



**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 – 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 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® 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 Record (CDR) port on Communication Manager to obtain call events to help provide the user with as much information as possible about each call recording.

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 Communication Manager environment. Communication Manager users logged into various Avaya endpoints (outlined in **Section 4**) were used to make and receive calls and by mirroring the external port of the SBCE all RTP was received on 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 Communication Manager CDR is used to obtain information on each recording such as CLID, DNIS and call duration, to provide the end user with additional information. Recordings in a Contact Center environment is also possible, see *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*.

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.
- **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.

## 2.3. Support

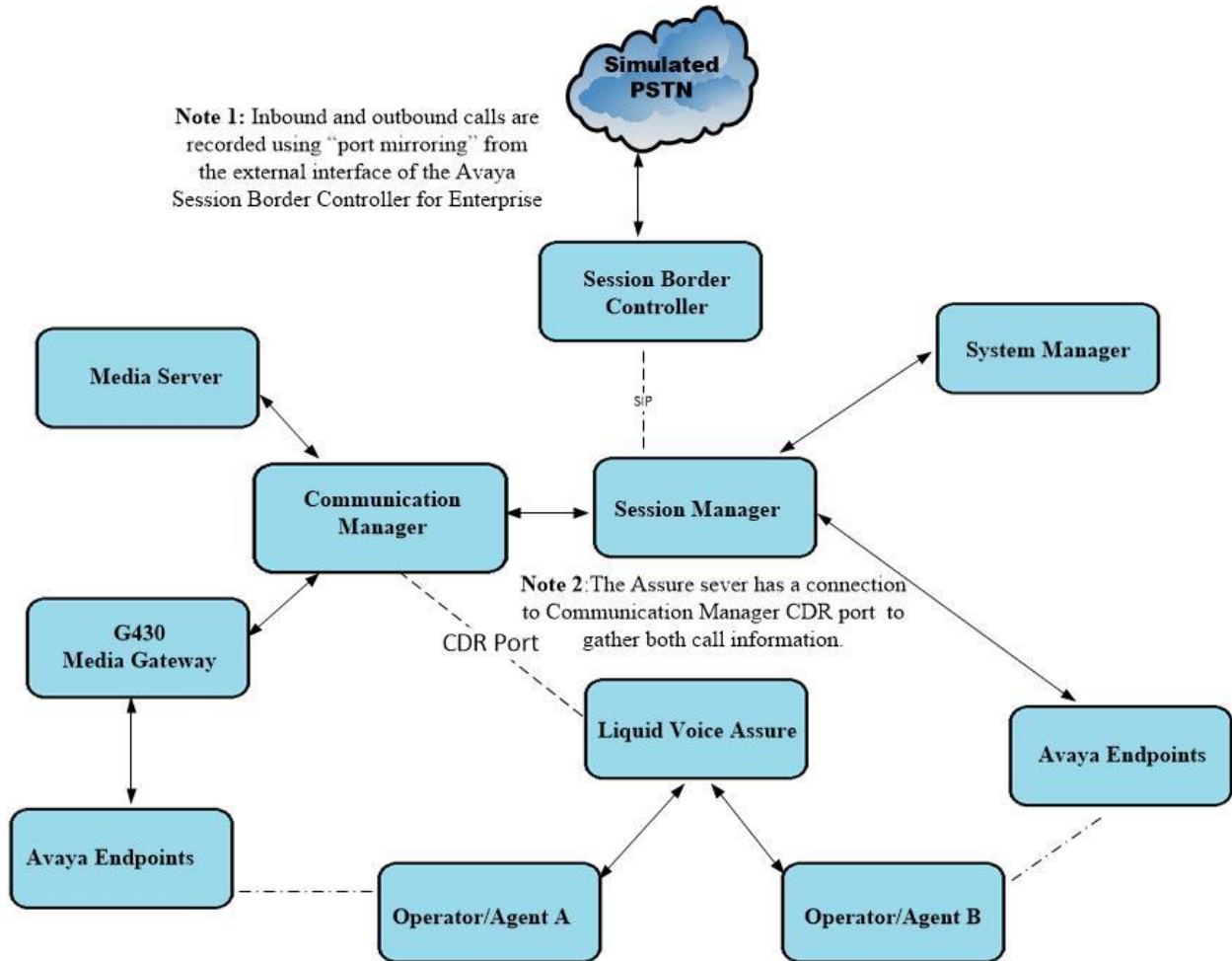
Technical support can be obtained for Liquid Assure from:

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

### 3. Reference Configuration

**Figure 1** below shows Avaya Aura® Communication Manager serving Digital, H.323 and SIP endpoints. Assure has a connection to Communication Manager to obtain Call Details, 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**

## 4. Equipment and Software Validated

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

<b>Avaya Equipment/Software</b>	<b>Release/Version</b>
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 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
<b>Liquid Voice Equipment/Software</b>	<b>Release/Version</b>
Liquid Voice Assure Interaction Recording	
- Interface version	7.5.1
- Recording Service version	8.3.4
- CDR Network Service	7.1

## 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 9**.

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                 0.0.0.0
g430                    10.10.40.15
procr                  10.10.40.37
procr6                  ::
sm81xvmpg              10.10.40.32

( 11 of 11 administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name '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-services Page 1 of 3
```

IP SERVICES					
Service Type	Enabled	Local Node	Local Port	Remote Node	Remote Port
CDR1		procr	0	LVoice	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	<b>Primary Output Endpoint: CDR1</b>
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	<b>Outg Trk Call Splitting? y</b>
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
<b>Inc Trk Call Splitting? y</b>	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.

change system-parameters cdr			<b>Page 2 of 2</b>		
CDR SYSTEM PARAMETERS					
Data Item - Length		Data Item - Length		Data Item - Length	
1: date	- 6	17: in-trk-code	- 4	33: line-feed	- 1
2: space	- 1	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	25: isdn-cc	- 11	41:	-
10: space	- 1	26: space	- 1	42:	-
11: code-used	- 4	27: attd-console	- 2	43:	-
12: space	- 1	28: space	- 1	44:	-
13: dialed-num	- 18	29: vdn	- 5	45:	-
14: space	- 1	30: space	- 1	46:	-
15: clg-num/in-tac	- 10	31: acct-code	- 15	47:	-
16: space	- 1	32: return	- 1	48:	-
Record length = 117					



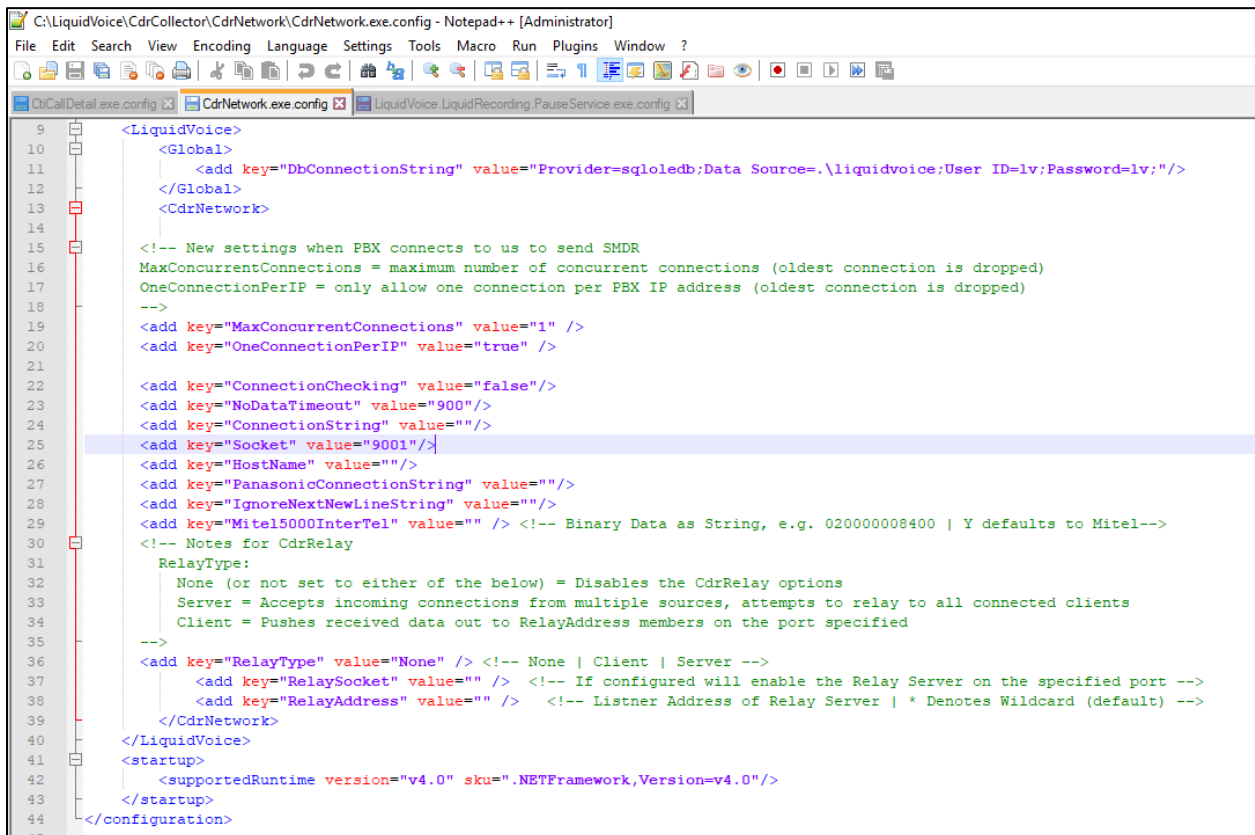
## 6. Configure Liquid Voice Assure

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

1. Connection to Avaya Aura® Communication Manager to collect CDR data.
2. Connection to Avaya Session Border Controller for Enterprise (ASBCE) via port mirror of the external interface on the ASBCE, to record all trunk calls.

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

A configuration file is used to connect to Communication Manager. The location of the file is displayed at the top of the screen shot below.



```
C:\LiquidVoice\CdrCollector\CdrNetwork\CdrNetwork.exe.config - Notepad++ [Administrator]
File Edit Search View Encoding Language Settings Tools Macro Run Plugins Window ?
C:\CallDetail.exe.config CdrNetwork.exe.config LiquidVoice.LiquidRecording.PauseService.exe.config
9 <LiquidVoice>
10 <Global>
11 <add key="DbConnectionString" value="Provider=sqloledb;Data Source=.\liquidvoice;User ID=lv;Password=lv;"/>
12 </Global>
13 <CdrNetwork>
14
15 <!-- New settings when PBX connects to us to send SMDR
16 MaxConcurrentConnections = maximum number of concurrent connections (oldest connection is dropped)
17 OneConnectionPerIP = only allow one connection per PBX IP address (oldest connection is dropped)
18 -->
19 <add key="MaxConcurrentConnections" value="1" />
20 <add key="OneConnectionPerIP" value="true" />
21
22 <add key="ConnectionChecking" value="false"/>
23 <add key="NoDataTimeout" value="900"/>
24 <add key="ConnectionString" value=""/>
25 <add key="Socket" value="9001"/>
26 <add key="HostName" value=""/>
27 <add key="PanasonicConnectionString" value=""/>
28 <add key="IgnoreNextNewLineString" value=""/>
29 <add key="Mitel5000InterTel" value="" /> <!-- Binary Data as String, e.g. 020000008400 | Y defaults to Mitel-->
30 <!-- Notes for CdrRelay
31 RelayType:
32 None (or not set to either of the below) = Disables the CdrRelay options
33 Server = Accepts incoming connections from multiple sources, attempts to relay to all connected clients
34 Client = Pushes received data out to RelayAddress members on the port specified
35 -->
36 <add key="RelayType" value="None" /> <!-- None | Client | Server -->
37 <add key="RelaySocket" value="" /> <!-- If configured will enable the Relay Server on the specified port -->
38 <add key="RelayAddress" value="" /> <!-- Listener Address of Relay Server | * Denotes Wildcard (default) -->
39 </CdrNetwork>
40 </LiquidVoice>
41 <startup>
42 <supportedRuntime version="v4.0" sku=".NETFramework,Version=v4.0"/>
43 </startup>
44 </configuration>
```

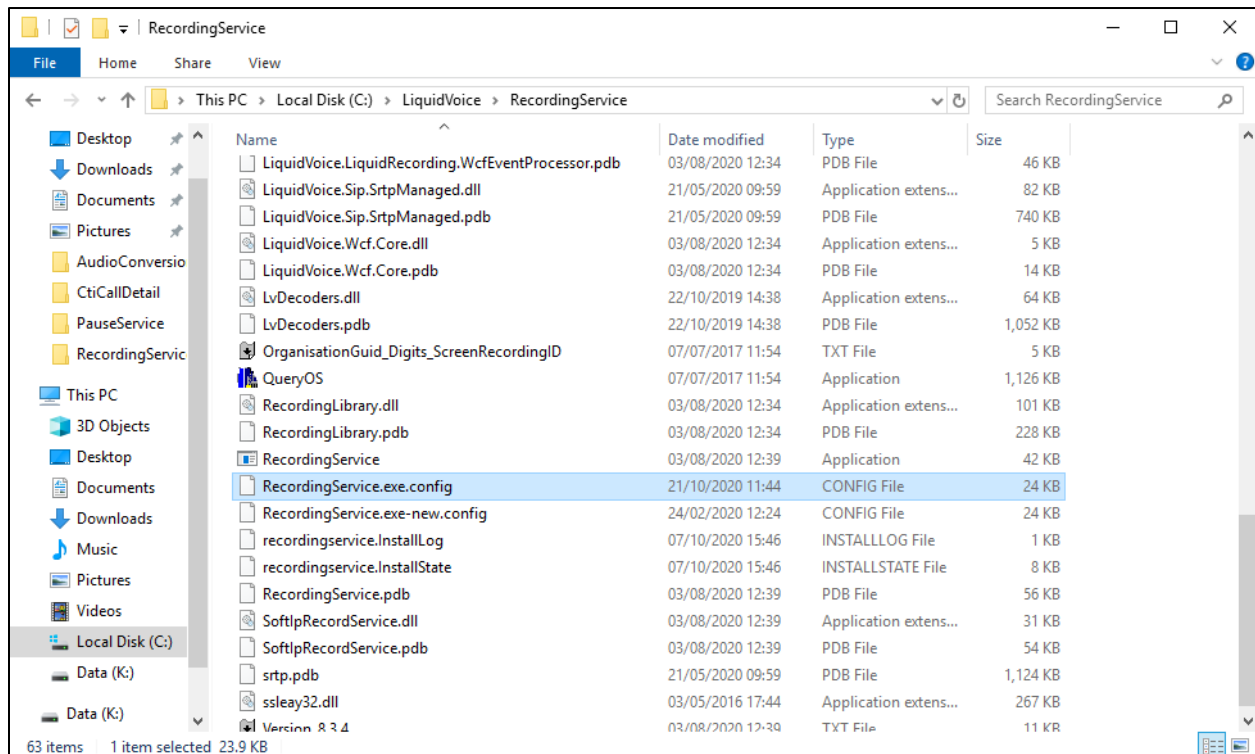
A closer look at the entry for the Socket will show the same port that was used in **Section 5**. Assume listen on port **9001** for the CDR output. If a host name or IP address is added to the **HostName** key the CDR service will connect to Communication Manager.

```
<add key="MaxConcurrentConnections" value="1" />
<add key="OneConnectionPerIP" value="true" />

<add key="ConnectionChecking" value="false"/>
<add key="NoDataTimeout" value="900"/>
<add key="ConnectionString" value=""/>
<add key="Socket" value="9001"/>
<add key="HostName" value=""/>
<add key="PanasonicConnectionString" value=""/>
<add key="IgnoreNextNewLineString" value=""/>
<add key="Mitel5000InterTel" value="" /> <!-- Binary Data
<!-- Notes for CdrRelay
```

## 6.2. 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**.



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="IsAccurateTimeoutEnabled" value="Y" />
<add key="LongTimeout" value="100"/>
<add key="ShortTimeout" value="3"/>
<add key="ReplayCaptureFile" value=""/>
<add key="ReplayCaptureFile" value="c:/temp2"/>
<add key="IsReplayCaptureFileProcessedRealtime" value="Y"/>
<add key="CustomDtmfPayload" value="100" />
<add key="StereoRecording" value="N"/>

<!-- If mixing to Mono - use alternative mixing algorithmn-->
<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="SipPort4" value=""/>
<add key="SipPort5" value=""/>
<add key="SipPort6" value=""/>
<add key="CustomSipHeader1" value="VIA" />
<!--<add key="CustomSipHeader1" value="X-Asterisk-GUID"/>-->
```

```

<!-- 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="InterfaceKernelBuffer" value="100000000"/>

<!-- 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
&amp;&amp; (udp[9] &amp; 0x7f == 0x0 || udp[9] &amp; 0x7f == 0x8 || udp[9] &amp; 0x7f == 0x9
|| udp[9] &amp; 0x7f == 0xd || udp[9] &amp; 0x7f == 0x67 || udp[9] &amp; 0x7f == 0x68)" />-->
<add key="PcapFilter" value="" />
<add key="PcapFilter1" value="" />
<add key="PcapFilter3" 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>

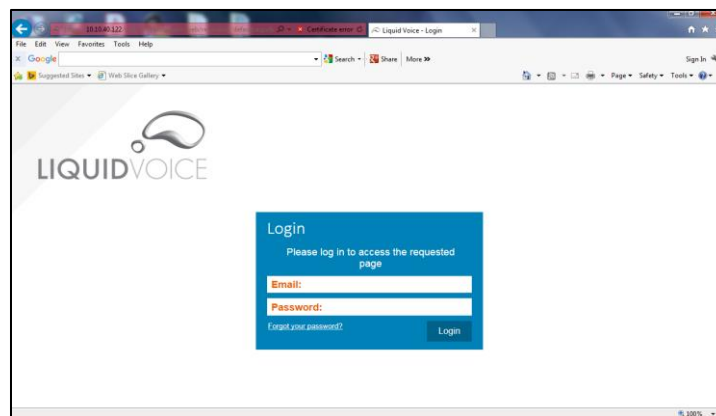
## 7. Verification Steps

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

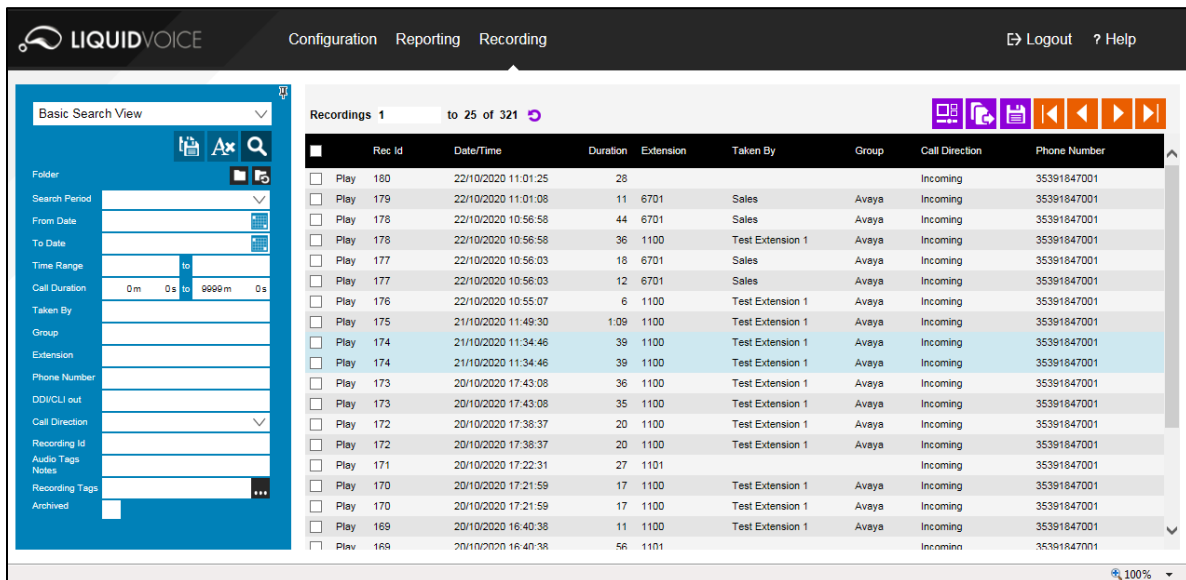
**Note:** Users can use any browser to access the interface and play back recordings. However, Assure uses Silverlight and so works best with Internet Explorer. If any other browser is used, the Liquid Voice Viewer will need to be downloaded. The Liquid Voice Viewer launches the Silverlight components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

### 7.1. 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**.

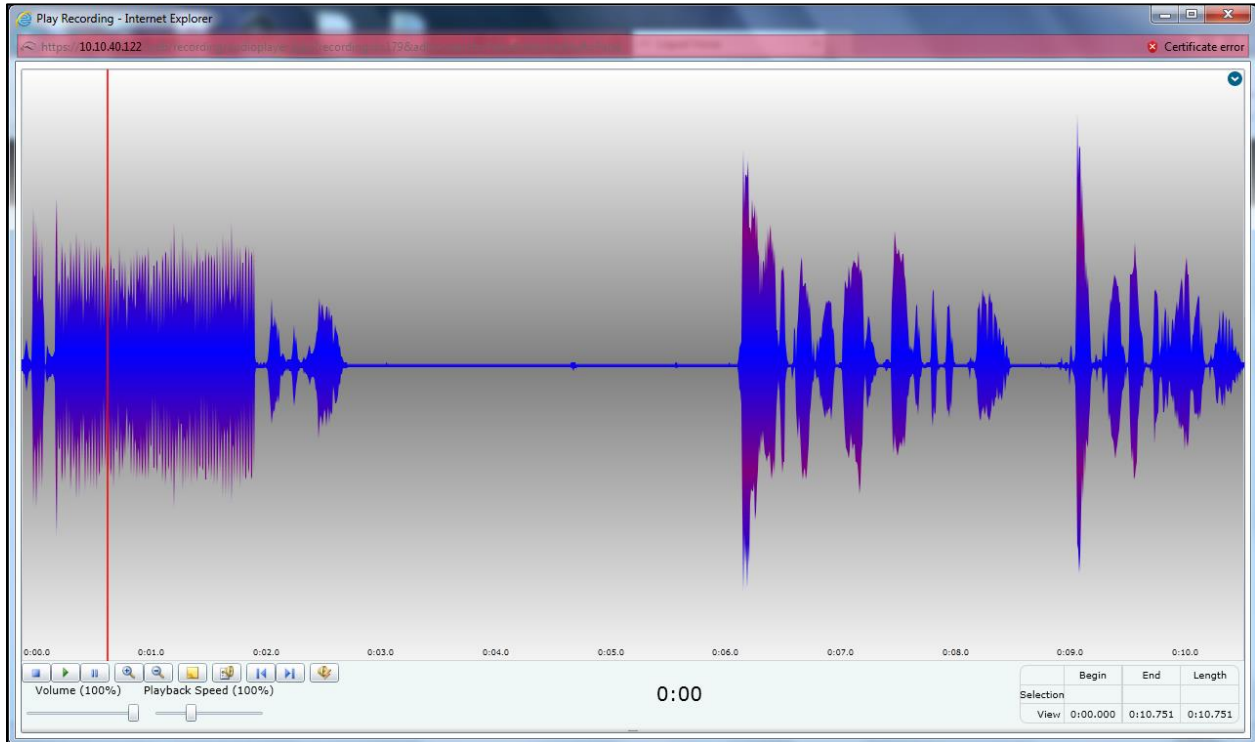


A list of recordings is displayed and by clicking on **Play** at the beginning of any of the lines will open a new window containing the playback of that particular recording.



Rec Id	Date/Time	Duration	Extension	Taken By	Group	Call Direction	Phone Number
180	22/10/2020 11:01:25	28				Incoming	35391847001
179	22/10/2020 11:01:08	11	6701	Sales	Avaya	Incoming	35391847001
178	22/10/2020 10:56:58	44	6701	Sales	Avaya	Incoming	35391847001
178	22/10/2020 10:56:58	36	1100	Test Extension 1	Avaya	Incoming	35391847001
177	22/10/2020 10:56:03	18	6701	Sales	Avaya	Incoming	35391847001
177	22/10/2020 10:56:03	12	6701	Sales	Avaya	Incoming	35391847001
176	22/10/2020 10:55:07	6	1100	Test Extension 1	Avaya	Incoming	35391847001
175	21/10/2020 11:49:30	1:09	1100	Test Extension 1	Avaya	Incoming	35391847001
174	21/10/2020 11:34:46	39	1100	Test Extension 1	Avaya	Incoming	35391847001
174	21/10/2020 11:34:46	39	1100	Test Extension 1	Avaya	Incoming	35391847001
173	20/10/2020 17:43:08	36	1100	Test Extension 1	Avaya	Incoming	35391847001
173	20/10/2020 17:43:08	35	1100	Test Extension 1	Avaya	Incoming	35391847001
172	20/10/2020 17:38:37	20	1100	Test Extension 1	Avaya	Incoming	35391847001
172	20/10/2020 17:38:37	20	1100	Test Extension 1	Avaya	Incoming	35391847001
171	20/10/2020 17:22:31	27	1101			Incoming	35391847001
170	20/10/2020 17:21:59	17	1100	Test Extension 1	Avaya	Incoming	35391847001
170	20/10/2020 17:21:59	17	1100	Test Extension 1	Avaya	Incoming	35391847001
169	20/10/2020 16:40:38	11	1100	Test Extension 1	Avaya	Incoming	35391847001
168	20/10/2020 16:40:38	56	1101			Incoming	35391847001

The following window shows the playing back of the recording that was chosen from the screen on the previous page.



## 7.2. 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.

```
status cdr-link
```

	Primary	CDR LINK STATUS	Secondary
Link State:	up		CDR not administered
Date & Time:	2020/10/16 14:33:53		0000/00/00 00:00:00
Forward Seq. No:	0		0
Backward Seq. No:	0		0
CDR Buffer % Full:	0.00		0.00
Reason Code:	OK		

## 8. Conclusion

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

## 9. 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] *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**.

## 10. 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
##
```



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