

Avaya Solution & Interoperability Test Lab

Application Notes for configuring Avaya Aura® Communication Manager R6.2, Avaya Aura® Session Manager R6.2 and Avaya Modular Messaging R5.2 with Orange Open Trade – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Orange Open Trade trading solution to interoperate with Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Modular Messaging. Orange Open Trade trading solution consists of trading turret endpoints attached to a server which communicates with Avaya Aura® Session Manager via a SIP trunk.

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

1. Introduction

These Application Notes describe the configuration used to enable Orange Business Services – Trading Solutions Open Trade to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Orange Open Trade trading solution consists of trading turret endpoints attached to proprietary Orange PBX which communicates with Avaya Aura® Session Manager via a SIP trunk.

Orange Business Services - Trading Solutions is a leading provider of convergent voice and electronic trading infrastructure and services for the trading communities. In 2010, Orange Business Services - Trading Solutions launched Open Trade. Open Trade is a complete solution that meets the mandatory high-demanding voice trading communication requirements. It is made of the Open Trade Smart Turrets (the terminal), the Open Trade Communication Manager and Plug-in Units. Open Trade is SIP compliant and interfaces with Avaya Aura® Session Manager via a SIP trunk

The Open Trade endpoints do not register with Session Manager. Calls to Open Trade endpoints from Avaya endpoints are established using appropriate call routing on both Avaya Aura® Communication Manager and Avaya Aura® Session Manager, as described in these application notes with Avaya Modular Messaging providing voicemail.

2. General Test Approach and Test Results

The general test approach was to validate correct operation of a variety of call handling scenarios and recovery from service interruption.

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

Interoperability compliance testing consisted of successful execution of the following features and link failure scenarios:

- Basic call incoming/outgoing, calling/called party terminates
- Call forwarding to/from Open Trade/Avaya endpoints
- Call Coverage
- Supervised Transfer/Blind transfer
- Name/number presentation
- Conferencing
- Voicemail related activities and MWI verification
- DTMF recognition
- Barge-In

Recovery from power failure and LAN failure to Open Trade PBX

2.2. Test Results

All test cases were performed successfully with the following observations:

• Open Trade is unable to propagate updates to the display name during a call. For example, where Avaya1 calls OT1 and OT1 transfers the call to Avaya2, both Avaya1 and Avaya2 show the connected party as OT1.

2.3. Support

Support for Open Trade is available at

- 0033-(0)144556699
- 0033-(0)670930326

3. Reference Configuration

An Avaya S8800 Server running Avaya Aura® Communication Manager R6.2 serving H323 endpoints with an Avaya G450 Media Gateway was configured along with Avaya Aura® Session Manager R6.2 hosted on an Avaya S8800 Server providing SIP endpoints. Etrali Open Trade was configured on a routable IP network and a SIP trunk was configured between the CAB interface of the Open Trade solution and the Avaya Aura® Session Manager SIP Signalling interface. Simulated connection to the PSTN was provided by en E1 QSIG trunk connected to the Avaya G450 Media Gateway. Open Trade endpoints comprised of both physical turrets and the Open Trace Soft Phone. Avaya Modular Messaging provided voicemail.

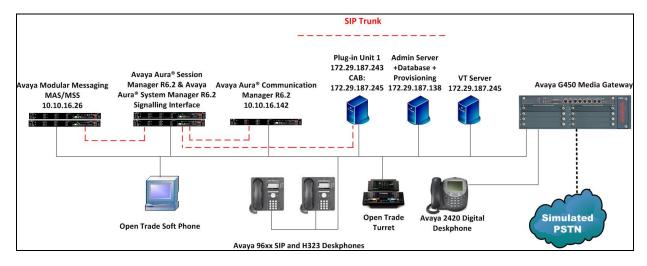


Figure 1: Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Modular Messaging with Open Trade Test Configuration

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	R6.2 SP4 build R016x.02.0.823.0-20199
running on Avaya S8800 Server	
Avaya Aura® Session Manager running on	R6.2 SP3
Avaya S8800 Server	
Avaya Aura® System Manager running on	R6.2 SP4
Avaya S8800 Server	
Avaya Modular Messaging running on	5.2 Patch 8
S3500 Servers	MAS - 9.2.150.13
Avaya 9630 IP Deskphone	• H323 S3.105S
	• SIP 2.6.8.4
Open Trade Administration	3.0.5.2
Open Trade Everywhere	1.0.4

5. Configure Avaya Aura® Communication Manager

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

5.1. Dialplan

In order that calls are routed to the extensions configured on the Open Trade solution, the dialplan must be configured accordingly using the **change dialplan analysis command**. In this case a **4** digit **Dialed String** beginning with **4** is routed to the uniform-dialplan (**udp**) table.

change diap	lan analysis	Page 1 of 12
		DIAL PLAN ANALYSIS TABLE
		Location: all Percent Full: 1
Dialed	Total Call	Dialed Total Call Dialed Total Call
String	Length Type	String Length Type String Length Type
1	3 fac	
2	10 udp	
3	11 udp	
4	4 udp	
5	4 ext	
6	4 ext	
7	3 dac	
8	4 udp	
9	1 fac	
*	3 fac	

5.2. Call Routing

The Application Notes assume that the relevant digital, SIP and H323 stations are configured and routing to Session Manager and the PSTN is in place. Use the **change uniform-dialplan 0** command and configure as shown below, where a matching patter of **4** with a **Length** of **4** digits is sent to the **aar** table.

change uniform-dialplan 0						Page	1 of	2	
UNIFORM DIAL PLAN TABLE					_		0		
							Perce	ent Full:	U
Matching			Insert			Node			
Pattern	Len	Del	Digits	Net	Conv	Num			
2	10	0		ars	n				
3	11	0		aar	n				
4	4	0		aar	n				
8	4	0		aar	n				
					n				

Use the **change aar analysis 0** command. Assign values for this command as shown in the following table. In this case the **Dialed String 4** digits in length is routed using **Route Pattern 1** where route pattern 1 is a preconfigured route to Session Manager.

change aar analysis 0						Page	1 of	2
AAR DIGIT ANALYSIS TABLE								
Location: all	Percent Full: 0							
Dialed	Tot	al	Route	Call	Node	ANI		
String	Min	Max	Pattern	Type	Num	Reqd		
3	11	11	1	unku		n		
4	4	4	1	unku		n		
402	4	4	4	aar		n		
5	4	4	1	aar		n		
5999	4	4	1	unku		n		
6000	4	4	1	unku		n		
6001	4	4	1	unku		n		
6002	4	4	1	unku		n		
6003	4	4	1	unku		n		
8000	4	4	1	unku		n		
8897	4	4	1	aar		n		

5.3. Configure Signalling Group

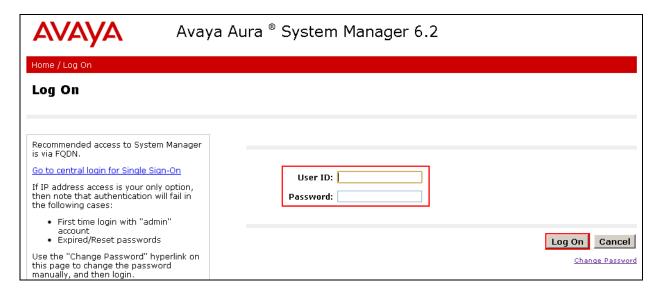
It is assumed the necessary signaling group and trunk configuration has been completed for the interface between Communication Manager and Session Manager. Enter the command **change signaling-group x** where **x** is the signaling group relevant to the trunk between Communication Manager and Session Manager, in this case **1.** Ensure that the items highlighted below are configured accordingly in order that the DTMF and shuffling features work as expected.

```
change signaling-group 1
                                                             Page
                                                                   1 of
                                                                          2
                               SIGNALING GROUP
Group Number: 1
                            Group Type: sip
 IMS Enabled? n
                      Transport Method: tcp
       Q-SIP? n
    IP Video? n
                                                Enforce SIPS URI for SRTP? y
 Peer Detection Enabled? y Peer Server: SM
  Near-end Node Name: procr
                                           Far-end Node Name: sm62sigint
Near-end Listen Port: 5060
                                        Far-end Listen Port: 5060
                                       Far-end Network Region: 1
Far-end Domain:
                                          Bypass If IP Threshold Exceeded? n
                                                 RFC 3389 Comfort Noise? n
Incoming Dialog Loopbacks: eliminate
        DTMF over IP: rtp-payload
                                          Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3
                                                    IP Audio Hairpinning? y
        Enable Layer 3 Test? y
                                              Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? y
                                             Alternate Route Timer(sec): 6
```

6. Configure Avaya Aura® Session Manager

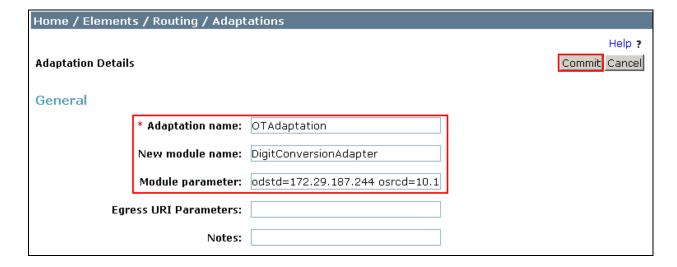
This section illustrates relevant aspects of the Avaya Aura® Session Manager configuration required for interoperating with Open Trade.

Session Manager is managed via Avaya Aura® System Manager. Using a web browser, access https://<ip-addr of System Manager>/SMGR. In the Log On screen, enter appropriate User ID and Password and press the Log On button



6.1. Configure Adaptation

In order for successful interoperation of Open Trade with Session Manager over a SIP trunk, an Adaptation must be configured. Click **Routing** → **Adaptations** → **New** enter a **Name** to identify this Adaptation, select **Click to add module** from the drop down box next to **Module Name** and set the **New Module Name** to **DigitConversionAdapter**. Set the **Module Parameter** to **odstd=172.29.187.244 osrcd=10.10.16.148 fromto=true**. For a SIP Entity using this Adaptation, the SIP messaging will be modified so that the destination domain (**odstd**) and source domain (**osrcd**) are changed to 172.29.187.244 and 10.10.16.148, the Open Trade CAB IP address and the Session Manager SIP signalling interface IP address respectively. Click **Commit** when done.



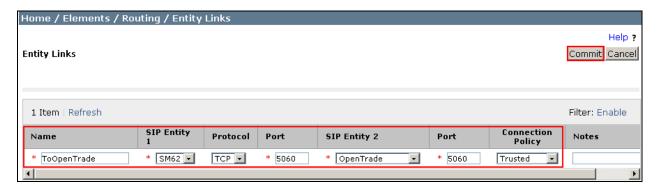
6.2. Configure Etrali Open Trade Entity

A SIP Entity must be created for the Open Trade CAB SIP interface. Click **Routing** → **SIP Entities** → **New** assign an identifying **Name**, the **FQDN or IP Address** for the Open Trade CAB SIP interface, set the **Type** to **SIP Trunk**, choose the Adaptation configured in **Section 6.1** from the drop down box and click **Commit** when done.



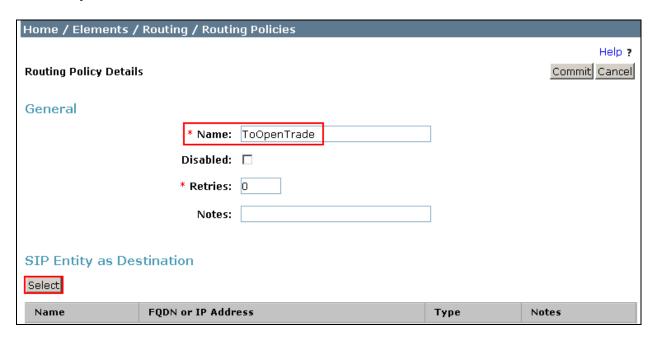
6.3. Configure Entity Link

The configuration of an Entity Link connects the Session Manager SIP Entity with the Open Trade CAB SIP Entity. Click **Routing** \rightarrow **Entity Links** \rightarrow **New** assign an identifying **Name** choose the entity assigned to the preconfigured Session Manager SIP Signaling Interface as **SIP Entity 1**, set the **Protocol** as **TCP**, enter **5060** for the Port, choose the Open Trade CAB SIP entity as **SIP Entity 2** and set the **Port** to **5060**, select **Trusted** from the **Connection Policy** drop down box. Click **Commit** when done. This establishes the Session Manager end of the SIP Trunk to Open Trade.

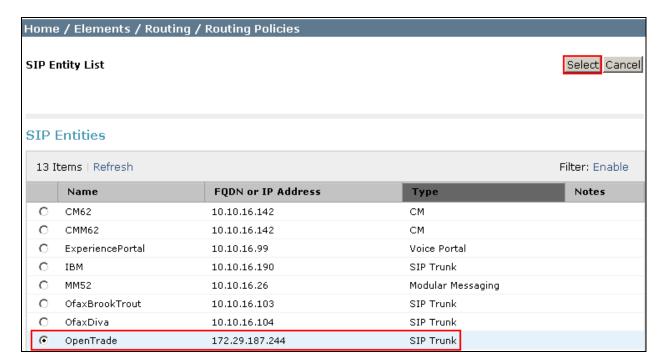


6.4. Create Routing Policy

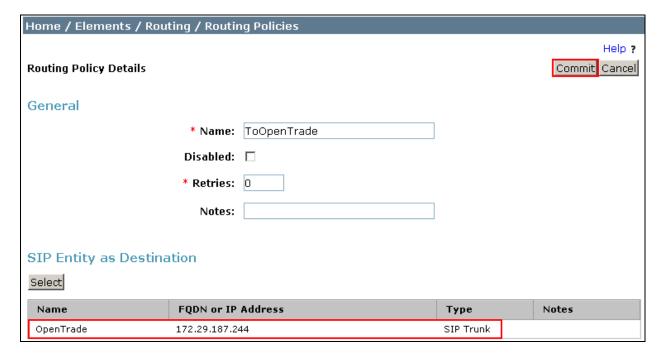
Click **Routing** → **Routing Polices** → **New** assign an indentifying **Name** for the route. Under the **SIP Entity as Destination** section, click on **Select**.



Choose the Open Trade Entity configured in **Section 6.2** and click **Select**.

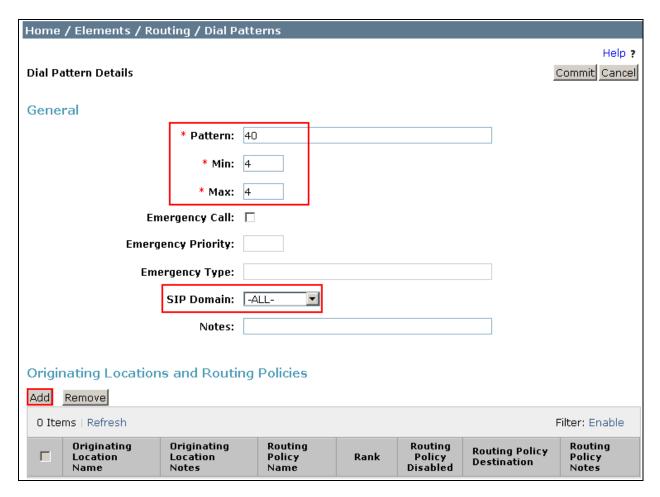


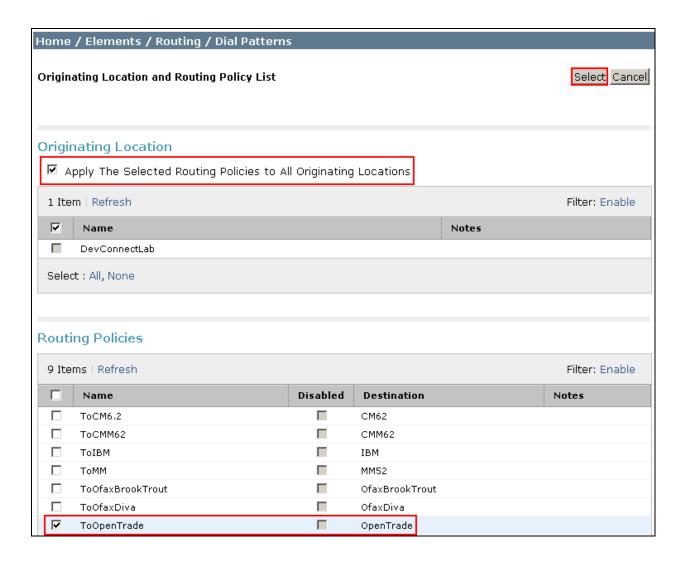
Review the configuration and click **Commit** when done.

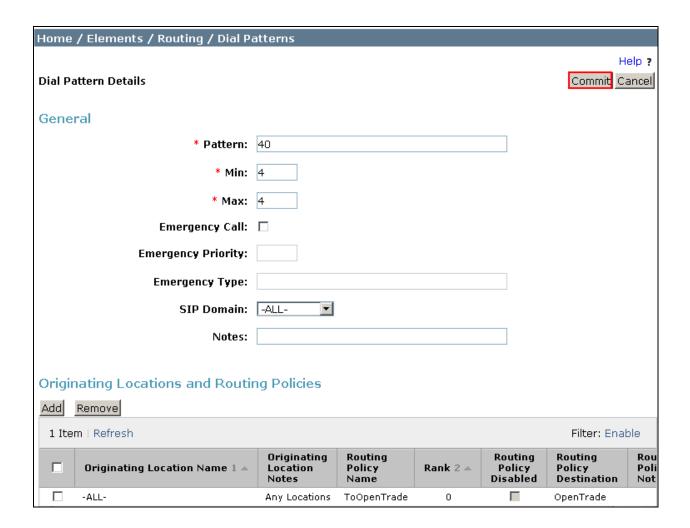


6.5. Administer Dial Patterns

Session Manager routes SIP traffic between connected devices. Dial Patterns are created as part of the configuration to manage SIP traffic routing, which will direct calls based on the number dialed to the appropriate destination. In **Section 5.2** Communication Manager is configured to route 4 digit strings beginning with 4 to Session Manager. To create a Dial Pattern to route these digits from Session Manager to Open Trade click **Routing** → **Dial Patterns** → **New**. Under Pattern enter the numbers presented to Session Manager by Communication Manager in the Patterns box. Set the Min and Max digit string length, and set SIP Domain to ALL. In the **Originating Locations and Routing Policies** section of the web page, click **Add**.

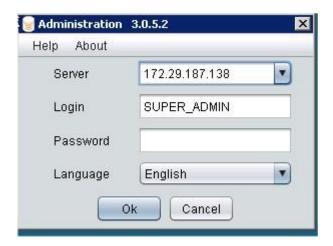






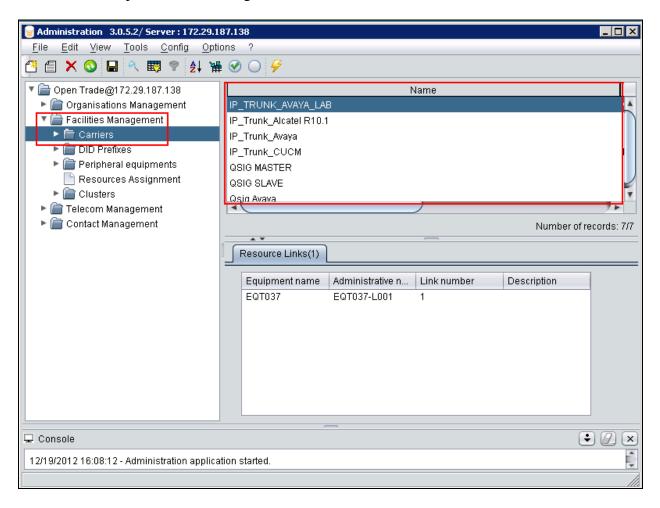
7. Configure Orange Open Trade Server

Start the Open Trade Administration program, and log in with the appropriate credentials.

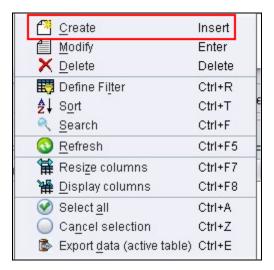


7.1. Create Carrier

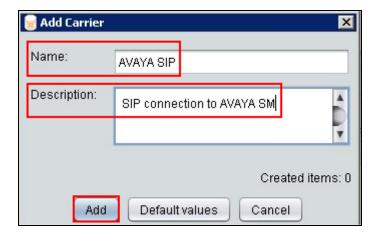
Expand the **Facilities Management** icon and select the **Carriers** menu element. Place the cursor under the **Name** pane header and right-click the mouse.



Select **Create** from the menu which appears.

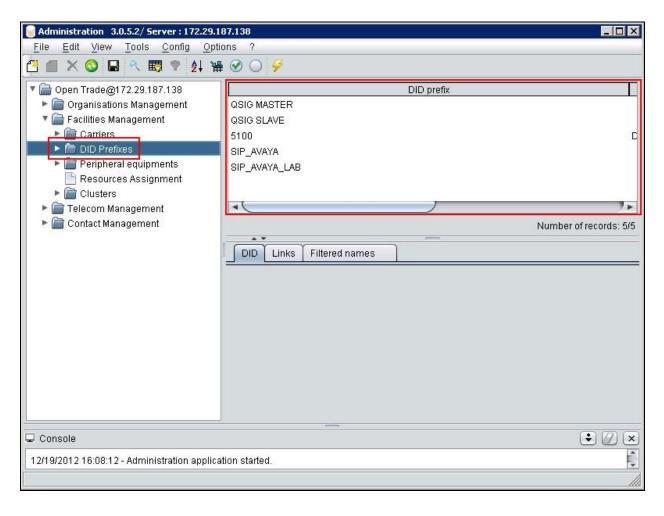


Enter an appropriate Name and Description and click Add.

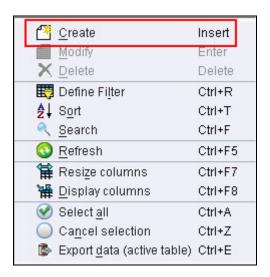


7.2. Create DID Prefixes

Expand the **DID Prefixes** menu item. Place the cursor in the **DID prefix** pane and right-click the mouse.



Select Create from the menu which appears

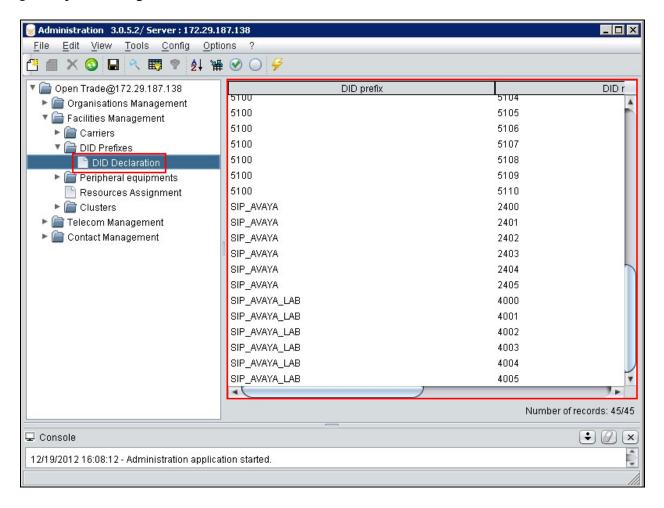


Enter an identifying name in the **DID prefix** field and click **Add.**

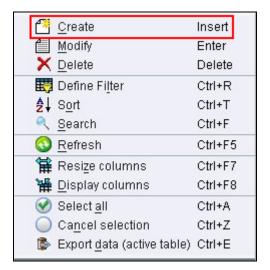


7.3. Create DID Declaration

Expand the **DID Declaration** menu item. Place the cursor under the first blank entry in the **DID prefix** pane and right-click the mouse.

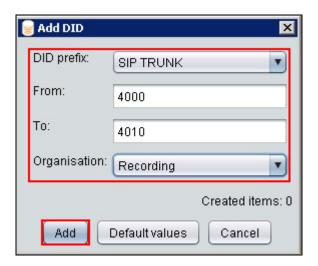


Select **Create** from the menu which appears.



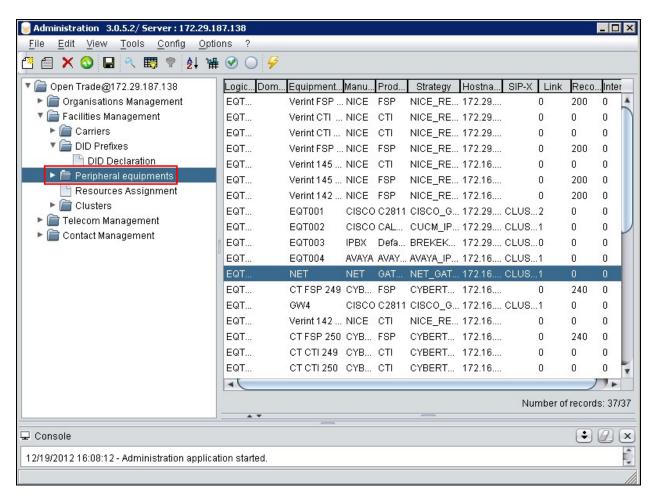
Enter the following values in the **Add DID** screen which appears and click **Add**.

- **DID prefix** select the DID Prefix created in **Section 7.2**
- **From** enter the first number of the range of numbers Open Trade will receive call for. In this case **4000**.
- To enter the last number of the range of numbers Open Trade will receive calls for. In this case 4010.
- **Organization** select the name of the appropriate Organization from the drop-down menu, this is preconfigured and the details are not covered here.

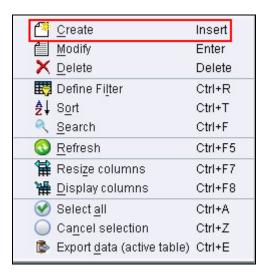


7.4. Create Peripheral Equipment

This section relates to the Avaya components from/to SIP traffic will be routed. Expand the **Peripheral equipments** menu item, Place the cursor under the first blank entry in the right hand pane and right-click the mouse.

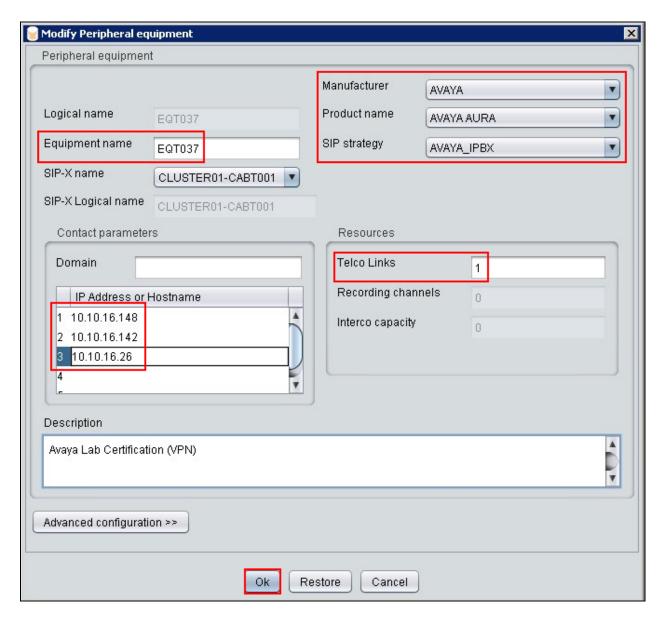


Select Create from the menu which appears.



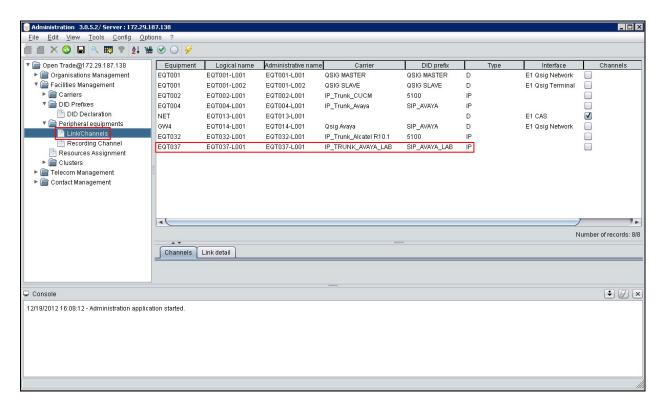
Enter the following values in the **Modify Peripheral equipment** screen which appears and click **Ok**.

- **Manufacturer** select **AVAYA** from the drop down list.
- **Product name** select **AVAYA AURA** from the drop down list.
- SIP Strategy select AVAYA_IPBX from the drop down list.
- **Equipment name** enter an identifying name.
- **IP Address or Hostname** enter the IP address of Communication Manager, Session Manager and Modular Messaging.
- Telco links enter 1



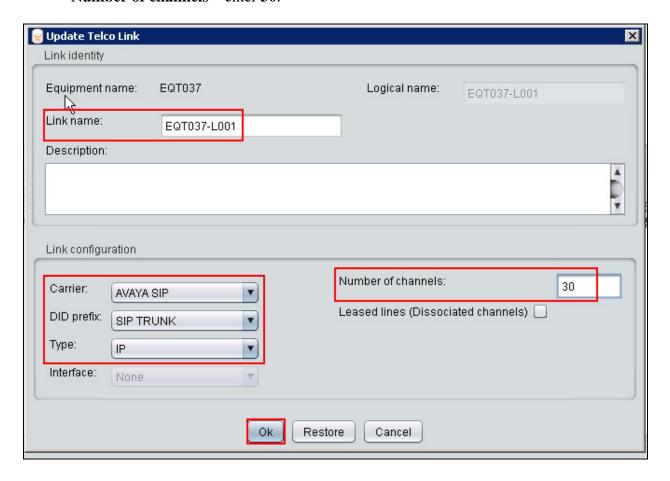
7.5. Create Link/Channels

Expand the **Link/Channels** menu item and double click the newly created Equipment created in **Section 7.4**.



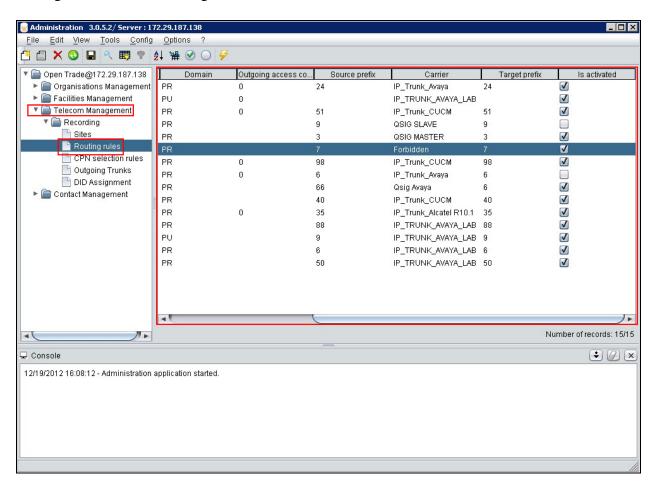
Enter the following values in the **Update Telco Link** screen which appears and click **Ok**.

- **Link name** enter an identifying name.
- Carrier select the Carrier created in **Section 7.1**.
- **DID prefix** select the DID prefix created in **Section 7.2**.
- **Type** select **IP** from the drop down list.
- Number of channels enter 30.

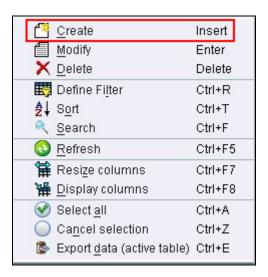


7.6. Create Routing Rule

Expand the **Telecom Management** item and select the **Routing rules** menu element. Place the cursor under the first blank entry in the right hand pane and right-click the mouse to create a new routing rule to Session Manager.

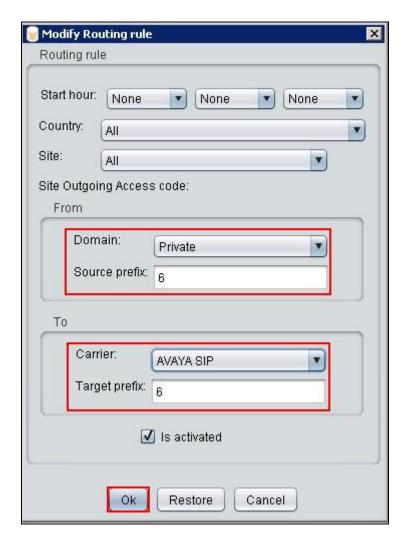


Select Create from the menu which appears.



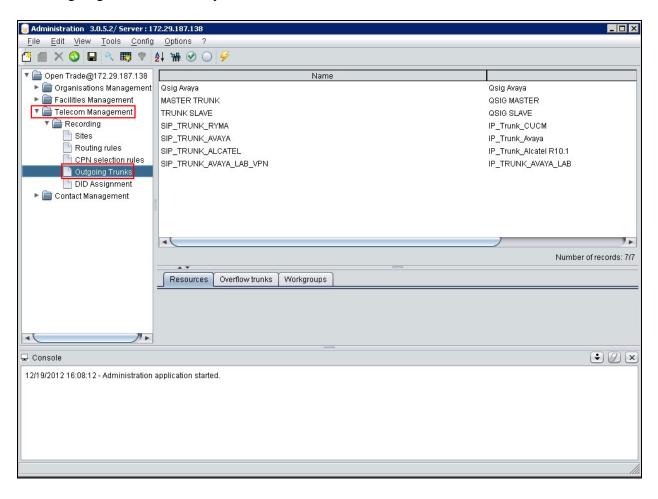
Enter the following values in the **Modify Routing rule** screen which appears and click **Ok**.

- **Domain** select **Private** from the drop down list.
- **Source prefix** enter the first digit of the extension range configured for Avaya endpoints.
- Carrier enter the Carrier configured in Section 7.1.
- **Target prefix** enter the first digit of the extension range configured for Avaya endpoints.

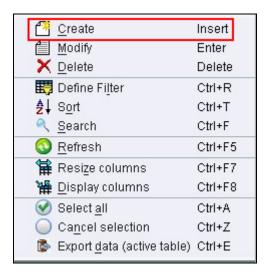


8. Create Outgoing Trunk

Expand the **Telecom Management** item and select the **Outgoing Trunks** menu element. Place the cursor under the first blank entry in the right hand pane and right-click the mouse to create a new outgoing trunk to the Avaya solution.

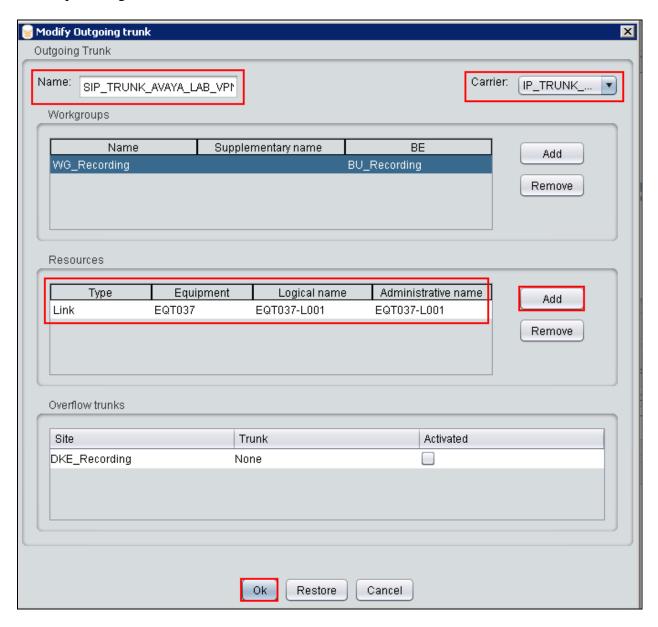


Select "Create" from the menu which appears.



Enter the following values in the **Modify Outgoing trunk** screen which appears and click **Ok**.

- Carrier select the carrier created in **Section 7.1** from the drop down list.
- **Resources** add the peripheral equipment created in **Section 7.4**.
- **Workgroup** add the appropriate Workgroup from the drop-down menu, this is preconfigured and the details are not covered here.



9. Verification Steps

The correct installation and configuration of Orange Open Trade trading solution can be verified by performing the following steps shown below.

9.1. Verify Communication Manager SIP Trunk

Using the SAT terminal, enter the **status signaling-group <n>** command, where <n> is the number of the SIP signaling group which connects to Session Manager. Verify that the signaling **Group State** is **in-service**.

```
Status signaling-group 1

STATUS SIGNALING GROUP

Group ID: 1

Group Type: sip

Group State: in-service
```

9.2. Verify Open Trade Softphone Functionality

Place a call to/from an Open Trade turret extension number, ensure the call can be answered, controlled and terminated as expected. Leave a voicemail on the mailbox of the Open Trade extension, ensure that the message waiting indicator illuminates, highlighted below, and extinguishes when the voicemail is retrieved.



10. Conclusion

These Application Notes describe the compliance testing of the Orange Open Trade trading solution with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Open Trade passed all of the tests performed with observations noted in **Section 2.2**.

11. References

This section references documentation relevant to these Applications. Avaya product documentation, including the following, is available at http://support.avaya.com

• Administering Avaya Aura® Communication Manager, Release 6.2, 03-300509, Issue 7.0 December 2012

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