



Avaya Solution & Interoperability Test Lab

Application Notes for DSR SoliDBank 601 VoIP Voice Recorder with Avaya Communication Manager – Issue 1.0

Abstract

These Application Notes describe the conformance testing of the DSR SoliDBank 601 VoIP voice recorder with Avaya Communication Manager and Avaya Application Enablement Services. These Application Notes contain an extensive description of the configurations for SoliDBank 601 VoIP, Avaya Communication Manager, and Avaya Application Enablement Services which were used for testing. The testing which was performed tested the major functions of the SoliDBank 601 VoIP product.

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 DSR SoliDBank 601 VoIP voice recording server to interoperate with Avaya Communication Manager and Avaya Application Enablement Services (AES). The SoliDBank 601 VoIP voice recorder uses the Avaya Communication Manager Service Observing feature to collect voice data for locally attached telephones which have been designated as monitored endpoints. SoliDBank 601 VoIP has an integrated web server which allows authorized users the ability to search for and play back voice records of interest remotely via a web browser and media player.

1.1. Interoperability Compliance Testing

The compliance testing included the following test scenarios:

- Basic incoming call
- Basic outgoing call
- Hold/retrieve
- Supervised transfer
- Blind transfer
- Conference
- Call forward
- Call to hunt group
- Call to/from bridged appearance
- Call to coverage
- Recovery from network interruptions and restarts.

Where applicable, tests were performed with local and external (PSTN) telephones. Calling scenarios were performed with various combinations of local monitored telephones, local unmonitored telephones, and external (PSTN) telephones.

1.2. Support

Phone: +36 1 205 3606

Email: support@dsr.hu e-mail

2. Reference Configuration

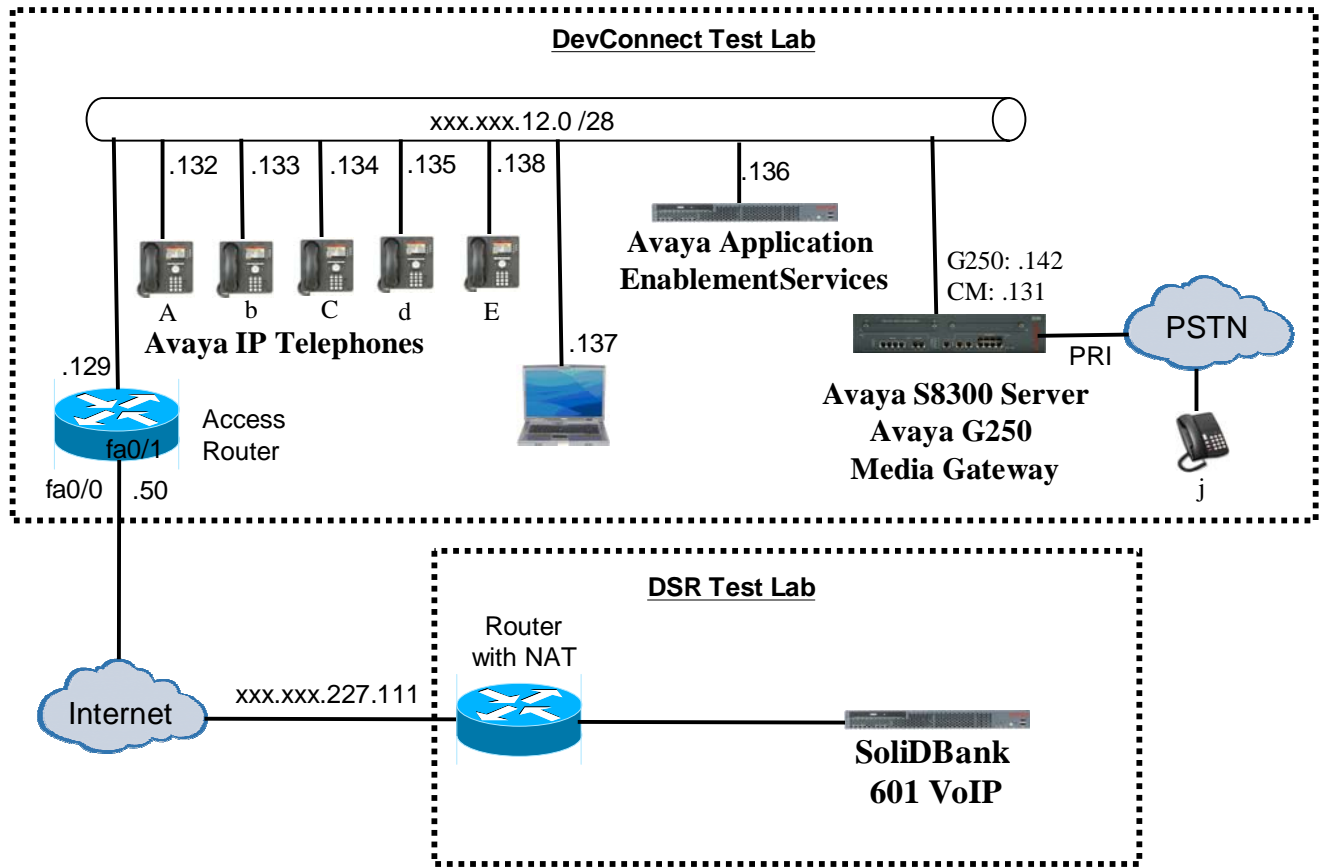


Figure 1: SoliDBank 601 VoIP Test Configuration

In the above diagram, the DSR SoliDBank 601 VoIP records voice conversations from telephones attached to Avaya Communication Manager. DSR SoliDBank 601 VoIP uses AE Services' Device, Media, and Call Control (DMCC) service to monitor call activity associated with Avaya Communication Manager.

The SoliDBank 601 VoIP voice recorder uses the Avaya Communication Manager Service Observing feature to monitor calls which it wishes to record. The voice stream for such calls is received via the LAN interface to Avaya Communication Manager.

The telephones depicted in these Application Notes are designated by an upper case letter if configured to be monitored by the SoliDBank 601 VoIP voice recorder. A lower case letter designates those terminals which have been configured to not be monitored or are possibly unable to be monitored, in the case of the terminal attached to the PSTN.

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

SolidBank 601 VoIP voice recorder. Note that one Virtual CTI Station is required for each telephone station which is to be monitored, as these are used by SolidBank 601 VoIP to initiate Service Observe operations.

Phone	Monitored	Model	Extension
A	*	Avaya 9620	7211
b		Avaya 9620	7212
C	*	Avaya 9650	7213
d		Avaya 9630	7214
E	*	Avaya 9630	7215
j			069 xxxx 6174

Table 1: Device Monitor Configuration

3. Equipment and Software Validated

Software Component	Version
Avaya Communication Manager	S8300-015-01.2.416.4 Update: 01.2.416.4-16770
Avaya Application Enablement Services	r4-2-1-20-5-0
Avaya 96xx Telephones (H.323)	2.00
DSR SolidBank 601 VoIPSW	1.03
DSR SolidBank 601 VoIP platform OS: Fedora 7	Kernel: 2.6.21-1.3194.fc7

Table 2: Hardware/Software Component Versions

4. Configuration

4.1. Configure and Verify Avaya Communication Manager

The configuration and verification operations illustrated in this section were all performed using the Avaya Communication Manager System Administration Terminal (SAT).

The information provided in this section describes the configuration of Avaya 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].

4.1.1. Verify Customer Options

Use the **display system-parameters customer options** command to verify that Avaya Communication Manager is configured to meet the minimum requirements to run SoliDBank 601 VoIP. 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 Stations (page2)	This must be sufficient to support the total number of IP stations. This is only required if IP stations are included in the configuration. Other station types can also be used.
Computer Telephony Adjunct Links? (page3)	This parameter must be set to "y".
IP Stations? (page4)	This parameter must be set to "y".
Enable 'dadmin' Login? (page 4)	This parameter must be set to "y".
IP_API_A (page10)	This parameter must be set the number of Virtual IP Stations
IP_Phone (page10)	This parameter must be set the number of IP stations plus 1 for each station which is to be monitored.

Table 3: System-Parameters Customer-Options Parameters

```

display system-parameters customer-options                               Page 2 of 11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                                    USED
      Maximum Administered H.323 Trunks: 10                          0
      Maximum Concurrently Registered IP Stations: 12              4
      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 Stations: 0                            0
      Maximum Video Capable IP Softphones: 10                      0
      Maximum Administered SIP Trunks: 10                          10
Maximum Administered Ad-hoc Video Conferencing Ports: 0              0
      Maximum Number of DS1 Boards with Echo Cancellation: 0        0
      Maximum TN2501 VAL Boards: 10                                0
      Maximum Media Gateway VAL Sources: 10                        0
      Maximum TN2602 Boards with 80 VoIP Channels: 0               0
      Maximum TN2602 Boards with 320 VoIP Channels: 0             0
      Maximum Number of Expanded Meet-me Conference Ports: 0       0
  
```

Figure 2: System-Parameters Customer-Options Screen, page 2

```

display system-parameters customer-options                               Page 3 of 11
                                OPTIONAL FEATURES

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

```

Figure 3: System-Parameters Customer-Options Screen, page 3

```

display system-parameters customer-options                               Page 4 of 11
                                OPTIONAL FEATURES

Emergency Access to Attendant? y          IP Stations? y
Enable 'dadmin' Login? y
Enhanced Conferencing? y          ISDN Feature Plus? n
Enhanced EC500? y          ISDN/SIP Network Call Redirection? y
Enterprise Survivable Server? n          ISDN-BRI Trunks? y
Enterprise Wide Licensing? n          ISDN-PRI? y
ESS Administration? n          Local Survivable Processor? n
Extended Cvg/Fwd Admin? y          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
Global Call Classification? n          Multimedia Call Handling (Basic)? y
Hospitality (Basic)? y          Multimedia Call Handling (Enhanced)? y
Hospitality (G3V3 Enhancements)? n          Multimedia IP SIP Trunking? y
IP Trunks? y

IP Attendant Consoles? y

```

Figure 4: System-Parameters Customer-Options Screen, page 4

```

display change system-parameters customer-options
MAXIMUM IP REGISTRATIONS BY PRODUCT ID
Page 10 of 11

Product ID  Rel. Limit      Used
IP_API_A   : 10           0
IP_API_B   : 10           0
IP_API_C   : 10           0
IP_Agent   : 10           0
IP_IR_A    : 10           0
IP_Phone  : 12           4
IP_ROMax   : 12           0
IP_Soft    : 2            0
IP_eCons   : 10           0
oneX_Comm  : 12           0
           : 0            0

```

Figure 5: System-Parameters Customer-Options Screen, page 10

4.1.2. Configure Features

Use the **display system-parameters features** command to set system features as shown in the following table.

Parameter	Usage
Service Observing: Warning Tone?	Set this parameter to “n”.
Allow Two Observers in Same Call?	Set this parameter to “y”.

Table 4: System-Parameters Features

```

change system-parameters features
FEATURE-RELATED SYSTEM PARAMETERS
CALL CENTER SYSTEM PARAMETERS
EAS

Direct Agent Announcement Extension:          Delay:

VECTORING
Converse First Data Delay: 0          Second Data Delay: 2
Converse Signaling Tone (msec): 100   Pause (msec): 70
Prompting Timeout (secs): 10

Reverse Star/Pound Digit For Collect Step? n
Available Agent Adjustments for BSR? n
BSR Tie Strategy: 1st-found
Store VDN Name in Station's Local Call Log? n
SERVICE OBSERVING
Service Observing: Warning Tone? n      or Conference Tone? n
Service Observing Allowed with Exclusion? n
Allow Two Observers in Same Call? y

```

Figure 6: System-Parameters Features, page 11

4.1.3. Configure Stations

4.1.3.1 Configure Class of Restriction

Use the **change cor** command to configure Service Observing Class of Restriction parameters as required for Service Observe monitoring. For the purpose of the tests described by these application notes, a common Class of Restriction was shared by the stations being monitored (**Figure 8**) and the Virtual CTI Stations which were used for monitoring (**Figure 9**). In this case, both the “Can Be Service Observed” and “Can Be A Service Observer” parameters in the following screen must be set to “y”.

```
change change cor 1                                     Page 1 of 23
                                     CLASS OF RESTRICTION
COR Number: 1
COR Description:
FRL: 0                                               APLT? y
Can Be Service Observed? y      Calling Party Restriction: none
Can Be A Service Observer? y    Called Party Restriction: none
Partitioned Group Number: 1     Forced Entry of Account Codes? n
Priority Queuing? n              Direct Agent Calling? n
Restriction Override: none       Facility Access Trunk Test? n
Restricted Call List? n          Can Change Coverage? n
Access to MCT? y                 Fully Restricted Service? n
Group II Category For MFC: 7
Send ANI for MFE? n
MF ANI Prefix:                    Automatic Charge Display? n
Hear System Music on Hold? y     PASTE (Display PBX Data on Phone)? n
Can Be Picked Up By Directed Call Pickup? n
Can Use Directed Call Pickup? n
Group Controlled Restriction: inactive
```

Figure 7: Class of Restriction Screen

4.1.3.2 Configure H.323 IP Stations

Use the **add station** command to create a station for extensions A through E, as shown in **Table 1**.

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 string to be used as a security code.
COR	Enter the number of the Class of Restriction which was defined in Figure 7 .

Table 5: Configuration IP Stations

```

add station 7211                                     Page 1 of 5
                                                    STATION
Extension: 7211                                Lock Messages? n          BCC: 0
Type: 9620                                    Security Code: 123456  TN: 1
Port: S00004                                       Coverage Path 1:         COR: 1
Name: extn 7211                               Coverage Path 2:         COS: 1
                                                    Hunt-to Station:
STATION OPTIONS
                                                    Time of Day Lock Table:
Loss Group: 19                                     Personalized Ringing Pattern: 1
                                                    Message Lamp Ext: 7211
Speakerphone: 2-way                               Mute Button Enabled? y
Display Language: english
Survivable GK Node Name:
Survivable COR: internal                          Media Complex Ext:
Survivable Trunk Dest? y                          IP SoftPhone? n
                                                    IP Video? n
                                                    Customizable Labels? y

```

Figure 8: IP Station Screen

4.1.3.3 Configure Virtual CTI Stations

Use the **add station** command to create a station for each of the Virtual CTI Stations listed in **Table 1**. A separate Virtual CTI Station is required for each station to be monitored.

Parameter	Usage
Extension	Create the Virtual CTI Stations listed in Table 1
Type (page 1)	Enter "CTI".
Name (page 1)	Any alphanumeric string can be assigned as an extension name.
Security Code (page 1)	Enter a numeric security code.
COR	Enter the number of the Class of Restriction which was defined in Figure 7 .

Table 6: Virtual CTI Station Parameters

```

add station 7311                                     Page 1 of 5
                                                    STATION
Extension: 7311                                Lock Messages? n          BCC: 0
Type: CTI                                     Security Code: 123456  TN: 1
Port: X                                           Coverage Path 1:         COR: 1
Name: CTI 1                                   Coverage Path 2:         COS: 1
                                                    Hunt-to Station:
STATION OPTIONS
    Loss Group: 1                                Time of Day Lock Table:
    Data Module? n                               Personalized Ringing Pattern: 1
    Display Module? n                           Message Lamp Ext: 7311
    Survivable COR: internal                     Media Complex Ext:
    Survivable Trunk Dest? Y
  
```

Figure 9: Virtual CTI Station Screen, page 1

4.1.4. Configure Feature Codes

Use the **change feature-access-codes** command to assign an access code to the Service Observe Listen Only Access Code, to allow SoliDBank 601 VoIP to activate the Service Observe operation.

Parameter	Usage
Service Observing Listen Only Access Code	Assign an otherwise unused access code which can be used by SoliDBank 601 VoIP to activate this feature.

Table 7: Parameters for the Feature Access Codes

```

change feature-access-codes                                     Page 5 of 9
                                FEATURE ACCESS CODE (FAC)

Service Observing Listen Only Access Code: *02
Service Observing Listen/Talk Access Code:
Service Observing No Talk Access Code:

Remote Logout of Agent Access Code:

```

Figure 10: Feature Access Codes Form, Page 5

4.1.5. Configure CTI Link

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. Use an unused extension as the value for the “Extension” parameter. The value chosen for the “Name” parameter is a matter of personal preference. A link type of “ADJ-IP” must be specified.

```

add cti-link 4                                               Page 1 of 3
                                CTI LINK
CTI Link: 4
Extension: 7999
Type: ADJ-IP
Name: AES-devcon223-tsapi
COR: 1

```

Figure 11: Cti-link Screen

4.2. Configure Avaya AES

The AES server is configured via a web browser by accessing the following URL:

`https://<AES server address>/`

Click “AE Server Administration”.

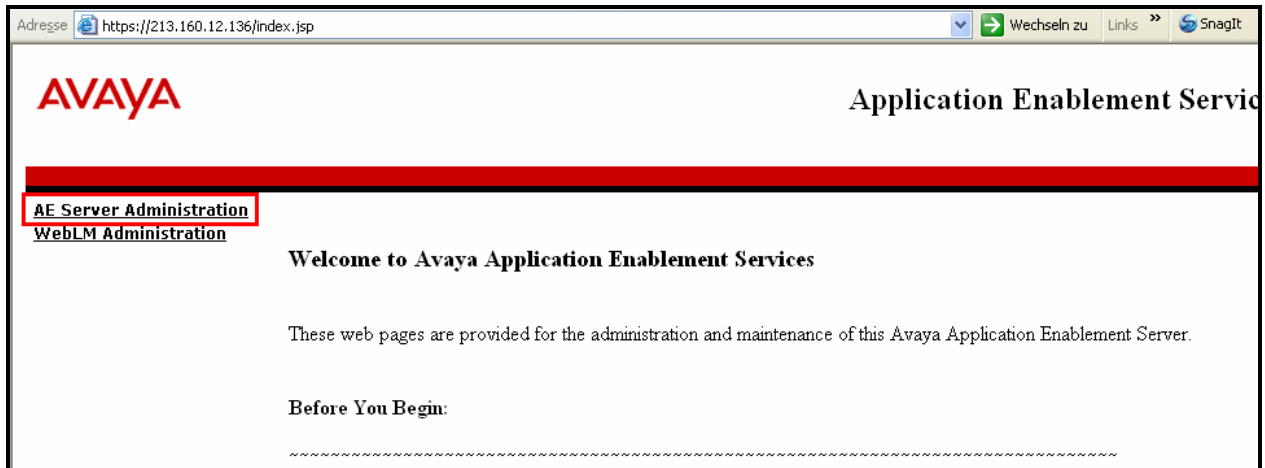


Figure 12: AES Access Screen

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



Figure 13: AES Login Screen

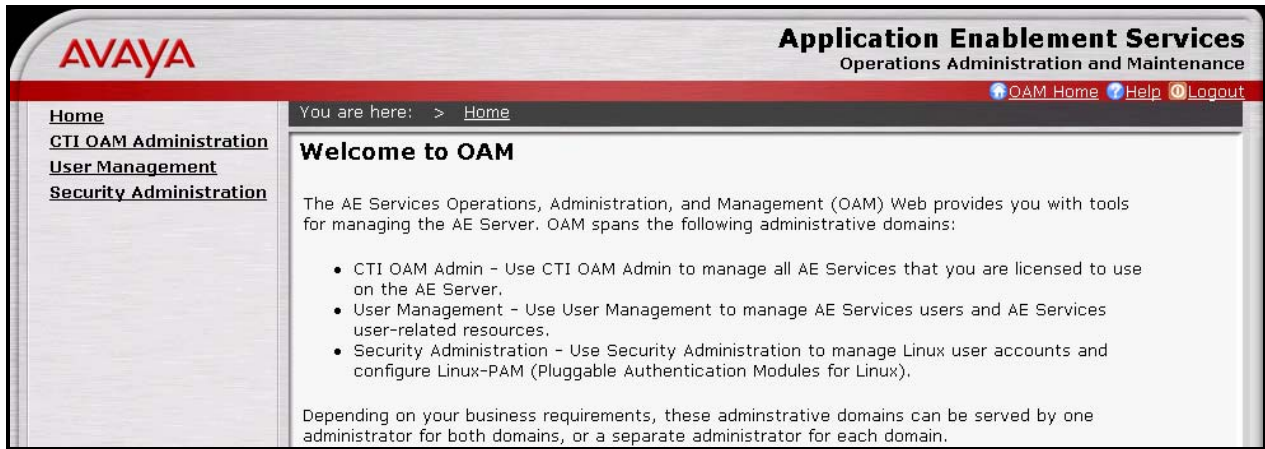


Figure 14: AES Welcome Screen

After logging in, select “CTI OAM Admin” which displays the following screen. Verify that the AES server installation has a TSAPI service license. If this is not the case, please contact an Avaya representative regarding licensing.

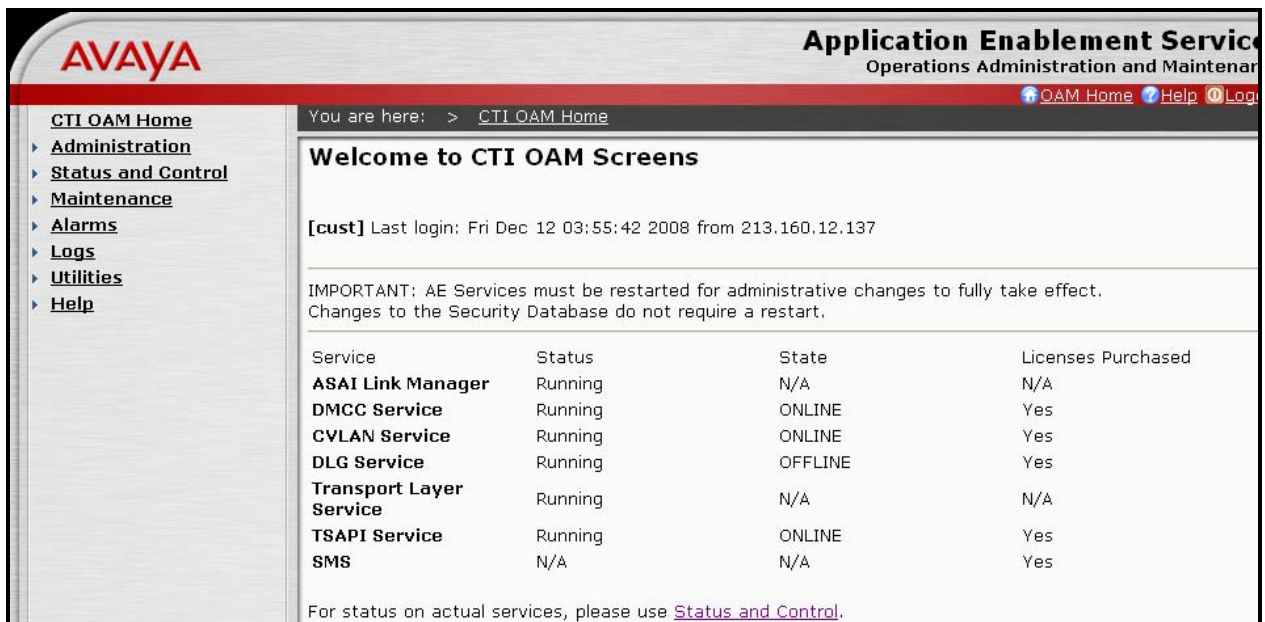


Figure 15: AES CTI OAM Welcome Screen

From the **Administration->Switch Connections** screen, click the “Edit CLAN IPs” button to display the screen shown below. Enter the IP address of the Avaya S8300 Server, and click the “Add Name or IP” button.

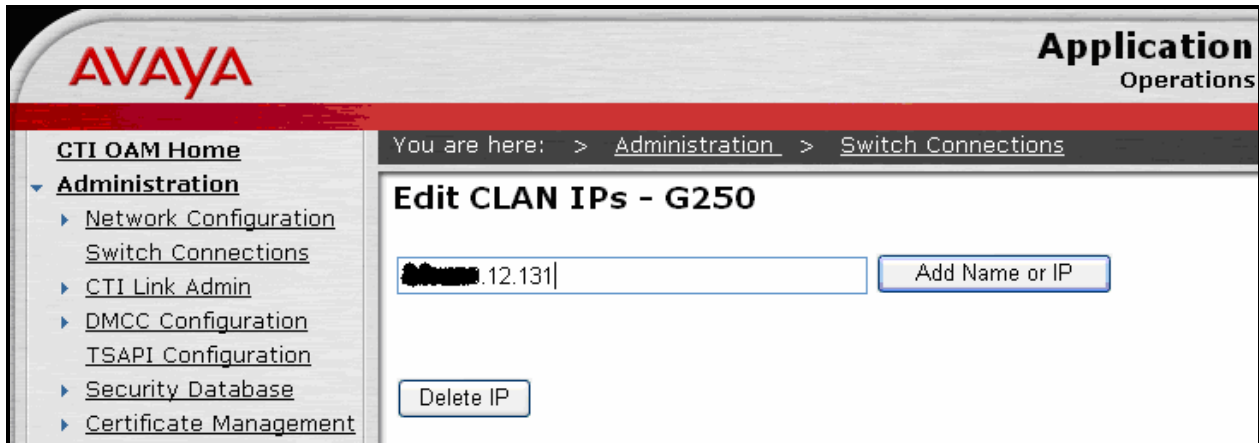


Figure 16: CLAN Screen

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 17: 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 Communication Manager “add cti-link” configuration command shown in **Figure 11**. Click the “Apply Changes” button.

The screenshot shows the Avaya Application Operations interface. The top left features the Avaya logo, and the top right says "Application Operations". A breadcrumb trail reads "You are here: > Administration > CTI Link Admin > TSAPI Links". The main content area is titled "Add / Edit TSAPI Links" and contains the following form fields:

- Link: 1
- Switch Connection: G250
- Switch CTI Link Number: 4
- ASAI Link Version: 4
- Security: Unencrypted

At the bottom of the form are two buttons: "Apply Changes" (highlighted with a blue border) and "Cancel Changes".

Figure 18: Add TSAPI Link Screen

Click “OAM Home” on the menu bar and Navigate to **User Management->Add User**.

Create a User which SoliDBank 601 VoIP can use to login to AES, assigning a “User ID” and “User Password” as credentials. The “CT User” field for this user must be set to “Yes”. Although the “Common Name” and “Surname” fields are required, the content of these fields can be set to any string.

AVAYA

User Management Home You are here: > User Management > Add User

Add User

Fields marked with * can not be empty.

* User Id SolIDBank

* Common Name CommonName

* Surname Surname

* User Password

* Confirm Password

Admin Note

Avaya Role None

Business Category

Car License

CM Home

Css Home

CT User Yes

Department Number

Figure 19: Add User Screen

Navigate to **Administration -> Security Database -> CTI Users -> List All Users**, and then click “Edit User” for the newly added user “SoliDBank”. Enable “Unrestricted Access” and click “Apply Changes”.



Figure 20: Edit CTI User Screen

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

Parameter	Usage
Unencrypted Port	Leave this port set to its default value of 4721, and click the “Enabled” radio button.

Table 8: SoliDBank 601 VoIP AES Server Interface Parameters

You are here: > Administration > Network Configuration > Ports

Ports

CVLAN Ports Enabled Disabled

Unencrypted TCP Port	9999	<input checked="" type="radio"/> <input type="radio"/>
Encrypted TCP Port	<input type="text" value="9998"/>	<input checked="" type="radio"/> <input type="radio"/>

DLG Port TCP Port

TCP Port	5678
----------	------

TSAPI Ports Enabled Disabled

TSAPI Service Port	450	<input checked="" type="radio"/> <input type="radio"/>
--------------------	-----	--

Local TLINK Ports

TCP Port Min	1024
TCP Port Max	1039

Unencrypted TLINK Ports

TCP Port Min	<input type="text" value="1050"/>
TCP Port Max	<input type="text" value="1065"/>

Encrypted TLINK Ports

TCP Port Min	<input type="text" value="1066"/>
TCP Port Max	<input type="text" value="1081"/>

DMCC Server Ports Enabled Disabled

Unencrypted Port	<input type="text" value="4721"/>	<input checked="" type="radio"/> <input type="radio"/>
Encrypted Port	<input type="text" value="4722"/>	<input checked="" type="radio"/> <input type="radio"/>
TR/87 Port	<input type="text" value="4723"/>	<input type="radio"/> <input checked="" type="radio"/>

H.323 Ports

TCP Port Min	<input type="text" value="3000"/>
TCP Port Max	<input type="text" value="4100"/>
Local UDP Port Min	<input type="text" value="7000"/>
Local UDP Port Max	<input type="text" value="8100"/>
RTP Local UDP Port Min	<input type="text" value="5000"/>
RTP Local UDP Port Max	<input type="text" value="5300"/>

Figure 21: AES Port Configuration

4.3. Configure SoliDBank 601 VoIP Server

Enter the URI of the DSR SoliDBank 601 VoIP into the address field of an Internet browser. When the login screen appears, enter an appropriate “User” name and “Password”. From the “Settings” menu of the following screen select “Avaya Settings”.



Figure 22: SoliDBank 601 VoIP Login Screen

This causes the screen shown in **Figure 24** to be presented. At this point, enter the screen fields as described in the following table, and click the “Apply” button.

Section	Parameter	Usage
General Parameters	AES Server IP Address	Enter the IP address of the AES server.
	AES Server Port	Enter the port number configured in Figure 21 .
	Media IP Address	Enter the address of the DSR SoliDBank 601 VoIP, or in this case the address of the NAT router, which assumes that router is configured to forward the media packets to DSR SoliDBank 601 VoIP.
	Media Port	Enter an otherwise unused port number.
	DMCC User Name	Enter the AES user name configured in Figure 19 .
	DMCC Password	Enter the AES password configured in Figure 19 .
	Switch Name	Enter the switch name which can be found on AES under Administration/Security Database/Tlinks, as shown in Figure 23 .
	Switch IP Interface	Enter the IP address of the Avaya S8300 Server.
	Observing FAC	Enter the Feature Access Code assigned in Figure 10 .
Recording Terminals	Terminals	Enter the list of virtual stations, with password, which were assigned in Section 4.1.3.3 , followed by the codec to be used for recording. These items should be separated from another with a “ ” character. There must be one Recording Terminal configured for each Recorded Terminal.
Recorded Terminals	Terminals	Enter the list of extensions of the stations which are to be monitored, as listed in Table 1 .

Table 9: SoliDBank 601 VoIP Avaya Settings Parameters

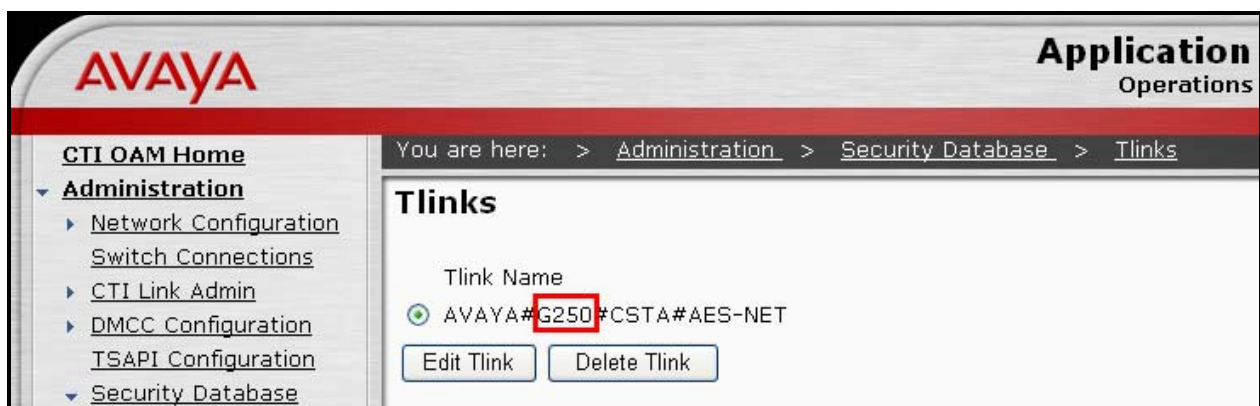


Figure 23: AES Tlinks Screen

RECORDS		PROFILE SETTINGS		USERS		SETTINGS		LOGOUT	
SETTINGS > AVAYA SETTINGS									
▼ GENERAL PARAMETERS:									
▪ AES SERVER IP ADDRESS:		<input type="text" value="██████████.12.136"/>							
▪ AES SERVER PORT:		<input type="text" value="4721"/>							
▪ MEDIA IP ADDRESS:		<input type="text" value="██████████.227.111"/>							
▪ MEDIA PORT:		<input type="text" value="17640"/>							
▪ DMCC USER NAME:		<input type="text" value="SolIDBank"/>							
▪ DMCC PASSWORD:		<input type="text" value="████████████████████"/>							
▪ SWITCH NAME:		<input type="text" value="G250"/>							
▪ SWITCH IP INTERFACE:		<input type="text" value="██████████.12.131"/>							
▪ OBSERVING FAC:		<input type="text" value="*02"/>							
▼ RECORDING TERMINALS:									
▪ TERMINALS:		<input type="text" value="7311 123456 g.711a
7312 123456 g.711a
7313 123456 g.711a"/>							
▪ EXTENSION:		<input type="text"/>							
▪ PASSWORD:		<input type="text"/>							
▪ CODEC:		<input type="text" value="g.711a"/>							
▼ RECORDED TERMINALS:									
▪ TERMINALS:		<input type="text" value="7211
7213
7215"/>							
▪ EXTENSION:		<input type="text"/>							

Figure 24: SoliDBank 601 Avaya Settings Screen

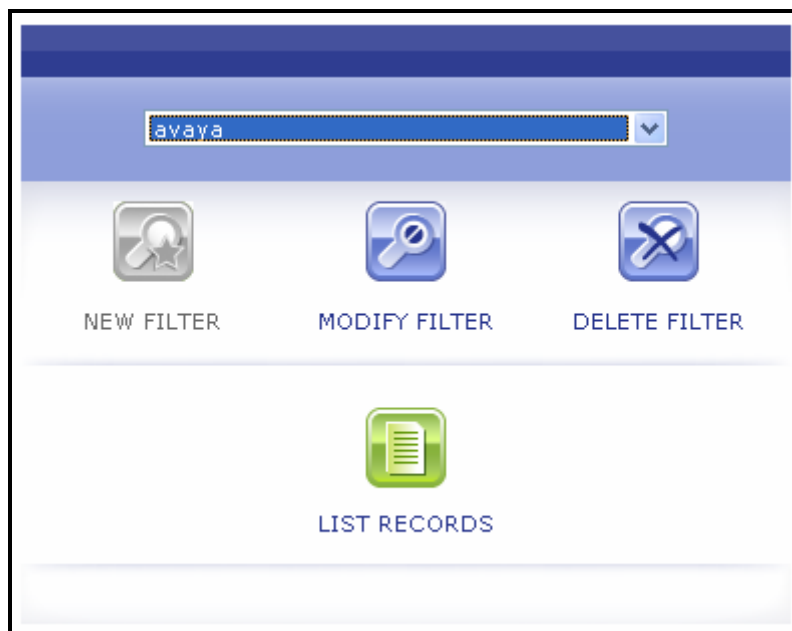
5. General Test Approach and Test Results

The compliance testing for DSR SoliDBank 601 VoIP was performed manually. The tests were all functional in nature, and no performance testing was done. All tests which were performed produced the expected result. **Section 1.1** contains a list of tests which were performed.

6. Verification Steps

The following steps can be performed to verify the correct installation and configuration of SoliDBank 601 VoIP:

- Verify that the Avaya AES and SoliDBank 601 VoIP systems can ping each other.
- Verify that the various telephones can call each other.
- Log into the Avaya AES as described in **Section 4.2** and perform the following:
 - Verify that CTI OAM Status and Control “Switch Connection Summary” shows that the connection between Avaya AES and Avaya Communication Manager is operational.
 - Verify that CTI OAM Status and Control “Services Summary” shows that TSAPI service is operational.
- Verify that it is possible to record a conversation from one of the stations that is being monitored. Login to SoliDBank 601 VoIP, as shown in **Figure 22**, select an appropriate record filter, and click the “List Records” button.



The call can then be played back and the call details can be verified.



The screenshot shows a web interface for SoliDBank 601. At the top, there is a navigation menu with options: RECORDS, PROFILE SETTINGS, USERS, SETTINGS, and LOGOUT. The main content area is titled 'RECORDS' and includes a 'FILTER' button and a 'LIST RECORDS' link. Below this is a table with the following columns: Record Number, Start Time, Marked, Duration, Extension, Direction, Dialed Num, Remote Party, Format, State, Origin, First ID, Call ID, and Comment. The table contains five rows of call records. At the bottom of the interface, there is a green circular arrow icon and the text 'BACK'.

Record Number	Start Time	Marked	Duration	Extension	Direction	Dialed Num	Remote Party	Format	State	Origin	First ID	Call ID	Comment
71033	2009-02-11 10:41:29		00:01:20	7211	out		7214	G.711Alaw	ready		71033		
71034	2009-02-11 11:11:07		00:00:06	7211	out		7212	G.711Alaw	ready		71034		
71035	2009-02-11 11:11:53		00:01:20	7213	in		7212	G.711Alaw	ready		71035		
71036	2009-02-11 11:19:25		00:01:45	7213	in		7212	G.711Alaw	ready		71036		
71037	2009-02-11 11:21:16		00:00:07	7215	in		7214	G.711Alaw	ready		71037		

7. References

- [1] *Administrator Guide for Avaya Communication Manager*, January 2008, Issue 4.0, Document Number 03-300509.
- [2] *Feature Description and Implementation for Avaya Communication Manager*, January 2008, Issue 6, Document Number 555-245-205.
- [3] *Avaya one-X™ Deskphone Edition for 9600 Series IP Telephones Installation and Maintenance Guide*, May 2008, Issue 5, Release 2.0, Document Number 16-300694
- [4] "User guide SB601VoIP" (3rd edition)

8. Conclusion

These Application Notes describe the conformance testing of the DSR SoliDBank 601 VoIP voice recorder with Avaya Communication Manager. Silent monitoring via the Service Observe recording method offered by the SoliDBank 601 VoIP was tested. A detailed description of the configuration required for both the Avaya and the DSR equipment is documented within these Application Notes. The SoliDBank 601 VoIP passed all of the tests performed, which included both functional and robustness tests.

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