



Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Avaya Aura® Communication Manager R6.0.1 and Avaya Aura® Session Manager R6.1 to interoperate with Netcall SEMAP+ – Issue 1.1

Abstract

The Application Notes describe the configuration steps for the Netcall SEMAP+ solution to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. The Netcall SEMAP+ system can host multiple applications such as IVR and Voicemail via connection to Avaya Aura® Session Manager over 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

The purpose of this document is to describe the compliance tested configuration between Avaya Aura® Session Manager and the Netcall SEMAP+ platform with Avaya Aura® Communication Manager providing telephony features configured as an Evolution Server. This document includes a description of the configuration of both the Avaya and the Netcall solutions, a description of the tests that were performed and a summary of the results of those tests.

Netcall SEMAP+ is a Speech Enabled Multi Application Platform which integrates with the Avaya solution using a SIP trunk. SEMAP+ enables the hosting of multiple solutions concurrently on the same platform.

2. General Test Approach and Test Results

The interoperability compliance testing evaluated the ability of Netcall SEMAP+ to carry out call handling and routing using in a variety of scenarios using the ContactPortal (ASR) and Messaging+ applications. For the purposes of the compliance test the Speech Recognition and Messaging applications were used to test the call handling functions of the SEMAP+ platform. These applications were used to give a broad overview of the call handling abilities of the Netcall SEMAP+ platform and are indicative of the way calls to other SEMAP+ applications are handled.

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 test included both feature functionality and serviceability testing. The feature functionality testing focused on placing calls in different call scenarios, to ensure successful routing of the call depending on the application used. Call handling and routing functions are common across SEMAP+ these tests are indicative of the way SEMAP+ handles calls. However for the purpose of the compliance testing for SEMAP+, the ContactPortal® ASR and Messenger+ voicemail modular applications were used, and the tests carried out included:

- Successful Speech recognition
- Barge-in
- Correct call routing
- Transfer to busy/engaged/dnd/restricted user
- SIP REFER usage
- Hold/Retrieve
- Call Transfer
- Intra switch call

- Inbound trunk call
- Outbound trunk call
- Out of band Signaling (DTMF)
- Logging

In the case of the Messaging+ application:

- Coverage to correct mailbox
- Message retrieval
- User recognition from CPN
- MWI
- Greeting recording
- Intra switch call
- Inbound trunk call

The serviceability testing focused on verifying the ability of Netcall SEMAP+ to recover from disconnection and reconnection to the Avaya solution and power failure.

2.2. Test Results

All functionality and serviceability test cases were completed successfully with the following observation:

- In the case where an extension is called which has an unconditional divert set to another extension, and the extension being forwarded to has a divert set to the SEMAP+ Messaging+ application, the voicemail will answer on the mailbox of the person originally called e.g. 4001 calls 4002, 4002 is on unconditional divert to 4003, 4003 has divert set to the SEMAP+ Messaging+ application, call from 4001 covers to the voicemail box of 4002. This is by design.

2.3. Support

Technical support can be obtained for the Netcall SEMAP+ solution as follows:

- Email: technical.support@netcall.com
- Website: <http://www.netcall.com>
- Phone: + 44 330 333 6100

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of an Avaya S8800 Server running Communication Manager with Avaya G650 Media Gateway. An Avaya S8800 Server hosts Session Manager. Another Avaya S8800 Server hosts System Manager. A variety of Avaya H.323, Digital and SIP endpoints (not shown) were used in the testing. SEMAP+ was hosted on a Microsoft Windows 2003 server provided by Netcall.

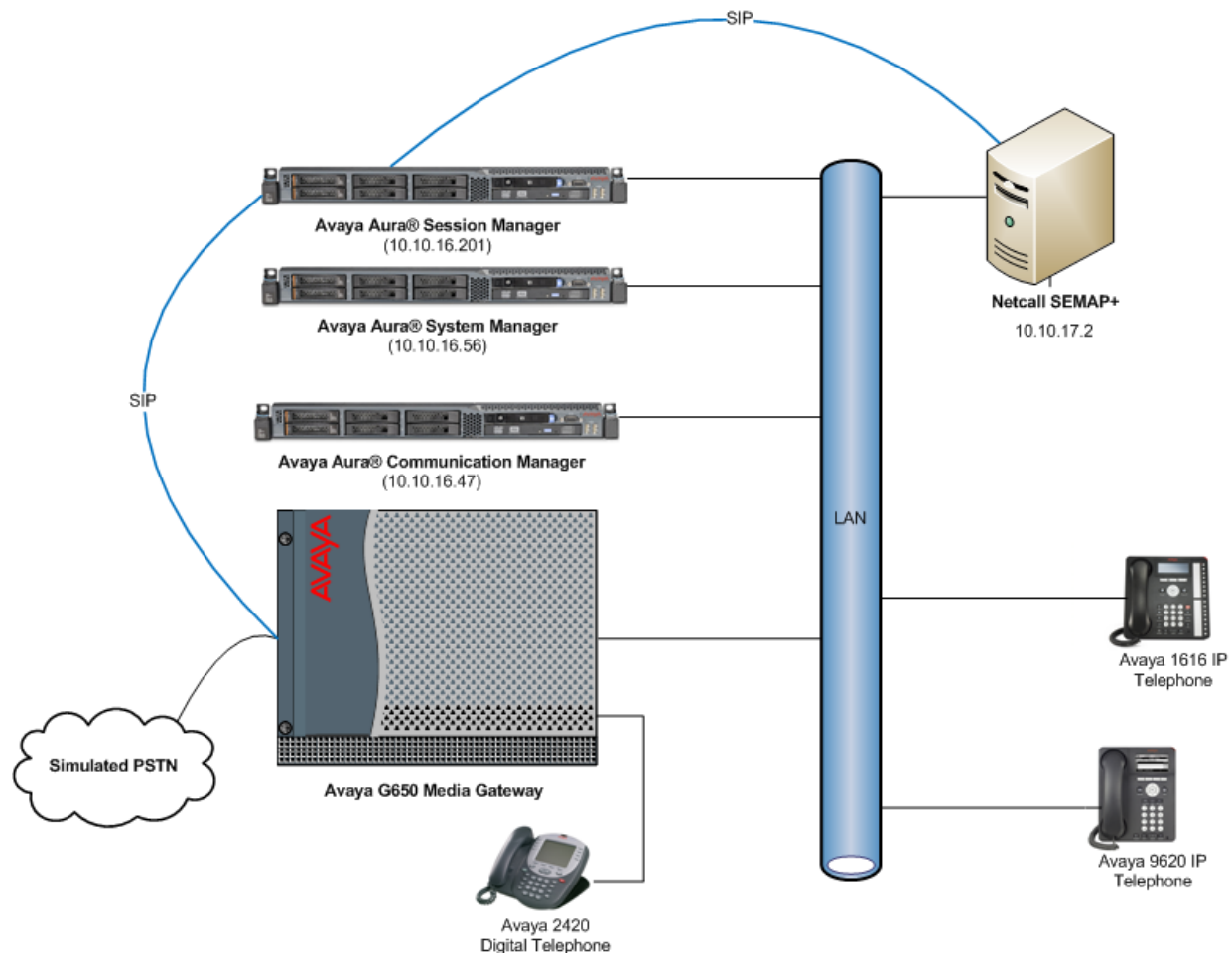


Figure 1: Avaya Aura® Communication Manager with Avaya Aura® Session Manager and Avaya Aura® System Manager and Netcall SEMAP+ Solution.

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 running on Avaya S8800 Server	R6.0.1 SP6 R16.00.1.510.1-19350
Avaya G650 Media	<ul style="list-style-type: none">• TN2602AP HW8 FW61• TN799DP HW01 FW040• TN 2464CP HW02 FW024
Avaya Aura® Session Manager running on Avaya S8800 Server	R6.1 SP6 6.1.6.0.616008
Avaya Aura® System Manager running on Avaya S8800 Server	R6.1 SP6 Build Number 6.1.0.0.7345-6.1.5.606 Software Update Revision Number 6.1.10.1.1774
Avaya H323 IP Telephones	<ul style="list-style-type: none">• 1616 FW 1.301S• 9620 FW 3.102S
Avaya Digital Telephone	<ul style="list-style-type: none">• 2420 REL 4.00 HWV 1 FWV 4
Netcall Server	Netcall SEMAP+ 18.16 Messaging+ / Contactportal

5. Configure Avaya Aura® Communication Manager

The configuration and verification operations illustrated in this section were all performed using Communication Manager System Administration Terminal (SAT). The information provided in this section describes the configuration of Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation as referenced in **Section 10**. The configuration operations described in this section can be summarized as follows:

- Configure node-name
- Configure SIP Signaling Group
- Configure SIP Trunk
- Configure Route Pattern
- Configure AAR
- Configure Coverage Paths
- Configure Station

5.1. Configure Node Names

The Application Note assumes a C-LAN interface or equivalent is administered on Communication Manager. These Application Notes also assume that the Session Manager SIP Signaling Interface has been configured on Session Manager. In order to create the SIP Trunk between Communication Manager and Session Manager a node-name needs to be specified for the Session Manager SIP Signaling Interface, this will be used in **Section 5.2**. Enter the command **change node-names ip** and enter an identifying **Name** for the Session Manager SIP Signaling Interface and its **IP address**, in this case **sesmgr-sm100** and **10.10.16.201** respectively. Take a note here of the C-LAN node name, **clancm601**.

```
change node-names ip                                     Page 1 of 2
2
                                                    IP NODE NAMES
      Name                IP Address
clancm601             10.10.16.31
default                 0.0.0.0
devconaes61            10.10.16.30
ipo7.0                  10.10.16.105
medprocm601            10.10.16.32
procr                   10.10.16.47
procr6                  ::
sesmgr-sm100          10.10.16.201

( 8 of 8      administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

5.2. Configure SIP Signaling Group

A signaling group must be used to specify the signaling type and node-names to be used for the SIP Trunk configured in **Section 5.3**. Enter the command **add signaling-group next**, take a note of the **Group Number**, set the **Group Type** to **sip**, **Transport Method** to **tcp**, set the **Near-end Node Name** to the node name of the C-LAN, in this case **clancm601**, and the **Far-end Node Name** to that configured in **Section 5.1**, in this case **sesmgr-sm100**. Leave all other settings as default.

add signaling-group next		Page 1 of 1
SIGNALING GROUP		
Group Number: 2	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: clancm601	Far-end Node Name: sesmgr-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? y	
	Alternate Route Timer(sec): 6	

5.3. Configure SIP Trunk

To route calls between Communication Manager and Session Manager the SIP trunk must use the signaling group setup in **Section 5.2**. Enter the command **add trunk-group next**, on **Page 1** take a note of the **Group Number**, this will be used when configuring route patterns. Set the **Group Type** to **sip** and assign the trunk an identifying **Group Name**. Set the **TAC** according to the dialplan. Set the **Service Type** as **public-ntwrk** and assign the **Signaling Group** as added in **Section 5.2**, set **Number of Members** according to requirements.

add trunk-group 2		Page 1 of 21	
TRUNK GROUP			
Group Number: 2	Group Type: sip	CDR Reports: y	
Group Name: SIP TRUNK TO SES-MGR	COR: 1	TN: 1	TAC: 702
Direction: two-way	Outgoing Display? y	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: public-ntwrk	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group: 2		
	Number of Members: 30		

On **Page 3** set the **Numbering Format** as **Private**.

add trunk-group 2		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private		UUI Treatment: service-provider	
		Replace Restricted Numbers? n	
		Replace Unavailable Numbers? n	
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? y			

On **Page 4** set the **Send Diversion Header** to **y** and the **Identity for Calling Party Display** to **From**.

```
add trunk-group 2
                                PROTOCOL VARIATIONS
                                Mark Users as Phone? n
                                Prepend '+' to Calling Number? n
                                Send Transferring Party Information? n
                                Network Call Redirection? n
                                Send Diversion Header? y
                                Support Request History? n
                                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: From
                                Enable Q-SIP? n
```

5.4. Configure Route Pattern

In order to place calls from via Communication Manager to Session Manager and on to SEMAP+ a route pattern must be created from Communication Manager to Session Manager. The route pattern will use the trunk created in **Section 5.3**. Enter the command **change route-pattern x** where **x** is an unused route pattern, configure an identifying **Pattern Name**, the **Grp No** setup in the previous Section, and an **FRL** accordingly.

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

5.5. Configure Dialplan

Enter the command **change dialplan analysis** to configure the digit string to be routed to Session Manager and onto SEMAP+. In the example below, **Dialed String** beginning with **85** and a **Total Length** of **4** is set to the **call type udp**, this will route the dialed string to the uniform dialplan for further classification.

change dialplan analysis						Page 1 of 12		
DIAL PLAN ANALYSIS TABLE								
Location: all						Percent Full: 2		
Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type	Dialed String	Total Length	Call Type
13	4	ext	#	3	fac			
15	4	ext						
16	4	ext						
18	4	ext						
2	3	ext						
30	4	ext						
4	4	ext						
5	1	fac						
70	3	dac						
71	3	dac						
77	3	ext						
85	4	udp						
9	1	fac						
*	3	fac						

5.6. Configure Uniform Dialplan

Enter the command **change uniform-dialplan 85**, set the **Matching Pattern** to **85**, the **Len** to **4** and the **Net** to **aar**, this will route the 4 digit number beginning with **85** to the **aar** table where a route pattern is defined.

change uniform-dialplan 85						Page 1 of 2	
UNIFORM DIAL PLAN TABLE							
						Percent Full: 0	
Matching			Insert			Node	
Pattern	Len	Del	Digits	Net	Conv	Num	
85	4	0		aar	n		
					n		

5.7. Configure AAR

The AAR table must be configured in order that calls to SEMAP+ are routed using the pattern configured in **Section 5.4**. Enter the command **change aar analysis 0**, in this example when a **4** digit number beginning with digits **85** is dialed, it will be placed using route pattern **2**. Configure the **Dialed String 85**, **Min 4** and **Max 4**, **Route Pattern 2** and **Call Type aar** as shown below.

change aar analysis 0							Page 1 of 2
AAR DIGIT ANALYSIS TABLE							
Location: all							Percent Full: 0
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num	ANI Req'd	
85	4	4	2	aar		n	
201	3	3	4	aar		n	
2456	4	4	2	aar		n	
400	3	3	5	aar		n	
5	7	7	999	aar		n	
6	7	7	999	aar		n	
7	7	7	999	aar		n	
8	5	5	3	aar		n	
9	7	7	999	aar		n	
						n	

5.8. Configure Coverage Paths

Cover paths must be administered in order to route calls to the SEMAP+ Messaging+ application. For the purposes of the compliance test, both a cover path and a remote cover path were added. Enter the command **change coverage remote 1**, enter the extension assigned in **Section 7.3** for voicemail deposit, in this case **8502** in an available field, in this example **02**. This is referred to as r2.

change coverage remote 1			Page 1 of 23
REMOTE CALL COVERAGE TABLE			
ENTRIES FROM 1 TO 1000			
01: 90857412987	16:	31:	
02: 8502	17:	32:	
03:	18:	33:	
04:	19:	34:	
05:	20:	35:	
06:	21:	36:	
07:	22:	37:	
08:	23:	38:	
09:	24:	39:	
10:	25:	40:	
11:	26:	41:	
12:	27:	42:	
13:	28:	43:	
14:	29:	44:	
15:	30:	45:	

Enter the command **add coverage path next**, take a note of the **Coverage Path Number**, in this case **5**, and enter **r2** in **Point 1**.

```

add coverage path next                                     Page 1 of 1

                                COVERAGE PATH

                                Coverage Path Number: 5
                                Cvg Enabled for VDN Route-To Party? n      Hunt after Coverage? n
                                Next Path Number:                          Linkage

COVERAGE CRITERIA
  Station/Group Status      Inside Call      Outside Call
      Active?                n                n
      Busy?                  Y                Y
      Don't Answer?          Y                Y      Number of Rings: 2
      All?                   n                n
  DND/SAC/Goto Cover?       Y                Y
  Holiday Coverage?         n                n

COVERAGE POINTS
  Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: r2                Rng:      Point2:
  Point3:                   Point4:
  Point5:                   Point6:

```

5.9. Configure Station

A station must be administered with the coverage path configured in the previous section for use with the SEMAP+ Messaging+ application. Enter the command **add station x** in this case station **4001** is used. Configure according to requirements and set the **Coverage Path** to **5**.

```

add station 4001                                         Page 1 of 5

                                STATION

Extension: 4001                Lock Messages? n          BCC: 0
  Type: 9620                   Security Code: 1234         TN: 1
  Port: S00018                 Coverage Path 1: 5          COR: 1
  Name: Richard Pope           Coverage Path 2:         COS: 1
                                Hunt-to Station:

STATION OPTIONS
                                Time of Day Lock Table:
      Loss Group: 19           Personalized Ringing Pattern: 1
                                Message Lamp Ext: 4001
      Speakerphone: 2-way      Mute Button Enabled? y
      Display Language: english
  Survivable GK Node Name:
      Survivable COR: internal      Media Complex Ext:
      Survivable Trunk Dest? y      IP SoftPhone? y

                                IP Video Softphone? n
                                Short/Prefixed Registration Allowed: default

                                Customizable Labels? y

```

6. Configure Avaya Aura® Session Manager

System Manager is used to configure Session Manager SIP Entities and manages the connections between related SIP trunks and endpoints. It is also used to configure dial patterns and route calls according to defined rules. The configuration and verification operations illustrated in this section were all performed using the System Manager Web Interface.

This section provides the procedure for configuring Session Manager. For further reference documents, refer to **Section 10** of this document. The procedures include the following areas:

- Log in to Avaya Aura® System Manager
- Administer Adaptation
- Administer SIP Entity
- Administer Entity Link
- Administer Routing Policies
- Administer Dial Patterns

6.1. Log in to Avaya Aura® System Manager

Log into the System Manager web interface using the System Manager IP address, in this case <https://10.10.16.56/index.jsp> enter the appropriate credentials and click **Log On**.

AVAYA Avaya Aura® System Manager 6.1

Home / Log On

Log On

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

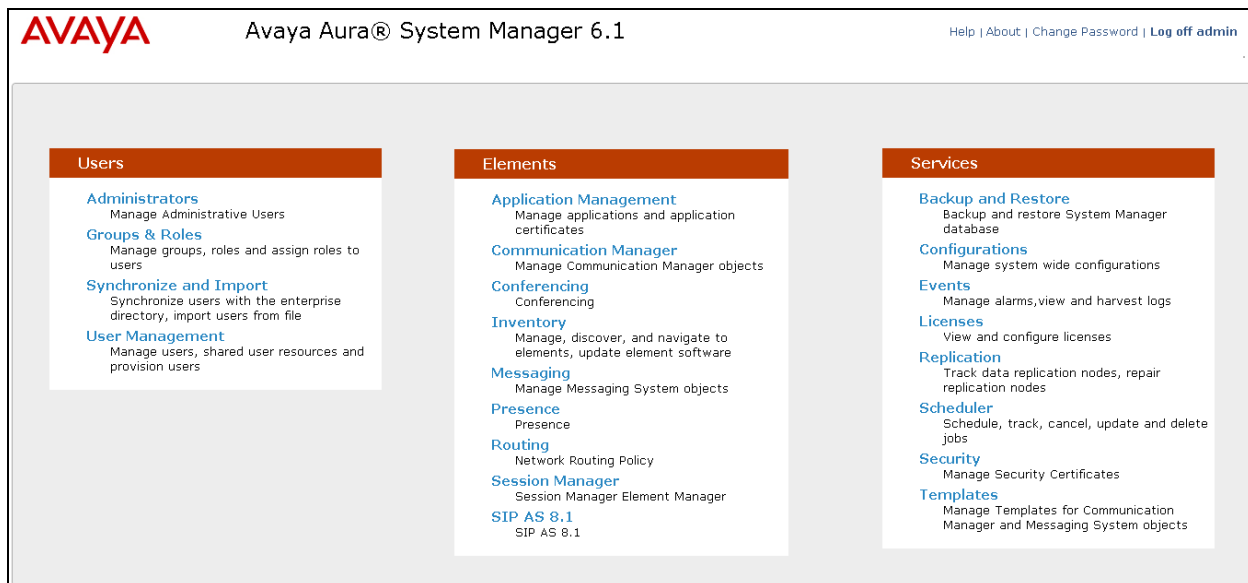
User ID: admin

Password: *****

Log On Cancel

[Change Password](#)

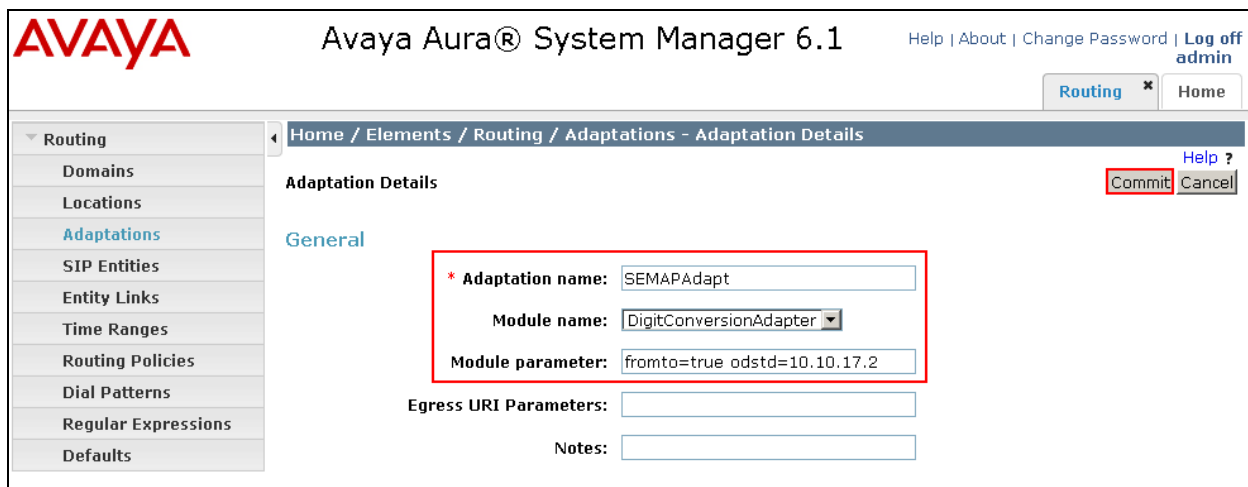
The home screen is divided into three sections with hyperlinked categories below.



The screenshot shows the Avaya Aura® System Manager 6.1 home screen. At the top, the Avaya logo is on the left, the title "Avaya Aura® System Manager 6.1" is in the center, and links for "Help | About | Change Password | Log off admin" are on the right. The main content area is divided into three columns: "Users", "Elements", and "Services". Each column contains a list of hyperlinked categories with brief descriptions. The "Users" column includes Administrators, Groups & Roles, Synchronize and Import, and User Management. The "Elements" column includes Application Management, Communication Manager, Conferencing, Inventory, Messaging, Presence, Routing, Session Manager, and SIP AS 8.1. The "Services" column includes Backup and Restore, Configurations, Events, Licenses, Replication, Scheduler, Security, and Templates.

6.2. Administer Adaptation

In order for successful interoperation of SEMAP+ with Session Manager over a SIP trunk, an Adaptation must be configured to replace the outbound domain destination with the IP address of the SEMAP+ server. Click **Home** → **Elements** → **Routing** → **Adaptations** → **New** assign and identifying **Adaptation name**, 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 **fromto=true odst=10.10.17.2** and click on **Commit**.



The screenshot shows the "Adaptation Details" form in the Avaya Aura® System Manager 6.1 interface. The breadcrumb trail at the top reads "Home / Elements / Routing / Adaptations - Adaptation Details". On the left is a sidebar menu with "Routing" expanded, showing sub-items like Domains, Locations, Adaptations, SIP Entities, etc. The main form area has a "General" tab selected. It contains fields for "Adaptation name" (filled with "SEMAPAdapt"), "Module name" (a dropdown menu showing "DigitConversionAdapter"), and "Module parameter" (filled with "fromto=true odst=10.10.17.2"). There are also empty fields for "Egress URI Parameters" and "Notes". At the top right of the form are "Commit" and "Cancel" buttons, with a "Help ?" link next to them.

6.3. Administer SIP Entity

Each SIP device (other than Avaya SIP Phones) that communicates with the Session Manager requires a SIP Entity configuration. This section details the steps to create SIP Entity for the SEMAP+ Solution.

6.3.1. Configure Netcall SEMAP+ Entity

Click **Home** → **Elements** → **Routing** → **SIP Entities** → **New** assign an identifying **Name**, the **FQDN or IP Address** for the SEMAP+ server, set the **Type** to **SIP Trunk**, choose the **Adaptation** configured in **Section 6.4** from the drop down box and set **Call Detail Recording** to **none**, leave all other settings default and click **Commit**.

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

Routing × Home

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

SIP Entity Details

General

* Name: SEMAP+

* FQDN or IP Address: 10.10.17.2

Type: SIP Trunk

Notes:

Adaptation: SEMAPAdapt

Location:

Time Zone: America/Fortaleza

Override Port & Transport with DNS SRV: ☐

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

Credential name:

Call Detail Recording: none

Commit Cancel

The screen below will now be displayed confirming the entry.

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

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

▼ Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / SIP Entities - SIP Entities

SIP Entities

[Edit](#) [New](#) [Duplicate](#) [Delete](#) [More Actions](#)

5 Items | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Name	FQDN or IP Address	Type	Notes
<input type="checkbox"/>	61sesmgr	10.10.16.201	Session Manager	
<input type="checkbox"/>	Commgr	10.10.16.31	CM	
<input type="checkbox"/>	SEMAP+	10.10.17.2	SIP Trunk	

Select : [All](#), [None](#)

6.4. Administer SIP Entity Link

A SIP Trunk between a Session Manager and a telephony system is described by an Entity Link. An Entity Link needs to be created between Session Manager and SEMAP+.

6.4.1. Administer SIP Entity Link from Avaya Aura® Session Manager to Netcall SEMAP+

Click on **Home** → **Elements** → **Routing** → **Entity Links** → **New** assign an identifying **Name** choose the entity assigned to the Session Manager SIP Signaling Interface as **SIP Entity 1**, set the **Protocol** as **TCP**, enter **5060** for the **Port**, choose the SEMAP+ 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 SEMAP+.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Entity Links

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* toSemap+	* 61sesmgr	TCP	* 5060	* SEMAP+	* 5060	Trusted	

* Input Required

Commit Cancel

The screen below will be shown confirming the entry.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing x Home

Entity Links

Edit New Duplicate Delete More Actions

4 Items | Refresh Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
<input type="checkbox"/>	toCM	61sesmgr	TCP	5060	Commgr	5060	Trusted	
<input type="checkbox"/>	toSemap+	61sesmgr	TCP	5060	SEMAP+	5060	Trusted	

Select : All, None

6.5. Administer Routing Policies

To complete the routing configuration, a Routing Policy is created. Routing policies direct how calls will be routed to an attached system. Two routing policies must be created, one for the Communications Manager and the second for SEMAP+. These will be associated with the Dial Patterns created in **Section 6.10**.

6.5.1. Create Routing Policy to Avaya Aura® Communication Manager

Click **Home** → **Elements** → **Routing** → **Routing Policies** → **New** assign an identifying **Name** for the route. Under the **SIP Entity as Destination** section, click on **Select** and choose the CM SIP Entity and click **Select**. Click **Commit** when done.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Routing * Home

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

Routing Policy Details

General

* Name: ToCM

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Commgr	10.10.16.31	CM	

The screen below will be shown confirming the entry.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Routing * Home

Home / Elements / Routing / Routing Policies - Routing Policies

Routing Policies

Edit New Duplicate Delete More Actions

2 Items | Refresh

Filter: Enable

Name	Disabled	Destination	Notes
ToCM	<input type="checkbox"/>	Commgr	

6.5.2. Create Routing Policy to Netcall SEMAP+

Click **Home** → **Elements** → **Routing** → **Routing Policies** → **New** assign an identifying **Name** for the route. Under the **SIP Entity as Destination** section, click on **Select** and choose the **SEMAP+** SIP Entity and click **Select**. Click **Commit** when done.

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

General

* Name: ToSemap+

Disabled: ☐

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
SEMAP+	10.10.17.2	SIP Trunk	

The screen below will be shown confirming the entry.

Avaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Routing Policies - Routing Policies

Routing Policies

Edit New Duplicate Delete More Actions

4 Items | Refresh Filter: Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	ToCM	<input type="checkbox"/>	Commgr	
<input type="checkbox"/>	ToSemap+	<input type="checkbox"/>	SEMAP+	

Select : All, None

6.6. Administer Dial Patterns

As one of its main functions, 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 system.

6.6.1. Create Dial Pattern for calls to Netcall SEMAP+

In **Section 5.7** Communication Manager is configured to route 4 digit strings beginning with 85 to Session Manager. To create a Dial Pattern to route these digits from Session Manager to SEMAP+ click **Home → Elements → Routing → Dial Patterns → New**. Under **Pattern** enter the numbers presented to Session Manager by Communication Manager in the **Patterns** box. Set **Min** and **Max** digit string length, and set **SIP Domain** to **ALL**.

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 ?](#)

Dial Pattern Details [Commit](#) [Cancel](#)

General

* Pattern: 85

* Min: 4

* Max: 4

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

In the **Originating Locations and Routing Policies** section of the web page, click **Add**. In the **Origination Location** section place a tick in the box next to **Apply The Selected Routing Policies to All Originating Locations**, in the **Routing Policies** section click the routing policy created for SEMAP+. Click **Select** when done.

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

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

Home / Elements / Routing / Dial Patterns - Originating Location and Routing Policy List

Originating Location and Routing Policy List [Select](#) [Cancel](#)

Origination Location

☒ Apply The Selected Routing Policies to All Originating Locations

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

<input checked="" type="checkbox"/>	Name	Notes
<input type="checkbox"/>	SessionMGR	

Select : All, None

Routing Policies

4 Items | [Refresh](#) Filter: [Enable](#)

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	ToCM	<input type="checkbox"/>	Commqr	
<input checked="" type="checkbox"/>	ToSemap+	<input type="checkbox"/>	SEMAP+	

Select : All, None

Click **Commit** when complete.

AVAYAAvaya Aura® System Manager 6.1

Help | About | Change Password | Log off admin

Routing *Home

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

Help ?

Dial Pattern Details

Commit

Cancel

General

* Pattern: 85

* Min: 4

* Max: 4

Emergency Call: ☐

SIP Domain: -ALL-

Notes:

Originating Locations and Routing Policies

Add

Remove

1 Item | Refresh

Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	-ALL-	Any Locations	ToSemap+	0	<input type="checkbox"/>	SEMAP+	

Select : All, None

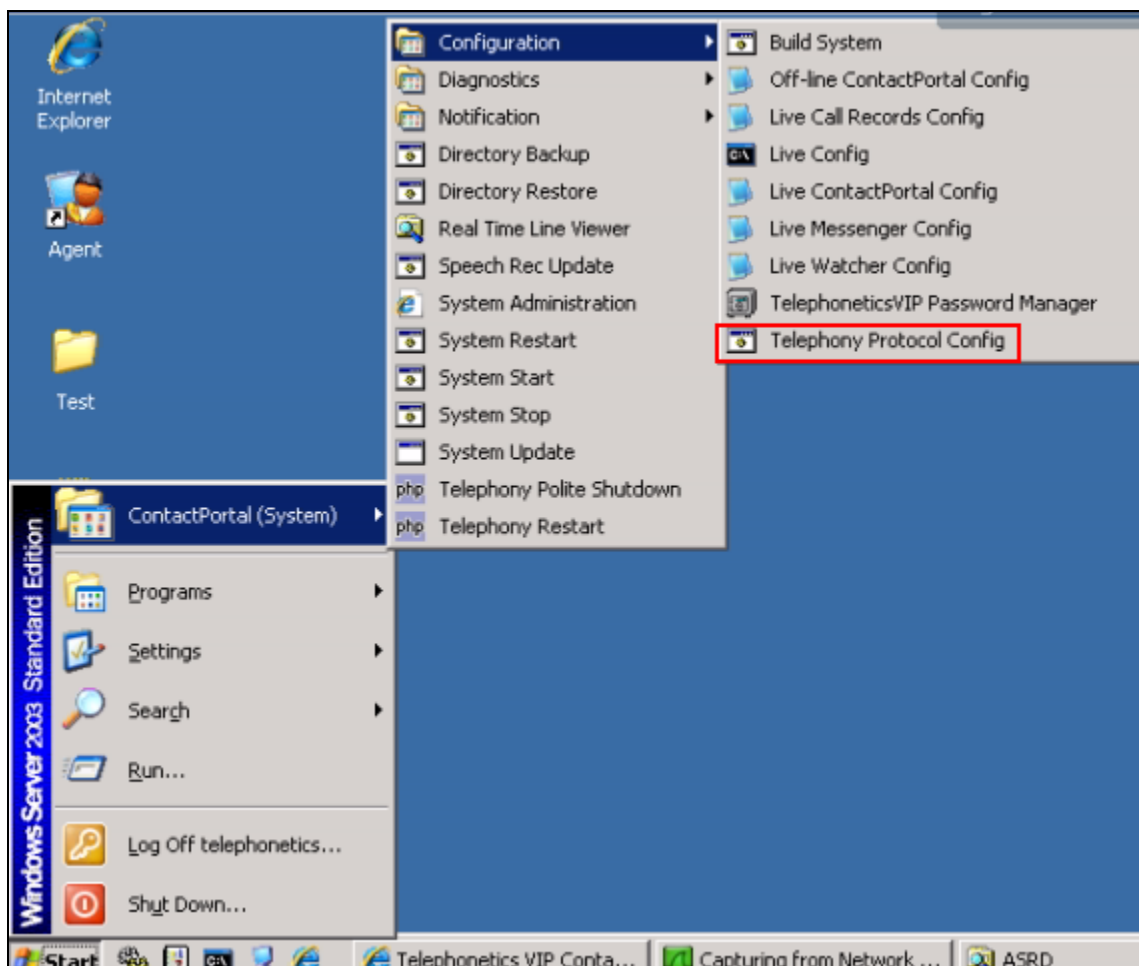
7. Configure Netcall SEMAP+

These Application Notes assume that a SIP build of SEMAP+ v18.16 with a G5 SEMAP+ processor or above is provided and installed by Netcall. Administration of the interface between SEMAP+ and the Avaya solution is summarized as follows:

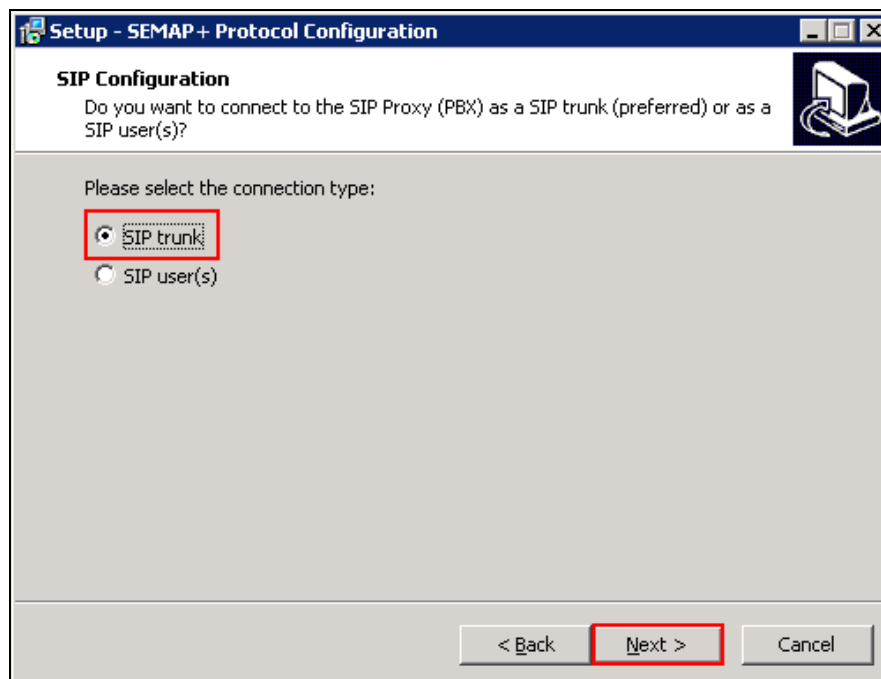
- Netcall SEMAP+ Protocol Configuration Wizard Setup
- Configure Netcall SEMAP+ Avaya MWI Special Settings
- Configure Netcall SEMAP+ Routing Rules
- Configure Netcall SEMAP+ Number Range Details
- Configure Netcall SEMAP+ Test User and Mailbox

7.1. Netcall SEMAP+ Protocol Configuration Wizard Setup

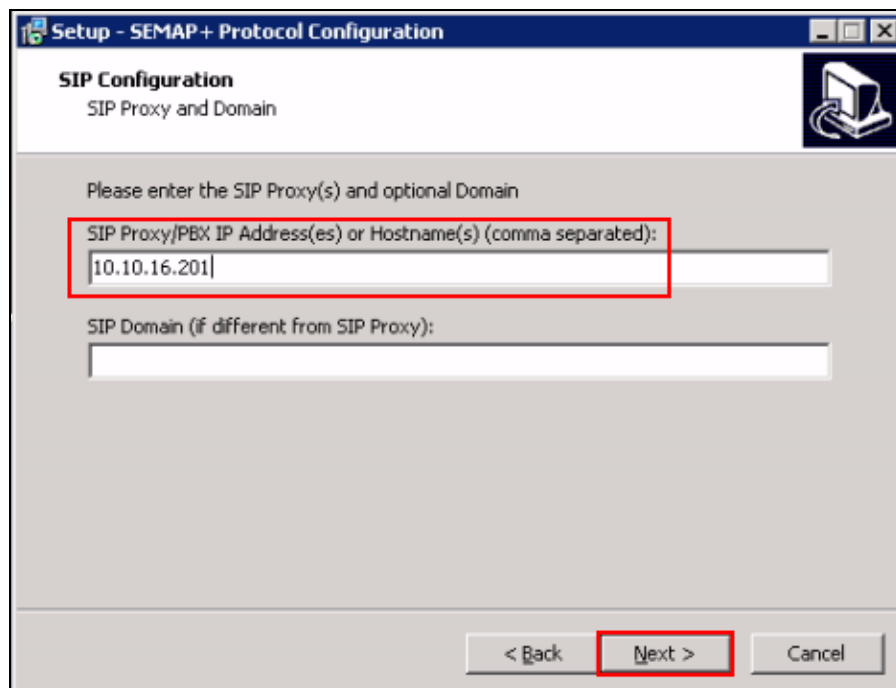
On the SEMAP+ server log in to the Windows OS and click **Start → ContactPortal → Configuration → Telephony Protocol Config.**



A familiar setup Microsoft Software Installer will start, choose the relevant SEMAP+ program option (not shown) and click next. At the **SIP Configuration** page click the radio button next to **SIP trunk** and click **Next**.



Enter the IP address of the Session Manager SIP Signalling interface in the **SIP Proxy/PBX IP Address(es) or Hostname(s) (comma separated):** box as shown below and click **Next**.



In the screen that appears, select **TCP** as the IP transport to use, and click **Next**.

Setup - SEMAP+ Protocol Configuration

SIP Configuration
SIP Transport

Please select the IP transport to use:

☒ TCP
☐ UDP

< Back **Next >** Cancel

Enter the **Inbound Channel Count** and **Outbound/Enquiry Channel Count**, according to requirements, as shown below and click **Next**.

Setup - SEMAP+ Protocol Configuration

SIP Configuration
Total Number of SIP Channels

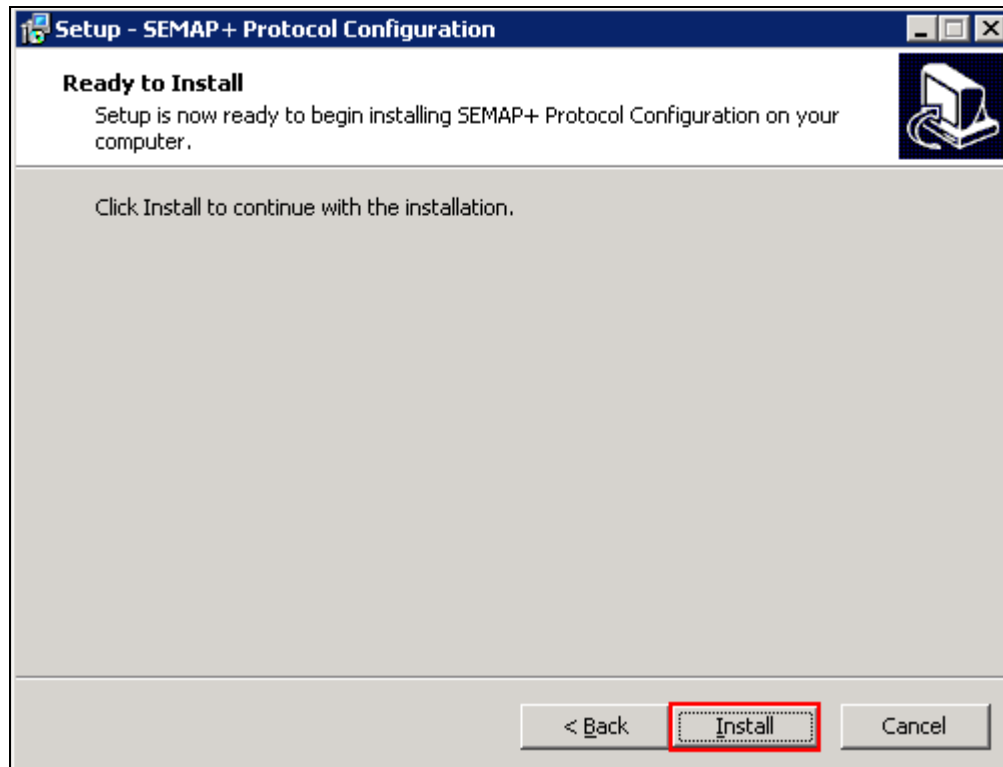
Please enter the number of inbound and outbound/enquiry SIP channels

Inbound Channel Count:
25

Outbound/Enquiry Channel Count:
25

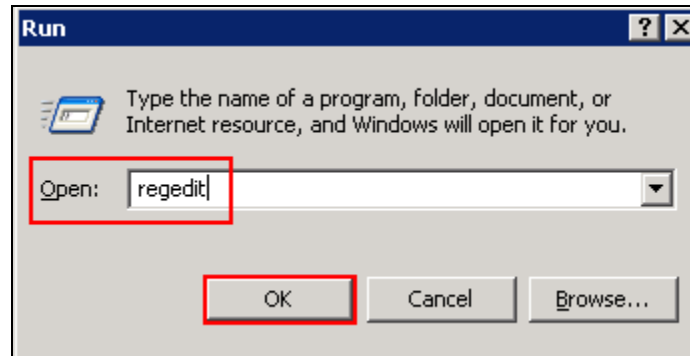
< Back **Next >** Cancel

Click **Install** to complete the installation with the configuration specified.



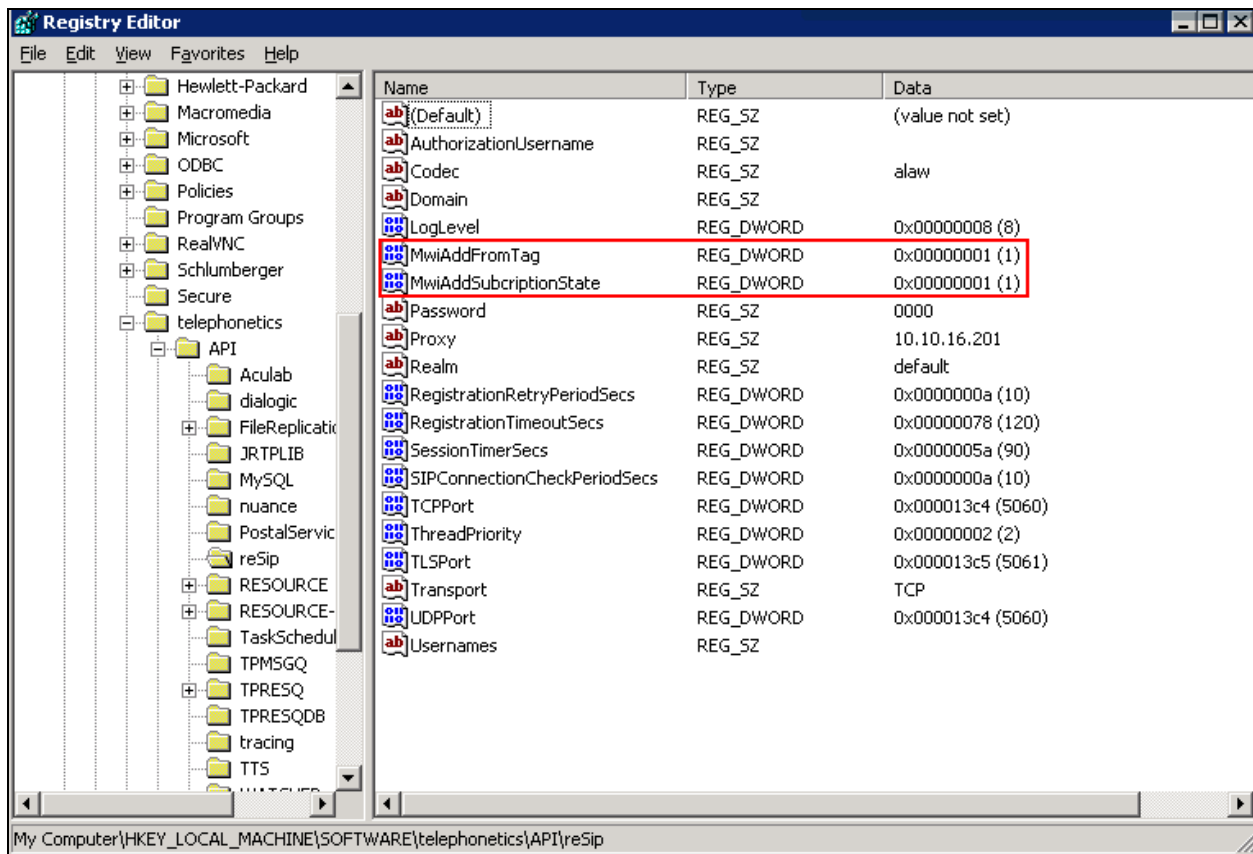
7.2. Configure Netcall SEMAP+ Avaya MWI Special Settings

On the SEMAP system two registry keys need to be added to ensure the MWI is sent to Session Manager in the correct format. From the SEMAP+ OS, click **Start** → **Run** type **regedit** in the Open box, and click **OK**.



Navigate to [**HKEY_LOCAL_MACHINE\SOFTWARE\Telephonetics\API\reSip**] and add the following as shown below:

- **MwiAddSubscriptionState=dword:00000001**
- **MwiAddFromTag=dword:00000001**



7.3. Configure Netcall SEMAP+ Routing Rules

The SEMAP+ routing rules must be configured with the access number assigned in **Section 5.6**. Access the web interface of the SEMAP+ server using `http://IP_OF_SEMAP+_SERVER/padmin` and log in using the credentials provided by Netcall. Click **Partition Administration** → **Log In** → **System Settings** → **Call Routing** the screen shown below will appear.

The screenshot shows the SEMAP+ ContactPortal web interface. The top header includes the 'Telephoneticsvip' logo on the left and 'SEMAP+ ContactPortal®' on the right. Below the header, a navigation bar contains 'Ayaya Test, Call Routing Rules', a search box labeled 'Search the Directory' with a 'Go' button, and links for 'Administration | User Log In | Help'. A 'back to: System Administration' link is also present. The main content area states 'The Call Routing Rules configure how calls are handled by this Partition.' and includes a button to 'Add a new routing rule:'. Below this is a dropdown menu with '<Please select...>' and an 'Add' button. At the bottom, there is a table with columns: 'Type', 'DDI Range', 'CLI Range', 'Line Range', 'Divert Range', and 'More details'. The table currently shows 'No call routing rules configured'.

Type	DDI Range	CLI Range	Line Range	Divert Range	More details
No call routing rules configured					

For the purposes of the compliance test, three call routing rules are created:

- 8500 – ContactPortal AutoAttendant
- 8502 – Voicemail Deposit
- 8503 - Voicemail Retrieval

Select **AutoAttendant** under the **Dialler** section, from the **Add a new routing rule** drop down box.

The screenshot shows the 'Ayaya Test, Call Routing Rules' page. A dropdown menu titled 'Add a new routing rule:' is open, displaying a list of options. The 'Dialler' section is expanded, and 'AutoAttendant' is highlighted. Other options include 'SpeedDial', 'Messenger+', 'DDI', 'ContactCentre 59R', 'Admin', 'Business Continuity', 'Custom', 'Other Services', and 'Meet Me Paging'.

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Send any feedback to support@telephoneticsvip.co.uk or visit our website at www.telephoneticsvip.co.uk

Enter **8500** in the **DDI Range** box, leave all other settings as default and click **Add Rule**.

The screenshot shows the 'Ayaya Test, Add AutoAttendant Rule' page. The 'Routing' section is highlighted, and the 'DDI Range' box contains the value '8500'. The 'Switch Integration' section is also visible, with 'Allow Route Optimisation?' checked and 'Initial Ring Time (secs)' set to '2'.

Routing:

DDI Range: 8500 to

CLI Range: to

Line Range: to

Divert Range: to

Caller Category(s):

Switch Integration:

Allow Route Optimisation? ☒ (Remember to configure route optimisation as part of the Number Ranges)

Initial Ring Time (secs): 2

Initial Silence Time (secs): 0.4

Inbound fax action: None

Add Rule

The screen below will appear showing the added **AutoAttendant** Rule.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Call Routing Rules Search the Directory Go Administration | User Log In | Help

back to: System Administration

The Call Routing Rules configure how calls are handled by this Partition.

Add a new routing rule:

<Please select...> Add

Type	DDI Range	CLI Range	Line Range	Divert Range	More details
AutoAttendant	8500	-	-	-	remove

Select **Check Messages** under the **Messenger+** section, from the **Add a new routing rule** drop down box.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Call Routing Rules Search the Directory Go Administration | User Log In | Help

back to: System Administration

The Call Routing Rules configure how calls are handled by this Partition.

Add a new routing rule:

<Please select...> Add

<Please select...>

Dialler

- AutoAttendant
- SpeedDial

Messenger+

- Check Messages
- General Messages
- Specific Mailbox

DDI

- Direct Dial
- DDI Divert Integration

ContactCentre 59R

- ACD Queue
- Outbound Agent Call

Admin

- User Administration
- Supervisor Administration
- System Administration

Business Continuity

- Business Continuity for Users
- Request DDI

Custom

- Dialogue
- Custom IVR Module

Other Services

- Number Translation
- Remote Extension Server
- Barred

Meet Me Paging

- Meet Me Paging - Send Page

Type	DDI Range	CLI Range	Line Range	Divert Range	More details
AutoAttendant	8500	-	-	-	remove

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Enter **8503** in the **DDI** field, leave all others settings as default and click **Add Rule**.

Telephoneticsvip

SEMAP+ ContactPortal®

Ayaya Test , Add General Messaging Rule for Contac...

Search the DirectoryGo | Administration | User Log In | Help

back to: Routing Rules

Calls to these Number Ranges will be routed into the *ContactPortal* VM Messaging System's General Messaging IVR. You can also choose a Dialler Interface and Caller Category to use if the caller transfers to the Dialler.

Routing:

DDI Range:

8503

to

CLI Range:

to

Line Range:

to

Divert Range:

to

Messaging System:

ContactPortal VM

Interface:

SpeedDial

Caller Category(s):

Virtual DDI for routing, when the diverted-from number does not match against any entities in the directory:

Switch Integration:

Allow Route Optimisation?

☒

(Remember to configure route optimisation as part of the Number Ranges)

Initial Ring Time (secs):

2

Initial Silence Time (secs):

0.4

Add Rule

The screen below will appear showing the added **Check Messages Rule**.

Type	DDI Range	CLI Range	Line Range	Divert Range	More details
AutoAttendant	8500	-	-	-	remove
Check Messages	8503	-	-	-	ContactPortal VM remove

Select **General Messages** under the **Messenger+** section, from the **Add a new routing rule** drop down box.

Type	DDI Range	CLI Range	Line Range	Divert Range	More details
AutoAttendant	8500	-	-	-	remove
Check Messages	8503	-	-	-	remove

Enter **8502** in the **DDI** field, leave all others settings as default and click **Add Rule**.

Telephoneticsvip

SEMAP+ ContactPortal®

Ayaya Test , Add General Messaging Rule for Contac...

Search the DirectoryGo | Administration | User Log In | Help

back to: Routing Rules

Calls to these Number Ranges will be routed into the *ContactPortal VM* Messaging System's General Messaging IVR. You can also choose a Dialler Interface and Caller Category to use if the caller transfers to the Dialler.

Routing:

DDI Range:

8502

to

CLI Range:

to

Line Range:

to

Divert Range:

to

Messaging System:

ContactPortal VM

Interface:

SpeedDial

Caller Category(s):

Virtual DDI for routing, when the diverted-from number does not match against any entities in the directory:

Switch Integration:

Allow Route Optimisation?

☒

(Remember to configure route optimisation as part of the Number Ranges)

Initial Ring Time (secs):

2

Initial Silence Time (secs):

0.4

Add Rule

The screen below will appear showing the added **General Messages** Rule.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Call Routing Rules Search the Directory Go Administration | User Log In | Help

back to: System Administration

The Call Routing Rules configure how calls are handled by this Partition.

Add a new routing rule:

<Please select...> Add

Type	DDI Range	CLI Range	Line Range	Divert Range	More details
AutoAttendant	8500	-	-	-	remove
Check Messages	8503	-	-	-	ContactPortal VM remove
General Messages	8502	-	-	-	ContactPortal VM remove

7.4. Configure Netcall SEMAP+ Number Range Details

Routing rules must be configured to define internal, external and barred numbers. Click **System Administration** → **Number Range Details** → **Add a new rule**.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Number Ranges Search the Directory Go Administration | User Log In | Help

Options: Add a new rule back to: System Settings

! There are currently no number ranges configured.

For the purposed of the compliance test, three new routing rules are added:

- Internal Numbers
- External Number
- Barred Number

In the **Name** field enter an identifying name for the rule, in this case **Avaya Internal Test Extensions**, enter the **Start of Number Range** and **End of Number Range** accordingly, in this case **1000** and **8999** respectively, Leave all other fields as default and click **Add**.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Add New Number Range: Search the Directory Go | Administration | User Log In | Help

back to: Number Range Summary

Enter the details for the new Number Range rule and then click 'Add'

Details:

Name: Avaya Internal Test Extensions

Start of Number Range: 1000

End of Number Range: 8999

Server Name:

Number Type: Internal

Prefix to dial:

Add

The following screen will appear displaying the new rule.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Number Ranges Search the Directory Go | Administration | User Log In | Help

Options: Add a new rule back to: System Settings

The following list shows the available Number Ranges:

Rule Name	Definition	Type	Prefix	Optimise	Recording
Avaya Internal Test Extensions	1000-8999	Internal		X	X

clone | remove

Note that Optimise is disabled, this rule can be edited to enable Optimise. This feature of SEMAP+ uses SIP REFER to free up SEMAP+ resources.

Click **Add a new rule**, in the **Name** field enter an identifying name for the rule, in this case **Outside Line**, set the **Start of Number Range** as **0**, this is the first digit for an external call, select **External** from the **Number Type** drop down box, and set **Prefix to dial** as **9** leave all other fields as default and click **Add**.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Add New Number Range: Search the Directory Go Administration | User Log In | Help

back to: Number Range Summary

Enter the details for the new Number Range rule and then click 'Add'

Details:

Name: Outside Line

Start of Number Range: 0

End of Number Range:

Server Name:

Number Type: External

Prefix to dial: 9

Add

The following screen will appear displaying the new rule.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Number Ranges Search the Directory Go Administration | User Log In | Help

Options: Add a new rule back to: System Settings

The following list shows the available Number Ranges:

Rule Name	Definition	Type	Prefix	Optimise	Recording	
Ayaya Internal Test Extensions	1000-8999	Internal		✗	✗	clone remove
Outside Line	0	External	9	✗	✗	clone remove

Click **Add a new rule**, in the **Name** field enter an identifying name for the rule, in this case **Barred Number**, set the **Start of Number Range** as **4002**, leave all other fields as default and click **Add**.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Add New Number Range: Search the Directory Go Administration | User Log In | Help

back to: Number Range Summary

Enter the details for the new Number Range rule and then click 'Add'

Details:

Name: Barred Number

Start of Number Range: 4002

End of Number Range:

Server Name:

Number Type: Internal

Prefix to dial:

Add

The following screen will appear displaying the new rule.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Number Ranges Search the Directory Go Administration | User Log In | Help

Options: Add a new rule back to: System Settings

The following list shows the available Number Ranges:

Rule Name	Definition	Type	Prefix	Optimise	Recording	
Avaya Internal Test Extensions	1000-8999	Internal		X	X	clone remove
Barred Number	4002	Internal		X	X	clone remove
Outside Line	0	External	9	X	X	clone remove

Once administered, the routing rules can be edited, click on the routing rule to edit, by default **Route Optimisation** is **Off** and **Delay** is set to **0**, change **Route Optimisation** to **On for all callers** and **Delay Before Route Optimising (secs)** to **2**. This will enable SIP REFER to take place 2 seconds until the established call.

Telephoneticsvip

SEMAP+ ContactPortal®

Ayaya Test, Number Range: Ayaya Internal Test Ext...

Search the DirectoryGo | Administration | User Log In | Help

back to: Number Range Summary

Update the details for the Number Range and then click 'Update'

Details:

Name:

Ayaya Internal Test Extensions

Start of Number Range:

1000

End of Number Range:

8999

Server Name:

Number Type:

Internal

Prefix to dial:

General Telephony Integration:

Route Optimisation:

On for all callers

Delay Before Route Optimising (secs):
(ignored if SIP reINVITE policy set to 'Never')

2

Use Single Step Call Transfer
(supported on QSig and SIP only)

☐

Scroll down the page and select **Always reINVITE** from the **SIP reINVITE Policy** drop down box, this will enable shuffling, it is not recommended that this is used where call recording is used in the solution. Click **Update** when done.

SIP Specific Telephony Integration:

SIP reINVITE Policy: Always reINVITE

IVR Options:

Default Ringlength (secs):

Force positive call accept:
(only if call screening is permitted) ☐

Internal Number Presentation Rule:

External Number Presentation Rule:

Bar external calls to this range: ☐

Call Recording:

Record calls to numbers in this range (secs):
(0 for no recording)

Stop recording if the call is remotely transferred: ☒

Messaging:

Can send SMS to numbers in this range: ☐

[Update](#)

7.5. Configure Netcall SEMAP+ Test User and Mailbox

For the purposes of the compliance test, users are added manually. Continuing from the Number Range Details page click **System Settings → Administration Home → Directory Management** the screen below will appear, click on **Add a new Person**.

Telephoneticsvip SEMAP+ ContactPortal®

Ayaya Test, Directory Management | [Administration](#) | [User Log In](#) | [Help](#)

[back to: Administration Home](#)

- [Browse the Directory](#)
- [Add a new Person](#) [Non-Person](#) | [Dialogue](#)
- [Configure Templates](#)
- [Class Of Service Levels](#)
- [Configure Link Groups](#)
- [Directory Synchronisation](#)
- [Export Directory](#)
- [Import Directory](#)
- [Missing Prompts](#)

Enter the details and extension number as required, in this case the **Title**, **First Name** and **Last Name** are defined. Under the **Contact Numbers** section the **Desk** number is set to **4001** and a tick is placed next to **Messenger+** to define that this user has a voicemail box. In the **General Information** section the **User ID** is set to **4001**, click **Add Entry** (not shown) when done.

Telephoneticsvip

SEMAP+ ContactPortal®

Ayaya Test, Add Person to the Directory

Search the DirectoryGo | Administration | User Log In | Help

back to: Directory Management

Name & Type (required)

Title:

First Name:

Last Name:

Gender:

Known by Title:

Name

Mr

Richard

Pope

Male


☐


(To add a department, company or resource, use [this form](#) instead)


Template


ContactCentre 59R


Contact Numbers (at least one required)

Desk:  4001

Mobile: 

Home: 

Secretary: 

Voicemail#: 

☐ External

☒ Messenger+

General Information (optional)

User ID: 4001

Email Address:

DDI:

Job Title:

Department:

Company:

Location:

Click **Search Directory**, enter the name of the added user and click **Search**.

Telephoneticsvip **SEMAP+ ContactPortal®**
Ayaya Test, Browse Directory Administration | User Log In | Help
Add: Person | Non-Person | Dialogue back to: Directory Management

Browse Directory:

Search: in:

☐ All Details ☒ Names only

Click **Messaging**.

Telephoneticsvip **SEMAP+ ContactPortal®**
Richard Pope Search the Directory Go Administration | User Log In | Help
back to: Directory

Calls are being forwarded to your desk Turn ON Do Not Disturb

Desk	4001	forward calls to my desk	edit call screening	remove
Mobile	(not set)			add
Home	(not set)			add
Secretary	(not set)			add
Pager	(not set)			add
Voicemail	ContactPortal VM (Leave a Message)	forward calls to my voicemail		remove

Personal Details | Call Options | Add Custom Number | Name Recordings | Speed-Dials | **Messaging**
Business Continuity | What callers can say | Contact Availability | Pronunciations | Links | Delete Entry

Click **Settings**.

Telephoneticsvip **SEMAP+ ContactPortal®**
Richard Pope Search the Directory Go Administration | User Log In | Help
back to: Contact Information

Delete | Save | Copy | Reply | Email | Mark as Read

Inbox | Saved Messages | Missed Calls
Deleted Items (6) **Settings**

From	Description	Received
You have no messages in your inbox		

Ensure a tick is placed next to **MWI** and click **Update Notification Settings**.

Telephoneticsvip

SEMAP+ ContactPortal®

Richard Pope

Search the DirectoryGoAdministration | User Log In | Help

back to: Contact Information

[Inbox](#) | [Saved Messages](#) | [Missed Calls](#)
[Deleted Items \(6\)](#) | [Settings](#)

Notification Settings:

Notify me via...	Announce on SpeedDial:	Email:	SMS:	Beeper/Pager:	MWI:	Dialout Delivery:
When I receive a new message	<input type="checkbox"/>	×	×	<input type="checkbox"/>	<input checked="" type="checkbox"/>	×
Or when I miss a call via an internal interface	<input type="checkbox"/>	×	×	<input type="checkbox"/>		
...via an external interface	<input type="checkbox"/>	×	×	<input type="checkbox"/>		
...call due to Do Not Disturb	<input type="checkbox"/>	×	×	<input type="checkbox"/>		

Sorry. This system does not support Email or SMS messaging!

Update Notification Settings

8. Verification Steps

This section provides the tests that can be performed to verify correct configuration of Avaya and SEMAP+.

8.1. Verify Avaya Aura® Communication Manager SIP Trunk

Enter the command **status trunk x** where x is the trunk configured in **Section 5.3**. Confirm all channels are **in-service/idle**.

status trunk 2				Page	1
TRUNK GROUP STATUS					
Member	Port	Service State	Mtce Connected Ports Busy		
0002/001	T00009	in-service/idle	no		
0002/002	T00010	in-service/idle	no		
0002/003	T00011	in-service/idle	no		
0002/004	T00012	in-service/idle	no		
0002/005	T00013	in-service/idle	no		
0002/006	T00014	in-service/idle	no		
0002/007	T00015	in-service/idle	no		
0002/008	T00016	in-service/idle	no		
0002/009	T00017	in-service/idle	no		
0002/010	T00018	in-service/idle	no		
0002/011	T00019	in-service/idle	no		
0002/012	T00020	in-service/idle	no		
0002/013	T00021	in-service/idle	no		
0002/014	T00022	in-service/idle	no		

8.2. Verify Avaya Aura® Communication Manager Entity Link Status

From the System Manager web interface click **Home → Elements → Session Manager → System Status → SIP Entity Monitoring → Commgr → Show** verify **Conn. Status** and **Link Status** is **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 displays a table of entity links for the "Commgr" SIP entity. 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" columns are highlighted with red boxes, both showing "Up". Below the table, there are sections for "Time Last Down", "Time Last Up", "Last Message Sent", "Last Message Response", and "Last Response Latency (ms)".

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▼ Hide	61sesmgr	10.10.16.31	5060	TCP	Up	200 OK	Up

Time Last Down	Time Last Up	Last Message Sent	Last Message Response	Last Response Latency (ms)
Never	Jan 19, 2012 2:48:50 PM GMT+00:00	Jan 24, 2012 10:46:15 AM GMT+00:00		57

8.3. Verify Netcall SEMAP+ Entity Link Status

From the System Manager web interface click **Home → Elements → Session Manager → System Status → SIP Entity Monitoring → SEMAP+ → Show** verify **Conn. Status** and **Link Status** is **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 displays a table of entity links for the "SEMAP+" SIP entity. 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" columns are highlighted with red boxes, both showing "Up". Below the table, there are sections for "Time Last Down", "Time Last Up", "Last Message Sent", "Last Message Response", and "Last Response Latency (ms)".

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
▼ Hide	61sesmgr	10.10.17.2	5060	TCP	Up	200 OK	Up

Time Last Down	Time Last Up	Last Message Sent	Last Message Response	Last Response Latency (ms)
Jan 23, 2012 11:06:40 AM GMT+00:00	Jan 23, 2012 2:09:32 PM GMT+00:00	Jan 24, 2012 10:46:15 AM GMT+00:00		228

8.4. Verify Avaya Aura® Session Manager SIP Call Routing

In order to verify the usage of the Session Manager entity link, adaptation and dial pattern configured, SSH into the Session Manager management interface, in this case, 10.10.16.54, and enter the command **traceSM**. Place a call to SEMAP+ and verify the use of the configured parameters as shown below.

```
61sesmgr - traceSM - FILTERED - Captured: 58 Displayed: 36
-----
Commgr:      SEMAP+
SM100
-----
15:40:56,513 | --INVITE--> | | (2) T:8500 F:anonymous@anonymous U:8500 P:terminating
15:40:56,514 | <--Trying-- | | (2) 100 Trying
15:40:56,515 | Remote host is trusted | Trusted
15:40:56,515 | Non registered contact | Handle: anonymous Domain: anonymous.invalid
15:40:56,515 | Location found | Location: SessionMGR
15:40:56,516 | Originating Location found | Location: SessionMGR
15:40:56,516 | Try routing to determine if eme | Location: SessionMGR
15:40:56,516 | Request Dial Pattern route | for: sip:8500@avaya.com Location: SessionMGR
15:40:56,516 | Dial Pattern route parameters | URI Domain: avaya.com Location: SessionMGR
15:40:56,516 | Trying Dial Pattern route | Domain: avaya.com Location: SessionMGR
15:40:56,516 | Dial Pattern route parameters | URI Domain: null Location: SessionMGR
15:40:56,516 | Trying Dial Pattern route | Domain: null Location: SessionMGR
15:40:56,516 | Dial Pattern route parameters | URI Domain: avaya.com Location: null
15:40:56,516 | Trying Dial Pattern route | Domain: avaya.com Location: null
15:40:56,516 | Dial Pattern route parameters | URI Domain: null Location: null
15:40:56,516 | Trying Dial Pattern route | Domain: null Location: null
15:40:56,516 | Dial Pattern found | for: 8500 Pattern: 85
15:40:56,516 | Route Policy found | Pattern: 85 RoutePolicyList: ToSemap+
15:40:56,516 | Route found | for: sip:8500@avaya.com SIPEntity: SEMAP+
15:40:56,517 | Entity link found | SIPEntity: SEMAP+ EntityLink: 61sesmgr->TCP, biDirId=null:5060
15:40:56,517 | Request Adaptation | Adapter: SEMAPAdapt
15:40:56,517 | Applied egress Adaptation | To=<sip:8500@10.10.17.2>, Request-URI=sip:8500@10.10.17.2;routeinfo=0-0
15:40:56,518 | Routing SIP request | SipEntity: SEMAP+ EntityLink: 61sesmgr->TCP:5060
15:40:56,519 | No hostname resolution required | Routing to: sip:10.10.17.2;transport=tcp;lr;phase=terminating
15:40:56,519 | Using Session Manager Location | Location: SessionMGR
15:40:56,520 | | --INVITE--> | | (2) T:8500 F:anonymous@anonymous U:8500 P:terminating
15:40:56,764 | | <--Trying-- | | (2) 100 Trying
15:40:56,923 | | <--Ringing-- | | (2) 180 Ringing
15:40:56,925 | Request Adaptation | Adapter: SEMAPAdapt
15:40:56,927 | <--Ringing-- | | (2) 180 Ringing
15:40:58,904 | <--200 OK-- | | (2) 200 OK (INVITE)
15:40:58,906 | Request Adaptation | Adapter: SEMAPAdapt
15:40:58,907 | <--200 OK-- | | (2) 200 OK (INVITE)
15:40:58,945 | ----ACK----> | | (2) sip:8500@10.10.17.2
15:40:58,947 | Request Adaptation | Adapter: SEMAPAdapt
15:40:58,948 | ----ACK----> | | (2) sip:8500@10.10.17.2
```

8.5. Verify SEMAP+ Subsystem Status

From the SEMAP+ web interface, click **System Status**, verify all systems are **UP**.

The screenshot shows the SEMAP+ ContactPortal® System-Status page. The page header includes the Telephoneticsvip logo and the SEMAP+ ContactPortal® logo. The main content area displays the status of various subsystems for the host WEBTEST. A red box highlights the 'System UP' status for each subsystem.

Host: WEBTEST	
Telephony Core (Including Notify)	System UP
Database Server	System UP
Speech Recognition Server	System UP
Speech Recognition Server	System UP
Speech Recognition Licence Manager	System UP
Speech Recognition Resource Manager	System UP
Speech Recognition Compilation Server	System UP
Queue Manager	System UP
Licence Manager	System UP
Messaging Server	System UP
Telephony Integration Server (MWI, DTMF Pager, ProCenter®)	System UP
Watcher (Configured to report events)	System UP

8.6. Verify SEMAP+ Access

Manually place a call to the SEMAP+ and verify successful interpretation of ASR and appropriate response of the Messaging+ application.

8.7. Verify Netcall SEMAP+ Logging and SEMAP+ Version

On the SEMAP+ server, browse to **E:\tptrace\<date>.log** and verify the file contains application logging information including SIP messaging. Confirm that the **User-Agent** is shown as **ContactPortal/18.16**.

```
OPTIONS sip:10.10.16.201:5060;transport=tcp SIP/2.0
Via: SIP/2.0/ ;branch=z9hG4bK-d8754z-9e3c2854a804f834-1---d8754z-;rport
Max-Forwards: 70
Contact: <sip:contactportal>
To: <sip:10.10.16.201:5060;transport=tcp>
From: <sip:contactportal@10.10.16.201>;tag=415d4964
Call-ID: YTYzOTA3OGM4MDQ2ZTVmZmFjNjQzMWUxMjA1MWUzY2E.
CSeq: 1 OPTIONS
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO, NOTIFY
Supported: timer
Content-Length: 0
```

```
Contact: <sip:8500@10.10.17.2>
To: <sip:8500@10.10.17.2>;tag=bc77c625
From: "PSTN,4000"<sip:anonymous@anonymous.invalid>;tag=0d84ae9625de117f4f475d9000
Call-ID: 0d84ae9625de118f4f475d9000
CSeq: 1 INVITE
Session-Expires: 1200;refresher=uac
Min-SE: 1200
Allow: INVITE, ACK, CANCEL, OPTIONS, BYE, INFO, NOTIFY
Content-Type: application/sdp
Supported: timer
User-Agent: ContactPortal/18.16
Content-Length: 207
```

Click **Start** → **ContactPortal (System)** → **Real Time Line Viewer**, place call to SEMAP+ and verify the **State, Direction, Session, Description, Call Duration** and **Called** are accurately presented.

[illegible]

On the same screen, use the bar at the bottom of the screen to scroll to the right and verify the **Port:Channel(session)** column displays a connection to the IP address of the Session Manager SIP Signaling Interface.

Enq.A	Enq.B	Description	Call Duration	Calling(ses...	Called(ses...	Called(enq...	Called(enq...	Port:Channel(session)	Port:Chan...	Port:Chan...	Call ID	Instance
Idle												0x01410048
Idle												0x01415748
Idle												0x0141B680
Idle												0x014220D0
Idle												0x01427BF8
Idle												0x0142EB78
Idle												0x01435718
Idle												0x0143B4A8
Idle												0x01441B40
Idle												0x014484A0
Idle												0x0144E9F8
Idle												0x01454918
Idle												0x0145B190
Idle												0x014618E8
Idle												0x01467E18
Idle												0x0146E5E0
Idle												0x01475018
Idle												0x0147B698
Idle												0x01481118
Idle												0x01486E48
Idle												0x0148D7C0
Idle	What name?...	12 secs	8500					10.10.16.201:8052f5...			16778625	0x01494010
Idle												0x0149AA10
Idle												0x014A2A88
Idle												0x014A86D8

9. Conclusion

These Application Notes describe the configuration steps required for Netcall SEMAP+ to successfully interoperate with Avaya Aura® Communication Manager, Avaya Aura® System Manager and Avaya Aura® Session Manager. All functionality and serviceability test cases were completed successfully with observations noted in **Section 2.2**.

10. Additional References

Product documentation for Avaya products may be found at <http://support.avaya.com>

- [1] *Administering Avaya Aura® Communication Manager – Release 6.0, Issue 6.0, June 2010*
- [2] *Administering Avaya Aura® Session Manager – Release 6.1, Issue 1, November 2010*

Netcall SEMAP+ Documentation can be found at <http://www.netcall.com>

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