



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.0.1 as an Evolution Server, Avaya Aura® Session Manager R6.1 and Avaya Aura® Session Border Controller to support Vodafone DE SIP Trunking Service for IP-PBX- Issue 1.0

Abstract

These Application Notes describe the steps used to configure Session Initiation Protocol (SIP) trunking between the Vodafone DE SIP Trunking service and an Avaya SIP enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Session Border Controller, Avaya Aura® Session Manager and Avaya Aura® Communication Manager as an Evolution Server. Vodafone DE is a member of the DevConnect Service Provider program.

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 steps used to configure Session Initiation Protocol (SIP) trunking between Vodafone DE SIP Trunking service for IP-PBX and an Avaya SIP-enabled Enterprise Solution. The Avaya solution consists of Avaya Aura® Session Border Controller Avaya Aura® Session Manager and Avaya Aura® Communication Manager Evolution Server. Customers using this Avaya SIP-enabled enterprise solution with Vodafone DE SIP trunking service are able to place and receive PSTN calls via a dedicated Internet connection and the SIP protocol. This converged network solution is an alternative to traditional PSTN trunks. This approach generally results in lower cost for the Enterprise customer.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise site using an Avaya SIP telephony solution consisting of Communication Manager, Session Manager and Session Border Controller. The enterprise site was configured to use the SIP Trunk Service provided by Vodafone DE.

2.1. Interoperability Compliance Testing

The interoperability test included the following:

- Incoming calls to the enterprise site from the PSTN routed to the DDI numbers assigned by Vodafone DE
- Incoming PSTN calls made to SIP, H.323 and Digital telephones at the enterprise
- Outgoing calls from the enterprise site completed via Vodafone DE to PSTN destinations
- Outgoing calls from the enterprise to the PSTN made from SIP, H.323 and Analogue telephones
- Calls using G.729, and G.711A codecs
- Fax calls to/from a group 3 fax machine to a PSTN connected fax machine using pass-through mode
- DTMF transmission using RFC 2833 with successful Voice Mail/Vector navigation for inbound and outbound calls
- User features such as hold and resume, transfer, conference, call forwarding, etc
- Caller ID Presentation and Caller ID Restriction
- Direct IP-to-IP media (also known as “shuffling”) with SIP and H.323 telephones
- Call coverage and call forwarding for endpoints at the enterprise site
- Transmission and response of SIP OPTIONS messages sent by Vodafone DE requiring Avaya response and sent by Avaya requiring Vodafone DE response

2.2. Test Results

Interoperability testing of the sample configuration was completed with successful results for the Vodafone DE SIP trunking service with the following observations:

- No inbound toll free numbers were tested as none were available from the Service Provider
- No Emergency services numbers tested as test calls to these numbers should be pre-arranged with the Operator
- G.711mu is not offered by Vodafone DE SIP trunking service and thus incoming calls were not tested
- Outgoing calls succeed where G.711mu is offered by the Avaya equipment
- G.711 pass-through was tested for fax as T.38 is not supported by Vodafone DE SIP trunking

2.3. Support

For technical support on Vodafone DE products please visit the website at www.vodafone.de or contact an authorized Vodafone DE representative.

3. Reference Configuration

Figure 1 illustrates the test configuration. The test configuration shows an Enterprise site connected to the Vodafone DE SIP Trunking Service. Located at the Enterprise site is a Session Border Controller, Session Manager and Communication Manager. Endpoints are Avaya 96x0 series and Avaya 96x1 series IP telephones (with SIP and H.323 firmware), Avaya 46xx series IP telephones (with H.323 firmware), Avaya 16xx series IP telephones (with SIP firmware) Avaya analogue telephones and an analogue fax machine. Also included in the test configuration was a one-X® Communicator soft phone running on a laptop PC configured for H.323.

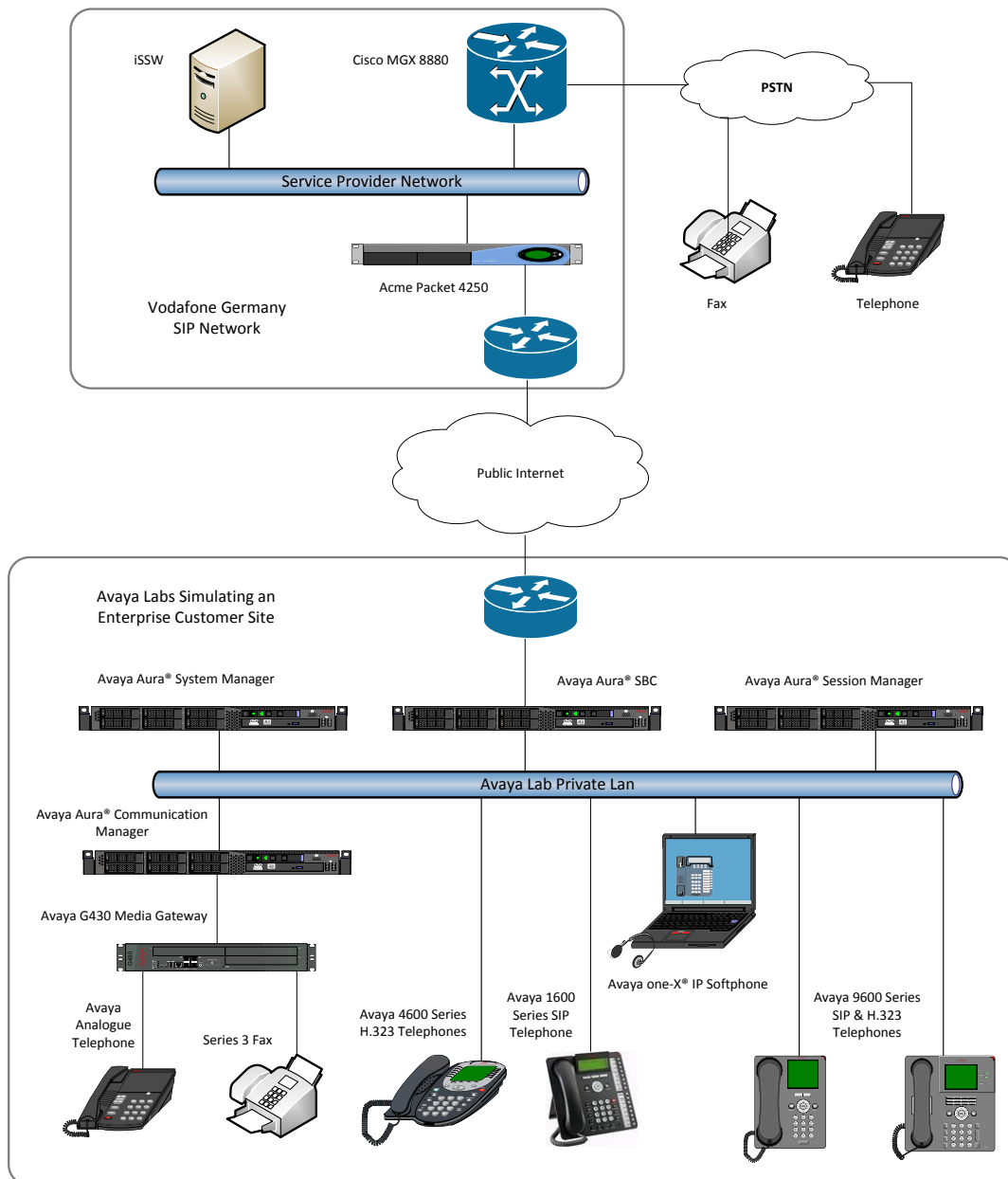


Figure 1: Test Setup Vodafone DE SIP Trunking to Avaya Enterprise

4. Equipment and Software Validated

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

4.1. Avaya Enterprise

Equipment	Software
Avaya S8800 Server	Avaya Aura® Communication Manager R6.0.1 (R016x.00.1.510.1) Service Pack 19009 (System Platform 6.0.3.1.3)
Avaya G430 Media Gateway	FW 30.12.1
Avaya S8800 Server	Avaya Aura® Session Manager R6.1 (6.1.0.0.610023)
Avaya S8800 Server	Avaya Aura® System Manager R6.1 (System Platform 6.0.3.1.3, Template 6.1.5.0)
Avaya S8800 Server	Avaya Aura® Session Border Controller 6.1 (System Platform 6.0.3.0.3, Template E362P4)
Avaya 1616 Phone (H.323)	1.22
Avaya 4621 Phone (H.323)	2.901
Avaya 9670 Phone (H.323)	2.0
Avaya 9601 Phone (SIP)	1.0.11.3
Avaya one-X® Communicator (H.323)	Avaya one-X® Communicator 6.0.1.16-SP1-25226
Analogue Phone	N/A

4.2. Vodafone DE SIP Trunking Service:

Equipment	Software
Softswitch: iSSW	20.50.34-olxi
SBC: Acme Packet 4250	6.1.0 MR5
MG: Cisco MGX 8880	5.5(10.204)P4

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring Communication Manager for SIP Trunking. SIP trunks are established between Communication Manager and Session Manager. These SIP trunks will carry SIP Signaling associated with the Vodafone DE SIP Trunking Service. For incoming calls, the Session Manager receives SIP messages from the SBC and directs the incoming SIP messages to Communication Manager. Once the message arrives at Communication Manager, further incoming call treatment, such as incoming digit translations and class of service restrictions may be performed. All outgoing calls to the PSTN are processed within Communication Manager and may be first subject to outbound features such as automatic route selection, digit manipulation and class of service restrictions. Once Communication Manager selects a SIP trunk, the SIP signaling is routed to the Session Manager. The Session Manager directs the outbound SIP messages to the Session Border Controller at the enterprise site that then sends the SIP messages to the Vodafone DE network. Communication Manager Configuration was performed using the System Access Terminal (SAT). Some screens in this section have been abridged and highlighted for brevity and clarity in presentation. The general installation of the Avaya S8800 Server and Avaya G430 Media Gateway is presumed to have been previously completed and is not discussed here.

5.1. Confirm System Features

The license file installed on the system controls the maximum values for these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative to add additional capacity. Use the **display system-parameters customer-options** command and on **Page 2**, verify that the **Maximum Administered SIP Trunks** supported by the system is sufficient for the combination of trunks to the Vodafone DE network, and any other SIP trunks used.

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:		12000	0
Maximum Concurrently Registered IP Stations:		18000	3
Maximum Administered Remote Office Trunks:		12000	0
Maximum Concurrently Registered Remote Office Stations:		18000	0
Maximum Concurrently Registered IP eCons:		414	0
Max Concur Registered Unauthenticated H.323 Stations:		100	0
Maximum Video Capable Stations:		18000	0
Maximum Video Capable IP Softphones:		18000	0
Maximum Administered SIP Trunks:		24000	10
Maximum Administered Ad-hoc Video Conferencing Ports:		24000	0
Maximum Number of DS1 Boards with Echo Cancellation:		522	0
Maximum TN2501 VAL Boards:		128	0
Maximum Media Gateway VAL Sources:		250	1
Maximum TN2602 Boards with 80 VoIP Channels:		128	0
Maximum TN2602 Boards with 320 VoIP Channels:		128	0
Maximum Number of Expanded Meet-me Conference Ports:		300	0

On **Page 4**, verify that **IP Trunks** field is set to **y**.

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? y	Local Survivable Processor? n	
Extended Cvg/Fwd Admin? y	Malicious Call Trace? y	
External Device Alarm Admin? y	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? y	Multifrequency Signaling? y	
Global Call Classification? y	Multimedia Call Handling (Basic)? y	
Hospitality (Basic)? y	Multimedia Call Handling (Enhanced)? y	
Hospitality (G3V3 Enhancements)? y	Multimedia IP SIP Trunking? y	
IP Trunks? y		
IP Attendant Consoles? y		

5.2. Administer IP Node Names

The node names defined here will be used in other configuration screens to define a SIP signaling group between Communication Manager and Session Manager. In the **IP Node Names** form, assign the node **Name** and **IP Address** for the Session Manager. In this case, **SM100** and **10.10.9.61** are the **Name** and **IP Address** for the Session Manager SIP interface. Also note the **procr** name as this is the processor interface that Communication Manager will use as the SIP signaling interface to Session Manager.

display node-names ip		IP NODE NAMES
Name	IP Address	
SM100	10.10.9.61	
default	0.0.0.0	
procr	10.10.9.52	
procr6	::	

5.3. Administer IP Network Region

Use the **change ip-network-region 1** command to set the following values:

- The **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is **avaya.com**.
- By default, **IP-IP Direct Audio** (both **Intra-** and **Inter-Region**) is enabled (**yes**) to allow audio traffic to be sent directly between endpoints without using gateway VoIP resources. When a PSTN call is shuffled the enterprise endpoint will talk directly to the public interface of the Vodafone DE Session Border Controller.
- The **Codec Set** is set to the number of the IP codec set to be used for calls within the IP network region. In this case, codec set **1** is used.

change ip-network-region 1		Page 1 of 20
IP NETWORK REGION		
Region: 1		
Location: 1	Authoritative Domain: avaya.com	
Name: default		
MEDIA PARAMETERS		
Codec Set: 1	Intra-region IP-IP Direct Audio: yes	
	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		
AUDIO RESOURCE RESERVATION PARAMETERS		
H.323 IP ENDPOINTS	RSVP Enabled? n	
H.323 Link Bounce Recovery? y		
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

5.4. Administer IP Codec Set

Open the **IP Codec Set** form for the codec set specified in the **IP Network Region** form, **Section 5.3**. Enter the list of audio codec's eligible to be used in order of preference. For the interoperability test the codec's supported by Vodafone DE were configured, namely **G.711A**, **G.726A-32K** and **G.729A**. During compliance testing, other codec set configurations were also verified.

change ip-codec-set 1		Page 1 of 2	
IP Codec Set			
Codec Set: 1			
Audio	Silence	Frames	Packet
Codec	Suppression	Per Pkt	Size (ms)
1: G.711A	n	2	20
2: G.726A-32K	n	2	20
3: G.729A	n	2	20

Vodafone DE SIP Trunk Service does not support the T.38 fax protocol. Fax pass-through was tested using G.711. Navigate to **Page 2** to configure fax pass-through by setting the **Fax Mode** to **pass-through** as shown below.

change ip-codec-set 1

Page 2 of 2

IP Codec Set

Allow Direct-IP Multimedia? n

	Mode	Redundancy
FAX	pass-through	0
Modem	off	0
TDD/TTY	US	3
Clear-channel	n	0

5.5. Administer SIP Signaling Groups

This signaling group (and trunk group) will be used for inbound and outbound PSTN calls to Vodafone DE SIP Trunking Service. During test, this was configured to use TCP and port 5060 to facilitate tracing and fault analysis. It is recommended however, to use TLS (Transport Layer Security) and the default TLS port of 5061 for security. Configure the **Signaling Group** using the **add signaling-group x** command as follows:

- Set **Group Type** to **sip**
- Set **Transport Method** to **tcp**
- Set **Peer Detection Enabled** to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager
- Set **Near-end Node Name** to the processor interface (node name **procr** as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Far-end Node Name** to the Session Manager (node name **SM100** as defined in the **IP Node Names** form shown in **Section 5.2**)
- Set **Near-end Listen Port** and **Far-end Listen Port** to 5060 (recommended TCP port value)
- Set **Far-end Network Region** to the IP Network Region configured in **Section 6.2**. (logically establishes the **far-end** for calls using this signaling group as network region 1)
- Leave **Far-end Domain** blank (removes the analysis of the far end domain name and subsequent handling of multiple signaling groups where it is not required)
- Set **Direct IP-IP Audio Connections** to **y**
- Leave **DTMF over IP** at default value of **rtp-payload** (Enables **RFC2833** for DTMF transmission from the Communication Manager)

The default values for the other fields may be used.

add signaling-group 1		Page 1 of 1
SIGNALING GROUP		
Group Number: 1	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n	SIP Enabled LSP? 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: SM100	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain:		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

5.6. Administer SIP Trunk Group

A trunk group is associated with the signaling group described in **Section 5.5**. Configure the trunk group using the **add trunk-group x** command, where **x** is an available trunk group. On **Page 1** of this form:

- Set the **Group Type** field to **sip**
- Choose a descriptive **Group Name**
- Specify a trunk access code (**TAC**) consistent with the dial plan
- The **Direction** is set to **two-way** to allow incoming and outgoing calls
- Set the **Service Type** field to **tie**
- Specify the signaling group associated with this trunk group in the **Signaling Group** field as previously configured in **Section 5.5**
- Specify the **Number of Members** supported by this SIP trunk group

add trunk-group 1		Page 1 of 21	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: y	
Group Name: Group 1	COR: 1	TN: 1	TAC: 101
Direction: two-way	Outgoing Display? y		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Member Assignment Method: auto	
		Signaling Group: 1	
		Number of Members: 10	

On **Page 2** of the trunk-group form the **Preferred Minimum Session Refresh Interval (sec)** field should be set to a value mutually agreed with Vodafone DE to prevent unnecessary SIP messages during call setup. Also note that the value for **Redirect On OPTIM Failure** can be increased to allow additional set-up time for calls destined for an EC500 destination.

add trunk-group 1		Page 2 of 21	
Group Type: sip			
TRUNK PARAMETERS			
Unicode Name: auto			
		Redirect On OPTIM Failure: 5000	
SCCAN? n	Digital Loss Group: 18		
		Preferred Minimum Session Refresh Interval(sec): 600	

On **Page 3**, set the **Numbering Format** field to **public**.

add trunk-group 1		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n		Measured: none	
		Maintenance Tests? y	
Numbering Format: public			
		UII Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	

On **Page 4**, set the **Convert 180 to 183 for Early Media** to **n**. If the 183 Session Progress message is received by Vodafone DE SIP Trunking, ring tone is expected by the terminating equipment. The Avaya G430 Media gateway does not play ring tone, so none is heard by the caller. The default value for **Network Call Redirection** is **n**, but the setting is changed on the test system to facilitate testing of User to User Information.

add trunk-group 1		Page 4 of 21	
PROTOCOL VARIATIONS			
Mark Users as Phone? n			
Prepend '+' to Calling Number? n			
Send Transferring Party Information? n			
Network Call Redirection? y			
Send Diversion Header? n			
Support Request History? y			
Telephone Event Payload Type: 101			
Convert 180 to 183 for Early Media? n			
Always Use re-INVITE for Display Updates? n			
Identity for Calling Party Display: P-Asserted-Identity			
Enable O-SIP? n			

5.7. Administer Calling Party Number Information

Use the **change public-unknown-numbering** command to configure Communication Manager to send the calling party number. In the sample configuration, individual stations are mapped to send numbers allocated from the Vodafone DE DDI range supplied. This calling party number will be sent in the SIP From, Contact and PAI headers, and displayed on display-equipped PSTN telephones.

change public-unknown-numbering 0					Page 1 of 2
NUMBERING - PUBLIC/UNKNOWN FORMAT					
Total					
Ext	Ext	Trk	CPN	CPN	
Len	Code	Grp(s)	Prefix	Len	
4	2291	1	49691382698104	14	
4	2296	1	49691382698103	14	
4	2316	1	49691382698105	14	
4	2346	1	49691382698102	14	
4	2396	1	49691382698101	14	
Total Administered: 5					
Maximum Entries: 9999					
Note: If an entry applies to					
a SIP connection to Avaya					
Aura(tm) Session Manager,					
the resulting number must					

5.8. Administer Route Selection for Outbound Calls

In these Application Notes, the Automatic Route Selection (ARS) feature will be used to route outbound calls via the SIP trunk to Vodafone DE SIP Trunking Service. In the sample configuration, the single digit **9** is used as the ARS access code. Avaya telephone users will dial **9** to reach an outside line. Use the **change feature-access-codes** command to configure or observe **9** as the **Auto Route Selection (ARS) - Access Code 1**.

change feature-access-codes		Page 1 of 10
FEATURE ACCESS CODE (FAC)		
Abbreviated Dialing List1 Access Code:		
Abbreviated Dialing List2 Access Code:		
Abbreviated Dialing List3 Access Code:		
Abbreviated Dial - Prgm Group List Access Code:		
Announcement Access Code: *69		
Answer Back Access Code:		
Attendant Access Code:		
Auto Alternate Routing (AAR) Access Code: 7		
Auto Route Selection (ARS) - Access Code 1: 9		Access Code 2:

Use the **change ars analysis** command to configure the routing of dialed digits following the first digit 9. A small sample of dial patterns is illustrated here. Further administration of ARS is beyond the scope of these Application Notes. The example entries shown will match outgoing calls to numbers beginning **0** or **00**. Calls are sent to route pattern **1**.

change ars analysis 0							Page 1 of 2
ARS DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 1
	Dialed	Total		Route	Call	Node	ANI
	String	Min	Max	Pattern	Type	Num	Reqd
0		9	14	1	pubu		n
00		10	17	1	pubu		n

Use the **change route-pattern x** command to add the SIP trunk group to the route pattern that ARS selects. In this configuration, route pattern **1** is used to route calls to trunk group **1**.

change route-pattern 1		Page 1 of 3
Pattern Number: 1 Pattern Name: all calls		
SCCAN? n Secure SIP? n		
Grp FRL NPA Pfx Hop Toll No. Inserted	DCS/ IXC	
No Mrk Lmt List Del Digits	QSIG	
	Intw	
1: 1 0	n user	
2:	n user	
3:	n user	
4:	n user	
5:	n user	
6:	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 intl-pub none	
2: y y y y y n n	rest none	

5.9. Administer Incoming Digit Translation

This step configures the settings necessary to map incoming DDI calls to the proper Communication Manager extension(s). The incoming digits sent in the INVITE message from Vodafone DE can be manipulated as necessary to route calls to the desired extension. In the examples used in the compliance testing, the incoming DDI numbers provided by Vodafone DE correlate to the internal extensions assigned within Communication Manager. The entries displayed below translate incoming DDI numbers 0691382698101 - 0691382698110 to a 4 digit extension by deleting all of the incoming digits and inserting an extension.

change inc-call-handling-trmt trunk-group 1				Page	1 of 30
INCOMING CALL HANDLING TREATMENT					
Service/ Feature	Number Len	Number Digits	Del	Insert	
tie	13	0691382698101	all	2396	
tie	13	0691382698102	all	2346	
tie	13	0691382698103	all	2296	
tie	13	0691382698104	all	2291	
tie	13	0691382698105	all	2316	
tie	13	0691382698106	all	6101	
tie	13	0691382698107	all	2000	
tie	13	0691382698108	all	2400	
tie	13	0691382698109	all	6102	
tie	13	0691382698110	all	2501	

5.10. EC500 Configuration

When EC500 is enabled on the Communication Manager station, a call to that station will generate a new outbound call from Communication Manager to the configured EC500 destination, typically a mobile phone. The following screen shows an example EC500 configuration for the user with station extension 2396. Use the command **change off-pbx-telephone station mapping x** where **x** is the Communication Manager station.

- The **Station Extension** field will automatically populate with station extension
- For **Application** enter **EC500**
- Enter a **Dial Prefix** (e.g., 9) if required by the routing configuration
- For the **Phone Number** enter the phone that will also be called (e.g. **01723456789**)
- Set the **Trunk Selection** to **1** so that Trunk Group 1 will be used for routing
- Set the **Config Set** to **1**
- Other parameters can retain default value

change off-pbx-telephone station-mapping 2396							Page	1 of 3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION								
Station Extension	Application	Dial Prefix	CC	Phone Number	Trunk Selection	Config Set	Dual Mode	
2396	EC500	-		01723456789	1	1		
		-						

Save Communication Manager changes by entering **save translation** to make them permanent.

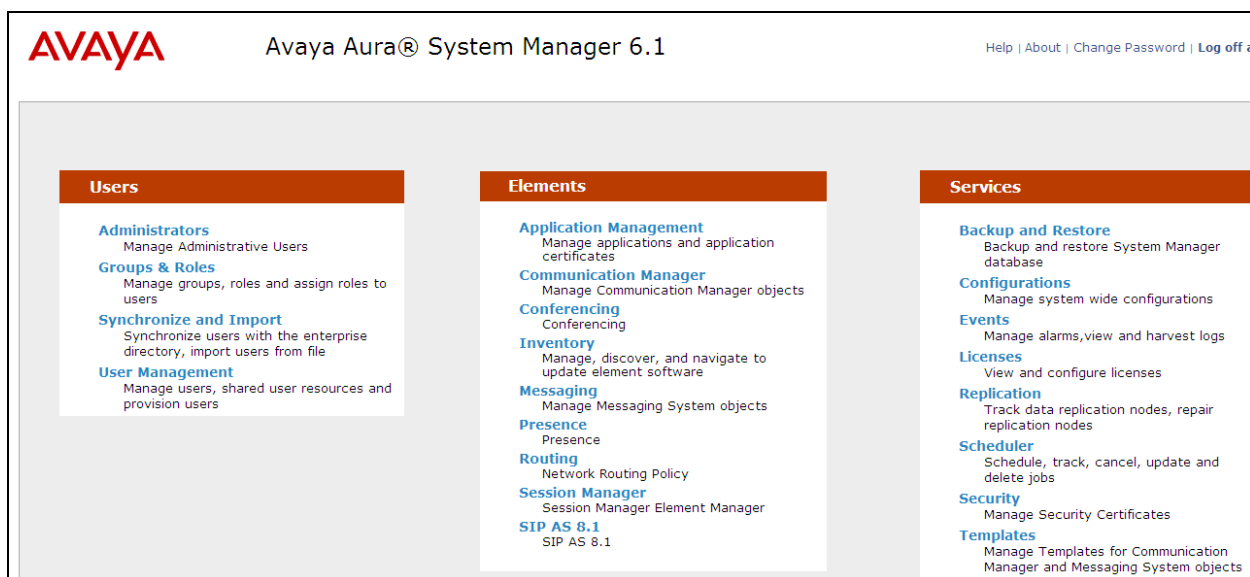
6. Configuring Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The Session Manager is configured via the Avaya Aura® System Manager. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer SIP domain
- Administer Locations
- Administer SIP Entities
- Administer Entity Links
- Administer Routing Policies
- Administer Dial Patterns
- Administer Application for Avaya Aura® Communication Manager
- Administer Application Sequence for Avaya Aura® Communication Manager
- Administer SIP Extensions

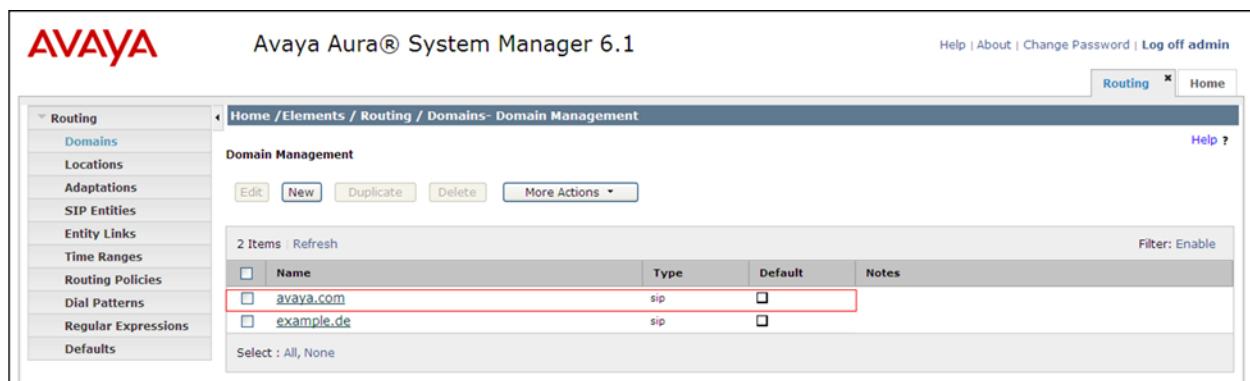
6.1. Log in to Avaya Aura® System Manager

Access the System Manager using a Web Browser by entering **http://<FQDN>/SMGR**, where **<FQDN>** is the fully qualified domain name of System Manager. Log in using appropriate credentials (not shown) and the Home tab will be presented with menu options shown below.



6.2. Administer SIP Domain

To add the SIP domain that will be used with Session Manager, select **Routing** from the **Home** tab menu and in the resulting tab select **Domains** from left hand menu. Click the **New** button to create a new SIP domain entry. In the **Name** field enter the domain name (e.g., **avaya.com**) and optionally a description for the domain in the **Notes** field. Click **Commit** to save changes.



6.3. Administer Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside, for the purposes of bandwidth management. One location is added to the sample configuration for all of the enterprise SIP entities. On the **Routing** tab select **Locations** from the left hand menu. Under **General**, in the **Name** field enter an informative name for the location. Scroll to the bottom of the page and under **Location Pattern**, click **Add**, then enter an **IP Address Pattern** in the resulting new row, * is used to specify any number of allowed characters at the end of the string. Below is the location configuration used for the simulated enterprise.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing **Home**

Home / Elements / Routing / Locations- Location Details

Location Details Help? Commit Cancel

General

* Name: Galvay

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units: Kbit/sec

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): 1000 Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): 1000 Kbit/Sec

Minimum Multimedia Bandwidth: 64 Kbit/Sec

* Default Audio Bandwidth: 80 Kbit/sec

Location Pattern

Add Remove

2 Items Refresh Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.10.9.*	Private

6.4. Administer SIP Entities

A SIP Entity must be added for each SIP-based telephony system, supported by a SIP connection to the Session Manager. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown). The following will need to be entered for each SIP Entity.

Under **General**:

- In the **Name** field enter an informative name
- In the **FQDN or IP Address** field enter the IP address of Session Manager or the signaling interface on the connecting system
- In the **Type** field use **Session Manager** for a Session Manager SIP entity, **CM** for a Communication Manager SIP entity and **Gateway** for the Session Border Controller SIP entity
- In the **Location** field select the appropriate location from the drop down menu
- In the **Time Zone** field enter the time zone for the SIP Entity

In this configuration there are three SIP Entities:

- Avaya Aura® Session Manager SIP Entity
- Avaya Aura® Communication Manager SIP Entity
- Avaya Aura® Session Border Controller SIP Entity

6.4.1. Avaya Aura® Session Manager SIP Entity

The following screens show the SIP entity for Session Manager. The **FQDN or IP Address** field is set to the IP address of the Session Manager SIP signaling interface.

The screenshot displays the Avaya Aura® System Manager 6.1 web interface. The left sidebar shows a navigation menu with 'Routing' selected. The main content area is titled 'SIP Entity Details' and 'General'. The form contains the following fields:

- Name:** Session Manager 1
- FQDN or IP Address:** 10.10.9.61
- Type:** Session Manager (dropdown)
- Notes:** (empty text area)
- Location:** Galway (dropdown)
- Outbound Proxy:** (empty text field)
- Time Zone:** Europe/Dublin (dropdown)
- Credential name:** (empty text field)
- SIP Link Monitoring:** Use Session Manager Configuration (dropdown)

Buttons for 'Commit', 'Cancel', and 'Help ?' are visible in the top right corner of the form area.

The Session Manager must be configured with the port numbers on the protocols that will be used by the other SIP entities. To configure these scroll to the bottom of the page and under **Port**, click **Add**, then edit the fields in the resulting new row.

- In the **Port** field enter the port number on which the system listens for SIP requests
- In the **Protocol** field enter the transport protocol to be used for SIP requests
- In the **Default Domain** field, from the drop down menu select **avaya.com** as the default domain

The screenshot shows a web-based configuration interface for the Session Manager. At the top, there is a 'Port' section with 'Add' and 'Remove' buttons. Below this is a table with 3 items, a 'Refresh' button, and a 'Filter: Enable' link. The table has five columns: a checkbox, 'Port', 'Protocol', 'Default Domain', and 'Notes'. Three rows are visible, each with a checkbox, a port number (5060, 5060, 5061), a protocol (TCP, UDP, TLS), a default domain (avaya.com), and a notes field. A red box highlights the first three rows. Below the table, there is a 'Select : All, None' dropdown and a '* Input Required' message. At the bottom right, there are 'Commit' and 'Cancel' buttons.

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	avaya.com	
<input type="checkbox"/>	5060	UDP	avaya.com	
<input type="checkbox"/>	5061	TLS	avaya.com	

6.4.2. Avaya Aura® Communication Manager SIP Entity

The following screens show the SIP entity for Communication Manager which is configured as an Evolution Server. The **FQDN or IP Address** field is set to the IP address of the interface on Communication Manager that will be providing SIP signaling.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / SIP Entities- SIP Entity Details

SIP Entity Details

Commit Help ? Cancel

General

* Name: Communication Manager

* FQDN or IP Address: 10.10.9.52

Type: CM

Notes:

Adaptation:

Location: Galway

Time Zone: Europe/Dublin

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

SIP Link Monitoring

SIP Link Monitoring: Use Session Manager Configurati...

Entity Links

Add Remove

1 Item Refresh

Filter Enable

6.4.3. Avaya Aura® Session Border Controller SIP Entity

The following screen shows the SIP Entity for the Session Border Controller. The **FQDN or IP Address** field is set to the IP address of the Session Border Controller private network interface (see **Figure 1**).

AVAYA Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / SIP Entities- SIP Entity Details

SIP Entity Details

General

* Name: SBC

* FQDN or IP Address: 10.10.9.67

Type: Gateway

Notes:

Adaptation: None

Location: Galway

Time Zone: Europe/Dublin

Override Port & Transport with DNS SRV: ☐

* SIP Timer B/F (in seconds): 4

Credential name:

Call Detail Recording: none

Commit Cancel

6.5. Administer Entity Links

A SIP trunk between a Session Manager and another system is described by an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- In the **Name** field enter an informative name
- In the **SIP Entity 1** field select **Session Manager 1**
- In the **Port** field enter the port number to which the other system sends its SIP requests
- In the **SIP Entity 2** field enter the other SIP Entity for this link, created in **Section 6.4**
- In the **Port** field enter the port number to which the other system expects to receive SIP requests
- Select the **Trusted** tick box to make the other system trusted
- In the **Protocol** field enter the transport protocol to be used to send SIP requests

Click **Commit** to save changes. The following screen shows the Entity Links used in this configuration.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with options like Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Entity Links' and shows a table with two items. The table has columns for Name, SIP Entity 1, Protocol, Port, SIP Entity 2, Port, Trusted, and Notes. Both 'CM Link' and 'SBC Link' are listed with 'Session Manager 1' as the SIP Entity 1, 'TCP' as the Protocol, and '5060' as the Port. 'CM Link' has 'Communication Manager' as the SIP Entity 2, and 'SBC Link' has 'SBC' as the SIP Entity 2. Both links are marked as 'Trusted'.

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted	Notes
<input type="checkbox"/>	CM Link	Session Manager 1	TCP	5060	Communication Manager	5060	<input checked="" type="checkbox"/>	
<input type="checkbox"/>	SBC Link	Session Manager 1	TCP	5060	SBC	5060	<input checked="" type="checkbox"/>	

6.6. Administer Routing Policies

Routing policies must be created to direct how calls will be routed to a system. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- Enter an informative name in the **Name** field
- Under **SIP Entity as Destination**, click **Select**, and then select the appropriate SIP entity to which this routing policy applies
- Under **Time of Day**, click **Add**, and then select the time range

The following screen shows the routing policy for Communication Manager

The screenshot displays the Avaya Aura System Manager 6.1 interface. The left sidebar shows the navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a breadcrumb trail: 'Home / Elements / Routing / Routing Policies - Routing Policy Details'. The 'General' tab is active, showing the 'Name' field set to 'CM Phones', a 'Disabled' checkbox, and a 'Notes' field. The 'SIP Entity as Destination' section has a 'Select' button. Below this is a table with one entry: 'Communication Manager' with FQDN '10.10.9.52' and Type 'CM'. The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons. A table below shows one time range: '24/7' with start time '00:00' and end time '23:59'. The 'Dial Patterns' section has 'Add' and 'Remove' buttons.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing x Home

Home / Elements / Routing / Routing Policies - Routing Policy Details

Routing Policy Details

General

* Name: CM Phones

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Communication Manager	10.10.9.52	CM	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

Dial Patterns

Add Remove

The following screen shows the routing policy for the Session Border Controller.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) [About](#) [Change Password](#) [Log off admin](#)

[Routing](#) [Home](#)

Home / Elements / Routing / Routing Policies- Routing Policy Details

Routing Policy Details [Help?](#) [Commit](#) [Cancel](#)

General

* Name:

Disabled: ☐

Notes:

SIP Entity as Destination

[Select](#)

Name	FQDN or IP Address	Type	Notes
SBC	10.10.9.67	Gateway	

Time of Day

[Add](#) [Remove](#) [View Gaps/Overlaps](#)

1 Item Refresh [Filter: Enable](#)

<input type="checkbox"/>	Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	ip	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select: All, None

Dial Patterns

[Add](#) [Remove](#)

7 Items Refresh [Filter: Enable](#)

6.7. Administer Dial Patterns

A dial pattern must be defined to direct calls to the appropriate telephony system. To configure a dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

Under **General**:

- In the **Pattern** field enter a dialed number or prefix to be matched
- In the **Min** field enter the minimum length of the dialed number
- In the **Max** field enter the maximum length of the dialed number
- In the **SIP Domain** field select the domain configured in **Section 6.2**

Under **Originating Locations and Routing Policies**. Click **Add**, in the resulting screen (not shown), under **Originating Location** select **ALL** and under **Routing Policies** select one of the routing policies defined in **Section 6.6** Click **Select** button to save. The following screen shows an example dial pattern configured for the Session Border Controller which will route the calls out to the Vodafone DE SIP Trunking Service.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

Routing Home

Home / Elements / Routing / Dial Patterns- Dial Pattern Details

Dial Pattern Details Commit Cancel Help?

General

* Pattern: 00353

* Min: 10

* Max: 17

Emergency Call: ☐

SIP Domain: ALL

Notes:

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	ALL	Any Locations	External	0	<input type="checkbox"/>	SBC	

Select: All, None

Denied Originating Locations

Add Remove

0 Items Refresh Filter Enable

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

The following screen shows an example dial pattern configured for Communication Manager.

AVAYA Avaya Aura® System Manager 6.1 [Help](#) [About](#) [Change Password](#) [Log off admin](#)

[Routing](#) [Home](#)

[Home](#) / [Elements](#) / [Routing](#) / [Dial Patterns](#) - Dial Pattern Details

[Help?](#)
[Commit](#) [Cancel](#)

Dial Pattern Details

General

* Pattern: 0691382698

* Min: 10

* Max: 13

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

[Add](#) [Remove](#)

1 Item Refresh [Filter](#) [Enable](#)

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	CM Phones	0	<input type="checkbox"/>	Communication Manager	

Select: All, None

Denied Originating Locations

[Add](#) [Remove](#)

0 Items Refresh [Filter](#) [Enable](#)

<input type="checkbox"/>	Originating Location	Notes
--------------------------	----------------------	-------

6.8. Administer Application for Avaya Aura® Communication Manager

From the home tab select **Session Manager** from the menu. In the resulting tab from the left panel menu select **Application Configuration → Applications** and click **New**.

- In the **Name** field enter a name for the application
- In the **SIP Entity** field select the SIP entity for the Communication Manager
- In the **CM System for SIP Entity** field select the SIP entity for the Communication Manager

Select **Commit** to save the configuration.

The screenshot shows the Avaya Aura® System Manager 6.1 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.1", and links for "Help", "About", "Change Password", and "Log off admin". Below the navigation bar, there are tabs for "Session Manager" and "Home". The left sidebar contains a menu with the following items: Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, Applications (highlighted), Application Sequences, Implicit Users, NRS Proxy Users, System Status, and System Tools. The main content area is titled "Application Editor" and contains the following fields: "Name" (text input with value "cm-app"), "SIP Entity" (dropdown menu with value "Communication Manager"), "CM System for SIP Entity" (dropdown menu with value "CM Instance" and a "Refresh" button), and "Description" (text input). There are also links for "View/Add CM Systems". Below these fields is a section for "Application Attributes (optional)" with a table containing two rows: "Application Handle" and "URI Parameters", each with a corresponding text input field. At the top right of the main content area, there are "Commit" and "Cancel" buttons.

Name	Value
Application Handle	
URI Parameters	

6.9. Administer Application Sequence for Communication Manager

From the left panel navigate to **Session Manager** → **Application Configuration** → **Application Sequences** and click on **New**.

- In the **Name** field enter a descriptive name
- Under **Available Applications**, click the + sign in front of the appropriate application instance. When the screen refreshes the application should be displayed under the **Applications in this Sequence** heading

Select **Commit**.

The screenshot shows the Avaya Aura System Manager 6.1 interface. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, Applications, Application Sequences, Implicit Users, NRS Proxy Users, System Status, and System Tools. The main content area is titled 'Application Sequence Editor' and includes a breadcrumb trail: Home / Elements / Session Manager / Application Configuration / Application Sequences - Application Sequences. The 'Application Sequence' section has a 'Name' field (containing 'cm-app-seq') and a 'Description' field. Below this is the 'Applications in this Sequence' section, which includes buttons for 'Move First', 'Move Last', and 'Remove'. A table lists the applications in the sequence, with one item highlighted: 'cm-app' (Communication Manager) with a checked 'Mandatory' box. The 'Available Applications' section at the bottom shows a table with one item: 'cm-app' (Communication Manager). The interface also includes 'Commit' and 'Cancel' buttons at the top right and bottom right, and a '*Required' label at the bottom left.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Session Manager x Home

Home / Elements / Session Manager / Application Configuration / Application Sequences - Application Sequences

Help ?

Application Sequence Editor

Application Sequence

*Name

Description

Applications in this Sequence

Move First Move Last Remove

1 Item

<input type="checkbox"/>	Sequence Order (first to last)	Name	SIP Entity	Mandatory	Description
<input type="checkbox"/>		cm-app	Communication Manager	<input checked="" type="checkbox"/>	

Select: All/None

Available Applications

1 Item Refresh Filter: Enable

	Name	SIP Entity	Description
+	cm-app	Communication Manager	

*Required

Commit Cancel

6.10. Administer SIP Extensions

SIP extensions are registered with the Session Manager and use Communication Manager for their feature and configuration settings. From the Home tab select **User Management** from the menu. Then select **Manage Users** and click **New** (not shown).

On the **Identity** tab:

- Enter the user's name in the **Last Name** and **First Name** fields
- In the **Login Name** field enter a unique system login name in the form of user@domain (e.g. **2296@avaya.com**) which is used to create the user's primary handle
- The **Authentication Type** should be **Basic**
- In the **Password/Confirm Password** fields enter an alphanumeric password

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

User Management x Home

Home / Users / User Management / Manage Users - New User Profile Help ?

User Management Manage Users Public Contacts Shared Addresses System Presence ACLs

New User Profile Commit Cancel

Identity * Communication Profile * Membership Contacts

Identity ▾

* Last Name: SIP

* First Name: 9630

Middle Name:

Description:

* Login Name: 2296@avaya.com

* Authentication Type: Basic ▾

* Password: *****

* Confirm Password: *****

Localized Display Name:

On the **Communication Profile** tab enter a numeric **Communication Profile Password** and confirm it, then expand the **Communication Address** section and click **New**. For the **Type** field select **sip** from the drop-down menu. In the **Fully Qualified Address** field, enter an extension number and select the relevant domain from the drop-down menu. Click the **Add** button.

AVAYA Avaya Aura® System Manager 6.1 Help | About | Change Password | Log off admin

[User Management](#) [Home](#)

Home / Users / User Management / Manage Users- New User Profile Help ?

New User Profile Commit Cancel

Identity **Communication Profile** Membership Contacts

Communication Profile

Communication Profile Password: Confirm Password:

New Delete Done Cancel

Name
Primary

Select: None

* Name: Default: ☒

Communication Address

New Edit Delete

Type	Handle	Domain
No Records found		

* Fully Qualified Address:

Expand the **Session Manager Profile** section.

- Make sure the **Session Manager** check box is checked
- Select the appropriate Session Manager instance from the drop-down menu in the **Primary Session Manager** field
- Select the appropriate application sequence from the drop-down menu in the **Origination Application Sequence** field configured in **Section 6.9**
- Select the appropriate application sequence from the drop-down menu in the **Termination Application Sequence** field configured in **Section 6.9**
- Select the appropriate location from the drop-down menu in the **Home Location** field

☒ Session Manager Profile

* Primary Session Manager: Session Manager 1

Secondary Session Manager: (None)

Origination Application Sequence: cm-app-seq

Termination Application Sequence: cm-app-seq

Survivability Server: (None)

* Home Location: Galway

Primary	Secondary	Maximum
2	0	2

Primary	Secondary	Maximum
---------	-----------	---------

Expand the **Endpoint Profile** section.

- Select the Communication Manager SIP Entity from the **System** drop-down menu
- Select **Endpoint** from the drop-down menu for **Profile Type**
- Enter the extension in the **Extension** field
- Select the desired template from the **Template** drop-down menu
- For the **Port** field select **IP**
- Select the **Delete Endpoint on Unassign of Endpoint from User or on Delete User** check box
- Select **Commit** to save changes and the System Manager will add the Communication Manager user configuration automatically

The screenshot shows a web-based configuration interface for an Endpoint Profile. The 'Endpoint Profile' section is expanded, revealing several required fields marked with a red asterisk. The 'System' dropdown is set to 'CM Instance', and the 'Profile Type' dropdown is set to 'Endpoint'. The 'Extension' field contains '2292', and the 'Template' dropdown is set to 'DEFAULT_9630SIP_CM_6_0'. The 'Set Type' field is '9630SIP', and the 'Security Code' is masked with six dots. The 'Port' dropdown is set to 'IP'. The checkbox 'Delete Endpoint on Unassign of Endpoint from User or on Delete User.' is checked. Below the Endpoint Profile section is a collapsed 'Messaging Profile' section. At the bottom of the form, there is a 'Required' label, a 'Commit' button, and a 'Cancel' button.

☒ Endpoint Profile

* System CM Instance

* Profile Type Endpoint

Use Existing Endpoints ☐

* Extension 2292 Endpoint Editor

* Template DEFAULT_9630SIP_CM_6_0

Set Type 9630SIP

Security Code *****

* Port IP

Voice Mail Number

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

☐ Messaging Profile

* Required


Commit Cancel

7. Configure Avaya Aura® Session Border Controller

This section describes the configuration of the Session Border Controller. The configuration is done using the Session Border Controller web interface.

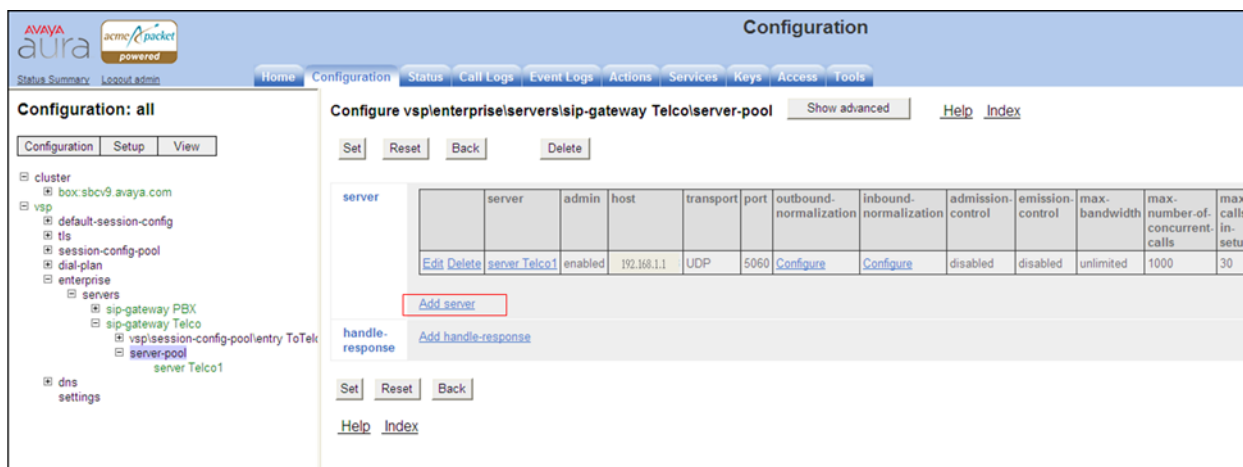
7.1. Access Avaya Aura® Session Border Controller

Access the Session Border Controller using a web browser by entering the URL **https://<ip-address>**, where **<ip-address>** is the private IP address configured at installation. Log in with the appropriate credentials.



7.2. Add Additional Service Provider IP Addresses

To add the additional IP addresses for the Vodafone DE SBCs click on the **Configuration** tab and browse to **vsp → enterprise → servers → sip-gateway Telco → server-pool**. A list of the IP addresses already configured in the server pool is displayed in the right hand pane. Click the **Add server** link.



In the resulting page enter a name for the server in the **server-name** field and an IP address in the **host** field. Click **Create** to continue.

The screenshot shows the Avaya Aura Configuration interface. On the left is a tree view under 'Configuration: all' showing a hierarchy from 'cluster' to 'server Telco2'. The main panel is titled 'Create vsplenterprise\servers\sip-gateway Telco\server-pool\server - Step 1 of 1: Edit server'. It contains a 'General' section with two fields: '* server-name' with the value 'Telco2' and '* host' with the value '192.168.2.1'. Below these fields are 'Create', 'Reset', and 'Cancel' buttons.

In the resulting page verify the details entered and click the **Set** button.

The screenshot shows the Avaya Aura Configuration interface for editing the server. The main panel is titled 'Configure vsplenterprise\servers\sip-gateway Telco\server-pool\server Telco2'. It includes buttons for 'Set', 'Reset', 'Back', 'Copy', and 'Delete'. The 'General' section contains five fields: '* server-name' (Telco2), 'admin' (a dropdown menu set to 'enabled' with the note '(Resource is active)'), '* host' (192.168.2.1), 'transport' (a dropdown menu set to 'UDP' with the note '(User Datagram Protocol)'), and 'port' (5060 with the note '(at minimum 1, default=5060)').

Repeat these steps for each additional IP address that needs to be added to the Session Border Controller server pool.

7.3. Save the Configuration

To save the configuration, click on **Configuration** in the left pane to display the configuration menu. Next, select **Update and save configuration**.

The screenshot displays the Avaya Aura Configuration web interface. The top navigation bar includes links for Home, Configuration, Status, Call Logs, Event Logs, Actions, Services, Keys, Access, and Tools. The left pane, titled 'Configuration: all', shows a tree structure with 'enterprise' expanded, revealing 'servers' and 'server-pool'. The 'server-pool' node is selected, and its sub-items 'server Telco1' and 'server Telco2' are visible. The 'Update and save configuration' option is highlighted in the left pane. The main pane shows the configuration for 'server Telco1' and 'server Telco2' in a table. The table has columns for server, admin, host, transport, port, outbound-normalization, and inbound-normalization. The 'server Telco1' row shows 'enabled', '192.168.1.1', 'UDP', '5060', and 'Configure' links. The 'server Telco2' row shows 'enabled', '192.168.2.1', 'UDP', '5060', and 'Configure' links. Below the table, there are buttons for 'Set', 'Reset', 'Back', and 'Delete'. At the bottom, there are links for 'Help' and 'Index'.

	server	admin	host	transport	port	outbound-normalization	inbound-normalization
▼	Edit Delete server Telco1	enabled	192.168.1.1	UDP	5060	Configure	Configure
▲	Edit Delete server Telco2	enabled	192.168.2.1	UDP	5060	Configure	Configure

8. Service Provider Configuration

The configuration of the Vodafone DE equipment used to support the Vodafone DE SIP Trunking service is outside of the scope of these Application Notes and will not be covered. To obtain further information on Vodafone DE equipment and system configuration please contact an authorised Vodafone representative.

9. Verification Steps

This section provides steps that may be performed to verify that the solution is configured correctly.

1. From System Manager Home Tab click on Session Manager and navigate to **Session Manager → System Status → SIP Entity Monitoring**. Select the relevant SIP Entity from the list and observe if the **Conn Status** and **Link Status** are showing as **up**.

The screenshot shows the Avaya Aura System Manager 6.1 web interface. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager Administration, Communication Profile Editor, Network Configuration, Device and Location Configuration, Application Configuration, and System Status. The main content area is titled 'SIP Entity, Entity Link Connection Status' and shows a table of entity links for the selected SIP Entity 'SBC'. The table has columns for Session Manager Name, SIP Entity Resolved IP, Port, Proto., Conn. Status, Reason Code, and Link Status. The 'Conn. Status' and 'Link Status' for the selected entity are both 'Up'.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
► Show	Session Manager 1	10.10.9.67	5060	TCP	Up	200 OK	Up

2. From the Communication Manager SAT interface run the command **status trunk n** where **n** is a previously configured SIP trunk. Observe if all channels on the trunk group display **In service/ idle**.

```
status trunk 1
```

TRUNK GROUP STATUS			
Member	Port	Service State	Mtce Connected Ports Busy
0001/001	T00001	in-service/idle	no
0001/002	T00002	in-service/idle	no
0001/003	T00003	in-service/idle	no
0001/004	T00004	in-service/idle	no
0001/005	T00005	in-service/idle	no
0001/006	T00006	in-service/idle	no
0001/007	T00007	in-service/idle	no
0001/008	T00008	in-service/idle	no
0001/009	T00009	in-service/idle	no
0001/010	T00010	in-service/idle	no

3. Verify that endpoints at the enterprise site can place calls to the PSTN and that the call remains active.
4. Verify that endpoints at the enterprise site can receive calls from the PSTN and that the call can remain active.
5. Verify that the user on the PSTN can end an active call by hanging up.
6. Verify that an endpoint at the enterprise site can end an active call by hanging up.

10. Conclusion

These Application Notes describe the configuration necessary to connect Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Aura® Session Border Controller to Vodafone DE SIP Trunking Service. Vodafone DE SIP Trunking Service is a SIP-based Voice over IP solution providing businesses a flexible, cost-saving alternative to traditional hardwired telephony trunks. The service was successfully tested with a number of observations listed in **Section 2.2**.

11. References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Installing and Configuring Avaya Aura® System Platform*, Release 6.0.3, February 2011.
- [2] *Administering Avaya Aura® System Platform*, Release 6.0.3, February 2011.
- [3] *Administering Avaya Aura® Communication Manager*, Release 6.0.1, April 2011.
- [4] *Avaya Aura® Communication Manager Feature Description and Implementation*, August 2010, Document Number 555-245-205.
- [5] *Installing and Upgrading Avaya Aura® System Manager Release 6.1*, November 2010.
- [6] *Installing and Configuring Avaya Aura® Session Manager*, April 2011, Document Number 03-603473
- [7] *Administering Avaya Aura® Session Manager*, May 2011, Document Number 03-603324.
- [8] *Avaya Aura® Session Border Controller System Administration*, September 2010
- [9] *Installing and Configuring Avaya Aura® Session Border Controller*, May 2011
- [8] RFC 3261 *SIP: Session Initiation Protocol*, <http://www.ietf.org/>

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