

Avaya Solution & Interoperability Test Lab

Application Notes for VoxSpectrum DGVox Voice Logger with Avaya AuraTM Communication Manager Using Single Step Conference – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for VoxSpectrum DGVox Voice Logger to interoperate with Avaya AuraTM Communication Manager using Avaya AuraTM Application Enablement Services. VoxSpectrum DGVox Voice Logger is a call recording solution. In the compliance testing, VoxSpectrum DGVox Voice Logger used the Telephony Services Application Programming Interface from Avaya AuraTM Application Enablement Services to monitor stations on Avaya AuraTM Communication Manager, and used the single Step Conference feature via the Avaya AuraTM Application Enablement Services Device, Media, and Call Control interface to capture the media associated with the monitored stations for call recording.

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

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1. Introduction

These Application Notes describe the configuration used to enable the VoxSpectrum DGVox Voice Logger to interoperate with Avaya AuraTM Communication Manager and Avaya AuraTM Application Enablement Services. The DGVox Voice Logger voice recorder offers various methods of voice recording. For the purpose of the tests described by these Application Notes, the Avaya AuraTM Communication Manager Single Step Conference facility was used.

1.1. Interoperability Compliance Testing

The following tests were performed as part of the compliance testing:

- The following test scenarios were used to test the various DGVox Voice Logger features:
 - o Basic call
 - Hold/retrieve
 - o Transfer / Blind transfer
 - Conference
 - Hunt group calls
 - Calls to/from bridged appearances
- DGVox Voice Logger's robustness was tested by verifying its ability to recover from interruptions to its external connections including:
 - o The LAN connection between and the DGVox Voice Logger and the network
 - The connection of the PBX to the network
- DGVox Voice Logger's robustness was further tested by verifying ability to recover from power interruptions to the following components:
 - o The DGVox Voice Logger.
 - Avaya AuraTM Communication Manager to which the DGVox Voice Logger is attached.

1.2. Support

Support for DGVox Voice Logger is available at: support@voxspectrum.com.

2. Reference Configuration

The following diagram shows the configuration used for conformance testing.

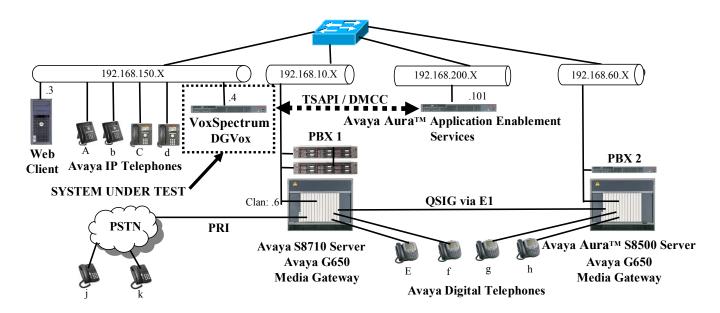


Figure 1: DGVox Voice Logger Test Configuration

In the above diagram, the DGVox Voice Logger records voice conversations from telephones attached to PBX 1. DGVox Voice Logger receives events from the Avaya Aura TM Application Enablement Services server when the state of calls associated with PBX 1 change, and informs the DGVox Voice Logger of these transitions. The DMCC service provided by Avaya Aura TM Application Enablement Services is used to monitor call activity associated with PBX 1. The DGVox Voice Logger is attached to PBX 1 via the local area network. PBX 2 is included in the configuration solely to test the ability to monitor conversations which traverse a trunk to a networked PBX.

The Web Client provides a web browser-based interface to administer the voice recording activity.

When a call is to be recorded, the DGVox Voice Logger initiates a single-step conference with the station being monitored using one of its virtual stations, and thus includes itself in the call which is to be recorded. The voice stream for such calls is received via the LAN interface to PBX 1.

The PBX 2 system is attached to PBX 1 via an IP/QSIG interface, and is used as a networked PBX system. This allows endpoints which are attached to a networked PBX (g, h) to be included in the test.

The telephones depicted in these Application Notes are designated by an upper case letter if configured to be monitored by the DGVox Voice Logger. A lower case letter designates those terminals which have been configured to not be monitored or are possibly unable to be monitored.

The following table contains additional information about each of the telephones shown in Figure 1. A "*" in the "Monitored" column indicated that the telephone is monitored by the DGVox Voice Logger. Note that one Virtual CTI Station is required for each conversation to be monitored, as these are used by DGVox Voice Logger to create single step conferences with the stations to be monitored. Since a Virtual CTI Station can be used to monitor only one call at a time, the number of virtual stations must be equal to the maximum number of simultaneous monitored calls. Note that calls between parties which are both monitored and conferences among monitored participants require additional virtual stations.

Phone	Monitored	Model	Extension
A	*	Avaya 1616	10071
b		Avaya 1608	10062
С	*	Avaya 9640G	10093
d		Avaya 9630G	10184
Е	*	Avaya 2410	10007
f		Avaya 2410	10008
g		Avaya 2420	60007
h		Avaya 2420	60008
j		N/A	069 xxxx 6176
k		N/A	069 xxxx 6630
X		CTI Station	11401
y		CTI Station	11402
Z		CTI Station	11403

Table 1: Device Monitor Configuration

3. Equipment and Software Validated

Software Component	Version
Avaya Aura TM Communication Manager	5.2.1 R015x.02.1.016.4 Patch: 17774
Avaya Aura TM Application Enablement Services	4.2.2
Avaya TN2312BP IP Server Interface	HW11/FW049
Avaya TN799DP Control LAN Interface	HW01/FW032
Avaya TN2302AP IP Media Processor	HW20/FW120
Avaya 1608 IP Telephone	3.0
Avaya 1616 IP Telephone	3.0
Avaya 9630G IP Telephone	3.02
Avaya 9640G IP Telephone	3.02
VoxSpectrum DGVox Voice Logger platform OS MS Windows XP Professional	SP3
VoxSpectrum DGVox Voice Logger	6.9

Table 2: Hardware/Software Component Versions

4. Configure Avaya Aura[™] Communication Manager

The configuration information in this section covers only PBX 1 – the system to which the DGVox Voice Logger is attached. The configuration and verification operations illustrated in this section were all performed using the Avaya Aura TM Communication Manager System Administration Terminal (SAT) via SSH port 5022.

The information provided in this section describes the configuration of Avaya AuraTM Communication Manager for this solution. For all other provisioning information such as installation and configuration, please refer to the product documentation in references [1] and [2].

The configuration of PSTN and QSIG trunks and associated switching is not described within this document.

4.1. Verify System Parameters Customer Options

Use the **display system-parameters customer options** command to verify that Avaya AuraTM Communication Manager is configured to meet the minimum requirements to run DGVox Voice Logger. Those items shown in **bold** indicate required values or minimum capacity requirements. If these are not met in the configuration, please contact an Avaya representative for further assistance.

Parameter	Usage
Maximum Concurrently Registered IP	This must be sufficient to support the total number of
Stations (p.2)	IP stations used.
IP Stations (p.4)	This parameter must be set to "y".
ID ADI A (p. 10)	This parameter must be set to the number of virtual
IP_API_A (p.10)	stations.
ID Dhana (n. 10)	This parameter must be set the number of IP stations
IP Phone (p.10)	plus the number of virtual stations.

Table 3: System-Parameters Customer-Options Parameters

display system-parameters customer-options OPTIONAL FEATURES		Page 2 of 11
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	1000	50
Maximum Concurrently Registered IP Stations:	18000	0 6
Maximum Administered Remote Office Trunks:	0	0
Maximum Concurrently Registered Remote Office Stations:	0	0
Maximum Concurrently Registered IP eCons:	0	0
Max Concur Registered Unauthenticated H.323 Stations:	0	0
Maximum Video Capable H.323 Stations:	0	0
Maximum Video Capable IP Softphones:	0	0
Maximum Administered SIP Trunks:	1000	30
Maximum Administered Ad-hoc Video Conferencing Ports:	0	0
Maximum Number of DS1 Boards with Echo Cancellation:	0	0
Maximum TN2501 VAL Boards:	10	1
Maximum Media Gateway VAL Sources:	0	0
Maximum TN2602 Boards with 80 VoIP Channels:	128	0
Maximum TN2602 Boards with 320 VoIP Channels:	128	0
Maximum Number of Expanded Meet-me Conference Ports:	0	0

Figure 2: System-Parameters Customer-Options Screen (p.2)

```
display system-parameters customer-options
                               OPTIONAL FEATURES
  Emergency Access to Attendant? y
                                                                IP Stations? y
          Enable 'dadmin' Login? y
          Enhanced Conferencing? y
                                                          ISDN Feature Plus? n
                                     ISDN/SIP Network Call Redirection? y
                Enhanced EC500? y
   Enterprise Survivable Server? n
                                                           ISDN-BRI Trunks? y
      Enterprise Wide Licensing? n
                                                                  ISDN-PRI? v
            ESS Administration? n
                                                 Local Survivable Processor? n
         Extended Cvg/Fwd Admin? n
                                                      Malicious Call Trace? n
    External Device Alarm Admin? n
                                                  Media Encryption Over IP? n
 Five Port Networks Max Per MCC? n
                                     Mode Code for Centralized Voice Mail? n
               Flexible Billing? n
  Forced Entry of Account Codes? n
                                                   Multifrequency Signaling? y
                                     Multimedia Call Handling (Basic)? n
Multimedia Call Handling (Enhanced)? n
     Global Call Classification? n
           Hospitality (Basic)? y
                                        Multimedia IP SIP Trunking? n
Hospitality (G3V3 Enhancements)? y
                      IP Trunks? y
          IP Attendant Consoles? n
```

Figure 3: System-Parameters Customer-Options Screen (p.4)

```
display system-parameters customer-options
                                                             Page 10 of 11
                   MAXIMUM IP REGISTRATIONS BY PRODUCT ID
Product ID Rel. Limit
                              Used
          : 1000
: 1000
IP API A
IP API B
                              0
             : 1000
IP API C
                              0
            : 1000
IP Agent
                              0
IP IR A
             : 1000
                              0
            : 18000
IP Phone
             : 18000
IP ROMax
                              Ω
            : 1000
IP Soft
IP_eCons
oneX Comm
            : 128
                              Ω
              : 18000
```

Figure 4: System-Parameters Customer-Options Screen (p.10)

4.2. Configure Dial Plan

Use the **change dialplan analysis** command to specify that dialed strings which begin with "1", or "6" are extensions. Include the strings "*" for Trunk Access Codes. Include the string "0" as a prefix for PSTN calls.

```
change dialplan analysis

DIAL PLAN ANALYSIS TABLE
Location: all Percent Full: 0

Dialed Total Call Dialed Total Call Dialed Total Call
String Length Type String Length Type String Length Type

1 fac
1 5 ext
6 5 ext
* 3 fac
```

Figure 5: Dialplan Analysis Screen

4.3. Configure IP Network Interface

Use the **change node-names ip** command to configure IP address, as shown in the following table.

Parameter	Usage
clan	Enter the IP address of the CLAN interface of PBX1.

Table 4: Node-Names IP Parameters

change change node-names ip

IP NODE NAMES

Name

IP Address

clan

192.168.10.6

default

0.0.0.0

Figure 6: Node-Names IP Screen

Use the **change ip-network-region** <**x**> command to designate a network region to be used for the IP telephone communications using the parameters shown in the following table, where <**x**> is the network region assigned to the clan IP interface.

Parameter	Usage
Name	Enter a name to identify the region.
Codec Set	Enter the number of the codec set defined in Figure 8 .

Table 5: IP-Network-Region Parameters

```
change ip-network-region 1
                                                                               Page
                                                                                       1 of 19
                                      IP NETWORK REGION
  Region: 1
Codec Set: 1 Intra-region IP-IP Direct Audio: yes
UDP Port Min: 2048 IP Audio Hairpinning? n

UDP Port Max: 3329

DIFFSERV/TOS PARAMETERS
Call Control PHB Value:
Location: 1
                    Authoritative Domain: ffm.com
 Call Control PHB Value: 46 RTCP MONITOR SERVER PARAMETERS
Audio PHB Value: 46 Use Default Server Parameters? y
         Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
          Audio 802.1p Priority: 6
         Video 802.1p Priority: 5
                                         AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS
                                                                     RSVP Enabled? n
  H.323 Link Bounce Recovery? y
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
              Keep-Alive Count: 5
```

Figure 7: IP-Network-Region Form, p.1

Use the **change ip-codec-set** command to designate a codec set to be used. Testing was done with the G.711A codec.

Parameter	Usage
Audio Codec	Enter "G.711A".

Table 6: IP-Codec-Set Parameters

```
change change ip-codec-set 1

IP Codec Set

Codec Set: 1

Audio Silence Frames Packet

Codec Suppression Per Pkt Size(ms)

1: G.711A n 2 20
```

Figure 8: IP-Codec-Set Form

4.4. Configure Interface to Avaya Aura[™] Application Enablement Services

Use the **change ip-services** command to configure the interface to Avaya AuraTM Application Enablement Services, as shown in the following table.

Parameter	Usage
Service Type (p.1)	Enter "AESVCS".
Enabled (p.1)	Enter "y" to enable the service.
Local Node (p.1)	Enter the IP node name for the CLAN interface.
Local Port (p.1)	Enter "8765".
AE Services Server (p.4)	Enter the name that was assigned to the Avaya Aura TM Application Enablement Services server when it was installed.
Password (p.4)	Enter the password that was assigned to the switch connection, as shown in Figure 17 .
Enabled (p.4)	Enter "y" to enable the connection.

Table 7: IP Services Parameters

change ip-s	services				Page	1 of	4		
Service Type AESVCS	Enabled Y cl	Local Node Lan	IP SERVICES Local Port 8765	Remote Node	Remote Port				

Figure 9: IP Services Screen, p. 1



Figure 10: IP Services Screen, p. 4

Use the **add cti-link** command to add a CTI link for use by TSAPI. The link number can be any value between 1 and 64 which is not currently assigned to another link. The link number specified must be the same value that is used in the "Switch CTI Link Number" field shown in **Figure 20**. Use an unused extension as the value for the "Extension" parameter. The value chosen for the "Name" parameter is a matter of personal preference.

add cti-link 4

CTI LINK

CTI Link: 4

Extension: 19996

Type: ADJ-IP

COR: 1

Name: AES-devcon223-tsapi

Figure 11: CTI-link Screen

4.5. Configure Stations

4.5.1. Configure H.323 IP Stations

Use the **add station** command to create an IP station for extensions A, b, C, and d in **Table 1**, using the values shown in the following table.

Parameter	Usage
Extension	Use an unused extension which is compatible with the dial plan.
Type	Use a type value which corresponds to the physical station to be used.
Name	Any alphanumeric string can be assigned as an extension name, which is used for identification purposes.
Security Code	Enter an appropriate numeric security code.

Table 8: Configuration IP Stations

```
add station 10071
                                                               Page 1 of
                                     STATION
                                                                      BCC: 0
Extension: 10071
                                        Lock Messages? n
                                       Lock Messages? n
Security Code: 123456
    Type: 1616
                                                                       TN: 1
                                      Coverage Path 1:
    Port: S00264
                                                                       COR: 1
    Name: extn 10071
                                      Coverage Path 2:
                                                                       cos: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
                                  Personalized Ringing Pattern: 1
             Loss Group: 19
       Speakerphone: 2-way
Display Language: english
able GK Node Name:
                                               Message Lamp Ext: 10071
                                             Mute Button Enabled? y
                                                  Button Modules: 0
Survivable GK Node Name:
         Survivable COR: internal
                                             Media Complex Ext:
  Survivable Trunk Dest? y
                                                    IP SoftPhone? n
                                              Customizable Labels? y
```

Figure 12: IP Station Screen

4.5.2. Configure Virtual Stations

Use the **add station** command to create a station for each of the virtual stations listed in **Table 1**. Sufficient virtual stations must be created to monitor the maximum number of simultaneous monitored conversations. Note that the DGVox Voice Logger requires that these station numbers be sequential.

Parameter	Usage
Туре	Enter "4620".
Name	Any alphanumeric string can be assigned as an extension name.
Security Code	Enter a security code which identical to the extension.
IP Softphone	Enter "y".

Table 9: Virtual CTI Station Parameters

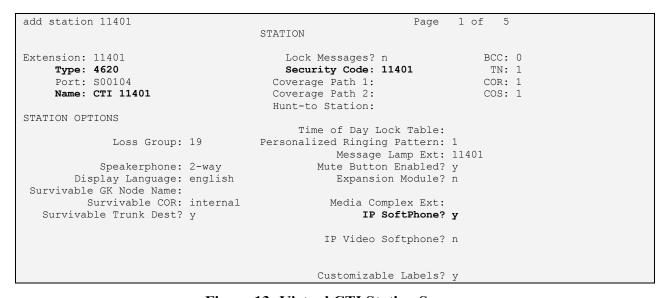


Figure 13: Virtual CTI Station Screen

5. Configure Avaya AuraTM Application Enablement Services

Avaya AuraTM Application Enablement Services is configured via a web browser by accessing the following URL:

https://<AES server address>:8443/MVAP/

Once the login screen appears, enter either the appropriate login ID/password for performing administrative activities or user management.

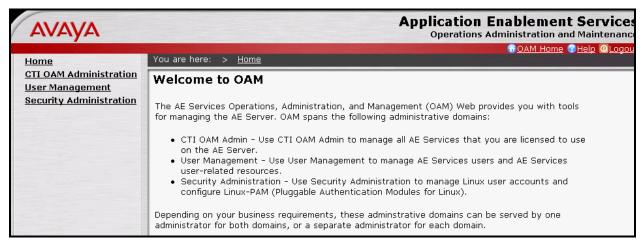


Figure 14: Avaya AuraTM Application Enablement Services Welcome Screen

After logging in, select "CTI OAM Admin" which displays the following screen. Verify that the Avaya AuraTM Application Enablement Services server installation has TSAPI and DMCC service licenses. If this is not the case, please contact an Avaya representative regarding licensing.

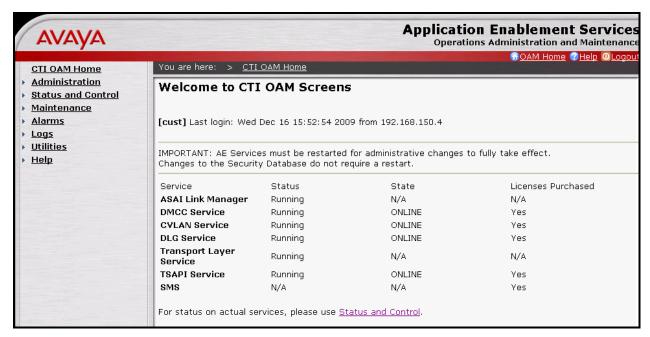


Figure 15: Avaya Aura TM Application Enablement Services CTI OAM Welcome Screen

Navigate to **Administration->Switch Connections**. Enter the name of the switch connection to be added, and click on the "Add Connection" button.

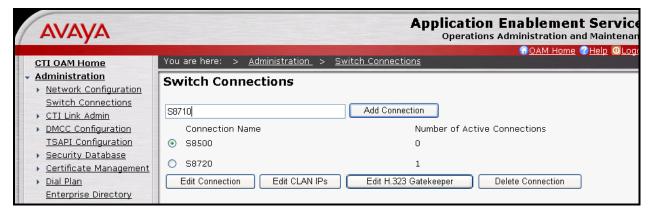


Figure 16: Switch Connection Screen

This causes the following screen to be presented. At this point, enter the screen fields as described in the following table, and click the "Apply" button.

Parameter	Usage
Switch Password	The Switch Password must be the same as was entered into the Avaya Aura TM Communication Manager AE Services Administration form via the "change ip-services" command, described in Figure 10 . Passwords must consist of 12 to 16 alphanumeric characters.
SSL	SSL (Secure Socket Layer) is enabled by default. Keep the default setting unless adding a switch connection for a DEFINITY Server CSI.

Table 10: Configuration of Switch Password



Figure 17: Set Switch Password Screen

From the **Administration->Switch Connections** screen, click the "Edit CLAN IPs" button to display the screen show below. Enter the IP address of the CLAN shown in **Figure 6** which Avaya AuraTM Application Enablement Services is to use for communication with the switch, and click the "Add Name or IP" button.

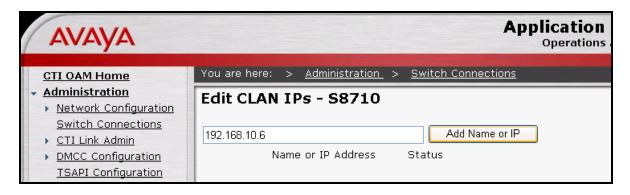


Figure 18: CLAN Screen

On the left margin of the screen, navigate to **Administration->CTI Link Admin->TSAPI Links.** The following screen is displayed. Click the "Add Link" button.

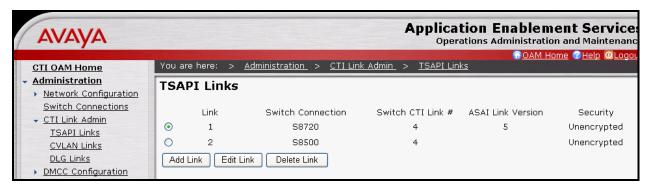


Figure 19: TSAPI Links Screen

Fill in the parameters for the link to be added. The "Link" parameter must be a value between 1 and 16 which is not assigned to another link. The "Switch Connection" parameter should be the name of the Avaya Server which is to be controlled by this link. The value for the TSAPI "Switch CTI Link Number" must be a value between 1 and 64, and must be the same as was used in the Avaya Aura TM Communication Manager "add cti-link" configuration command in **Figure 11**. Click the "Apply Changes" button.



Figure 20: Add TSAPI Link Screen

Navigate to **User Management->Add User**. In this case, the Avaya AuraTM Application Enablement Services user is the DGVox Voice Logger application, which uses Avaya AuraTM Application Enablement Services to monitor stations and initiate switching operations. Repeat this operation to add a DMCC user. Enter the parameters shown in the following table.

Parameter	Usage
User Id	Enter an appropriate user name.
Common Name	Enter the user's first name.
Surname	Enter the user's surname.
User Password	Enter an appropriate user password.
Avaya Role	Set this field to "None" from the drop-down menu.
CT User	Set this field to "Yes" from the drop-down menu.

Table 11: Avaya AuraTM Application Enablement Services User Parameters

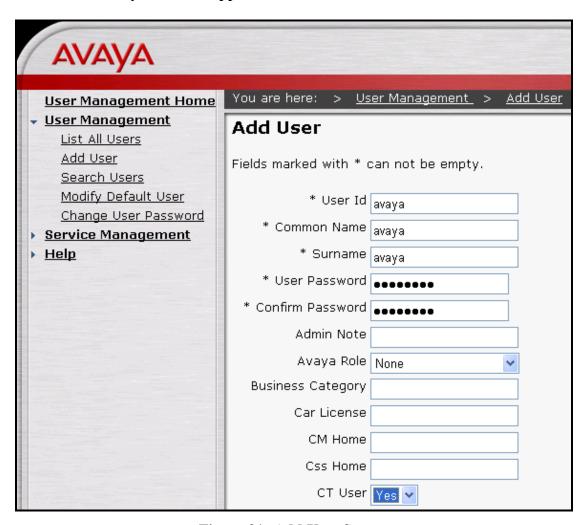


Figure 21: Add User Screen

Navigate to **Administration -> Network Configuration -> Ports** and configure the DMCC Server Ports as shown in the following table.

Parameter	Usage
Unencrypted Port	Enable this port using the default value of 4721.

Table 12: Avaya AuraTM Application Enablement Services Server Interface Parameters

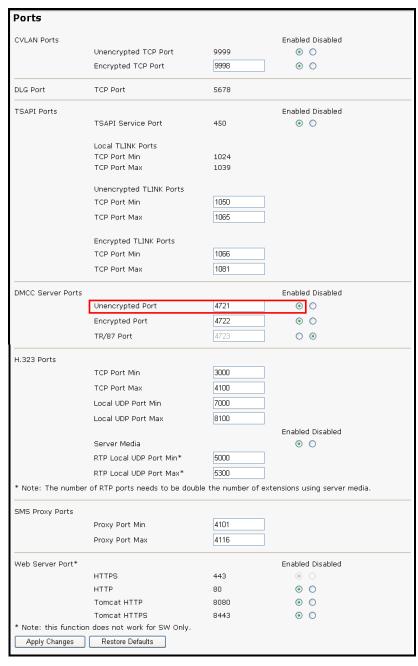


Figure 22: Avaya AuraTM Application Enablement Services Port Configuration

Navigate to **Utilities -> TSAPI Test** and select the newly configured PBX from the "Tlink" drop-down menu. This name will be used in further configuration activities.

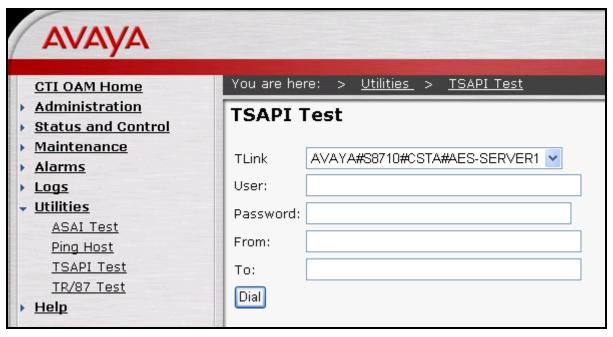


Figure 23: Avaya AuraTM Application Enablement Services TSAPI Test

6. Prepare DGVox Voice Logger Platform

Although the procedure for installing the DGVox Voice Logger software is outside the scope of this document, this section contains a few helpful hints regarding this procedure.

Any third-party HTTP servers running on the server (i.e. Apache) should be disabled or uninstalled prior to installation of DGVox Voice Logger. Subsequently, the Microsoft XP IIS package must be installed prior to installation of the DGVox Voice Logger package. The DGVox Voice Logger installation procedure installs the Microsoft .Net package, and will automatically uninstall any out-of-date versions of this package which may have been present.

If there are other LAN interfaces on the DGVox Voice Logger server (in addition to the one used by the DGVox Voice Logger connection to the network), they must be disabled.

7. Configure DGVox Voice Logger

Start ManagementConsole.exe, which is installed in C:\Voxspectrum\DGVox if the default install path was not changed during installation, and enter the appropriate credentials.



Figure 24: DGVox Voice Logger Management Console Login

From Management Console, double click on "Avaya CTI Settings" icon.

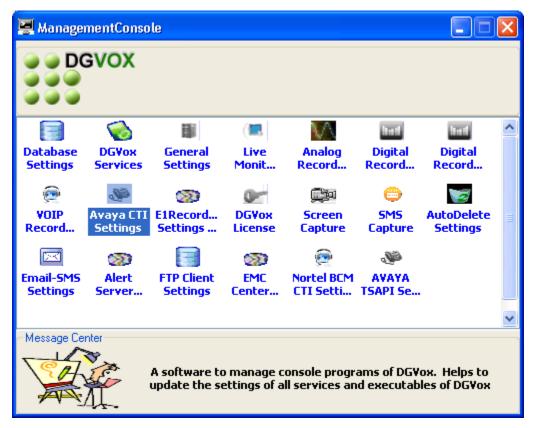


Figure 25: DGVox Voice Logger Management Console

Select the "General Settings" tab and enter the parameters shown in the following table.

Parameter	Usage
CM IP	Enter the IP address of the CLAN interface, or processor interface if is
CWI	Processor Ethernet is used.
AES IP	Enter the IP address of Avaya Aura TM Application Enablement
ALS II	Services.
MediaForwarding IP	Enter the IP address of the server on which the DGVox Voice Logger
Wicdian of warding if	server is running.
TSAPI Server	Enter the TLink identifier for the S8710, as shown in Figure 23 .
AES Port	Enter the number of the Unencrypted DMCC port shown in Figure 22 .
Codec	Select the codec which is used, as show in Figure 8 .
TSAPI LoginID/PW	Enter the user credentials allocated in Figure 21 .
DMCC LoginID/PW	Enter the user credentials allocated in Figure 21 .

Table 13: CTI Settings / General Settings Tab Parameters

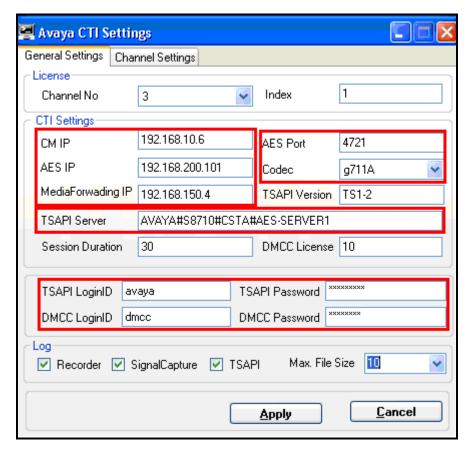


Figure 26: CTI Settings / General Settings Tab Screen

Select the "Channel Settings" tab and enter the parameters shown in the following table.

Section	Parameter	Usage
	~	Use this drop-down menu to allocate an Extension
	Channel ID	and Extension Password for each of the licensed
Channel-Extension		voice recording channels.
	Extension	Enter the extension of a station to be monitored, as
Mapping	Extension	shown in Table 1 .
	Extension Password	Enter the password of a station to be monitored,
		which was configured in Table 8 .
Virtual Extension	Extension	Enter each of the Virtual Extensions which were
Details	EXTERISION	allocated in Figure 13 into this drop-down menu.

Table 14: CTI Settings / Channel Settings Tab Parameters

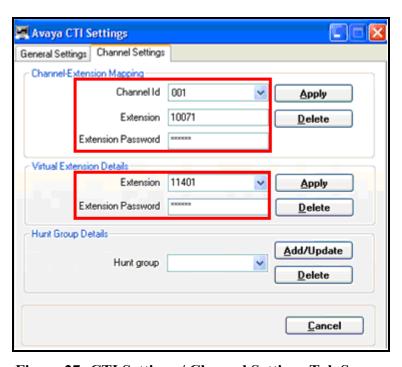


Figure 27: CTI Settings / Channel Settings Tab Screen

8. General Test Approach and Test Results

The compliance testing done between DGVox Voice Logger was performed manually. The tests were all functional in nature. The test method employed can be described as follows:

- Avaya AuraTM Communication Manager was configured to support various local IP telephones, as well as a networked PBX connection and a PSTN connection.
- A PSTN interface was attached to Avaya AuraTM Communication Manager.
- DGVox Voice Logger was configured to monitor various telephones attached to Avaya AuraTM Communication Manager.
- The functionality of DGVox Voice Logger was verified using the above-mentioned local and external telephones, including the ability to record and playback calls participated in by monitored local endpoints and
 - o Locally attached IP and digital telephones
 - o Telephones attached to the PSTN
 - Telephones attached to a networked PBX

The tests which were performed are shown is **Section 1.1**. All tests which were performed produced the expected result.

9. Verification Steps

The correct installation and configuration of DGVox Voice Logger can be verified by performing the following steps using the SAT terminal from PBX 1.

 Use the "status aesvcs cti-link" command to verify that the TSAPI link allocated in Figure 11 is "established".

status aesvcs cti-link							
AE SERVICES CTI LINK STATUS							
CTI Link	Version	Mnt Busy	AE Services Server	Service State	Msgs Sent	Msgs Rcvd	
1 2 3		no no no		down down down	0 0 0	0 0 0	
4	4	no	aes-server1	established	15	15	

Figure 28: Status Aesvcs Cti-link Screen

 Login to Avaya AuraTM Application Enablement Services, and navigate to the "CTI OAM Home" screen. Verify that the DMCC and TSAPI Services are licensed, ONLINE, and Running.

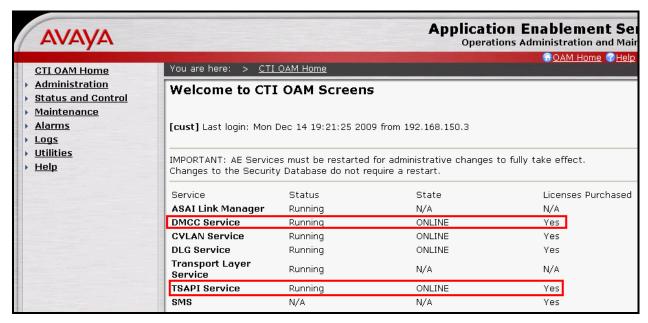


Figure 29: Avaya AuraTM Application Enablement Services CTI OAM Home Screen

• Navigate to **OAM Home -> Status and Control -> Switch Connections Summary** select the PBX 1, and click "Switch Connection Details". Verify that the connection state is "Online" and "Talking".



Figure 30: Avaya AuraTM Application Enablement Services Switch Connection Details Screen

• Navigate to **OAM Home -> Status and Control -> Services Summary** and click "Details" for "TSAPI Service". Verify that the TSAPI service for PBX 1 is "Online" and "Talking".

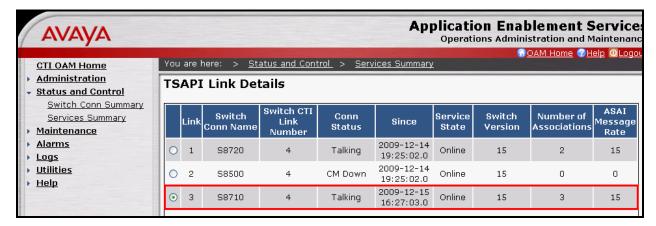


Figure 31: TSAPI Link Details Screen

Navigate to OAM Home -> Status and Control -> Services Summary and click "Details" for "TSAPI Service". Verify that there is a session for the DMCC user allocated in Figure 21 with a "Far-end Identifier" indicating the DGVox Voice Logger.

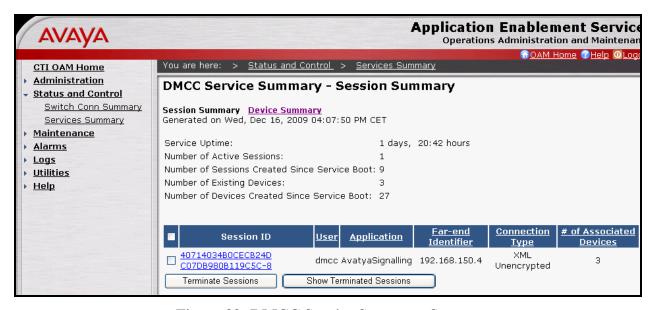


Figure 32: DMCC Service Summary Screen

• Click on the "Device Summary" control and verify that the virtual stations allocated in **Figure 13** are "REGISTERED".

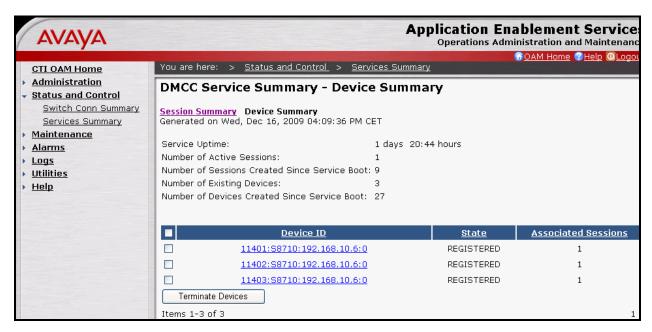


Figure 33: Status Aesvcs Interface Screen

• Make a call from a monitored telephone and use the web client to verify that the call was recorded correctly.

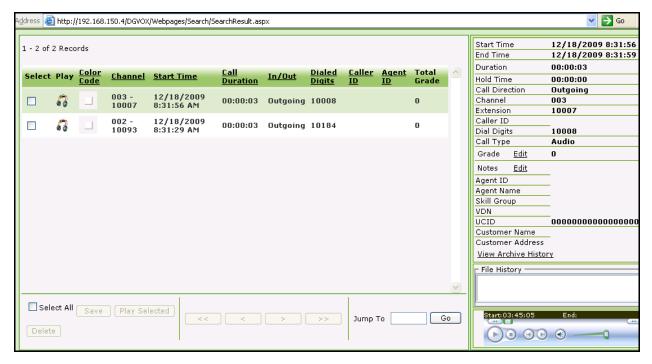


Figure 34: DGVox Voice Logger Web Client Screen

10. Conclusion

These Application Notes describe the compliance testing of the VoxSpectrum DGVox Voice Logger with Avaya AuraTM Communication Manager. A detailed description of the configuration required for both the Avaya and the VoxSpectrum equipment is documented within these Application Notes. The DGVox Voice Logger passed all of the tests performed, which included both functional and robustness tests

11. Additional References

- [1] *Administering Avaya Aura* TM *Communication Manager*, January 2009, Issue 5.0, Document Number 03-300509.
- [2] Avaya AuraTM Communication Manager Feature Description and Implementation, May 2009, Issue 7, Document Number 555-245-205.
- [3] Avaya AuraTM Application Enablement Services Administration and Maintenance Guide, November 2009, Issue 11, Release 5.2, Document Number 02-300357
- [4] DGVox User Manua, l Release 6.9,
- [5] DGVox Installation Manual, Release 6.9,
- [6] DGVox Maintenance Manual, Release 6.9,
- [7] DGVox Technical Manual, Release 6.9

12. Change History

Issue	Date	Reason
1.0	12/20/2009	Initial issue

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