference of other Agents.
- **PCI silencing client, DTMF testing** Calls to agents using URL's to initiate DTMF tones to "pause" the recording.
- **Serviceability testing** The behaviour of Liquid Voice Recording solution under different simulated failure conditions on the Avaya platform will also be observed.

## 2.2. Test Results

All test cases were executed successfully, with the following observations noted.

- 1. When a call is resumed from being placed on hold from a 9641G SIP phone, the recording breaks down at the time resume is pressed. This only seems to occur when the phone is dialed directly and not when the skillset is called. Looking at a trace of the RTP from the phone, it appears as though a burst of G.722 audio is being sent out from the 9641G SIP phone at the time resume is being pressed. At this point it is unknown why this is occurring as there is no G.722 programmed on the system as a usable CODEC. Avaya are investigating the issue and G.722 can be taken out of the 46xxsettings file to ensure that this will not occur (see Appendix for further information on this). Liquid Voice also have a workaround in place using a patch to accommodate the G.722 audio being sent.
- 2. If the PSTN calls to the extension and the Avaya deskphone places the PSTN caller on hold, all that is being recorded is the MOH coming from Communication Manager. However, if the same PSTN caller calls to the CDN/Skillset and is being answered by the same extension but this time on a skillset call and being placed on hold by the same Avaya deskphone now both the MOH and the PSTN caller are recorded. The PSTN caller calls to the heard shouting down the phone on top of the MOH but only when on a skillset call, not when a call is made directly to the agent's phone.

## 2.3. Support

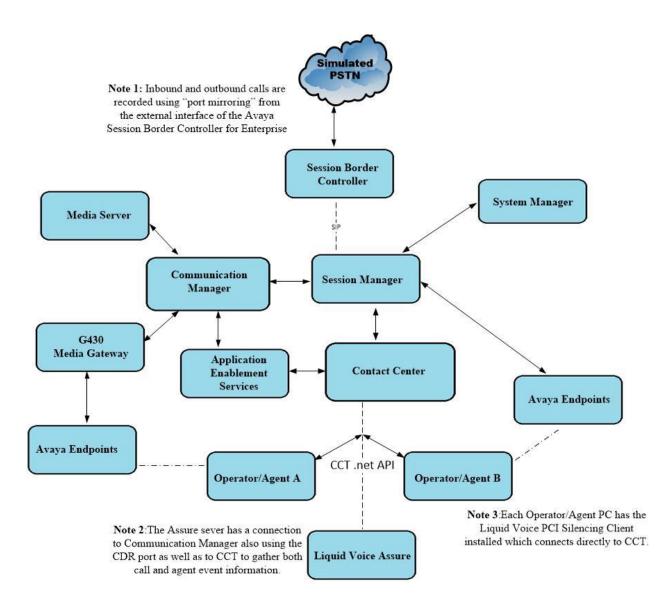
Technical support can be obtained for Liquid Assure from:

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

# 3. Reference Configuration

**Figure 1** below shows Avaya Aura® Communication Manager serving Digital, H.323 and SIP endpoints with Avaya Aura® Contact Center used to receive skillset calls. Assure has three connections to Communication Manager to obtain CDR, to CCT web services to obtain agent events and to initiate DTMF and to Session Border Controller for Enterprise to obtain all IP packets through a mirroring of its external port.

Note: SIP, H.323 and Digital endpoints were used during compliance testing.



#### Figure 1: Connection of Liquid Voice Assure Interaction Recording with Avaya Aura® Communication Manager R8.1 and Avaya Aura® Contact Center R7.1

PG; Reviewed: SPOC 1/10/2021

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# 4. Equipment and Software Validated

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

Avaya Equipment/Software	Release/Version
Avaya Aura® System Manager running on a virtual server	System Manager 8.1.2.0 Build No. – 8.1.0.0.733078 Software Update Revision No: 8.1.2.0.0611261 Feature Pack 2
Avaya Aura® Session Manager running on a virtual server	Session Manager R8.1.2 Build No. – 8.1.2.0.812039
Avaya Aura® Communication Manager running on a virtual server	R8.1.2.0 – FP2 R018x.00.0.890.0 Update ID 01.0.890.0-26095
Avaya Aura® Contact Center	7.1.0.3
Avaya Session Border Controller for Enterprise	8.1.1.0-26-19214
Avaya Aura® Application Enablement Services	8.1.2
Avaya Aura® Media Server	8.0.0.169
Avaya G430 Media Gateway	41.16.0/1
Avaya J179 H.323 Deskphone	6.8304
Avaya 96x1 SIP Deskphone	7.1.2.0.14
Avaya Digital 9408	2.00
Liquid Voice Equipment/Software	Release/Version
Liquid Voice Assure Interaction Recording <ul> <li>Interface version</li> <li>Recording Service version</li> <li>Cti Call Detail version</li> <li>CDR Network Service</li> </ul>	7.5.1 8.3.4 3.1.1.3 7.1
Liquid Voice PCI silencing client running on Windows 10 PC	10.0.1.43

# 5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section are performed using the Communication Manager System Access Terminal (SAT). The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation as referenced in **Section 10**.

The following sections illustrate the steps required to allow CDR data to be sent to the Assure server. The first step is to add the Assure server as a name and IP address in the **IP NODE NAMES**. Use the command **change node-names ip** to add the Assure server. This was added using the name **LVoice** with IP address of **10.10.40.122**, as highlighted below.

change node-names	ip	Page 1 of 2
	IP NODE NAMES	
Name	IP Address	
G430-Home	192.168.40.15	
IP500V2	192.168.40.20	
IPOffice	10.10.40.25	
LVoice	10.10.40.122	
aes81xvmpg	10.10.40.38	
ams81vmpg	10.10.40.39	
default	0.0.0	
g430	10.10.40.15	
procr	10.10.40.37	
procr6	::	
sm81xvmpg	10.10.40.32	
( 11 of 11 admi	nistered node-names were displayed	d )
Use 'list node-na	mes' command to see all the admini	stered node-names
Use 'change node-	names ip xxx' to change a node-nam	ne 'xxx' or add a node-name

Add the CDR service into IP Services by typing **change ip-services**. Note the following as this information may be needed when setting up the Assure server.

- Local Node is procr
- **Remote Node** is that of the **LVoice** entered as it was configured above
- Service Type is CDR1
- **Remote Port** number in this example shown as **9001** but can be any free port number

change ip-s	services				Page	1 of	3
Service Type CDR1	Enabled	Local Node procr	IP SERVICES Local Port 0	Remote Node LVoice	:	Remote Port 9001	

Use the command **change system-parameters cdr** to make changes to the way the CDR data is sent out. The following changes on **Page 1** were made specifically for this testing.

- Primary Output Endpoint is set to CDR1
- Outg Trk Call Splitting is set to y
- Inc Trk Call Splitting is set to y

change system-parameters cdr Page 1 of 2
CDR SYSTEM PARAMETERS
Node Number (Local PBX ID): CDR Date Format: month/day
Primary Output Format: customized Primary Output Endpoint: CDR1
Secondary Output Format:
CDR Retention (days): 20
Use ISDN Layouts? n Enable CDR Storage on Disk? n
Use Enhanced Formats? n Condition Code 'T' For Redirected Calls? n
Use Legacy CDR Formats? y Remove # From Called Number? n
Modified Circuit ID Display? n Intra-switch CDR? n
Record Outgoing Calls Only? n Outg Trk Call Splitting? y
Suppress CDR for Ineffective Call Attempts? y Outg Attd Call Record? y
Disconnect Information in Place of FRL? n Interworking Feat-flag? n
Force Entry of Acct Code for Calls Marked on Toll Analysis Form? n
Calls to Hunt Group - Record: member-ext
Record Called Vector Directory Number Instead of Group or Member? n
Record Agent ID on Incoming? y Record Agent ID on Outgoing? y
Inc Trk Call Splitting? y Inc Attd Call Record? n
Record Non-Call-Assoc TSC? n Call Record Handling Option: warning
Record Call-Assoc TSC? n Digits to Record for Outgoing Calls: dialed
Privacy - Digits to Hide: 0 CDR Account Code Length: 15
Remove '+' from SIP Numbers? y

**Page 2** of the system-parameters cdr shows the various fields and lengths that were set specifically for Liquid Voice Assure to obtain the records correctly.

char	nge system-parame	ters c	dr				Page	2	of	2
	CDR SYSTEM PARAMETERS									
	Data Item - Len									gth
1:	date	- 6	17:	in-trk-code	- 4	33:	line-feed			- 1
2:	space		18:	space	- 1	34:				-
3:	time	- 4	19:	in-crt-id	- 3	35:				-
4:	space	- 1	20:	space	- 1	36:				-
5:	sec-dur	- 5	21:	out-crt-id	- 3	37:				-
6:	space	- 1	22:	space	- 1	38:				-
7:	cond-code	- 1	23:	ppm	- 5	39:				-
8:	space	- 1	24:	space	- 1	40:				-
9:	code-dial	- 4		isdn-cc	- 11	41:				-
10:	space	- 1	26:	space	- 1	42:				-
	code-used		27:	attd-console	- 2	43:				-
12:	space	- 1	28:	space	- 1	44:				-
13:	dialed-num	- 18	29:	vdn	- 5	45:				-
	space		30:	space	- 1	46:				_
	-			acct-code	- 15	47:				_
	space	- 1		return	- 1					_
	-									
			]	Record length =	= 117					

# 6. Configure Avaya Aura® Contact Center

This section provides the procedures for configuring Contact Center. The procedures fall into the following areas:

- Add a Windows User for Liquid Voice
- Configure CCT Web Services
- Add a CCT User for Liquid Voice

**Note:** The Contact Center and its workstations must be part of a domain. The Liquid Voice PCI silencing client uses the user's domain credentials on the client PC for Single Sign on to access the CCT service. When a pause request is generated, the PCI silencing client sends a request to the Avaya CCT service to generate DTMF tones on any of the user's terminals that are in a state that is capable of DTMF tone generation.

**Note:** It is assumed that a fully working Contact Center is already in place, with all the necessary agents configured and skillset routing in place. An overview of one agent that was used for compliance testing can be found in **Appendix 12**.

## 6.1. Add a Windows User for Liquid Voice

A user for Liquid Voice is added to the Contact Center server, this user will be used by Assure to connect to CCT as per **Section 7.1**. This user is also used to Enable SIP Call Recording on Web Services in **Section 6.2**.

From **Computer Management**, navigate to **Local Users and Groups** in the left window and selected **Users**. From the main window, right click and select **New User**.

Enter a suitable **User name** and **Password**, and best to choose **Password never expires** as shown below. Click on **Create** to complete. This user will be used as part of the Assure configuration in **Section 7.1**.

		New User	?	x			
User name:	9						
Full name:	Liqui	d Voice					
Description:	Used	for CCT connection					
Password:		•••••					
Confirm passwo	rd:	•••••					
User must ch	nange pa	assword at next logon					
User cannot	change	password					
Password ne	ever expi	res					
Account is d	isabled						
Help Create Close							

The new user is now visible at the bottom of the users' page, as shown.

Name	Full Name	Description
Mathematica Mat	r	Built-in account for administering
👧 Guest		Built-in account for guest access t
iceAdmin	iceAdmin	Built-in account for Avaya Contac
💭 IUSR_SWC	IUSR_SWC	Built-in account for Avaya Contac
Ivoice	Liquid Voice	Used for CCT connection

## 6.2. Configure CCT Web Services

Open the **CCT Console** as shown below.

	ŀ	$Apps$ by name $\sim$						م
		Windows PowerShell ISE	А	Contact Center ProviderApplicati	Δ	Multicast Address and Port Confi.		System Control and Monitor Utili
		Windows PowerShell ISE (x86)	А	Contact Center Ref Client	Α	Multicast Stream Control	Δ	TraceControl
√izard		Windows Server Update Services	А	Contact Center Server Utility	Α	Multimedia Dashboard	А	Update Manager
	Avaya		А	Data Management	Α	Network Configuration	А	WorkspacesHAConfigurator
	Α	Agent Certificate Configuration	А	Database Integration Wizard		Orchestration Designer - Contac.	 Caché	
	Δ	Agent Desktop Display Configur	Д	Database Maintenance	Α	Process Monitor		Launcher [CCDSInstance]
	Δ	Archive Restore	Д	High Availability	Α	REST Configurator		Management Portal [CCDSInsta
	Δ	CCT Console	Д	License Grace Period Reset	Α	Security Manager		Start Caché [CCDSInstance]
dvanced	Α	Computer Update Utility	Д	License Manager Configuration	Α	Server Configuration	8	Stop Caché [CCDSInstance]
nostic	Α	Contact Center Logfile Collector	Д	Log Archiver	Α	SIP Gateway Management Client	Activate	Studio [CCDSInstance] Windows
6)	Δ	Contact Center OI Ref Client	Д	Manager Administration Configu	Α	SMMC SystemTray		em in Control Panel to activate Terminal [CCDSInstance]
	Ċ	Ð						

Navigate to **CCT Web Services** in the left window, this should be under **Server Configuration**. From the main window, ensure that **Enable CCT Web Services** box is **ticked**, note the port number shown. Ensure that **Enable SIP Call Recording** is also **ticked** and click on Browse Users. Select **Local Machine** as the **Location** and search for the user added in **Section 6.1**.

	e Root\Communication Cont	rol Toolkit\Server Configuration\/	CCT Web Services]
	nices window help		
Console Root Console Root Bulk Provisioning To Bulk Provisioning To Server Configuration Deployment Typ License Configu CCT Web Servico Client Applications	<ul> <li>✓ Enable CCT Web Services</li> <li>SOA Configuration</li> <li>Host Name:</li> <li>AACC71</li> <li>Ports:</li> <li>9080</li></ul>	TLS Security Min TLS Level TLS Encryption Step 1: Certificate Signing Request (CSF Generate CSR Step 2: Import Trusted Certificate Authori CA Alias:	View  New Window from Here Help
		Brov	vse Users
	Enable SIP Call Recording     Call Recording User Account:     Browse Users	Search Parameters Location: Local Machine (AACC71 Search Type: All Users V	
		Login Name       First Name         AACC71ViceAdmin       iceAdmin         AACC71VIUSR_SWC       IUSR_SWC         AACC71Vivoice       Liquid         Search       5 users found	Last Name none None Voice OK Cancel Activate
< III >			Windows.

Enable CCT Web Services	TLS Security	Actions				
SOA Configuration	Min TLS Level TLSv1	CCT Web Services				
Host Name:	TLS Encryption	Apply changes				
AACC71	- Step 1: Certificate Signing Request (CSF	Discard changes				
Ports:	Generate CSR	View				
9080 🗘 - 9083		New Window from Here				
Session Timeout:	Step 2: Import Trusted Certificate Authori	👔 Help				
	Warning Information X					
Enable SIP Call Recording     Call Recording User Account:	You need to restart CCT s	services to apply any changes made.				
AACC71\voice Browse Users		ОК				

This message will appear and the CCT services will need to be shut down and started again.

## 6.3. Add a Communication Control Toolkit User for Liquid Voice

Log into **Contact Center Manager** by opening Internet Explorer and navigating to the Contact Center FQDN or IP address.

				_ 0 ×
A https://aacc71.devconnect.local/CCM	ALogin/Home/Login	P → 🗎 C 🗛 aacc71.devconnect.local	× A CCT Administration	☆ ☆
AVAYA		Contact Center - Manager	,	About
Contact Center - Manager				
	Login			
	User ID Password			
				Login

#### Click on Configuration.

Αναγα	Contac	ct Center - Manager		About   Audit Trail   C	hange Password	Logout
Launchpad	Launch	npad				
	0	Contact Center Management	0	Configuration		
	Ô	Access and Partition Management	Ô	Scripting		
	0	Real-Time Reporting	0	Emergency Help		
	0	Historical Reporting	0	Outbound		
	Ô	Call Recording and Quality Monitoring	Ô	Multimedia		
	( <u>©</u> )	Prompt Management				
	Last successf	ful login: 19/10/2020 16:17:11				

Open CCT Administration and select the secure or unsecure URL, whichever is preferred.

AVAYA	Configuration	Logged in user:
Server Download Status La	unchpad Help	
CCMS1     CCT     CCT1     CCT Administration     CCMM1	CCT Administration	
	CCT Administration HTTP URL CCT Administration HTTPS URL	http://aacc71:8081/WebAdmin/ https://aacc71:8445/WebAdmin/

Αναγα		_	ССТ	A	dministration		
	CCT Users	•					
Wo View Details	Login User Name	First Name	Last Name				
GraAdd new User	DEVCONNECT\aacc1	AACC1	Agent one				
Providers	DEVCONNECT\aacc2	AACC 2	Agent two				
	DEVCONNECT/lvoice	Liquid	Voice				
	AACC71\Administrator	Administrator	Local AACC71				
	K     Image: Delete       H     Image: Delete						

Right click on **Users** in the left window and select **Add new User**.

Enter the same user details as in **Section 6.1**. The configuration file in **Section 7.1** will use this username to obtain events from CCT.

AVAYA	CCT Administration
<b>C</b> ()	Update CCT User
<ul> <li>Users</li> <li>Workstations</li> <li>Groups</li> <li>Providers</li> </ul>	Image: Optimized state         Login User Name       aacc71\voice         First Name       Liquid         Last Name       Voice         Image: Optimized state       Voice         Image: Optimized state       Voice         Image: Optimized state       Voice
	Address Assignments  Terminal Assignments  Address Group Assignments  Address Group Assignments  Address Group Assignments  Save  Resource aacc71\/voice was created.

# 7. Configure Liquid Voice Assure

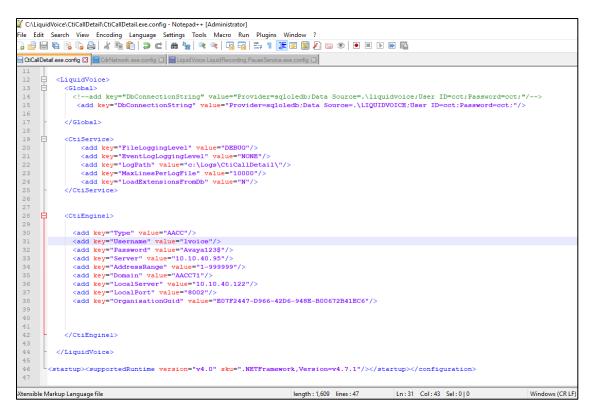
This section describes the steps required for Liquid Voice Assure Interaction Recording to interoperate with the Avaya solution. There are four connections to the Avaya solution required for this setup.

- 1. Liquid Voice Call Detail service connection to Avaya Aura® Contact Center, specifically to the Communication Control Toolkit (CCT) to gather events surrounding agent calls.
- 2. Connection to Avaya Aura® Contact Center, specifically to the Communication Control Toolkit (CCT) to allow Assure generate DTMF tones when a recording is being paused.
- 3. Connection to Avaya Aura® Communication Manager to collect CDR data.
- 4. Connection to Avaya Session Border Controller for Enterprise (ASBCE) via port mirror of the external interface on the ASBCE, to record all trunk calls.

**Note:** On most production setups the Assure server will have a separate connection to two different CCT servers in a geographical redundancy setup, however this was not tested during compliance testing and only one connection to one CCT server was configured.

# 7.1. Configure Liquid Voice Call Detail service connection to Avaya Aura® Contact Center

Assure connects to the CCT module of Contact Center. A user for CCT was setup in **Section 6.1** and **Section 6.3**, to allow Assure collect the necessary agent events from CCT. The location for the configuration file on the Assure server is shown in the top of the screen shot below. The file is named **CtiCallDetail.exe.config**.



Solution & Interoperability Test Lab Application Notes ©2021 Avaya Inc. All Rights Reserved. A closer examination of the **CtiEngine** section of the CtiCallDetail.exe.config configuration file shows the **Username** and **Password** that was used as per **Section 6.1**, as well as the IP address of the CCT server. The **Domain** entered should be the hostname of the CCT server. The **Server** address is the CCT server IP and the **LocalServer** is the Liquid Voice recorder IP.

```
<CtiEnginel>

<add key="Type" value="AACC"/>

<add key="Username" value="lvoice"/>

<add key="Password" value="999999"/>

<add key="Server" value="10.10.40.95"/>

<add key="AddressRange" value="1-999999"/>

<add key="Domain" value="AACC71"/>

<add key="LocalServer" value="10.10.40.122"/>

<add key="LocalServer" value="8002"/>

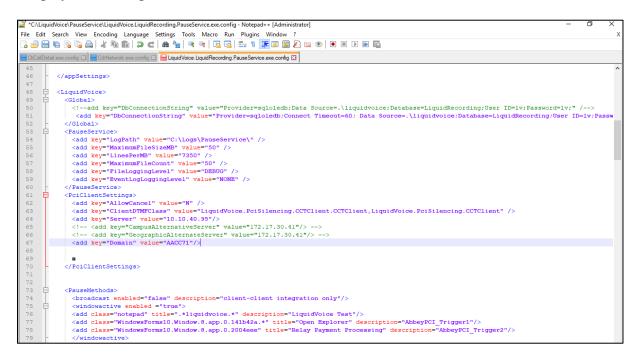
<add key="LocalPort" value="8002"/>

<add key="OrganisationGuid" value="E07F2447-D966-42D6-948E-B00672B41EC6"/>

</CtiEnginel>
```

#### 7.2. Liquid Voice Pause Solution connection to Avaya Aura® Contact Center

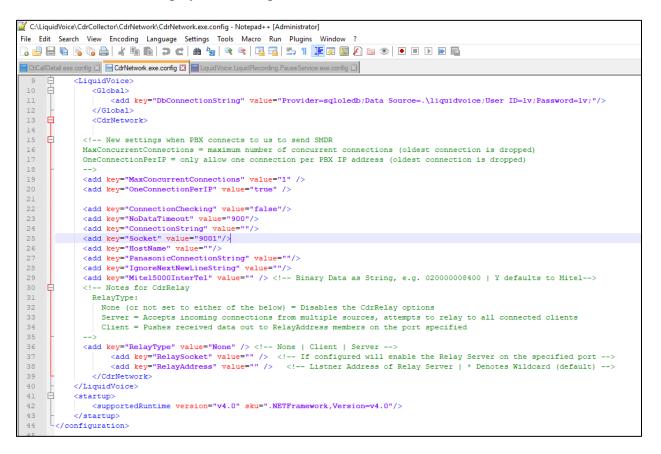
The pause solution is used to stop and start the recording automatically when an agent opens a certain URL. The Pause solution consist of two parts. The Pause Service and the PCI Clients. The PCI Clients take their configuration from the setting in the Pause Service configuration. When a pause request is received it is the client software that generates the DTMFs via the CCT connection. The Pause Service configuration file is used for that purpose. The location of this file is displayed at the top of the screen shot below.



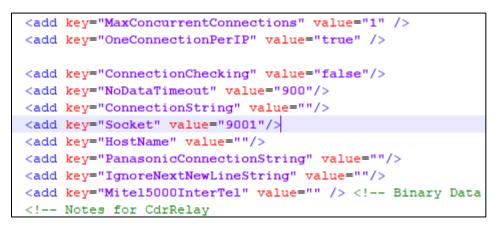
Solution & Interoperability Test Lab Application Notes ©2021 Avaya Inc. All Rights Reserved. 18 of 36 LVoice\_AACC71 A closer look at the file will show the CCT servers IP address along with the **Domain**, which is the host name of the CCT server, in that case both being that of the Contact Center server also.

#### 7.3. Configure connection to Avaya Aura® Communication Manager

Similarly, another configuration file is used to connect to Communication Manager. Again, the location of the file is displayed at the top of the screen shot below.



A closer look at the entry for the Socket will show the same port that was used in **Section 5.** Assure listen on port **9001** for the CDR output. If a host name or IP address is added to the **HostName** key the CDR service will connect to Communication Manager.



#### 7.4. Configure connection to Avaya Session Border Controller for Enterprise

A configuration file is used to connect to record the voice calls. These calls are recorded by mirroring the port on the Session Border Controller. The configuration file shows how this is setup. The location of the file is displayed at the top of the screen shot below i.e., **C:/LiquidVoice/RecordingService/** and the file is called **RecordingService.exe.config**.

	ngService			- 0	
File Home Shar					~
÷ → * ↑ 📴 > 1	his PC > Local Disk (C:) > LiquidVoice > RecordingService		v ⊙	Search RecordingService	م
📃 Desktop 🛛 🖈 ^	Name	Date modified	Туре	Size	
🕂 Downloads 🖈	LiquidVoice.LiquidRecording.WcfEventProcessor.pdb	03/08/2020 12:34	PDB File	46 KB	
🛱 Documents 🖈	LiquidVoice.Sip.SrtpManaged.dll	21/05/2020 09:59	Application extens	82 KB	
Pictures *	LiquidVoice.Sip.SrtpManaged.pdb	21/05/2020 09:59	PDB File	740 KB	
	LiquidVoice.Wcf.Core.dll	03/08/2020 12:34	Application extens	5 KB	
AudioConversio	LiquidVoice.Wcf.Core.pdb	03/08/2020 12:34	PDB File	14 KB	
CtiCallDetail	🗟 LvDecoders.dll	22/10/2019 14:38	Application extens	64 KB	
PauseService	LvDecoders.pdb	22/10/2019 14:38	PDB File	1,052 KB	
RecordingServic	OrganisationGuid_Digits_ScreenRecordingID	07/07/2017 11:54	TXT File	5 KB	
_	🔥 QueryOS	07/07/2017 11:54	Application	1,126 KB	
This PC	RecordingLibrary.dll	03/08/2020 12:34	Application extens	101 KB	
🧊 3D Objects	RecordingLibrary.pdb	03/08/2020 12:34	PDB File	228 KB	
E Desktop	RecordingService	03/08/2020 12:39	Application	42 KB	
Documents	RecordingService.exe.config	21/10/2020 11:44	CONFIG File	24 KB	
Downloads	RecordingService.exe-new.config	24/02/2020 12:24	CONFIG File	24 KB	
Music	recordingservice.InstallLog	07/10/2020 15:46	INSTALLLOG File	1 KB	
-	recordingservice.InstallState	07/10/2020 15:46	INSTALLSTATE File	8 KB	
Pictures	RecordingService.pdb	03/08/2020 12:39	PDB File	56 KB	
Videos	SoftlpRecordService.dll	03/08/2020 12:39	Application extens	31 KB	
🏪 Local Disk (C:)	SoftlpRecordService.pdb	03/08/2020 12:39	PDB File	54 KB	
👝 Data (K:)	srtp.pdb	21/05/2020 09:59	PDB File	1,124 KB	
<b>D</b>	ssleay32.dll	03/05/2016 17:44	Application extens	267 KB	
🔜 Data (K:) 🗸 🗸	Version 834	03/08/2020 12:39	TXT File	11 KR	
3 items   1 item selecte	d 23.9 KB				B==

Solution & Interoperability Test Lab Application Notes ©2021 Avaya Inc. All Rights Reserved. In the config file, it is the **SoftIpRecorder** section that is configured for SIP.

The following in red are the important bits to observe.

- <add key="Enabled" value="Y"/> turns on the Soft IP Recorder.
- <add key="SipPort1" value="5060"/> Specifies the SIP port number.
- <add key="NetworkInterface1" value="10.10.40.121"/> IP address or name of local NIC that the voice data is port mirrored to.
- <add key="IsTcpSipEnabled" value="Y"/> Enables TCP SIP recording.
- <add key="IsUdpSipEnabled" value="N"/> Enables UDP SIP recording.

<SoftIpRecorder>

```
<add key="Enabled" value="Y"/>
<add key="DisableJitterBuffer" value="N"/>
<add key="DisableTimeStampResetting" value="N"/>
<add key="RawRtpRecording" value="N"/>
<!-- This adds silence to SoftIP Mitel recording -->
<add key="AddSilentPadding" value="Y"/>
<add key="AddSilentPadding" value="Y"/>
<add key="IsAccurateTimeoutEnabled" value="Y"/>
<add key="LongTimeout" value="100"/>
<add key="ShortTimeout" value="3"/>
<add key="ReplayCaptureFile" value=""/>
<add key="ReplayCaptureFile" value=""/>
<add key="IsReplayCaptureFile" value="c:/temp2"/>
<add key="LongTimeout" value="100" />
<add key="CustomDtmfPayload" value="100" />
<add key="StereoRecording" value="N"/>
```

```
<!-- If mixing to Mono - use alternative mixing algorythmn-->
<add key="AlternativeMixing" value="Y" />
<add key="G72216KHZEnabled" value="Y" />
```

```
<!-- Used for SIP recording -->
<add key="PbxAddress" value=""/>
<add key="PbxAddressRegex" value=""/>
<add key="IpMappingFrom1" value=""/>
<add key="IpMappingTo1" value=""/>
<add key="SipPort1" value="5060"/>
<add key="SipPort2" value=""/>
<add key="SipPort3" value=""/>
<add key="SipPort3" value=""/>
<add key="SipPort4" value=""/>
<add key="SipPort5" value=""/>
<add key="SipPort6" value=""/><<add key="SipPort6" value=""/><<<add key="SipPort6" value=""/><<add key="SipPort6" value=""/><<<add key="SipPort6" value=""/><<<<add key="SipPort6" value=""/><<<<add key="SipPort6" value=""/><<<</a>
```

```
<!-- For SoftIP this is the nic device name, or ip address we are listening for RTP on -->
<add key="NetworkInterface1" value="10.10.40.121"/>
<add key="NetworkInterface2" value="-1"/>
<add key="NetworkInterface3" value="-1"/>
<add key="NetworkInterface4" value="-1"/>
<add key="NetworkInterface5" value="-1"/>
<add key="NetworkInterface5" value="-1"/>
<add key="NetworkInterface5" value="-1"/></add key="InterfaceKernelBuffer" value="10000000"/>
```

```
<!-- an example of a rather complicated rtp filter to filter rtp -->
<!--<add key="PcapFilter" value="udp[8] &gt;&gt; 6 == 0x02 &amp;&amp; length &lt; 250
&& (udp[9] & 0x7f == 0x0 || udp[9] & 0x7f == 0x8 || udp[9] & 0x7f == 0x9
|| udp[9] & 0x7f == 0xd || udp[9] & 0x7f == 0x67 || udp[9] & 0x7f == 0x68)" />-->
<add key="PcapFilter" value="" />
<add key="PcapFilter1" value="" />
```

```
<!-- Although SoftIP handlers can now be loaded dynamically, the following will load some defaults
<add key="IsTcpSipEnabled" value="Y"/>
<add key="IsUdpSipEnabled" value="N"/>
<add key="IsSkinnyEnabled" value="N"/>
<add key="IsLyncEnabled" value="N"/>
<add key="UseFullMacAddress" value="Y" />
<add key="UseIpForChannel" value="N" />
```

```
<!--<add key="MatchRule1" value="RtpRule"/>-->
<!--<add key="MatchRule2" value="IpAndPort"/>-->
<!--<add key="MatchRule3" value="IpAndNoDestinationPort"/>-->
<!--<add key="MatchRule4" value="IpAndSwapPort"/>-->
<!--<add key="MatchRule5" value="DestinationIpAndPort"/>-->
```

```
<!-- This is the softip engines load of extension. -->
<add key="LoadExtensionsFromDb" value="N"/>
<add key="CustomRtp1" value=""/>
```

```
<!-- We can silence pad using the timestamp. Only use if an IVR or other device is causing issues by sending no RTP when 'silent' --> <add key="TimeStampPadding" value="N"/>
```

</SoftIpRecorder>

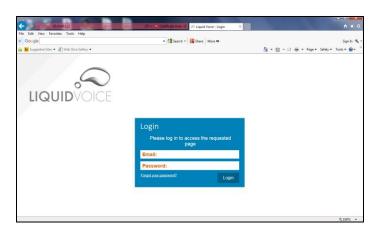
# 8. Verification Steps

The correct configuration of the solution can be verified as follows.

**Note**: Users can use any browser to access the interface and play back recordings. However, Assure uses Silverlight and so works best with Internet Explorer. If any other browser is used, the Liquid Voice Viewer will need to be downloaded. The Liquid Voice Viewer launches the Silverlight components out of browser enabling seamless use of the system in browsers that do not support Silverlight.

## 8.1. Verify Liquid Voice Assure

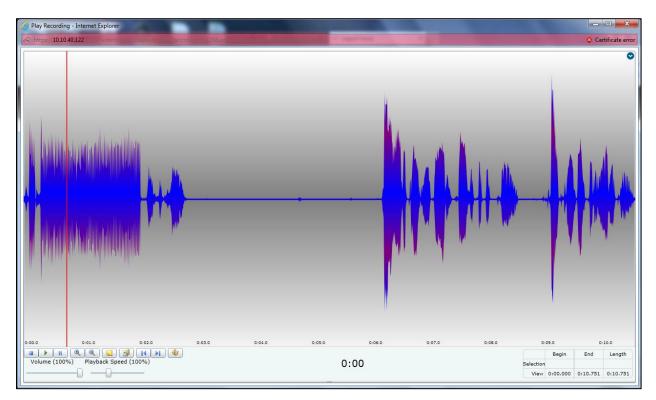
Open a URL to the Assure sever as shown below with Internet Explorer. Enter the appropriate credentials and click on **Login**.



A list of recordings is displayed and by clicking on **Play** at the beginning of any of the lines will open a new window containing the playback of that particular recording.

D 1 0 1 1 C		#									_		
Basic Search View		$\sim$	Reco	rdings 1	1	to 25 of 321 🍤						🛗 🖪 🖣 🕨	
	t≌i A×	Q		Re	ec Id	Date/Time	Duration	Extension	Taken By	Group	Call Direction	Phone Number	
older	- C	5		Play 18	80	22/10/2020 11:01:25	28				Incoming	35391847001	
Search Period		$\sim$		Play 17	79	22/10/2020 11:01:08	11	6701	Sales	Avaya	Incoming	35391847001	
From Date		<b></b>		Play 17	78	22/10/2020 10:56:58	44	6701	Sales	Avaya	Incoming	35391847001	
o Date				Play 17	78	22/10/2020 10:56:58	36	1100	Test Extension 1	Avaya	Incoming	35391847001	
ime Range	to			Play 17	7	22/10/2020 10:56:03	18	6701	Sales	Avaya	Incoming	35391847001	
Call Duration 0 m	0 s to 9999 m	0s		Play 17	7	22/10/2020 10:56:03	12	6701	Sales	Avaya	Incoming	35391847001	
aken By				Play 17	76	22/10/2020 10:55:07	6	1100	Test Extension 1	Avaya	Incoming	35391847001	
Group		_		Play 17	75	21/10/2020 11:49:30	1:09	1100	Test Extension 1	Avaya	Incoming	35391847001	
Extension		_		Play 17	'4	21/10/2020 11:34:46	39	1100	Test Extension 1	Avaya	Incoming	35391847001	
		_		Play 17	74	21/10/2020 11:34:46	39	1100	Test Extension 1	Avaya	Incoming	35391847001	
Phone Number		_		Play 17	'3	20/10/2020 17:43:08	36	1100	Test Extension 1	Avaya	Incoming	35391847001	
DDI/CLI out				Play 17	73	20/10/2020 17:43:08	35	1100	Test Extension 1	Avaya	Incoming	35391847001	
Call Direction		$\sim$		Play 17	2	20/10/2020 17:38:37	20	1100	Test Extension 1	Avaya	Incoming	35391847001	
Recording Id				Play 17	2	20/10/2020 17:38:37	20	1100	Test Extension 1	Avaya	Incoming	35391847001	
Audio Tags Notes				Play 17	71	20/10/2020 17:22:31	27	1101			Incoming	35391847001	
tecording Tags		•••		Play 17	70	20/10/2020 17:21:59	17	1100	Test Extension 1	Avaya	Incoming	35391847001	
rchived				Play 17	70	20/10/2020 17:21:59	17	1100	Test Extension 1	Avaya	Incoming	35391847001	
				Play 16	39	20/10/2020 16:40:38	11	1100	Test Extension 1	Avaya	Incoming	35391847001	

Solution & Interoperability Test Lab Application Notes ©2021 Avaya Inc. All Rights Reserved. 23 of 36 LVoice\_AACC71 The following window shows the playing back of the recording that was chosen from the screen on the previous page.



## 8.2. Verify CDR from Avaya Aura® Communication Manager

The **status cdr-link** command can be used to display the status of the CDR link from Communication Manager. The Primary **Link State** is shown as **up** below, which is good.

```
      CDR LINK STATUS

      Primary
      Secondary

      Link State: up
      CDR not administered

      Date & Time: 2020/10/16 14:33:53
      0000/000 00:00:00

      Forward Seq. No: 0
      0

      Backward Seq. No: 0
      0

      CDR Buffer % Full: 0.00
      0.00

      Reason Code: OK
      0K
```

## 8.3. Verify Avaya Aura® Contact Center

From the Contact Center server, open the **System Control and Monitor Utility**. The following window should be displayed showing all applications as **Started**.

10	System Control and Monitor Utility
avay	Contact Center System Control and Monitor Utility
	M CCMS CCMA CCT CCMM wn the selected Contact Center applications Start Shut down
	Contact Center Contact Center act Center applications to control
Manager	License Manager status: Started CCMS status: Started
Security Framework	Not installed
CCMA	CCMA status: Started
✓ ССТ	CCT status: Started
ССММ	CCMM status: Started
	Help View log Close

# 9. Conclusion

These Application Notes describe the compliance testing of Liquid Voice Assure Interaction Recording R7.5 with Avaya Aura® Contact Center R7.1 and Avaya Aura® Communication Manager R8.1. All test cases were executed successfully with any observations noted in **Section 2.2**.

# 10. Additional References

This section references the product documentations 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, Release 8.1
- [2] Avaya Aura® Communication Manager Feature Description and Implementation, Release 8.1
- [3] Avaya Aura® Application Enablement Services Administration and Maintenance Guide, Release 8.1
- [4] Administering Avaya Aura® Session Manager, Release 8.1
- [5] Deploying Avaya Aura® Contact Center DVD for Avaya Aura® Unified Communications Release 7.1 Issue 02.04 October 2020
- [6] Avaya Aura® Contact Center commissioning for Avaya Aura® Unified Communications Release 7.1 Issue 02.04 December 2019
- [7] Avaya Aura® Contact Center Server Administration Release 7.1 Issue 07.05 October 2020
- [8] Administering Avaya Session Border Controller for Enterprise, Release 8.1.x, Issue 3, August 2020

Product documentation for Assure can be found by contacting Liquid Voice as per Section 2.3.

# 11. Appendix A

To ensure a workaround is in place for the issue described in **Section 2.2** with the 9641G SIP phone, the 46xxsettings file for this SIP phone must be altered as shown below. Use the command **SET ENABLE\_G722 0** to disable the use of G.722 from the phoneset in question.

```
##
## ENABLE_G722 specifies whether the G.722 codec is enabled.
## Value Operation
## 0 Disabled
## 1 Enabled
## This parameter is supported by:
## 96x1 SIP R6.2 and later; the default value is 1.
## 96x1 SIP R6.0.x; the default value is 0.
## 96x0 SIP R2.0 and later; the default value is
## Hlxx SIP R1.0 and later; the default value is 1.
SET ENABLE_G722 0
##
```

# 12. Appendix B

This section illustrates the Contact Center user setup that was used during compliance testing and how that user is added for CCT.

## 12.1. Add a Windows Domain User

On most sites running Contact Center, a domain will have been configured with an Active Directory containing windows users. A 'Domain Administrator' will be on hand to provide windows users information to configure the CCT user on Contact Center.

For compliance testing a domain was already in place with users previously added to this domain specifically for this solution test. To add or display users, open Computer Management and select Active Directory Users and Computers (not shown). The following window is opened where new users are added by right-clicking on Users and selecting New  $\rightarrow$  User as shown below.

Active Directory Users and Computers							
File Action Vie	File Action View Help						
← → 2 元 ¼ □ X □ Q → 2 3							
-	Active Directory Users and Com Name Type						
> 🧾 Saved Queri V 🏥 devcor	Delegate Control						
> 🖬 724	Find						
> 📔 Bui	New	>	Computer				
> 🖬 Dor	All Tasks	>	Contact				
> 🤗 For > 🧖 Ma	View	>	Group				
> 🖬 Oce	Refresh		InetOrgPerson msDS-KeyCredential				
📔 Use	Export List		msDS-ResourcePropertyList				
	Properties		msDS-ShadowPrincipalContainer				
	Help		msImaging-PSPs				
bject.		_	MSMQ Queue Alias				
			Printer				
			User				
			Shared Folder				

Enter the details as shown in the example below for **aacc1** and click on **Next**.

New Object - User		$\times$
Create in:	devconnect.local/Users	
First name:	Aspire Initials:	
Last name:	Agent one	
Full name:	Aspire Agent one	
User logon name:		
aacc1	@devconnect.local ~	
User logon name (pre-	-Windows 2000):	
DEVCONNECT\	aacc1	
	< Back Next > Cance	4

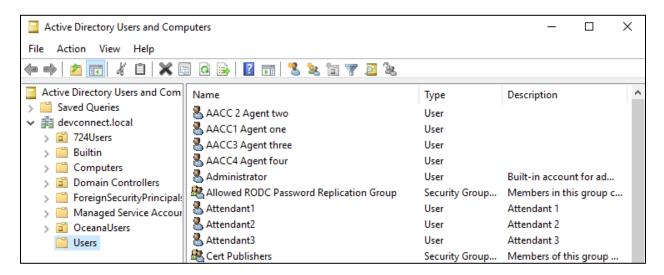
Enter the required **Password** and click on **Next** to continue.

New Object - User	×
Create in: devconnect.local/Users	
Password: Confirm password:	
<ul> <li>User must change password at next logon</li> <li>User cannot change password</li> <li>Password never expires</li> <li>Account is disabled</li> </ul>	
< Back Next > Cancel	

Solution & Interoperability Test Lab Application Notes ©2021 Avaya Inc. All Rights Reserved. Click on **Finish** to complete the addition of the new user that will be used for CCT.

New Object - User       X         Image: Aspire Agent one       Image: Aspire Agent one         User logon name: aacc 1@devconnect.local       Image: Aspire Agent one         The password never expires.       Image: Agent one         Image: Agent one       Image: Agent one         User logon name: aacc 1@devconnect.local       Image: Agent one         Image: Agent one       Image: Agent one         Image: Agent one       Image: Agent one         User logon name: aacc 1@devconnect.local       Image: Agent one         Image: Agent one       Image: Agent one         Image: Agent one       Image: Agent one         User logon name: aacc 1@devconnect.local       Image: Agent one         Image: Agent one       Image: Agent one		
When you click Finish, the following object will be created:         Full name: Aspire Agent one         User logon name: aacc 1@devconnect.local         The password never expires.	New Object - User	×
Full name: Aspire Agent one User logon name: aacc 1@devconnect.local The password never expires.	Create in: devconnect.local/Users	
User logon name: aacc 1@devconnect.local The password never expires.	When you click Finish, the following object will be created:	
The password never expires.	Full name: Aspire Agent one	~
	User logon name: aacc 1@devconnect.local	
< Back Finish Cancel	The password never expires.	
< Back Finish Cancel		~
< Back Finish Cancel		
	< Back Finish	Cancel

Four users were added specifically for compliance testing, these are **AACC1**, **AACC2**, **AACC3** and **AACC4**. These were added as there were four different phone set types used for the agent phones.



#### 12.2. Skillset, Route Point and Call Presentation Information

Log into Contact Center as per Section 6.3. From the Launchpad click on Configuration.

Launch	ıpad		
000000000000000000000000000000000000000	Contact Center Management Access and Partition Management Real-Time Reporting Historical Reporting Call Recording and Quality Monitoring Prompt Management	0 0 0	Configuration Scripting Emergency Help Outbound Multimedia

Expend the AACC server in the left window and click on **Skillsets**. The **Skillset Name**s are highlighted, and these will be assigned in **Section 12.3**.

VAYA	Logged in user: weba	admin   Change Pa	ssword					
ver Download Status Launc	hpad Help							
AACC70VMPG								
	Skillsets						Server: AA	CC70VMP
Blending Configuration								
Call Presentation Classes	Contact Type	Prefix	Skillset Name	Default Activity Code	Threshold Class	Call Age Preference	Out Of Service Mode	Target Ser
Call Recording and Quality Monitoring	POM_Outbound	PO_	Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	0
CDNs (Route Points)	Social_Networking	SN_	Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Contact Types	Voice_Mail		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
DNISs	SMS		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
E Formulas	Fax		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Global Settings	Scanned_Document		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Historical Statistics	OpenQ		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Media Servers	IM	IM_	Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Media Services and Routes	Video		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Multiplicity Presentation Classes	Outbound		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Real-time Statistics	Web_Communications		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Routes	EMail		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Skillsets	Voice		Default_Skillset	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	
Threshold Classes	Voice		Sales	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	0
CCT70vmpg	Voice		Support	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	0
CCMM70vmpg	POM_Outbound	PO_	POM_OUT	00, Skillset_Default_Activity_Code	Skillset_Template	First In Queue	N/A	0
POM3vmpg	Voice		Humanresources	00, Skillset Default Activity Code	Skillset Template	First In Queue	N/A	0

Click on **CDNs** (**Route Points**) to display the numbers that are to be dialed to reach the required skillsets.

AVAYA		Config	guration		Logged	in user: <b>webadmi</b> r
Server Download Status Lau	nchpad Help					
COMPG     Activity Codes     Blending Configuration     Call Presentation Classes	CDNs (Route Points)				Acquire All CDNs	De-acquire All CDNs
Call Recording and Quality Monitoring	Name	Number	URI	Call Type	Acquired?	Status
CDNs (Route Points)	Sales6500	6500	sip:6500@devconnect.local	Local		Acquired
Contact Types	Support6501	6501	sip:6501@devconnect.local	Local	<u>र</u>	Acquired
DNISs	HR6502	6502	sip:6502@devconnect.local	Local	<b>v</b>	Acquired
Formulas Global Settings Historical Statistics	*					

Solution & Interoperability Test Lab Application Notes ©2021 Avaya Inc. All Rights Reserved. Click on **Call Presentation Classes** to show calls are presented to the agents. This can be used to force the call to the agent or to let the phone ring at the agents set or a set amount of time before returning the call to the queue. This Call Presentation Class will be assigned to the agent in **Section 12.3**.

Call Presentation Classes     Call Presentation Classes     Call Recording and Quality Monitorin     Colls (Route Points)     Contact Types     ONISS	C	all Pres	sentation	Classe	S		Serve	er: CCM	S1
Formulas Global Settings						1			
Historical Statistics		Name	Presentation Option	Call Force Delay Timer	Return To Queue After N Seconds	After Return to Queue, Make Phoneset	After Call, Break for N seconds	Prompt On Answer	Pro Sile
Media Servers		Call_Centre_Admini:	Return To Queue	0	18	Not Ready	0	None	0
Media Services and Routes		APD	Return To Queue	0	30	Not Ready	5	None	0
Multiplicity Presentation Classes	*								
Real-time Statistics			1	1		1			
Routes									
Skillsets									
Threshold Classes									
CCT1									
б ссмм1									

## **12.3. Contact Center Agent information**

From the Launchpad, click on Contact Center Management.

Launch	npad		
0	Contact Center Management	0	Configuration
0	Access and Partition Management	0	Scripting
Ô	Real-Time Reporting	Ô	Emergency Help
0	Historical Reporting	0	Outbound
0	Call Recording and Quality Monitoring	Ô	Multimedia
0	Prompt Management		

An existing agent can be associated with the domain user created in **Section 12.1**. Right-click on the desired user and select **View Agent Details**.

AVA	yΑ			Cont	act Center Management
View/Edit	Add	Status	Launchpad	Help	
і́—́⊚ ссм і́—́ё́	S1 Superviso			Agent:	
	Agen	t Default t four AACC4			Skillset Name
÷	📲 Agen	t one AACCAL t three AAC	View Agent D	etails	
+	🎳 Agen	t two AACC	Delete Agent		
			Create Copy		
			Add Many Us	ers	

Tick the **Create CCT Agent** and expand the **Associate User Account** and selecting the **Search domain users**. Enter the domain user created in **Section 12.1** and click on list all (not shown). The **Login ID** and **Voice URI** were set to that of the Communication Manager extension that is associated with this agent.

Agent Details:	AACC1 Agent one			
First Name: Last Name: Title: Department:	AACC1 Agent one CCT Agent Sales	User Type: Login ID: Voice URI: IM URI:	Agent * 1050 sip:1050@devconnect.local sip:	0
Language:	English V	Account Type	:	
Comment:	$\bigcirc$	✓ Create C	CT Agent	
		CCT Ag	ent Login Details 🛛 🕕	
		Domain User ID:	DEVCONNECT aacc1	
_ ○ Sea	User Account rch local operating system O Search local security server	Search domain us	ers	
□ □ De	omain Details	Domain Account —		
Se	erver Name or IP * 10.10.40.5	(Domain\User ID) *	DEVCONNECT\aacc1	
	Specify Domain Account	Password *	•••••	*
	g speery bonnan secoure	Base DN		
		Port Number	Use Secure Connection	
Sear	ch all user accounts where:		A stir	ato Mindous

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Search local operating						
Domain Details			Domain Account			
Server Name or IP * 1	0.10.40.5		(Domain\User ID)	DEVCONNECT\aao	c1	
_			Password *	•••••		
Specify Domain Acc	count		Base DN			
			Port Number			
				Use Secure Con	nection	
earch all user accounts w Last Name 🗸 starts		and includes	all users	✓ All		
Last Name 🗸 starts	s with		Search List	All	Descrin	tion
					Descrip	
Last Name V starts	Last Name (24)	<ul> <li>Fir</li> </ul>	Search List	All	-	tion
Last Name V starts User Name O tagent2	Last Name (24)	Test	Search List /	All <u>Status</u> Available	(j)	
Last Name V starts User Name C tagent2 C tagent3	Last Name (24) Agent2 Agent3	Test	Search List /	All <u>Status</u> Available Available	it. it.	
Last Name Starts User Name Cagent2 Cagent3 Cag	Last Name (24) Agent2 Agent3 Agent one	<ul> <li>Fir</li> <li>Test</li> <li>AACC1</li> </ul>	Search List /	All Status Available Available Assigned	U U U	
Last Name Starts User Name tagent2 acc1 acc2	Last Name (24) × Agent2 Agent3 Agent one Agent two	<ul> <li>Fin</li> <li>Test</li> <li>AACC1</li> <li>AACC2</li> </ul>	Search List /	All Status Available Available Assigned Available	(1) (1) (1) (1)	

Other information such as the **Call Presentation** (described in **Section 12.2**) and assigned **Skillsets** are displayed and can be changed. Once all is complete, click on **Submit**.

rimary Supervisor: * Supervisor Default 🗸		Call Presentation:	APD
ogin status Logged Out		Multiplicity Presentation C	
		Threshold:	Agent_Template 🗸
Contact Types			
Contact Type			
SMS			
Social_Networking			
Video			
Voice		✓	
Voice_Mail			
Web_Communications		¥	
Default_Skillset	Voice	Standby 🗸	
EM_Default_Skillset	EMail	10 🗸	
Humanresources	Voice	5 🗸	
OQ_Default_Skillset	OpenQ	10 🗸	
Sales	Voice	1 🗸	
Support	Voice	20 🗸	
WC_Default_Skillset	Web_Communications	10 🗸	
Assign Skillsets			
witing			
artitions			

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