



Avaya Solution & Interoperability Test Lab

Application Notes for CallTech Dali® Processing Billing system with Avaya Aura® Communication Manager R6.3 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the CallTech Dali® Processing Billing system to successfully interoperate with Avaya Aura® Communication Manager.

CallTech Dali® CDR Processing Billing system is a call accounting software that interoperates with Avaya Aura® Communication Manager over the TCP/IP protocol. Call records can be generated for various types of calls. CallTech Dali® collects, and processes the call records.

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

The overall objective of this interoperability compliance test is to verify that the CallTech Dali® CDR Processing Billing system software can interoperate with Avaya Aura® Communication Manager R6.3. CallTech Dali® CDR Processing Billing system (herein referred to as Dali) connects to Avaya Aura® Communication Manager over the local or wide area network using a CDR link running on the TCP/IP protocol. Avaya Aura® Communication Manager is configured to send CDR records to Dali using a specific port.

Dali provides traditional call collection, rating, and reporting for any size businesses. Dali can interface with most telephone systems - in particular, with the Avaya Aura® Communication Manager - to collect and interpret the detailed records of inbound, outbound, tandem, and internal telephone calls. Dali then calculates the appropriate charge for local, long distance, international & special calls and allocates them to responsible parties.

During testing, SIP endpoints were included, the SIP endpoints registered with Avaya Aura® Session Manager. An assumption is made that Avaya Aura® Session Manager and Avaya Aura® System Manager are already installed and basic configuration has been performed.

Only steps relevant to this compliance test will be described in this document. In these Application Notes, the following topics will be described:

- Avaya Aura® Communication Manager – A SIP trunk configuration between Avaya Aura® Communication Manager and Avaya Aura® Session Manager. A CDR link configuration on Avaya Aura® Communication Manager.
- Avaya Aura® Session Manager – SIP trunk configuration between Avaya Aura® Communication Manager and Avaya Aura® Session Manager.
- Dali – A CDR link configuration on Dali.

2. General Test Approach and Test Results

The general test approach was to manually place intra-switch calls, inbound trunk and outbound trunk calls, transfer, conference, and verify that Dali collects the CDR records, and properly classifies and reports the attributes of the call.

For serviceability testing, physical and logical links were disabled/re-enabled, Avaya Servers were reset and Dali was restarted.

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

2.1. Interoperability Compliance Testing

The interoperability compliance testing included features and serviceability tests. The focus of the compliance testing was primarily on verifying the interoperability between Dali and Communication Manager.

2.2. Test Results

All executed test cases passed successfully. Dali successfully collected the CDR records from Communication Manager via a TCP/IP connection for all types of calls generated including intra-switch calls, inbound/outbound PSTN trunk calls, inbound/outbound private IP trunk calls, transferred calls, and conference calls.

For serviceability testing, Dali was able to resume collecting CDR records after failure recovery including buffered CDR records for calls that were placed during the outages.

2.3. Support

Technical support for Dali can be obtained through the following:

- support@calltechsa.com
- +57 (1) 6 35 65 35 Ext: 911

3. Reference Configuration

Figure 1 illustrates a configuration used during the compliance test. For completeness, Avaya 96xx Series SIP IP Telephones on the Avaya S8300D Server and Avaya G450 Media Gateway side have been registered to Session Manager. These endpoints are included in Figure 1 to demonstrate calls between the SIP IP telephones that are going through the IP/PRI trunk between two Communication managers. The solution described herein is also extensible to other Avaya Servers and Media Gateways.

Note: Avaya S8300D Server with an Avaya G430 Media Gateway was included in the test only to provide an inter-switch scenario. Thus, there will not be any discussion on configuring Avaya S8300D Server with an Avaya G430 Media Gateway.

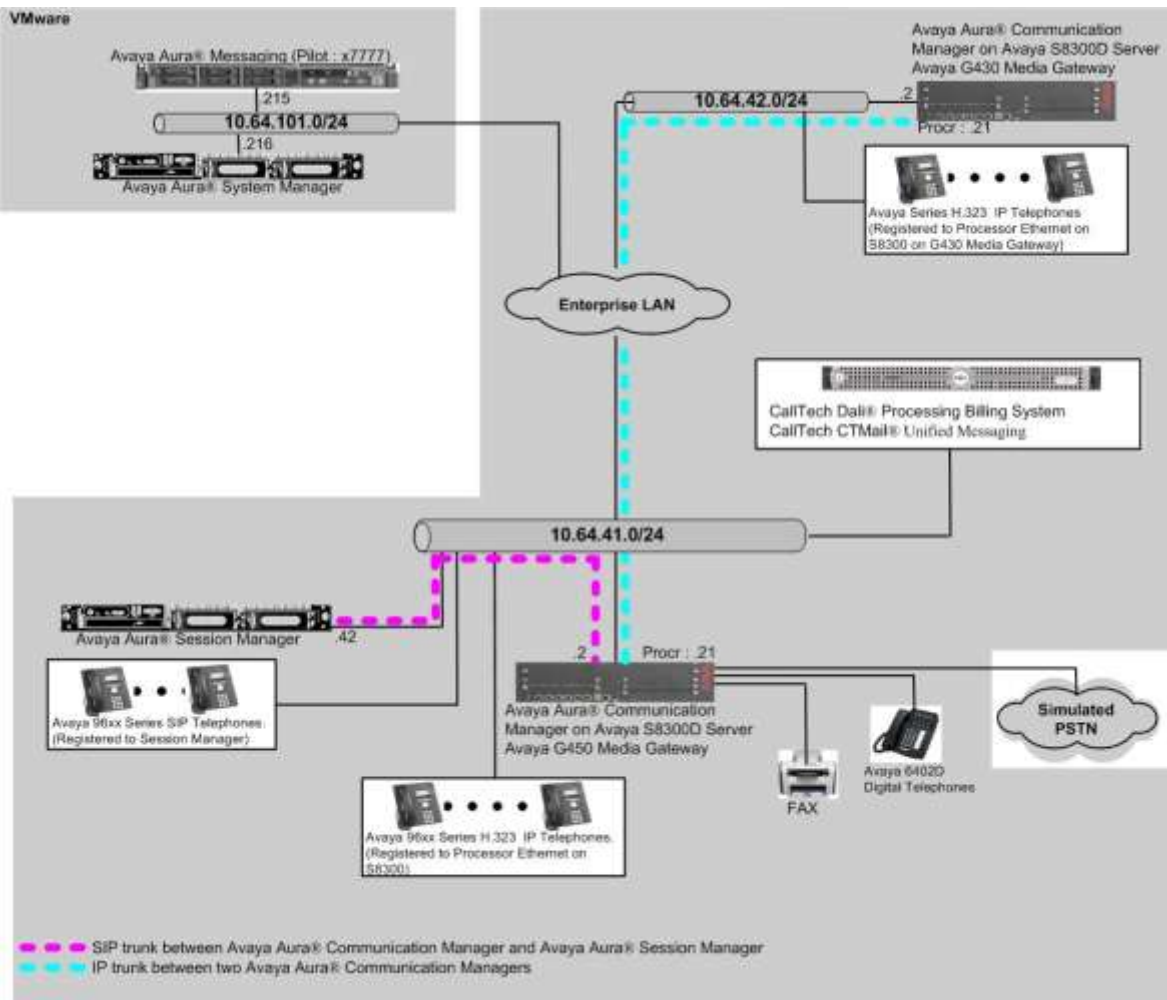


Figure 1. Test configuration of Dali with Avaya Aura® Communication Manager

4. Equipment and Software Validated

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

Equipment		Software
Avaya S8300D Server with Avaya G450 Media Gateway		Avaya Aura® Communication Manager 6.3 (R016x.03.0.124.0) with Patch 03.0.124.0-21754
Avaya Aura® System Manager		6.3.13
Avaya Aura® Session Manager		6.3.13.0.631304
Avaya S8300D Server with Avaya G430 Media Gateway		Avaya Aura® Communication Manager 6.01 (R015x.02.1.016.4)
Avaya 9600 Series SIP IP Telephone		
	9620	2.6.12
	9641G	6.4.1.25
Avaya 9600 and 96X1 Series H.323 IP Telephone		
	9620	3.22
	9621G	6.23
	9650	3.23
CallTech Dali on Windows 8.1 Enterprise		Dali 2.0

5. Configure Avaya Aura® Communication Manager

This section describes the procedure for configuring call detail recording (CDR) in Communication Manager. These steps are performed through the System Access Terminal (SAT). These steps describe the procedure used for the Avaya S8300D Server. All steps are the same for the other Avaya Servers.

Communication Manager will be configured to generate CDR records using TCP/IP to the IP address of the PC running Dali. For the Avaya S8300D Media Server, the TCP/IP link originates at the IP address of the local processor (with node-name - “procr”). For the Avaya G650 Media Gateway, the SIP trunk terminates at the IP address of the CLAN board.

5.1. Configure CDR

Use the **change node-names ip** command to create a new node name, for example, **calltech**. This node name is associated with the IP Address of the PC running the Dali application. Also, take note of the node name – “procr”. It will be used in the next step. The “procr” entry on this form was previously administered.

```
change node-names ip Page 1 of 2
```

IP NODE NAMES	
Name	IP Address
calltech	10.64.101.241
default	0.0.0.0
procr	10.64.41.21
procr6	::
rdtt-1	10.64.40.14
SM-1	10.64.41.42

Use the **change ip-services** command to define the CDR link to use TCP/IP. To define a primary CDR link, provide the following information:

- **Service Type:** “CDR1” [If needed, a secondary link can be defined by setting Service Type to CDR2.]
- **Local Node:** “procr” [For the Avaya G650 Media Gateway set the Local Node to the node name of the CLAN board.]
- **Local Port:** “0” [The Local Port is fixed to 0 because Communication Manager initiates the CDR link.]
- **Remote Node:** “calltech” [The Remote Node is set to the node name previously defined.]
- **Remote Port:** “9000” [The Remote Port may be set to a value between 5000 and 64500 inclusive, and must match the listening port configured in Dali.]

```
change ip-services Page 1 of 4
```

IP SERVICES						
Service Type	Enabled	Local Node	Local Port	Remote Node	Remote Port	
AESVCS	y	procr	8765			
CDR1		procr	0	calltech	9000	
CDR2		procr	0	rdtt-1	9005	

On **Page 3** of the ip-services form, disable the Reliable Session Protocol (RSP) for the CDR link by setting the **Reliable Protocol** field to “n”.

```
change ip-services Page 3 of 4
```

SESSION LAYER TIMERS							
Service Type	Reliable Protocol	Packet Timer	Resp	Session Message	Connect Cntr	SPDU Cntr	Connectivity Timer
CDR1	n	30		3		3	60
CDR2	y	30		3		3	60

Enter the **change system-parameters cdr** command from the SAT to set the parameters for the type of calls to track and the format of the CDR data. The example below shows the settings used during the compliance test. Provide the following information:

- **CDR Date Format:** “month/day”
- **Primary Output Format:** “customized”
- **Primary Output Endpoint:** “CDR1”

The remaining parameters define the type of calls that will be recorded and what data will be included in the record. See reference [2] for a full explanation of each field. The test configuration used some of the more common fields described below.

- **Use Legacy CDR Formats?:** “n” [Allows CDR formats to use 4.x CDR formats. If the field is set to “y”, then CDR formats utilize the 3.x CDR formats.]
- **Intra-switch CDR:** “y” [Allows call records for internal calls involving specific stations. Those stations must be specified in the INTRA-SWITCH CDR form.]
- **Record Outgoing Calls Only?:** “n” [Allows incoming trunk calls to appear in the CDR records along with the outgoing trunk calls.]
- **Outg Trk Call Splitting?:** “y” [Allows a separate call record for any portion of an outgoing call that is transferred or conferenced.]
- **Inc Trk Call Splitting?:** “y” [Allows a separate call record for any portion of an incoming call that is transferred or conferenced.]
- **Call Account Code Length:** “6” [The length may be set to a value between 1 and 15. However, during the compliance test, “6” was used.]

```

change system-parameters cdr                                     Page 1 of 1
                                CDR SYSTEM PARAMETERS

Node Number (Local PBX ID): 1                                CDR Date Format: month/day
Primary Output Format: customized                            Primary Output Endpoint: CDR1
Secondary Output Format: Customized                        Secondary Output Endpoint: CDR2
  Use ISDN Layouts? n                                     Enable CDR Storage on Disk? y
  Use Enhanced Formats? n                               Condition Code 'T' For Redirected Calls? n
  Use Legacy CDR Formats? n                             Remove # From Called Number? n
Modified Circuit ID Display? n                            Intra-switch CDR? y
  Record Outgoing Calls Only? n                          Outg Trk Call Splitting? y
  Suppress CDR for Ineffective Call Attempts? n          Outg Attd Call Record? n
  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: 6
  
```

If the **Intra-switch CDR** field is set to “y” on **Page 1** of the **system-parameters cdr** form, then use the **change intra-switch-cdr** command to define the extensions that will be subject to call detail records. In the Assigned Members field, enter the specific extensions whose usage will be tracked.

Note: Using the Communication Manager firmware (Section 4), it recognized SIP endpoints. Thus SIP endpoints also need to be included in the INTRA-SWITCH CDR form. The extension 72021 and 72022 are SIP extensions.

```
change intra-switch-cdr                                     Page 1 of 3
                                     INTRA-SWITCH CDR
                                     Assigned Members: 9 of 1000 administered
Extension      Extension      Extension      Extension
72001
72002
72003
72021
72022
```

5.2. Configure IP Network Region

This section describes the steps for administering an IP network region in Communication Manager for communication between Communication Manager and Session Manager. Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive, and configure the following:

- **Authoritative Domain** – Enter the appropriate name for the Authoritative Domain. Set to the appropriate domain. During the compliance test, the authoritative domain is set to “avaya.com”.
- **Codec Set** – Set the codec set number as provisioned in the **IP Codec Set** form.

```
change ip-network-region 1                               Page 1 of 20
                                     IP NETWORK REGION
Region: 1
Location:      Authoritative Domain: avaya.com
Name:
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
Codec Set: 1          Inter-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: 46
Audio PHB Value: 46
Video PHB Value: 26
802.1P/Q PARAMETERS
Call Control 802.1p Priority: 6
Audio 802.1p Priority: 6
Video 802.1p Priority: 5
H.323 IP ENDPOINTS      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y      RSVP Enabled? n
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```


5.3. Configure IP Node Name

This section describes the steps for setting IP node name for Session Manager in Communication Manager. Enter the **change node-names ip** command, and add a node name for **SM-1** (Session Manager) along with its IP address.

```
change node-names ip                               Page 1 of 2
                                                    IP NODE NAMES
Name          IP Address
Dali          10.64.43.249
default       0.0.0.0
procr         10.64.41.21
procr6        ::
rdtt          10.64.40.14
SM-1          10.64.41.42
```

5.4. Configure SIP Signaling

This section describes the steps for administering a signaling group in Communication Manager for signaling between Communication Manager and Session Manager. Enter the **add signaling-group <s>** command, where **s** is an available signaling group and configure the following:

- **Group Type** – Set to “sip”.
- **Transport Method** – Set to “tls”.
- **Near-end Node Name** - Set to “procr” as displayed in **Section 5.3**.
- **Far-end Node Name** - Set to the “SM-1” configured in **Section 5.3**.
- **Far-end Network Region** - Set to the region configured in **Section 5.2**.
- **Far-end Domain** - Set to “avaya.com”.
- **Direct IP-IP-Audio Connections:** Set to “y”

```
add signaling-group 92                             Page 1 of 2
                                                    SIGNALING GROUP
Group Number: 92
IMS Enabled? n
Q-SIP? n
IP Video? y
Peer Detection Enabled? y
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Near-end Listen Port: 5061
Far-end Listen Port: 5061
Far-end Network Region: 1
Far-end Domain:avaya.com
Group Type: sip
Transport Method: tls
Priority Video? y
Enforce SIPS URI for SRTP? y
Peer Server: SM
Bypass If IP Threshold Exceeded? n
RFC 3389 Comfort Noise? n
Direct IP-IP Audio Connections? y
DTMF over IP: rtp-payload
IP Audio Hairpinning? n
Session Establishment Timer(min): 3
Enable Layer 3 Test? y
Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n
Alternate Route Timer(sec): 6
```

5.5. Configure SIP Trunk

This section describes the steps for administering a trunk group in Communication Manager for trunking between Communication Manager and Session Manager. Enter the **add trunk-group <t>** command, where **t** is an unallocated trunk group and configure the following:

- **Group Type** – Set the Group Type field to “sip”.
- **Group Name** – Enter a descriptive name.
- **TAC (Trunk Access Code)** – Set to any available trunk access code.
- **Signaling Group** – Set to the Group Number field value configured in **Section 5.4**.
- **Number of Members** – Allowed value is between 0 and 255. Set to a value large enough to accommodate the number of SIP telephone extensions being used.

```
add trunk-group 92                                     Page 1 of 21
                                                    TRUNK GROUP
Group Number: 92                                     Group Type: sip          CDR Reports: y
Group Name: SM 41 42                                COR: 1                  TN: 1                TAC: 1092
Direction: two-way                                Outgoing Display? n
Dial Access? n                                    Night Service:
Queue Length: 0
Service Type: tie                                  Auth Code? n
                                                    Member Assignment Method: auto
                                                    Signaling Group: 92
                                                    Number of Members: 10
```

5.6. Configure Uniform Dial Plan

This section describes the steps for administering a uniform dial plan in Communication Manager. Enter **change uniform-dialplan <u>**, where **u** is the uniform-dialplan number. The following screen shows the Uniform Dial Plan configuration. The 5-digit extension range starting with 4200x was used for the Avaya S8300D Server with G430 Media Gateway side IP telephones, and utilized Automatic Alternate Routing (AAR).

```
change uniform-dialplan 2                             Page 1 of 2
                                                    UNIFORM DIAL PLAN TABLE
                                                    Percent Full: 0
Matching
Pattern      Len Del      Insert      Net Conv Num
4200         5  0         Digits     aar  n
```

5.7. Configure Automatic Alternate Routing

Enter **change aar analysis <a>**, where **a** is the AAR number. Automatic Alternate Routing (AAR) was used to route calls to the appropriate route pattern. The 5-digit extension range starting with 42 used the route pattern 10. 4200x extensions are H.323 IP phones in Avaya S8300D Server with Avaya G430 Media Gateway side. To call these H.323 IP phones from S8300D Server with Avaya G450 Media Gateway side, route pattern 10 will be used, which is an ISDN/PRI or IP trunk.

```
change aar analysis 2
```

Page 1 of 2

AAR DIGIT ANALYSIS TABLE
Location: all Percent Full: 3

Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Reqd
20004	5	5	91	unku		n
4200	5	5	10	aar		n
33	5	5	91	unku		n
415	10	10	92	aar		n
50000	5	5	92	unku		n
53005	5	5	91	unku		n

5.8. Configure Route Pattern

Enter **change route-pattern <r>**, where **r** is the route-pattern number. The route pattern 10 routes calls to the trunk group 10, which is either the IP or PRI trunk to Avaya S8300D Server with Avaya G430 Media Gateway.

```
change route-pattern 10
```

Page 1 of 3

Pattern Number: 10 Pattern Name: ToG430
SCCAN? n Secure SIP? n

Grp No	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/ IXC
		Mrk	Lmt	List	Del	Digits	Dgts	QSIG Intw
1:	10	0						n user
2:								n user
3:								n user

BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR
0	1	2	M	4	W	Request		Dgts	Format	Subaddress
1:	y	y	y	y	y	n	n		rest	none
2:	y	y	y	y	y	n	n		rest	none
3:	y	y	y	y	y	n	n		rest	none

5.9. Configure Off-PBX-Telephone Configuration-Set

SIP endpoints and off-pbx-telephone stations will be automatically created in Communication manager when users (SIP endpoints) were created in Session Manager.

However, the **off-pbx-telephone configuration-set** form needs to be modified. Enter **change off-pbx-telephone configuration-set**. Disable the **CDR for Calls to EC500 Destination?** field by setting it to “n”. Set the **CDR for Origination** field to “none”.

```
change off-pbx-telephone configuration-set 1                               Page 1 of 1

                                CONFIGURATION SET: 1

                                Configuration Set Description:
                                Calling Number Style: network
                                CDR for Origination: none
                                CDR for Calls to EC500 Destination? n
                                Fast Connect on Origination? n
                                Post Connect Dialing Options: dtmf
                                Cellular Voice Mail Detection: timed (seconds): 4
                                Barge-in Tone? n
                                Calling Number Verification? y
                                Call Appearance Selection for Origination: primary-first
                                Confirmed Answer? n

                                Use Shared Voice Connections for Second Call Answered? n
                                Use Shared Voice Connections for Second Call Initiated? n
                                Provide Forced Local Ringback for EC500? n
                                Apply Ringback upon Receipt of: Call-Proceeding
```

6. Configure Avaya Aura[®] Session Manager

This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

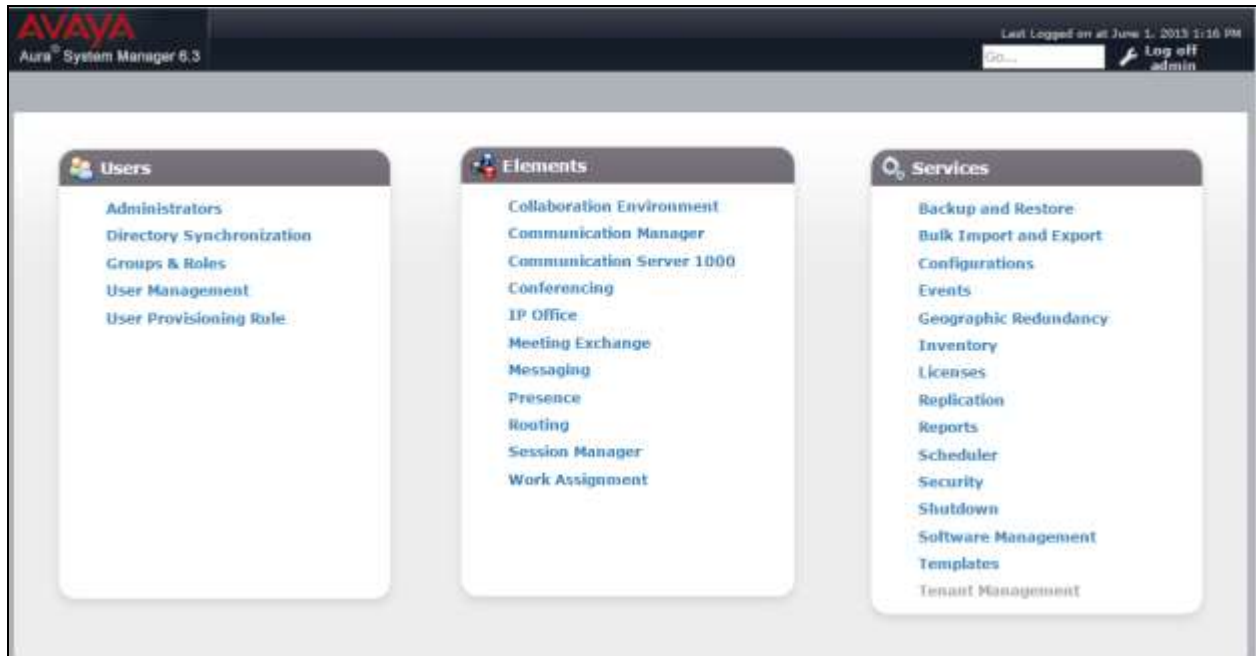
It is assumed that Session Manager and System Manager have been installed, network connectivity exists between the two platforms, and following topics are already configured:

- **SIP Domains**
- **Locations**
- **SIP Entities**
- **Entity Links**
- **Time Ranges**
- **Routing Policy**
- **Dial Patterns**
- **Manage Element**
- **Applications**
- **Application Sequence**

This section only discusses the User Management process to add SIP users that will be used during the compliance test.

6.1. Configure SIP Users

Launch a web browser, enter <http://<IP address of System Manager>> in the URL, and log in with the appropriate credentials.



During the compliance test, no special users were created for this solution. All users were created prior to the compliance test. However, steps to configure a user are included. When adding a new SIP user, use the option to automatically generate the SIP station in Communication Manager, after adding the new SIP user.

To add new SIP users, Navigate to **Home** → **Users** → **User management** → **Manage Users**. Click **New** (not shown) and provide the following information:

- Identity section
 - **Last Name** – Enter last name of user.
 - **First Name** – Enter first name of user.
 - **Login Name** – Enter extension number@sip domain. The sip domain is defined as Authoritative Domain in **Section 5.2**.
 - **Authentication Type** – Verify **Basic** is selected (default).

The screenshot shows the 'New User Profile' configuration page. The 'Identity' section is expanded, showing the following fields and values:

- User Provisioning Rule: [Dropdown]
- Last Name: 72021
- Last Name (Latin Translation): 72021
- First Name: 72021
- First Name (Latin Translation): 72021
- Middle Name: [Empty]
- Description: [Empty]
- Login Name: 72021@avaya.com
- Authentication Type: Basic
- Password: [Empty]
- Confirm Password: [Empty]
- Localized Display Name: SIP-1
- Endpoint Display Name: SIP-1

- Communication Profile section
 - **Communication Profile Password** – Enter a numeric value used to logon to SIP telephone.
 - **Confirm Password** – Repeat numeric password
 - Verify there is a default entry identified as the **Primary** profile for the new SIP user. If an entry does not exist, select **New** and enter values for the following required attributes:
 - **Name** – Enter **Primary**.
 - **Default** – Enter

The screenshot shows the 'New User Profile' configuration page. On the left is a navigation menu with options: Manage Users, Public Contacts, Shared Addresses, System Presence ACLs, Communication Profile, and Password Policy. The main area is titled 'New User Profile' and has tabs for Identity, Communication Profile, Membership, and Contacts. The 'Communication Profile' tab is active. It contains two password fields: 'Communication Profile Password' and 'Confirm Password', both masked with dots and highlighted with a red box. Below these is a toolbar with 'New', 'Delete', 'Done', and 'Cancel' buttons. A dropdown menu for 'Name' is open, showing 'Primary' as the selected option, also highlighted with a red box. Below the dropdown, the text 'Select : None' is visible. At the bottom, there is a field for '* Name:' containing 'Primary' and a 'Default:' checkbox which is checked.

- Communication Address sub-section

Select **New** to define a **Communication Address** for the new SIP user, and provide the following information.

- **Type** – Select **Avaya SIP** using the drop-down menu.
- **Fully Qualified Address** – Enter same extension number and domain used for Login Name, created previously.

Click the **Add** button to save the Communication Address for the new SIP user.

Communication Address ▼

<input type="checkbox"/>	Type	Handle	Domain
No Records found			

Type: Avaya SIP ▼

* Fully Qualified Address: 72021 @ avaya.com ▼

- Session Manager Profile section
 - **Primary Session Manager** – Select one of the Session Managers from the drop down list.
 - **Origination Application Sequence** – Select Application Sequence for Communication Manager from the drop down list.
 - **Termination Application Sequence** – Select Application Sequence for Communication Manager from the drop down list.
 - **Home Location** – Select a location already defined in the **Location** form.

Session Manager Profile ▾

SIP Registration

* Primary Session Manager ▾

Primary	Secondary	Maximum
7	0	7

Secondary Session Manager ▾

Survivability Server ▾

Max. Simultaneous Devices ▾

Block New Registration When Maximum Registrations Active?

Application Sequences

Origination Sequence ▾

Termination Sequence ▾

Call Routing Settings

* Home Location ▾

Conference Factory Set ▾

Call History Settings

Enable Centralized Call History?

- Endpoint Profile section
 - **System** – Select “Managed Element”, using the drop down menu.
 - **Profile Type** – Select “Endpoint”, using the drop down menu.
 - **Use Existing Endpoints** - Leave unchecked to automatically create new endpoint when new user is created. Or else, check the box if endpoint is already defined in Communication Manager.
 - **Extension** - Enter same extension number used in this section.
 - **Template** – Select template for type of SIP phone, using the drop down menu
 - **Security Code** – Enter numeric value used to logon to SIP telephone.
 - **Port** – Verify “IP” is shown for this field.
 - **Voice Mail Number** – Enter **Pilot Number** for Avaya Modular Messaging if installed. Or else, leave field blank. This feature is not used during the compliance test.
 - **Delete Endpoint on Unassign of Endpoint from User or on Delete User** – Check the box to automatically delete station when Endpoint Profile is unassigned from user.

CM Endpoint Profile

* System

* Profile Type

Use Existing Endpoints

* Extension

Template

Set Type

Security Code

Port

Voice Mail Number

Preferred Handle

Enhanced Callr-Info display for 1-line phones

Delete Endpoint on Unassign of Endpoint from User or on Delete User.

Override Endpoint Name and Localized Name

Click **Commit** (not shown) to save definition of the new user.

The following screen shows the created SIP users during the compliance test.

The screenshot displays the Avaya System Manager 6.3 User Management interface. The breadcrumb navigation shows 'Home / Users / User Management / Manage Users'. A search bar is present at the top. The main content area is titled 'User Management' and shows a list of 8 users. The table below contains the user details:

	Last Name	First Name	Display Name	Login Name	SIP Handle	Last Login
<input type="checkbox"/>	72026	SIP-softclient	72026, SIP-softclient	72026@avaya.com	72026	
<input type="checkbox"/>	Polycom	IP5000	72029, 72029	72029@avaya.com	72029	
<input type="checkbox"/>	72874	72874	72874, 72874	72874@avaya.com	72874	
<input type="checkbox"/>	Agent	97	Agent, 57	72097@avaya.com	72097	
<input type="checkbox"/>	admin	admin	Default Administrator	admin		June 3, 2015 11:00:24 AM -06:00
<input checked="" type="checkbox"/>	SIP	Station 1	SIP, Station 1	72021@avaya.com	72021	
<input checked="" type="checkbox"/>	SIP	Station 2	SIP, Station 2	72022@avaya.com	72022	
<input checked="" type="checkbox"/>	SIP	Station 3	SIP, Station 3	72023@avaya.com	72023	

The three SIP users (Station 1, 2, and 3) are highlighted with a red box in the original image. The interface also includes a 'Select: All, None' option at the bottom left of the table area.

7. Configure CallTech Dali® CDR Processing Billing system

This section describes the operation of Dali to receive CDR data from Communication Manager. Installation of the Dali software was performed by a CallTech engineer prior to the actual compliance test. In this section, the following topics are discussed:

- Configure Dali
- View Dali CDR report

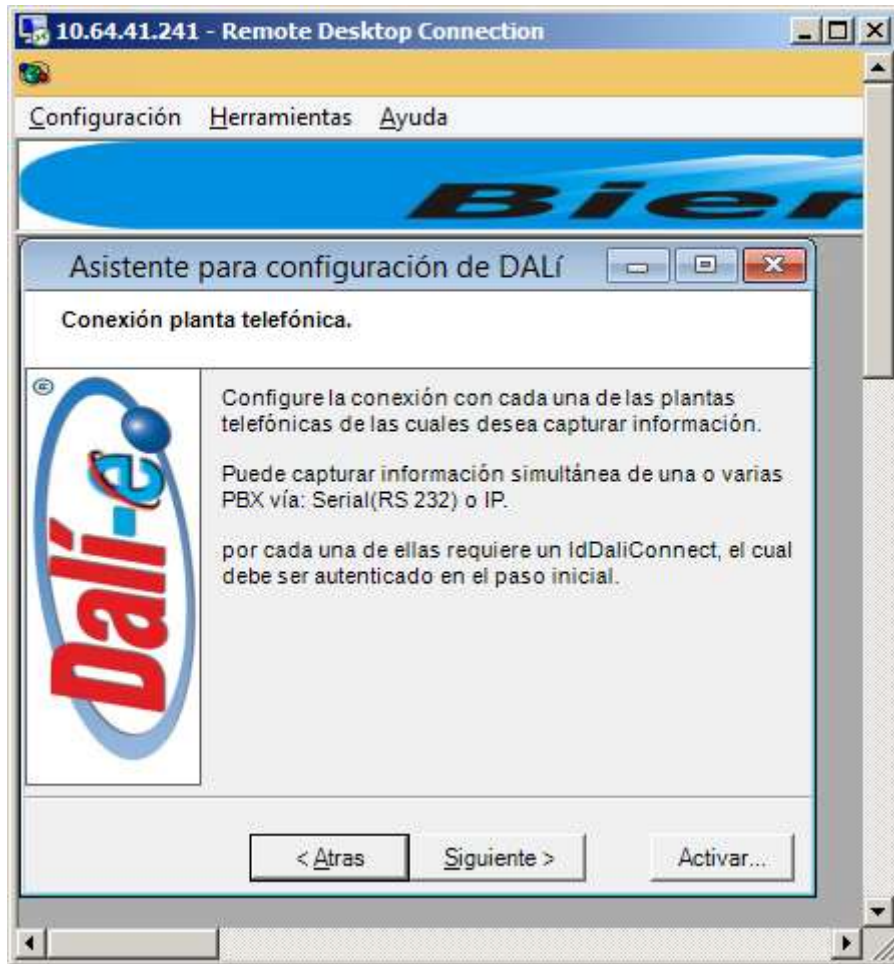
Note: At the time of testing, the Dali software was in Spanish, an English version was not available. A CallTech engineer installed the application in Spanish.

7.1. Configure Dali for Avaya Communication Manager

From the Windows 8 system, select **Administrador DaliConnect**.



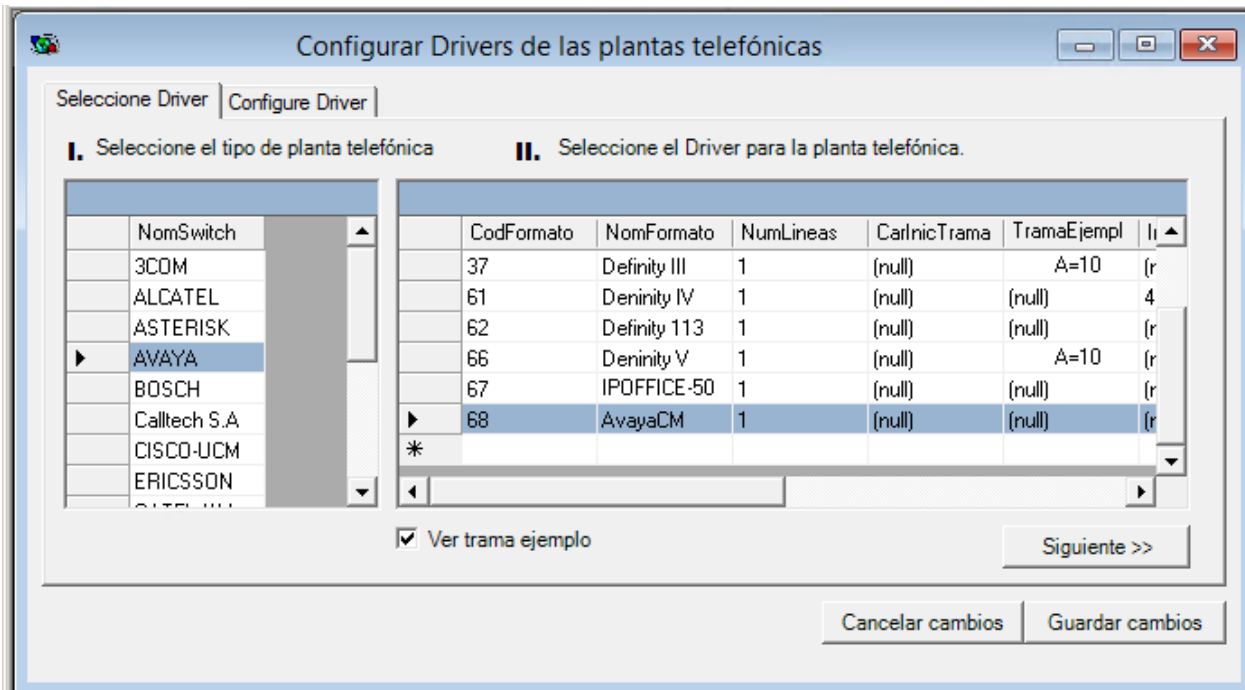
In the Remote Desktop Connection window, select the **Activar** button.



The **Configuración** page is displayed. Select **LAN/IP** tab, and click **NomFomato**.



On the following page, click on the **Seleccione Driver** tab. Select **Avaya** from the NomSwitch list on the left pane, and select **68** on the CodFomato of the right pane to select the proper driver. Select **Next (Siguiete)**.



From the previous page, click on the **Configure Driver** tab. Configure each field of the format under NombreCamp.

The screenshot shows a software interface with two tabs: 'Seleccione Driver' and 'Configure Driver'. The 'Configure Driver' tab is active. At the top, it says 'III. Revise el Driver de la planta telefónica.' followed by 'Switch: AVAYA' and 'Driver: AvayaCM'. Below this is a table with the following columns: NombreCamp, InicioCampo, LongCampo, Condicion, and CodFormato. The table contains the following data:

NombreCamp	InicioCampo	LongCampo	Condicion	CodFormato
ANI	50	14	(null)	68
Año	5	2	(null)	68
CodigoCuent	81	5	(null)	68
Día	3	2	(null)	68
DigitosMarca	29	20	(null)	68
DuracionHora	12	1	(null)	68
DuracionMinu	13	2	(null)	68

At the bottom right of the table area is a button labeled '<< Regresar'. At the bottom of the window are two buttons: 'Cancelar cambios' and 'Guardar cambios'.

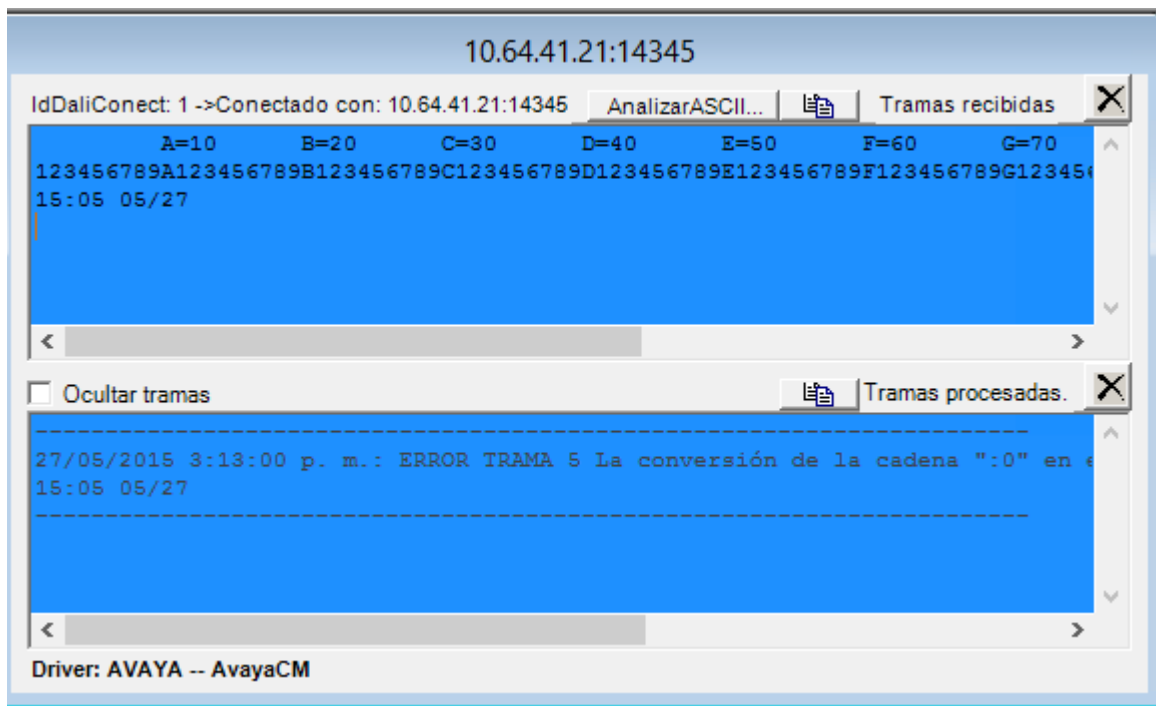
Note4: A CallTech engineer configured setting up a **format type** and **TCP/IP configuration** prior to the actual test. Please contact CallTech for above configuration issues.

7.2. View Dali CDR Report

To view the CDR report, launch the **DaliConnect** icon from the Main menu.



The next page shows raw data and processed data.



The next page shows when the call is in process.



To generate a report using the raw data, select the **Alarmas y Monitoreo** icon from the main menu.



The following page is displayed. Select initial and final dates for report.

Facturar

Facturación de llamadas telefónicas por demanda

Fecha Inicial:

◀ **mayo de 2015** ▶

lun.	mar.	mié.	jue.	vie.	sáb.	dom.
27	28	29	30	1	2	3
4	5	6	7	8	9	10
11	12	13	14	15	16	17
18	19	20	21	22	23	24
25	26	27	28	29	30	31
1	2	3	4	5	6	7

Hoy: 27/05/2015

Hora Inicial: :

Fecha Final:

◀ **mayo de 2015** ▶

lun.	mar.	mié.	jue.	vie.	sáb.	dom.
27	28	29	30	1	2	3
4	5	6	7	8	9	10
11	12	13	14	15	16	17
18	19	20	21	22	23	24
25	26	27	28	29	30	31
1	2	3	4	5	6	7

Hoy: 27/05/2015

Hora Final: :

Para volver a tarificar seleccione el rango de fechas y haga click en 'Volver a Facturar'

Facturación de llamadas telefónicas....

03:16:33 : Rango de fechas a mover entre 27/05/2015 3:07:00 p. m. y 27/05/2015 3:07:00 p. m.

03:16:33 : LlamadasSalientes a Mover: 1

03:16:33 : Ejecutando Proceso Movimiento de Llamadas Entrantes....

03:16:34 : MoverLlamadas-> UsarProcMoverLlamadas =

03:16:34 : LlamadasEntrantes a Mover: 0

03:16:34 : Verificando Sistema Prepago 27/05/2015 3:16:34 p. m.

03:16:34 : Fin Verificacion Sistema Prepago 27/05/2015 3:16:34 p. m.

Aplicar tarifas larga distancia, celulares, locales, etc.

CRK; Reviewed:
SPOC 8/25/2015

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Launch a web browser, enter <http://<IP address of Dali/DaliEnterprise/Login.aspx>> in the URL, and log in with the appropriate credentials.

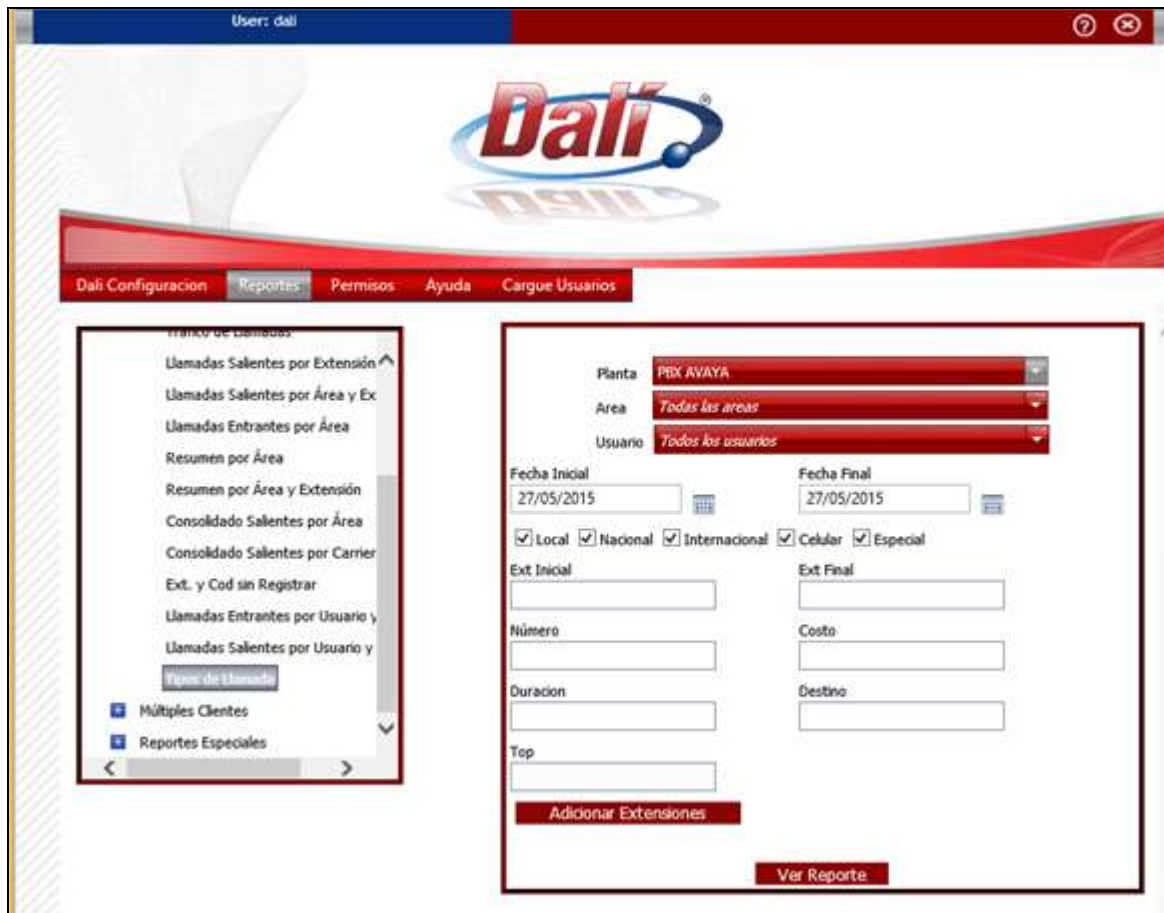


Navigate to **Reportes** → **Generar Reportes**.



Provide the following information on the Reportes page:

- On the right pane, select **PBX AVAYA** on the Planta field, using the drop-down menu.
- On the top menu, navigate to **Reportes** → **Multiples PBX** → **Tipos de Llamada** (not shown).
- Select **Ver Reporte** from the bottom of the right pane



The following shows the sample reports.

1 / 3 Main Report 100% Objects

Tipos de Llamada Planta telefónica: **PBX AVAYA**
 Periodo reporte: **miércoles, 27-may.-2015 -- miércoles, 27-may.-2015**

FechaHora	Origen	Destino	Duracion	CodCuenta	Pstntronal	Description
27/05/2015 8:54:00a.m.	72002	72022	8			Intra Switch
27/05/2015 8:55:00a.m.	72022	72001	9	88888		Intra Switch
27/05/2015 10:03:00a.m.	42001	72022	4			Incoming/Tandem
27/05/2015 10:26:00a.m.	3035382324	72021	2		1080	Ringin Station
27/05/2015 10:25:01a.m.	3035382324	72021	48		1080	Conference
27/05/2015 10:25:00a.m.	3035382324	72022	15		1080	Conference
27/05/2015 10:30:00a.m.	72021	13035382324	49	88888	1080	Conference
27/05/2015 10:38:00a.m.	72001	7777	14		1082	AAR:ARS
27/05/2015 10:49:00a.m.	72001	7777	10		1082	AAR:ARS
27/05/2015 10:52:00a.m.	72021	13035382324	0			Unsuccessful
27/05/2015 10:57:00a.m.	3035382324	72021	12		1080	Ringin Station
27/05/2015 10:57:01a.m.	3035382324	72021	7		1080	Incoming/Tandem
27/05/2015 10:59:00a.m.	3035382324	72021	14		1080	Abandoned
27/05/2015 11:01:00a.m.	72021	7777	300		1082	AAR:ARS
27/05/2015 11:01:01a.m.	3035382324	72021	300		1080	Incoming/Tandem
27/05/2015 11:02:00a.m.	3035382324	72021	0		1080	Busy Station
27/05/2015 11:04:00a.m.	72021	72001	145	88888		Intra Switch
27/05/2015 11:06:00a.m.	42001	72021	3			Incoming/Tandem
27/05/2015 11:06:01a.m.	42001	72022	5			Incoming/Tandem
27/05/2015 11:07:00a.m.	42001	72022	3			Incoming/Tandem
27/05/2015 11:08:00a.m.	42001	72084	5			Incoming/Tandem
27/05/2015 11:08:01a.m.	42001	72084	4			Incoming/Tandem
27/05/2015 11:09:00a.m.	42001	72084	5			Incoming/Tandem
27/05/2015 11:11:00a.m.	3035382324	72022	7		1080	Incoming/Tandem
27/05/2015 11:11:01a.m.	3035382324	72022	9		1080	Incoming/Tandem
27/05/2015 11:12:00a.m.	72022	13035382324	12	88888	1080	AAR:ARS
27/05/2015 11:19:00a.m.	72022	7777	0			Unsuccessful
27/05/2015 11:19:01a.m.	72022	72002	78	88888		Conference
27/05/2015 11:19:02a.m.	72022	72001	98	88888		Conference
27/05/2015 11:19:03a.m.	72022	72005	20	88888		Conference
27/05/2015 11:24:00a.m.	72022	42001	38	88888	1010	Conference
27/05/2015 11:24:01a.m.	72022	72001	47	88888		Conference
27/05/2015 11:25:00a.m.	72022	42001	29	88888	1010	Conference
27/05/2015 11:25:01a.m.	72022	72000	42	88888		Conference
27/05/2015 11:27:00a.m.	72022	42001	12		1010	Conference
27/05/2015 11:27:01a.m.	72022	72022	47	88888		Conference
27/05/2015 11:27:02a.m.	72022	42001	20		1010	Conference
27/05/2015 11:30:00a.m.	42001	72001	11			Conference
27/05/2015 11:30:01a.m.	42001	72021	34			Conference
27/05/2015 11:33:00a.m.	72021	42001	40	88888	1010	Conference
27/05/2015 11:33:01a.m.	72021	72001	30	88888		Conference
27/05/2015 11:34:00a.m.	72021	42001	14	88888	1010	Conference
27/05/2015 11:34:01a.m.	72021	72001	30	88888		Conference
27/05/2015 11:34:02a.m.	72021	42001	22	88888	1010	Conference

8. Verification Steps

The following steps may be used to verify the configuration:

- Check the CDR status, by running the **status cdr** command in Communication Manager, and verify the **Link State** is “up” and **Reason Code** is “OK”.

```
status cdr-link
                CDR LINK STATUS
                Primary                Secondary
Link State: up                down
Number of Retries:                999
Date & Time: 2015/05/28 14:36:47    2015/06/03 15:02:30
Forward Seq. No:11                0
Backward Seq. No: 0                0
CDR Buffer % Full: 0.82                0.00
Reason Code: OK                CDR connection is closed
```

- Make several SIP calls between two Communication Managers, and verify that call records were collected from Dali.

9. Conclusion

These Application Notes describe the procedures for configuring Dali to collect call detail records from Avaya Aura® Communication Manager. All executed test cases passed successfully.

10. References

This section references the Avaya and Dali documentation that are relevant to these Application Notes.

[1] *Administering Avaya Aura® Communication Manager*, Document 03-300509, Issue 10 Release 6.3, available at <http://support.avaya.com>.

[2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document 555-245-205, Release 6.3, Issue 12, available at <http://support.avaya.com>.

The Dali Solution and Product information is available from Dali. Visit https://www.Dali.com/literature_request.html

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