



Avaya Solution & Interoperability Test Lab

Application Notes for configuring novamail from novalink with Avaya Aura® Communication Manager R7.0 – Issue 1.0

Abstract

These Application Notes describe the configuration for connecting the novalink novamail voicemail system via SIP trunks to Avaya Aura® Communication Manager using Avaya Aura® Session Manager.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results

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 configuration for connecting the novalink novamail voicemail system via a SIP trunk interface to Avaya Aura® Session Manager, to allow endpoints on Avaya Aura® Communication Manager route voice calls to novamail to allow callers leave a voicemail and Communication Manager Users retrieve these voicemails.

novamail lets user's record individual welcome messages. These can be manually activated or permanently assigned to a call reason. In the latter case, the system knows why the call has reached the VoiceBox and informs the caller that user is not in the building, is temporarily absent from the workplace, are on the phone, or that the call is being received outside office hours. In all cases, a distinction can be made if required between internal and external calls, with calls connected to various messages accordingly. This ensures that callers are informed at all times as to why you are unable to take the call personally, and told when they can expect a return call.

2. General Test Approach and Test Results

This section describes the compliance testing used to verify interoperability of novamail with Communication Manager and covers the general test approach and the test results. Calls were made to novamail over SIP trunks connecting Communication Manager and novamail. novamail was configured as a SIP Entity on Session Manager allowing calls route between novamail and Communication Manager via Session Manager.

novamail was manually configured using the web interface to receive, store, alert and playback voicemail messages.

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

2.1. Interoperability Compliance Testing

The interoperability compliance testing evaluated the ability of novamail to carry out a variety voicemail functions in various conditions to multiple types of endpoint according to the configuration made via the web interface. These included:

- Forwarding to voicemail.
- Leaving and retrieving voicemail to/from PSTN/SIP/H.323/Digital endpoints.
- Message Waiting Indication (MWI).
- Use of DTMF for retrieval and menu navigation.
- novamail calling to local and PSTN endpoints.
- Serviceability testing consisted of verifying the ability of novamail to recover from power or network interruption to both Session Manager and novamail.

2.2. Test Results

All test cases were executed successfully.

2.3. Support

Technical support can be obtained for novamail from the website <http://www.novalink.ch/en/> or from the following.

novalink GmbH
Business tower
Zuercherstrasse 310
8500 Frauenfeld
Switzerland
helpdesk@novalink.ch
Phone: +41 52 762 66 77
Fax: +41 52 762 66 99

3. Reference Configuration

The configuration in **Figure 1** is used to compliance test novamail with Communication Manager registering with Session Manager as a third party SIP entity. Calls are passed to novamail using SIP trunks.

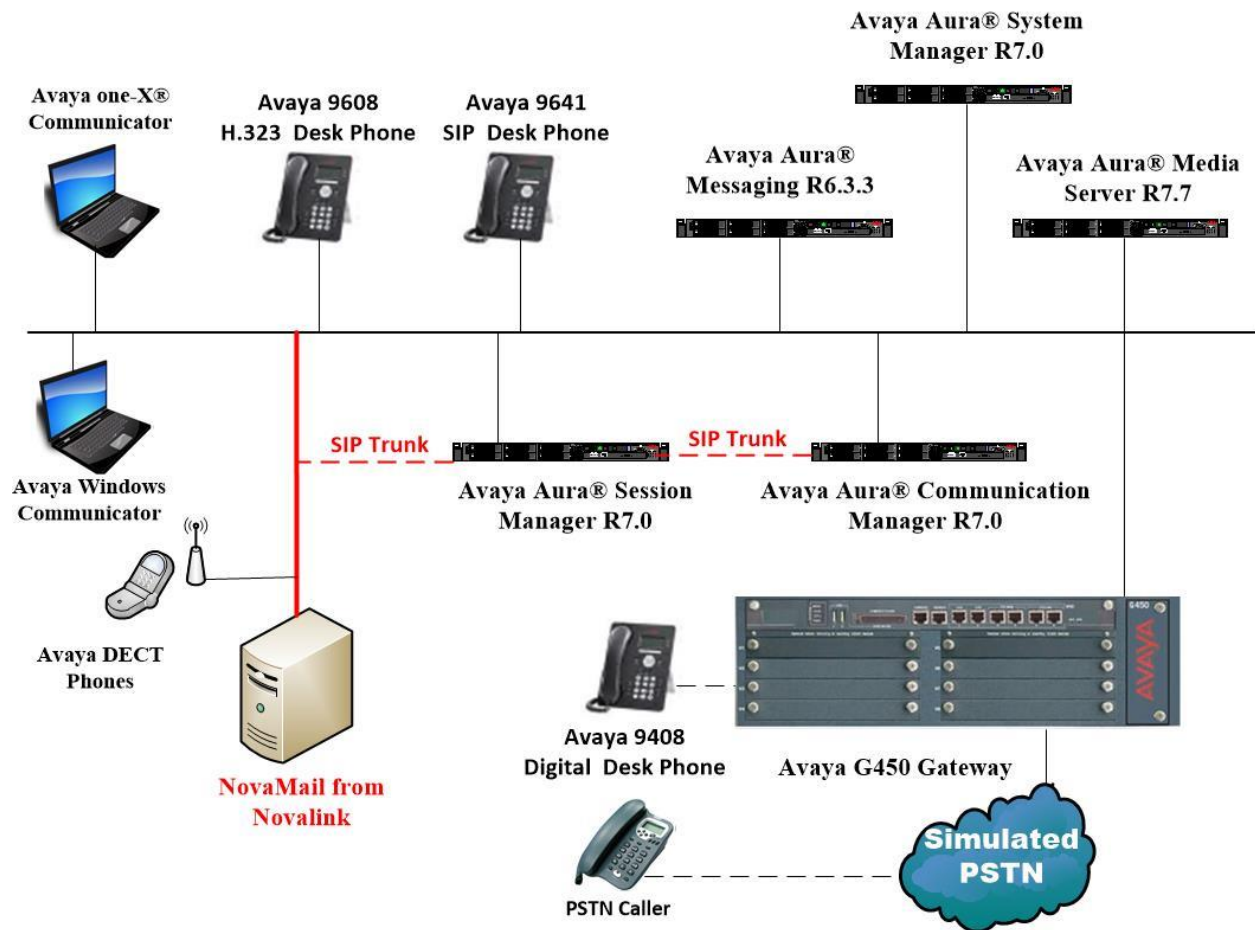


Figure 1: Connection of novamail from novalink with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® System Manager running on a virtual server	System Manager 7.0.1.1 Build No. - 7.0.0.0.16266 Software Update Revision No: 7.0.1.1.065378 Service Pack 1
Avaya Aura® Session Manager running on a virtual server	Session Manager R7.0 SP1 Build No. – 7.0.1.1.701114
Avaya Aura® Communication Manager running on a virtual server	R7.0 R017x.00.0.441.0 00.0.441.0-23169
Avaya Media Server running on a virtual server	Media Server SYSTEM R7.7.0.8 Media Server R7.7.0.200
Avaya G450 Gateway	37.19.0 /1
Avaya Aura® Messaging	R6.3.3
Avaya 9608 H323 Deskphone	96x1 H323 Release 6.6.028
Avaya 9641 SIP Deskphone	96x1 SIP Release 7.0.0.39
Avaya 9408 Digital Deskphone	V2.0
Avaya DECT Handsets	3725 DH4 (R3.3.11) 3720 DH3 (R3.3.11)
Avaya one-X® Communicator H.323	R6.2.4.07-FP4
Avaya Communicator for Windows	R2.1.3.80
novamail running on a Windows 2012 virtual server	9.8

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 in **Section 10**.

The configuration operations described in this section can be summarized as follows:

- Verify System Parameters Customer Options.
- System Features and Access Codes.
- Administer Dial Plan.
- Administer Route Selection for calls to novamail.
- Configure Network Region and IP Codec.

Note: The configuration of PSTN trunks and routes are outside the scope of these Application Notes.

5.1. Verify System Parameters Customer Options

The license file installed on the system controls these attributes. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya sales representative. Use the **display system-parameters customer-options** command to determine these values. On **Page 2**, verify that **Maximum Administered SIP Trunks** has sufficient capacity. Each call that receives IVR treatment from novamail uses a minimum of one SIP trunk. Calls that are routed back to stations commissioned on Communication Manager, or calls that are routed back to Communication Manager to access the PSTN, use 2 SIP trunks.

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES		USED	
Maximum Administered H.323 Trunks:		12000	250
Maximum Concurrently Registered IP Stations:		18000	2
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	319
Maximum Administered Ad-hoc Video Conferencing Ports:		24000	0

On **Page 3**, ensure that both **ARS** and **ARS/AAR Partitioning** are set to **y**.

display system-parameters customer-options	Page 3 of 11
OPTIONAL FEATURES	
Abbreviated Dialing Enhanced List? y	Audible Message Waiting? y
Access Security Gateway (ASG)? n	Authorization Codes? y
Analog Trunk Incoming Call ID? y	CAS Branch? n
A/D Grp/Sys List Dialing Start at 01? y	CAS Main? n
Answer Supervision by Call Classifier? y	Change COR by FAC? n
ARS? y	Computer Telephony Adjunct Links? y
ARS/AAR Partitioning? y	Cvg Of Calls Redirected Off-net? y
ARS/AAR Dialing without FAC? y	DCS (Basic)? y

On **Page 5**, ensure that **Uniform Dialing Plan** is set to **y**.

display system-parameters customer-options	Page 5 of 11
OPTIONAL FEATURES	
Multinational Locations? n	Station and Trunk MSP? y
Multiple Level Precedence & Preemption? n	Station as Virtual Extension? y
Multiple Locations? n	System Management Data Transfer? n
Personal Station Access (PSA)? y	Tenant Partitioning? y
PNC Duplication? n	Terminal Trans. Init. (TTI)? y
Port Network Support? y	Time of Day Routing? y
Posted Messages? y	TN2501 VAL Maximum Capacity? y
Private Networking? y	Uniform Dialing Plan? y
	Usage Allocation Enhancements? y

5.2.System Features and Access Codes

For the testing, **Trunk-to Trunk Transfer** was set to **all** on **page 1** of the **system-parameters features** page. This is a system wide setting that allows calls to be routed from one trunk to another and is usually turned off to help prevent toll fraud. An alternative to enabling this feature on a system wide basis is to control it using COR (Class of Restriction). See **Section 10** for supporting documentation.

display system-parameters features	Page 1 of 19
FEATURE-RELATED SYSTEM PARAMETERS	
Self Station Display Enabled? n	
Trunk-to-Trunk Transfer: all	
Automatic Callback with Called Party Queuing? n	
Automatic Callback - No Answer Timeout Interval (rings): 3	
Call Park Timeout Interval (minutes): 10	
Off-Premises Tone Detect Timeout Interval (seconds): 20	
AAR/ARS Dial Tone Required? y	
Music (or Silence) on Transferred Trunk Calls? no	
DID/Tie/ISDN/SIP Intercept Treatment: attd	
Internal Auto-Answer of Attnd-Extended/Transferred Calls: transferred	
Automatic Circuit Assurance (ACA) Enabled? n	
Abbreviated Dial Programming by Assigned Lists? n	
Auto Abbreviated/Delayed Transition Interval (rings): 2	
Protocol for Caller ID Analog Terminals: Bellcore	
Display Calling Number for Room to Room Caller ID Calls? n	

Use the **display feature-access-codes** command to verify that a FAC (feature access code) has been defined for both AAR and ARS. Note that **8** is used for AAR and **9** for ARS routing.

display 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:	
Answer Back Access Code:	
Attendant Access Code:	
Auto Alternate Routing (AAR) Access Code: 8	
Auto Route Selection (ARS) - Access Code 1: 9	Access Code 2:
Automatic Callback Activation: *25	Deactivation: #25

5.3.Administer Dial Plan

It was decided for compliance testing that all calls beginning with 49 with a total length of 4 digits were to be sent across the SIP trunk to Session Manager and therefore to novamail. In order to achieve this, automatic alternate routing (aar) would be used to route the calls. The dial plan and aar routing analysis need to be changed to allow this.

Type **change dialplan analysis**, in order to make changes to the dial plan. Ensure that **4** is added with a **Total Length** of **4** and a **Call Type** of **udp**.

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
2	4	ext						
3	4	ext						
4	4	udp						
5	4	ext						
6	4	udp						
7	3	dac						
8	1	fac						
9	1	fac						
*	3	fac						
#	3	fac						

5.4.Administer Route Selection for novamail Calls

As digits **49xx** were defined in the dial plan as udp (**Section 5.3**) use the **change uniform-dialplan** command to configure the routing of the dialed digits. In the example below calls to numbers beginning with **49** that are **4** digits in length will be matched. No further digits are deleted or inserted. Calls are sent to **aar** for further processing.

change uniform-dialplan 4									
UNIFORM DIAL PLAN TABLE									
Page 1 of 2									
Percent Full: 0									
Matching			Insert			Node			
Pattern	Len	Del	Digits	Net	Conv	Num			
49	4	0		aar	n				
						n			

Use the **change aar analysis x** command to further configure the routing of the dialed digits. Calls to novamail begin with **49** and are matched with the AAR entry shown below. Calls are sent to **Route Pattern 1**, which contains the outbound SIP Trunk Group.

change aar analysis 49							Page 1 of 2		
AAR DIGIT ANALYSIS TABLE									
Location: all							Percent Full: 1		
Dialed	Total		Route	Call	Node	ANI			
String	Min	Max	Pattern	Type	Num	Reqd			
49	4	4	1	unku		n			

Use the **change route-pattern n** command to add the SIP trunk group to the route pattern that AAR selects. In this configuration, **Route Pattern Number 1** is used to route calls to trunk group (**Grp No**) **1**, this is the SIP Trunk configured in **Appendix**.

change route-pattern 1										Page	1 of	3				
Pattern Number: 1										Pattern Name: SIPTRK						
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	unre					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			
6:	y	y	y	y	y	n	n	rest					none			

5.5. Configure Network Region and IP Codec

In the Node Names IP form, note the IP Address of the **procr** and the Session Manager (**sm70vmpg**). The host names will be used throughout the other configuration screens of Communication Manager and Session Manager. Type **display node-names ip** to show all the necessary node names.

display node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
AMS77vmpg	10.10.40.17	
CMS18vmpg	10.10.40.36	
IPO500V2	10.10.40.20	
IPOSE	10.10.40.25	
PGDECT	10.10.40.50	
aes70vmpg	10.10.40.26	
default	0.0.0.0	
procr	10.10.40.13	
procr6	::	
sm70vmpg	10.10.40.12	

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager in **Section 6.2**. In this configuration, the domain name is **devconnect.local**. The **IP Network Region** form also specifies the **IP Codec Set** to be used. This codec set will be used for calls routed over the SIP trunk to Session manager as **ip-network region 1** is specified in the SIP signaling group.

display ip-network-region 1		Page 1 of 20
IP NETWORK REGION		
Region: 1		
Location: 1	Authoritative Domain: devconnect.local	
Name: Default region		
MEDIA PARAMETERS	Intra-region IP-IP Direct Audio: yes	
Codec Set: 1	Inter-region IP-IP Direct Audio: yes	
UDP Port Min: 2048	IP Audio Hairpinning? n	
UDP Port Max: 3329		
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46		
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5		
H.323 IP ENDPOINTS	AUDIO RESOURCE RESERVATION PARAMETERS	
H.323 Link Bounce Recovery? y	RSVP Enabled? n	
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

In the **IP Codec Set** form, select the audio codec's supported for calls routed over the SIP trunk to novamail. The form is accessed via the **change ip-codec-set n** command. Note that IP codec set 1 was specified in IP Network Region 1 shown above. Multiple codecs may be specified in the **IP Codec Set** form in order of preference; the example below includes **G.711A** (a-law), which is supported by novamail. Note the **Media Encryption** has been set to **none**. This ensures that no media is encrypted.

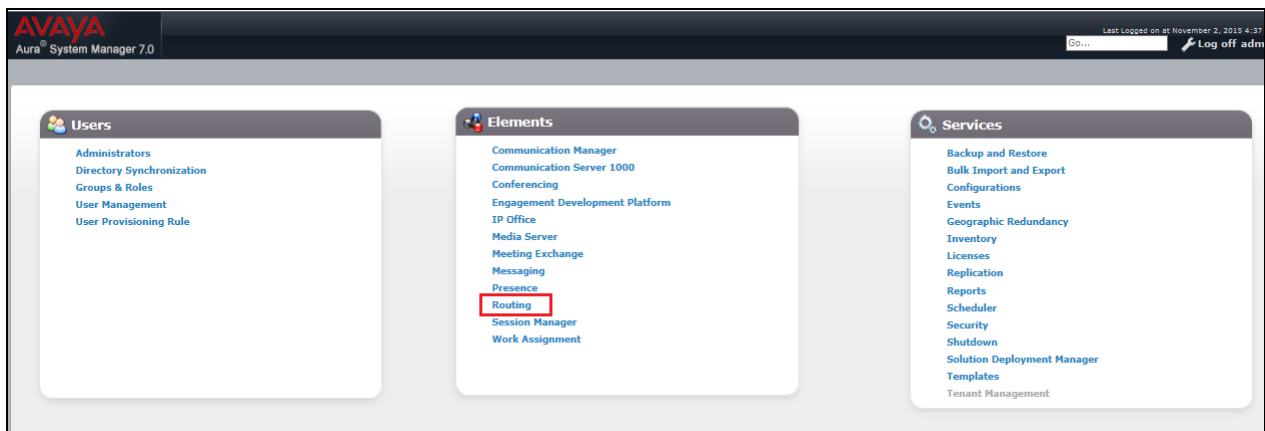
change ip-codec-set 1				Page 1 of 2
IP CODEC SET				
Codec Set: 1				
Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)	
1: G.711A	n	2	20	
2:				
3:				
4:				
5:				
6:				
7:				
Media Encryption		Encrypted SRTP:		
1: none				
2:				
3:				
4:				
5:				

6. Configure Avaya Aura® Session Manager

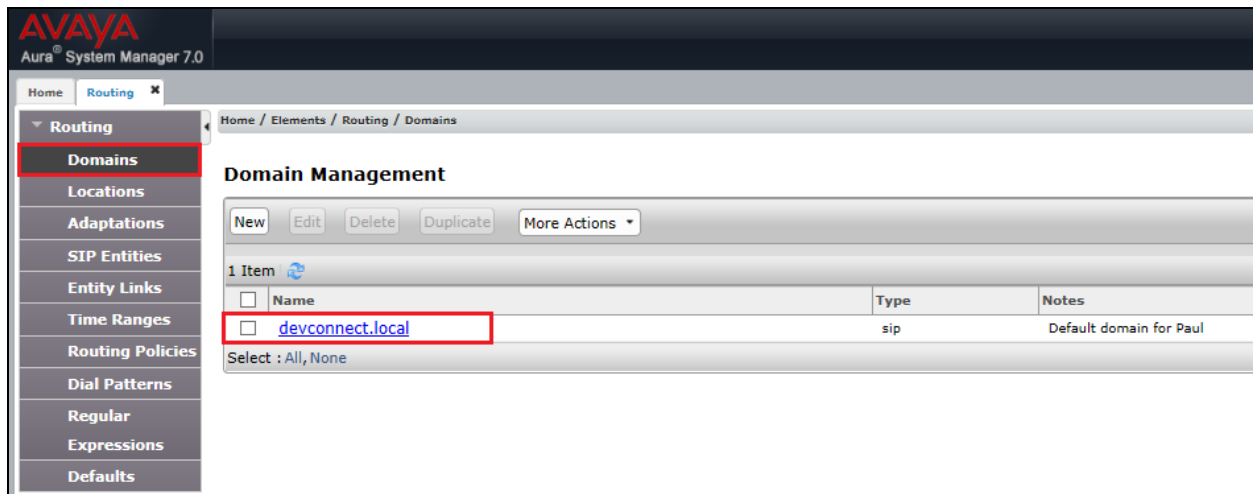
The NEC DECT handsets are added to Session Manager as SIP Users. In order to make changes in Session Manager, a web session to System Manager is opened. Navigate to <http://<System Manager IP Address>/SMGR>, enter the appropriate credentials and click on **Log On** as shown below.

6.1. Configuration of a Domain

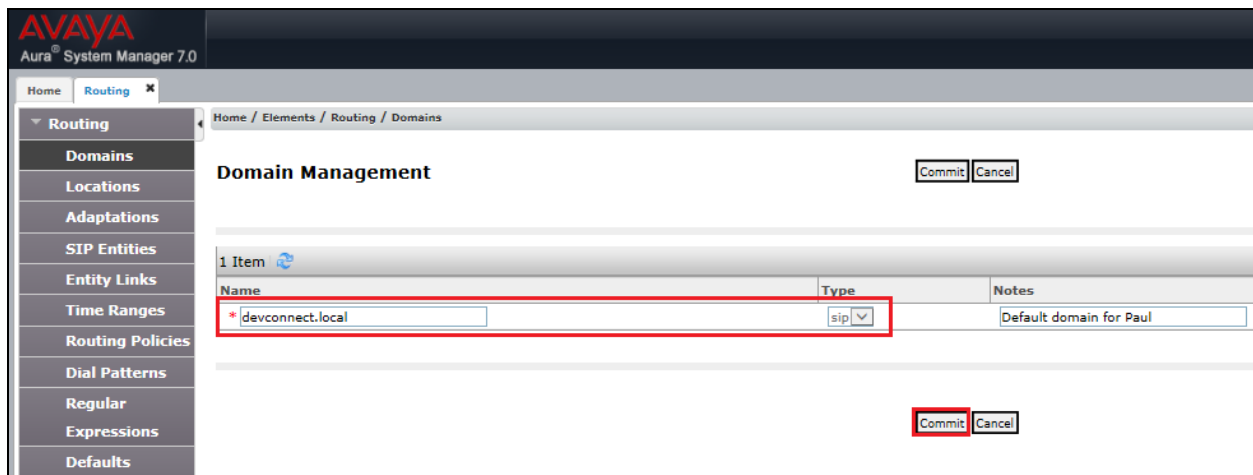
Click on **Routing** highlighted below.



Click on **Domains** in the left window. If there is not a domain already configured click on **New**. In the example below there exists a domain called devconnect.local which has been already configured.

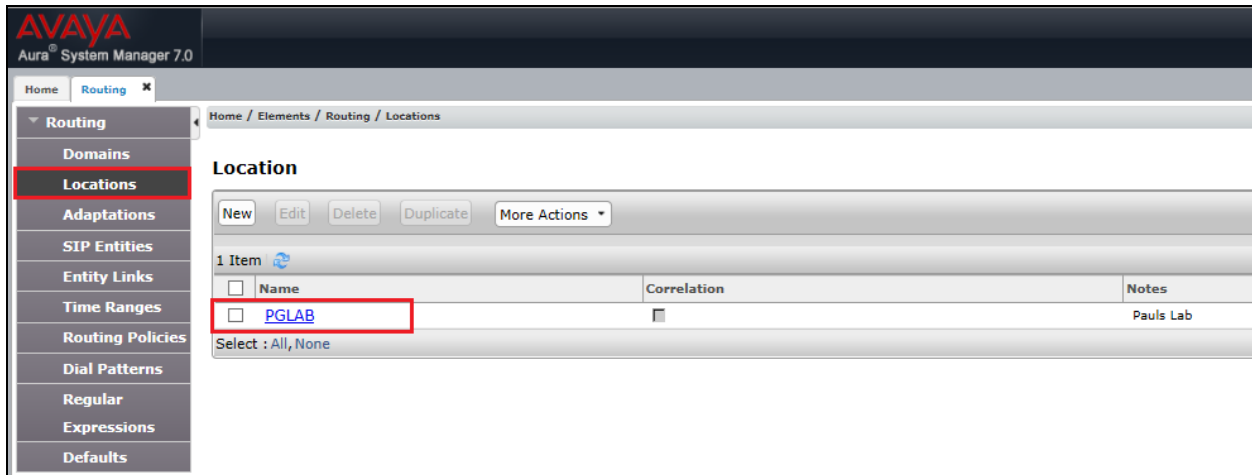


Clicking on the domain name above will open the following window; this is simply to show an example of such a domain. When entering a new domain the following should be entered, once the domain name is entered click on **Commit** to save this.



6.2. Configuration of a Location

Click on **Locations** in the left window and if there is no Location already configured then click on **New**, however in the screen below a location called **PGLAB** is already setup and configured and clicking into this will show its contents.



The screenshot displays the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with the following items: Routing, Domains, **Locations** (highlighted with a red box), Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Location' and includes a breadcrumb trail: Home / Elements / Routing / Locations. Below the title, there are buttons for 'New', 'Edit', 'Delete', 'Duplicate', and 'More Actions'. A table lists the configured locations:

Name	Correlation	Notes
<input type="checkbox"/> PGLAB		Pauls Lab

Below the table, there is a 'Select : All, None' option.

The Location below shows a suitable **Name** with a **Location Pattern** of **10.10.40.***. Once this is configured, click on **Commit**.

AVAYA
Aura® System Manager 7.0

Home / Elements / Routing / Locations

Location Details [Commit] [Cancel]

General

* Name: PGLAB
Notes: Pauls Lab

Dial Plan Transparency in Survivable Mode

Enabled: ☐
Listed Directory Number:
Associated CM SIP Entity:

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): 2000 Kbit/Sec
Maximum Multimedia Bandwidth (Inter-Location): 2000 Kbit/Sec
* Minimum Multimedia Bandwidth: 64 Kbit/Sec
* Default Audio Bandwidth: 80 Kbit/sec

Alarm Threshold

Overall Alarm Threshold: 80 %
Multimedia Alarm Threshold: 80 %
* Latency before Overall Alarm Trigger: 5 Minutes
* Latency before Multimedia Alarm Trigger: 5 Minutes

Location Pattern

Add Remove

1 Item

IP Address Pattern	Notes
10.10.40.	Pauls subnet

Select : All, None

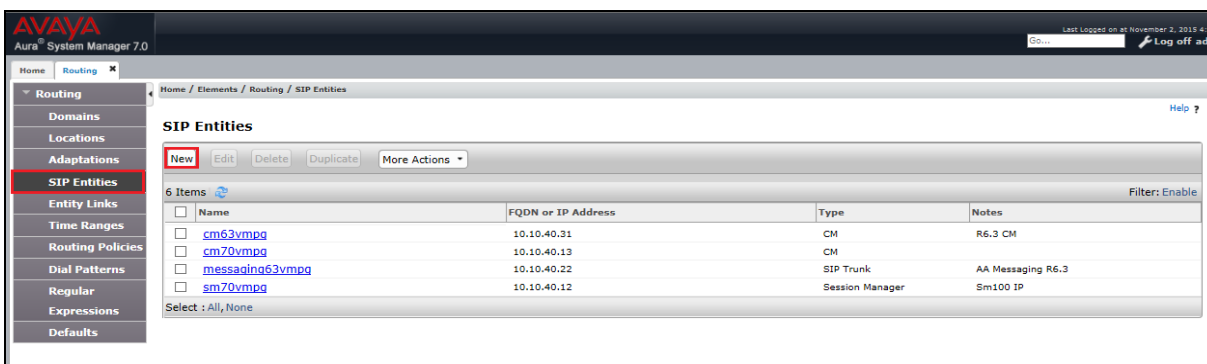
[Commit] [Cancel]

6.3. Configuration of SIP Entities

Clicking on **SIP Entities** in the left window shows what SIP Entities have been added to the system and allows the addition of any new SIP Entity that may be required. Please note the SIP Entities already present for the Compliance Testing of novamail.

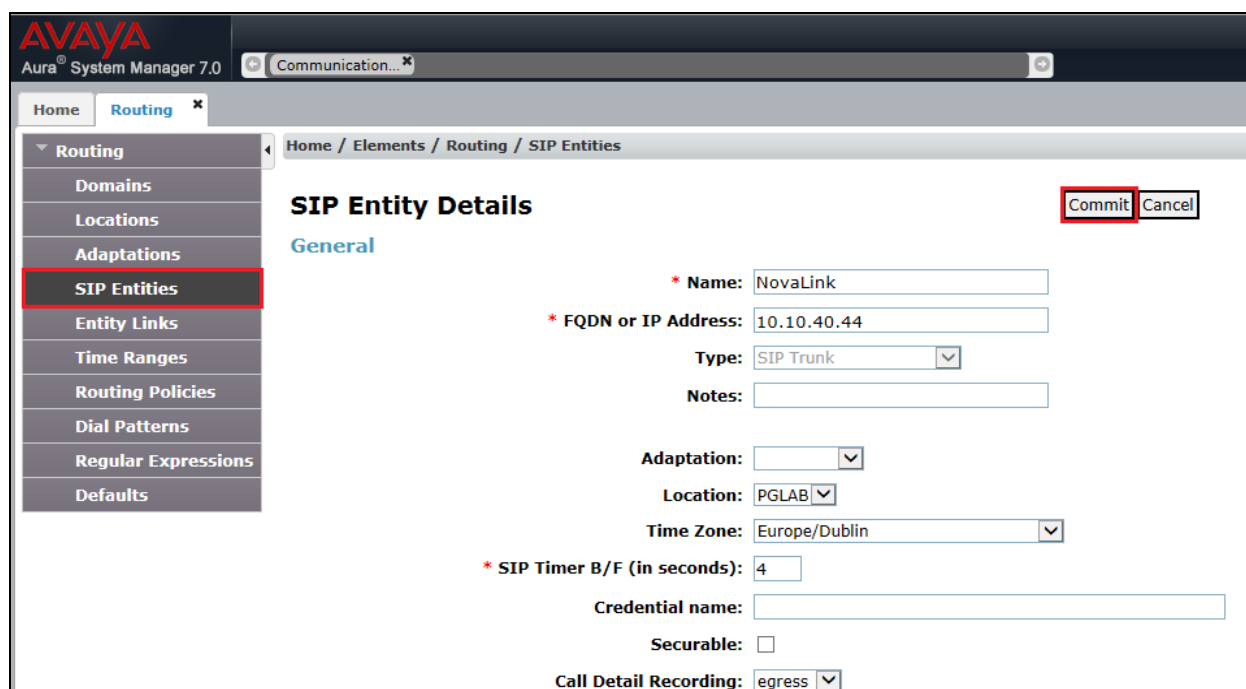
- Communication Manager SIP Entity (cm70vmpg)
- Session Manager SIP Entity (sm70vmpg)

To add a SIP entity, click on **New**.

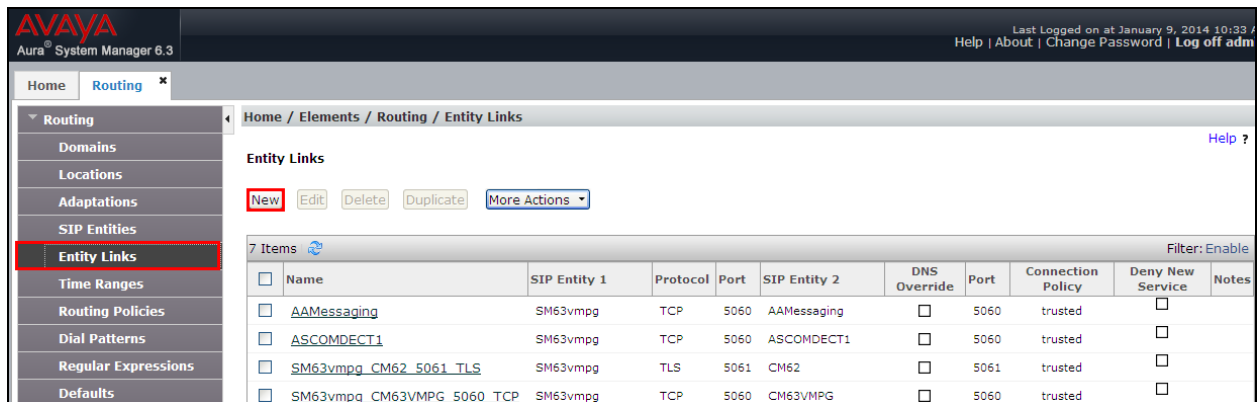


Enter a suitable **Name** as well as the **IP Address** of novamail. Select **SIP Trunk** as the **Type**. Click on **Commit** once completed.

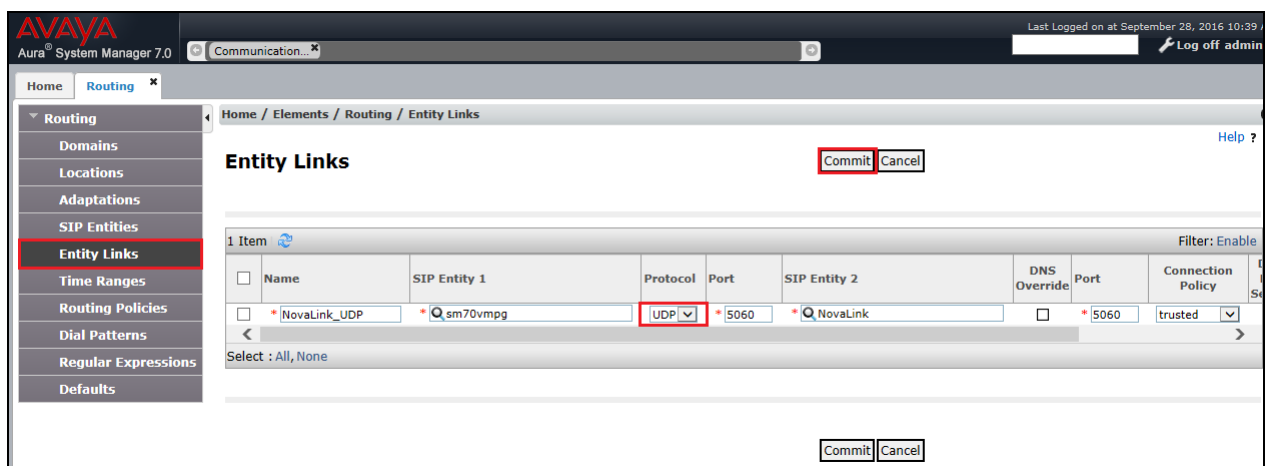
Note: In the remainder of this section including the screen shots below novamail may also be referred to as novalink.



An Entity Link between novamail and Session Manager is required, click on Entity Links in the left column and then on **New**.

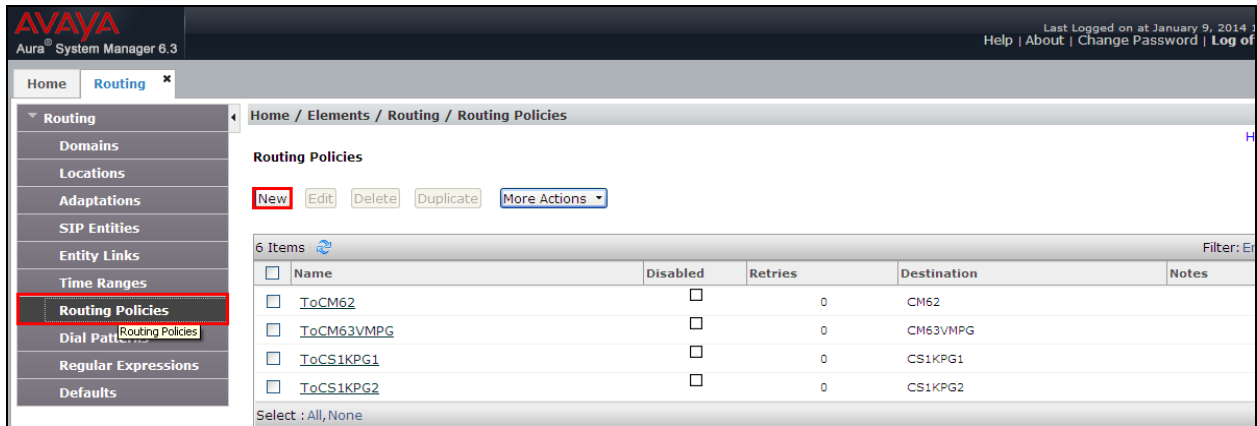


Enter a suitable **Name** and ensure that **UDP** is selected for the **Protocol** and **5060** for the **Port**. The **Connection Policy** must be setup as **trusted** as shown below. Click on **Commit** once completed.

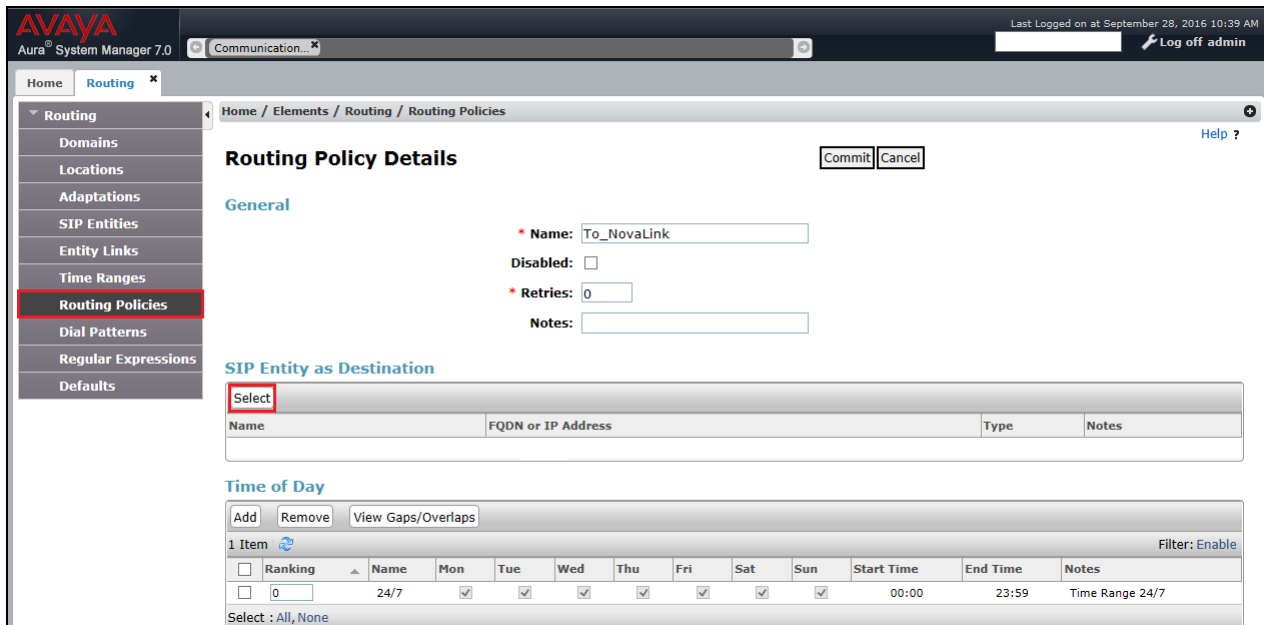


6.4. Configure Routing Policy for novalink

Select **Routing Policies** from the left window and click on **New** in the main window.



Enter a suitable **Name** and click on **Select** highlighted in order to associate this routing policy with a SIP Entity.



Select the **novalink** SIP Entity created in **Section 6.3** and click on **Commit** when done (not shown).

SIP Entities Help ?

Select Cancel

SIP Entities

12 Items Filter: Enable

	Name	FQDN or IP Address	Type	Notes
<input type="radio"/>	aacc64SIPvmppg	10.10.40.55	SIP Trunk	
<input type="radio"/>	AACC70vmppg	10.10.40.80	SIP Trunk	AACC70vmppg
<input type="radio"/>	cm63vmppg	10.10.40.31	CM	R6.3 CM
<input type="radio"/>	cm70vmppg	10.10.40.13	CM	
<input type="radio"/>	CS1000E	10.10.40.111	Other	CS1KPG1
<input type="radio"/>	EnghouseCP	10.10.40.106	SIP Trunk	EnghouseCP
<input type="radio"/>	Etrali_OT	172.29.187.244	SIP Trunk	
<input type="radio"/>	IPOS00V2	10.10.40.20	SIP Trunk	
<input type="radio"/>	messaging63vmppg	10.10.40.22	SIP Trunk	AA Messaging R6.3
<input type="radio"/>	NECDAP011	10.10.40.208	Endpoint Concentrator	DAP 1
<input checked="" type="radio"/>	NovaLink	10.10.40.44	SIP Trunk	
<input type="radio"/>	sm70vmppg	10.10.40.12	Session Manager	Sm100 IP

Select : None

6.5. Configure Dial Pattern for novalink

In order to route calls to the novamail a dial pattern is created pointing to the SIP Entity. Select **Dial Patterns** from the left window and click on **New** in the main window.

AVAYA
Aura® System Manager 6.3

Last Logged on at January 9, 2014
Help | About | Change Password | Log out

Home **Routing**

Routing

- Domains
- Locations
- Adaptations
- SIP Entities
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns**
- Regular Expressions
- Defaults

Home / Elements / Routing / Dial Patterns

Dial Patterns

New Edit Delete Duplicate More Actions

6 Items Filter:

	Pattern	Min	Max	Emergency Call	Emergency Type	Emergency Priority	SIP Domain	Notes
<input type="checkbox"/>	10	4	4	<input type="checkbox"/>			devconnect.local	
<input type="checkbox"/>	2	4	4	<input type="checkbox"/>			devconnect.local	CM63
<input type="checkbox"/>	30	4	4	<input type="checkbox"/>			-ALL-	CS1KPG1
<input type="checkbox"/>	5999	4	5	<input type="checkbox"/>			-ALL-	AURA_Messaging
<input type="checkbox"/>	70	4	4	<input type="checkbox"/>			devconnect.local	CS1KPG1

Select : All, None

Enter the number to be routed noting this will be the same number outlined in **Section 5.4**. Note the **SIP Domain** is that configured in **Section 6.2**. Click on **Add** to select the SIP Entity.

Dial Pattern Details
Commit Cancel

General

* Pattern:

* Min:

* Max:

Emergency Call: ☐

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

Add Remove

0 Items Filter: Enable

<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes	
<								>

Tick on the **Originating Location** as shown below and select the **novalink** Routing Policy. Click on **Select** once complete.

Originating Location
Select Cancel
Help ?

Originating Location

☐ Apply The Selected Routing Policies to All Originating Locations

1 Item Filter: Enable

<input checked="" type="checkbox"/>	Name	Notes
<input checked="" type="checkbox"/>	PGLAB	Pauls Lab

Select : All, None

Routing Policies

10 Items Filter: Enable

<input type="checkbox"/>	Name	Disabled	Destination	Notes
<input type="checkbox"/>	To_aacc64SIPvmppg	<input type="checkbox"/>	aacc64SIPvmppg	aacc64SIPvmppg
<input type="checkbox"/>	To_AACC70vmppg	<input type="checkbox"/>	AACC70vmppg	To_AACC70vmppg
<input type="checkbox"/>	To_cm63vmppg	<input type="checkbox"/>	cm63vmppg	Routing to CM63
<input type="checkbox"/>	To_cm70vmppg	<input type="checkbox"/>	cm70vmppg	
<input type="checkbox"/>	To_CS1000E	<input type="checkbox"/>	CS1000E	Routing to CS1KPG1
<input type="checkbox"/>	To_EnghouseCP	<input type="checkbox"/>	EnghouseCP	
<input type="checkbox"/>	To_Etrali	<input type="checkbox"/>	Etrali_OT	Etrali
<input type="checkbox"/>	To_IP0500V2	<input type="checkbox"/>	IP0500V2	To_IP0500V2
<input type="checkbox"/>	To_Messaging	<input type="checkbox"/>	messaging63vmppg	AA Messaging R63
<input checked="" type="checkbox"/>	To_NovaLink	<input type="checkbox"/>	NovaLink	

Select : All, None

With the new Routing Policy in place, click on **Commit** as shown below.

Dial Pattern DetailsHelp ?

CommitCancel

General

* Pattern:49

* Min:4

* Max:4

Emergency Call:☐

Emergency Priority:1

Emergency Type:

SIP Domain:devconnect.local

Notes:To NovaLink 10.10.40.44

Originating Locations and Routing Policies

AddRemove

1 Item

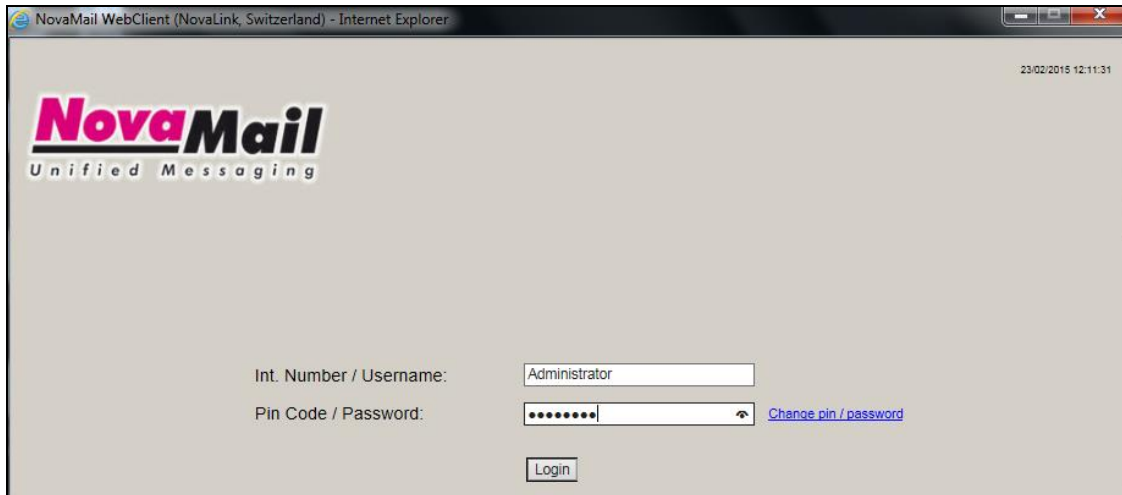
Filter: Enable

<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"/>	PGLAB	Pauls Lab	To_NovaLink	0	<input type="checkbox"/>	NovaLink	

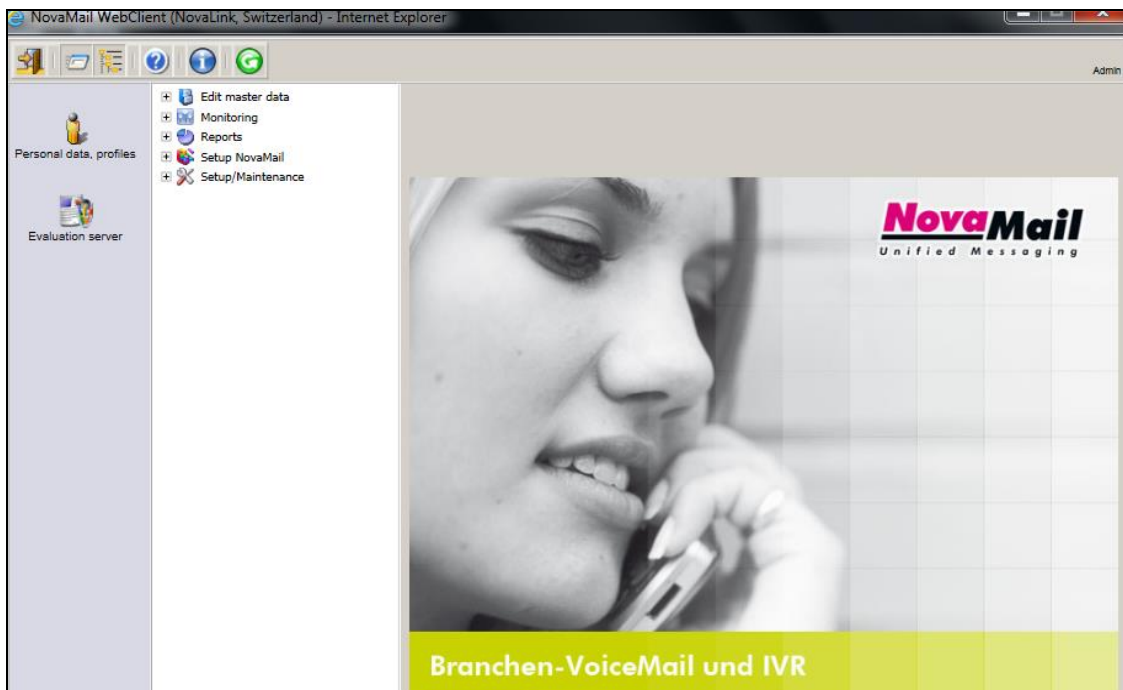
Select : All, None

7. Configure novamail

The following sections describe the steps required to configure novamail in order to successfully connect to Session Manager using SIP trunks. All configuration changes are made to novamail using a web browser session to the novamail server. Open a web browser session to the IP Address of the novamail server followed by /novamail. For example what was used for compliance testing was **http://10.10.40.44/novamail**. The following screen is shown asking for the **User Name** and **Password**. Enter these and click on the tick box as shown and click on the **Login** button.

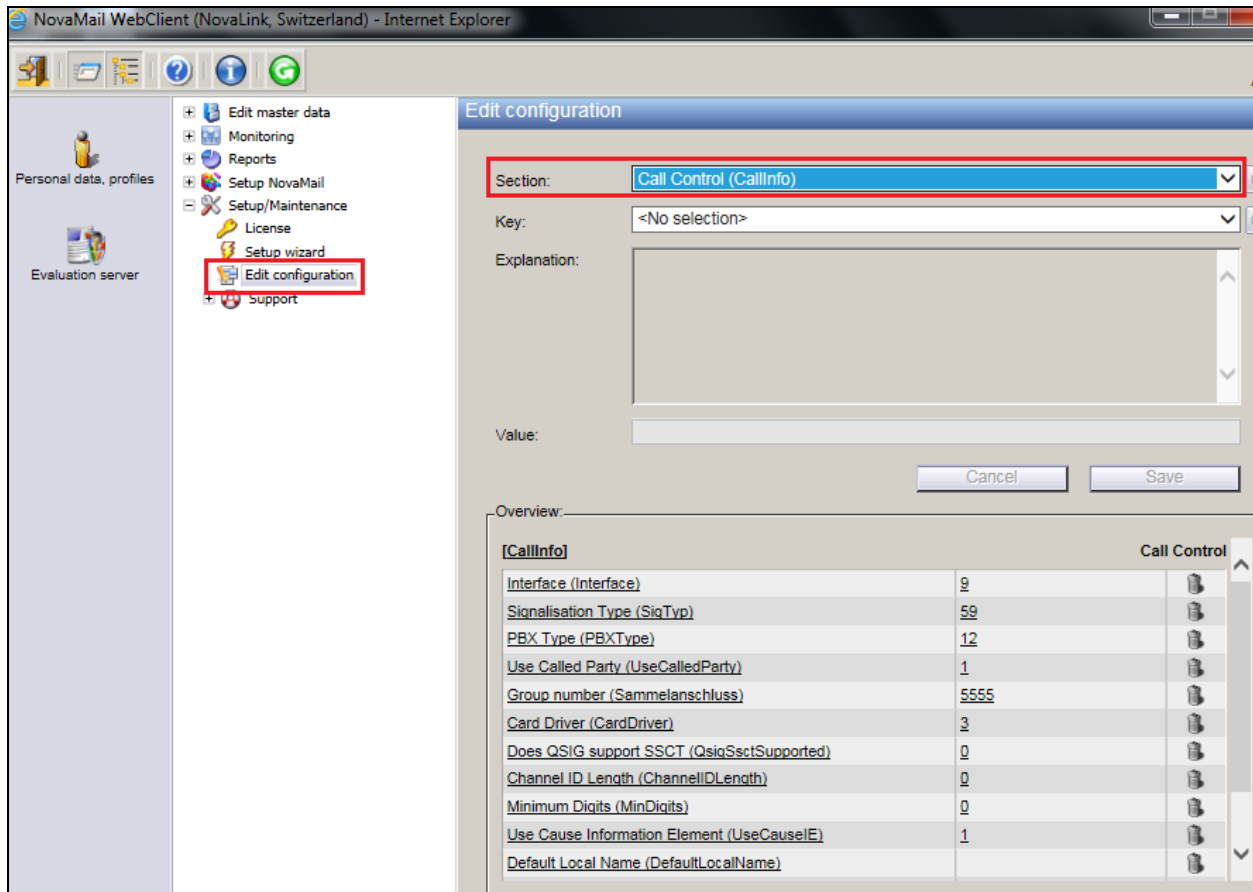


Once logged in the following screen is presented to the user.



7.1. Configure novamail SIP Trunk Connection

To begin the configuration of novamail, from the main menu, expand **Setup/Maintenance** and click on **Edit configuration**. From the main window select the **Section, Call Control (CallInfo)**, from the drop-down menu.



Select **Interface** from the **Key** drop-down menu. Ensure that **Value** is set to **VoIP** and click on **Save**.

The screenshot shows the 'Edit configuration' window with the 'Call Control (CallInfo)' section selected. The 'Key' dropdown is set to 'Interface' and the 'Value' dropdown is set to 'VoIP'. The 'Save' button is highlighted. Below the main configuration area, an 'Overview' table lists various parameters for the 'Call Control' section.

[CallInfo]		Call Control
Interface (Interface)	9	
Signalisation Type (SigTyp)	59	
PBX Type (PBXType)	12	
Use Called Party (UseCalledParty)	1	
Group number (Sammelanschluss)	5555	
Card Driver (CardDriver)	3	
Does QSIG support SSCT (QsigSctSupported)	0	
Channel ID Length (ChannelIDLength)	0	
Minimum Digits (MinDigits)	0	
Use Cause Information Element (UseCauseIE)	1	
Default Local Name (DefaultLocalName)		

Remaining in the same **Section**, select **Signalisation Type (SigTyp)** from the **Key** drop-down menu and ensure that **Value** is set to **SIP (MW Standard)**. Click on **Save** to complete.

The screenshot shows the 'Edit configuration' window with the 'Call Control (CallInfo)' section selected. The 'Key' dropdown is set to 'Signalisation Type (SigTyp)' and the 'Value' dropdown is set to 'SIP (MW Standard)'. The 'Save' button is highlighted.

Remaining in the same **Section**, select **PBX Type (PBXType)** from the **Key** drop-down menu and ensure that **Value** is set to **Avaya CM**. Click on **Save** to complete.

The screenshot shows the 'Edit configuration' dialog box with the following fields:

- Section:** Call Control (CallInfo)
- Key:** PBX Type (PBXType)
- Explanation:** Which PBX Type do you use (only PBX-typs requiring special paramters are listed)?
- Value:** Avaya CM
- Value:** 11
- Buttons:** Cancel, Save

In the same **Section**, select the **Use Called Party (UseCalledParty)** **Key**. Set **Value** to **Yes** and click on **Save**. This will allow novamail use the called party number for voicemail.

The screenshot shows the 'Edit configuration' dialog box with the following fields:

- Section:** Call Control (CallInfo)
- Key:** Use Called Party (UseCalledParty)
- Explanation:** Should NovaMail also use the called party number to select the mailbox?
- Value:** Yes
- Value:** 1
- Buttons:** Cancel, Save

In the same **Section**, select the **Group number (Sammelanschluss) Key**. Set **Value** to **x**, where x is the voicemail number that Communication Manager users will call. This happens to be **4999** for compliance testing. Click on **Save** to continue.

The screenshot shows the 'Edit configuration' dialog box. The 'Section' dropdown is set to 'Call Control (CallInfo)'. The 'Key' dropdown is set to 'Group number (Sammelanschluss)'. The 'Explanation' text area contains the text 'Do you use a trunk group for incoming calls? If yes, enter the phone number here:'. The 'Value' text field contains '4999'. The 'Save' button is highlighted with a red border.

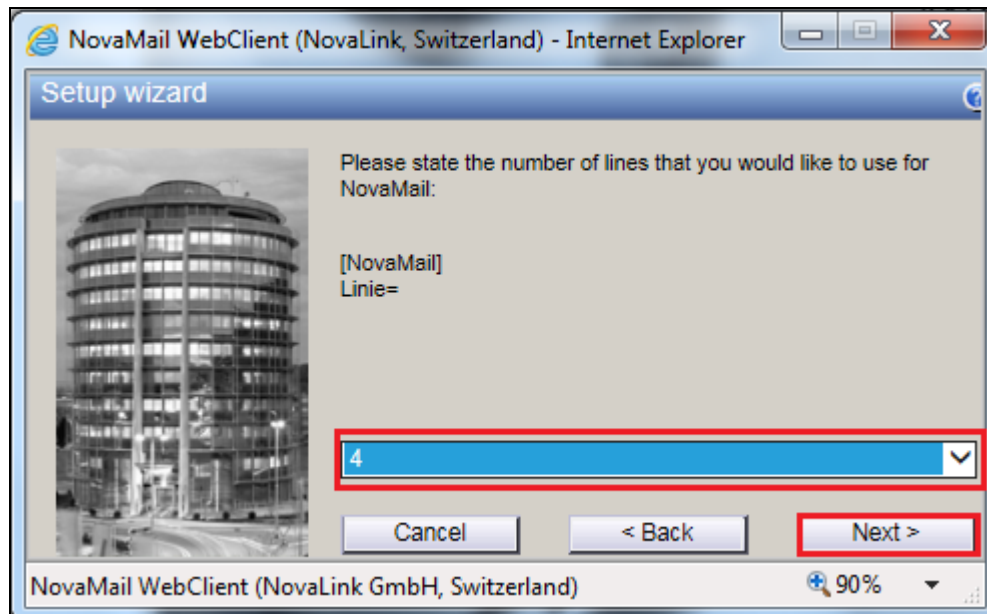
In the same **Section**, select the **Card Driver (CardDriver) Key**. Set **Value** to **VoIP (H.323/SIP)** and click on **Save**.

The screenshot shows the 'Edit configuration' dialog box. The 'Section' dropdown is set to 'Call Control (CallInfo)'. The 'Key' dropdown is set to 'Card Driver (CardDriver)'. The 'Explanation' text area contains the text 'Interface to use?'. The 'Value' dropdown is set to 'VoIP (H.323/SIP)'. The 'Save' button is highlighted with a red border.

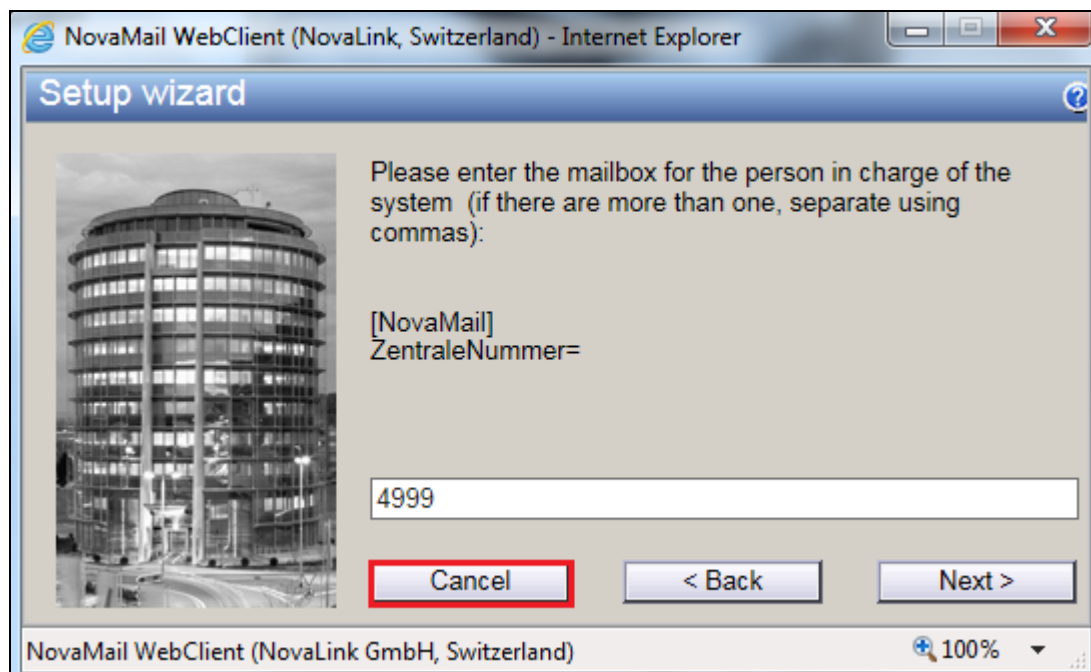
In order to add the SIP channels or lines a wizard must be run from the **Setup/Maintenance** in the left menu. Navigate to **Setup/Maintenance** and click on **Setup wizard**. This will bring up the **Setup wizard** window and click on **Next** to continue.



Enter the number of lines that are to be added in the drop-down box and click on **Next** to continue. The number of lines chosen will depend on the license granted.



Enter the Voice Mail number and click on **Cancel**. The lines will still be added from above.



These lines are now clearly seen as been added.

Edit configuration

Section: NovaMail Configuration (NovaMail)

Key: <No selection>

Explanation:

Value:

Cancel Save

Overview:

Message Waiting clear (MWLöschen)	*85*	
Message Waiting Dial Tone (MWWählton)	1	
Message Waiting Acknowledge (MWQuittung)	1	
Line from (MWLinieVon)	4	
Line to (MWLinieBis)	4	
Line from (CallLinieVon)	3	
Line to (CallLinieBis)	4	
Line 1 (Linie1)	1	
Line 2 (Linie2)	2	
Line 3 (Linie3)	3	
Line 4 (Linie4)	4	

Change the **Section** drop-down to **Voice over IP Configuration (VoIP)**.

The screenshot shows the 'Edit configuration' dialog box. On the left is a sidebar with icons for 'Edit master data', 'Monitoring', 'Reports', 'Setup NovaMail', 'Setup/Maintenance', 'License', 'Setup wizard', 'Edit configuration' (highlighted with a red box), and 'Support'. The main area has a 'Section:' dropdown menu set to 'Voice over IP Configuration (VoIP)'. Below it, the 'Key:' dropdown is set to '<No selection>'. The 'Explanation:' field is a large text area that is currently empty. At the bottom, there is a 'Value:' field, which is also empty. The 'Cancel' and 'Save' buttons are located at the bottom right of the dialog.

Select **Driver Preferences (DriverPref)** from the **Key** drop-down menu. Ensure that **Only SIP** is chosen for **Value** and click on **Save** to continue.

The screenshot shows the 'Edit configuration' dialog box with the following settings: 'Section:' is 'Voice over IP Configuration (VoIP)'; 'Key:' is 'Driver Preferences (DriverPref)'; 'Explanation:' is 'Which VoIP protocol should be used?'; and 'Value:' is 'Only SIP'. The 'Cancel' and 'Save' buttons are at the bottom right. The 'Key' and 'Value' dropdowns, along with the 'Save' button, are highlighted with red boxes.

Change the **Key** to **SIP Gateway (SIP_Gateway)** and enter the Server Editions IP address for **Value** in the format <IP Address>,<IP Address>. Click on **Save** to continue.

Edit configuration

Section: Voice over IP Configuration (VoIP)

Key: SIP Gateway (SIP_Gateway)

Explanation: SIP-Gateways with [Realm,IP,Prefix] (Prefix can be omitted) (separate multiple gateways with ";") (novalink.ch,192.168.25.1;novamail.ch,192.168.25.200)?

Value: 10.10.40.12,10.10.40.12

Cancel Save

7.2. Add an new Mailbox on novamail

In order to add a new mailbox for a Communication Manager extension, navigate to **Edit master data** → **Mailboxes** in the left window. Select **New mailbox** in the mail window.

Mailboxes

New mailbox Search mailbox Show all

Internal Number	Name
999	Standard Voicebox
3000	PSTN QSIG
5100	1140e SIP
5101	9608 SIP
5151	1608 H323
5201	9408 Digital
5220	9611 SIP
5250	9630 H323

Click on the **General** tab and enter a suitable **Surname / First name** and **Pin code**. Enter the Communication Manager extension number for **Internal phone number**. **From own unit without Pin** can be ticked (not below) in order to avoid having to type in a password every time one calls from their own telephone.

The screenshot shows the 'Edit Mailbox' window with the 'General' tab selected. The 'Number' field contains '7000' and the 'Name' field contains 'H323 7000'. The 'Client' dropdown is set to 'All'. The 'General' tab is highlighted with a red box. Below the tabs, the 'Internal phone number' field contains '7000', the 'Surname / First name' field contains 'H323 7000', the 'Pin code' field contains '2580', and the 'From own unit without Pin' checkbox is unchecked. The 'Language' dropdown is set to 'English'. The 'Internal fax number' field is empty, the 'Outg. fax authorization' checkbox is unchecked, and the 'Fax priority' dropdown is set to 'Normal'.

An additional participant needs to be added in order to route calls correctly to voicemail when using call forward. Click on the **Additional participants** tab and enter the addition mailbox number for this user, typically this number is logically associated in some way to the original mailbox/extension number. Click on **Add** to add this to the users mailbox setup.

The screenshot shows the 'Edit Mailbox' window with the 'Additional participants' tab selected. The 'Number' field contains '7000' and the 'Name' field contains 'H323 7000'. The 'Client' dropdown is set to 'All'. The 'Additional participants' tab is highlighted with a red box. Below the tabs, the 'Participant' field contains '4980' and the 'Activate MW' checkbox is unchecked. The 'Add' button is highlighted with a red box. The 'Cancel' and 'Save' buttons are also visible.

The following screen shows this extra mailbox number added correctly.

The screenshot shows the 'Edit Mailbox' window with the 'Additional participants' tab selected. At the top, the 'Number' field contains '7000' and the 'Name' field contains 'H323 7000'. The 'Client' dropdown is set to 'All'. Below the tabs, there is a form for adding a participant with a text input field, an 'Activate MW' checkbox, and 'Cancel', 'Save', and 'Add' buttons. At the bottom, a table lists the added participants.

Participant	Activate MW	
4980	<input type="checkbox"/>	

Click on the **Profiles** tab. Click on the **Standard** Profile that is already assigned to the mailbox.

The screenshot shows the 'Edit Mailbox' window with the 'Profiles' tab selected. The 'Number' field contains '5201' and the 'Name' field contains '9408 Digital'. The 'Client' dropdown is set to 'All'. Below the tabs, there are input fields for 'Alternative Phone number 1', 'Alternative phone number 2', 'Deputy's phone number', and 'Fixed diversion dest. for messages:'. A 'New profile' button is present. At the bottom, a table lists the assigned profiles.

Name	Active	
Standard	<input checked="" type="checkbox"/>	

In order to allow the voicemail system alert a remote user that they have a new voice message the profile of the user must be changes to allow for **Notification**. Click on the **Notification** tab and enter the mobile or alternative number for this user into the **Call to** box and ensure this is also ticked as shown below. Click on **Save data** to save this mailbox.

The screenshot shows the 'Process profile' window with the following details:

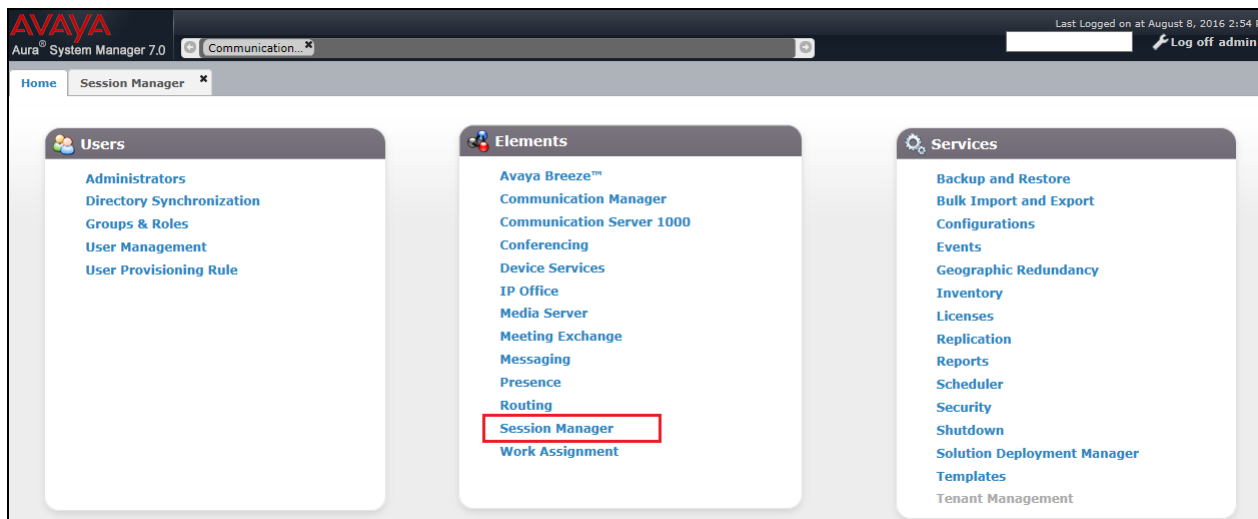
- Profile name:** Standard
- Participant:** 5201 / 9408 Digital
- Tabs:** Recorded messages, **Notification** (highlighted), Times, Fax
- Notification for:** Only voice messages (dropdown menu)
- Notification for unanswered calls:** None (dropdown menu)
- Internal e-mail:** ☐
- External e-mail:** ☐
- Display on telephone:** ☒
- SMS to:**
- Call to:** 0871234567 (highlighted with a red box, with a checked checkbox next to it)
- Number of SMS-licenses:** 500, currently used: 0
- Buttons:** **Save data** (highlighted with a red box), Discard

8. Verification Steps

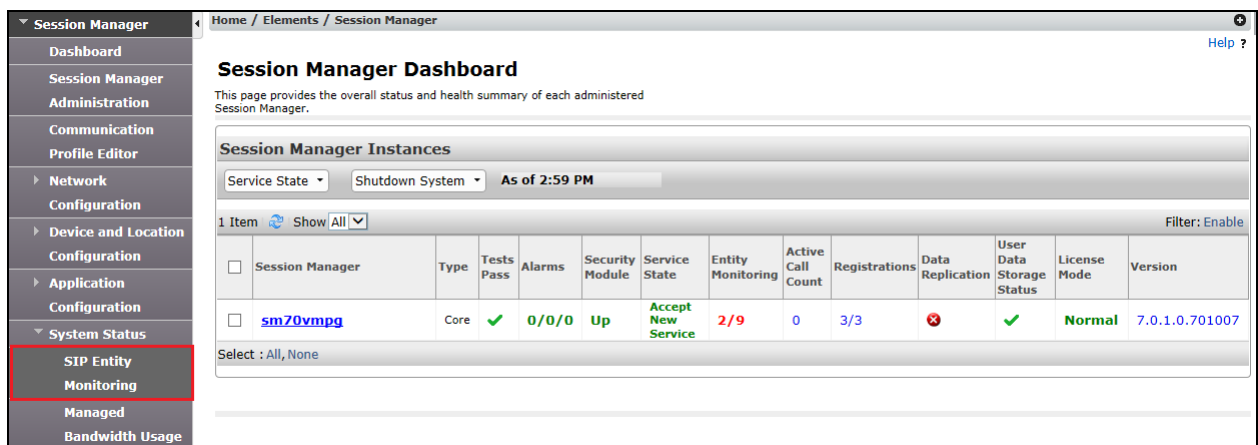
This section illustrates the steps necessary to verify that the novamail is configured correctly to allow extensions on Communication Manager dial in and use the voicemail facilities over SIP trunks.

8.1. Verify Link on Session Manager

Log in to System Manager as per **Section 6**. From the main menu select Session Manager as shown below.



Navigate to **System Status** → **SIP Entity Monitoring**.



Choose the **novalink** SIP entity as shown below.

Application Configuration
System Status
SIP Entity Monitoring
Managed Bandwidth Usage
Security Module Status
SIP Firewall Status
Registration Summary
User Registrations
Session Counts
User Data Storage
System Tools
Performance

Session Manager	Type	Monitored Entities					Deny	Total
		Down	Partially Up	Up	Not Monitored			
<input type="checkbox"/> sm70vmpg	Core	2	0	8	0	0	10	

Select: All, None

All Monitored SIP Entities

Run Monitor

10 Items | Refresh Filter: Enable

SIP Entity Name
<input type="checkbox"/> cm70vmpg
<input type="checkbox"/> messaging63vmpg
<input type="checkbox"/> cm63vmpg
<input type="checkbox"/> aacc64SIPvmpg
<input type="checkbox"/> AACC70vmpg
<input type="checkbox"/> Novalink
<input type="checkbox"/> Etrali_OT
<input type="checkbox"/> EnghouseCP

Select: All, None < Previous | Page 1 of 2 | Next >

The **Link Status** and **Conn. Status** should both show as **UP** as is shown below.

Session Manager
Dashboard
Session Manager Administration
Communication Profile Editor
Network Configuration
Device and Location Configuration
Application Configuration
System Status
SIP Entity Monitoring
Managed Bandwidth Usage
Security Module Status
SIP Firewall Status

Home / Elements / Session Manager / System Status / SIP Entity Monitoring

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: **Novalink**

Status Details for the selected Session Manager:

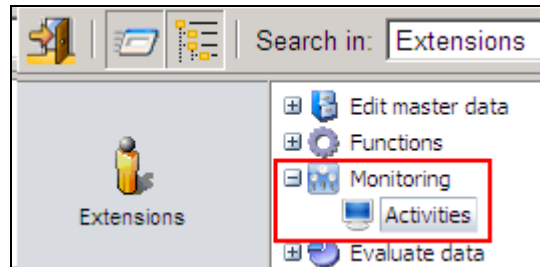
Summary View

1 Items | Refresh Filter: Enable

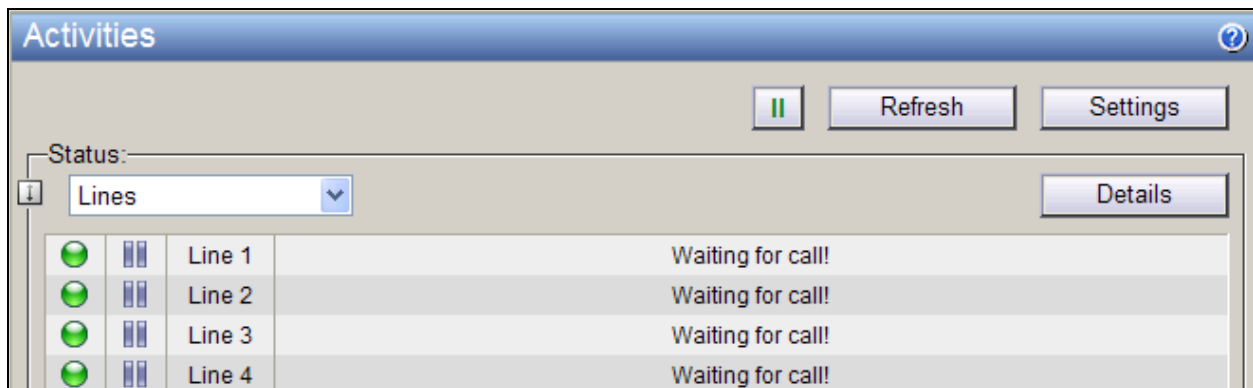
Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/> sm70vmpg	10.10.40.44	5060	UDP	FALSE	UP	200 OK	UP

8.2. Verify novamail Status

From the novamail web interface (not shown), navigate to **Monitoring** → **Activities** in the left column.



Verify that the icon in the left column is green indicating that the SIP trunks are in service and Session Manager can be reached.



8.3. Verify Successful Delivery of Voicemail

Place a call to a Communication Manager user with forwarding to voicemail configured. Ensure that novamail answers the call with the appropriate mailbox greeting and a message can be left. Verify that the message waiting indicator on the endpoint is illuminated.

8.4. Verify Successful Retrieval of Voicemail

Dial the voicemail retrieval access number from a Communication Manager user. Ensure that novamail automatically recognizes the user and may be prompted for a PIN. Verify that the audio prompts advise a message has been left and use the buttons on the telephone keypad to navigate the menu, listen to, and delete the message. Verify that the message waiting indicator is extinguished once all messages have been played back.

9. Conclusion

These Application Notes describe the configuration steps required for novamail from novalink to successfully interoperate with Avaya Aura® Communication Manager using Avaya Aura® Session Manager. All feature test cases were completed successfully with any observations noted in **Section 2.2**.

10. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com> where the following documents can be obtained.

- [1] *Administering Avaya Aura® Communication Manager*, Document ID 03-300509
- [2] *Avaya Aura® Communication Manager Feature Description and Implementation*, Document ID 555-245-205
- [3] *Implementing Avaya Aura® Session Manager* Document ID 03-603473
- [4] *Administering Avaya Aura® Session Manager*, Doc ID 03-603324

Technical support can be obtained for novamail from the website <http://www.novalink.ch/en/> or from [ftp://support.novalink.ch/Technikerhandbuch/English/Technikerhandbuch novalink GmbH EN.chm](ftp://support.novalink.ch/Technikerhandbuch/English/Technikerhandbuch%20novalink%20GmbH%20EN.chm) (please request Login and Password from novalink).

Appendix

Configure SIP Trunk between Session Manager and Communication Manager

The following shows the SIP Signalling Group and SIP trunk that was used during compliance testing.

- Set the **Group Type** field to **sip**.
- For compliance testing **Transport Method** was set to **tls**.
- The **Peer Detection Enabled** field should be set to **y** allowing the Communication Manager to automatically detect if the peer server is a Session Manager.
- Specify the node names for the procr and the Session Manager node name as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively.
- Set the **Near-end Node Name** to **procr**. Set the **Far-end Node Name** to the node name defined for the Session Manager (node name **sm70vmppg**), as per **Section 5.5**.
- Ensure that the recommended TLS port value of **5061** is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- In the **Far-end Network Region** field, enter the IP Network Region configured in **Section 5**. This field logically establishes the **far-end** for calls using this signaling group as network region 1.
- **Far-end Domain** was set to the domain used during compliance testing.
- The **DTMF over IP** field should remain set to the default value of **rtp-payload**. This value enables Communication Manager to send DTMF transmissions using RFC 2833.
- The **Direct IP-IP Audio Connections** field is set to **y**.
- **Initial IP-IP Direct Media** was set to **N** for compliance testing.
- The default values for the other fields may be used.

change signaling-group 1		Page	1 of	2
SIGNALING GROUP				
Group Number: 1	Group Type: sip			
IMS Enabled? n	Transport Method: tls			
Q-SIP? n				
IP Video? n	Enforce SIPS URI for SRTP? n			
Peer Detection Enabled? y	Peer Server: SM			
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y				
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n				
Alert Incoming SIP Crisis Calls? n				
Near-end Node Name: procr		Far-end Node Name: sm70vmppg		
Near-end Listen Port: 5061		Far-end Listen Port: 5061		
		Far-end Network Region: 1		
Far-end Domain: devconnect.local				
		Bypass If IP Threshold Exceeded? n		
Incoming Dialog Loopbacks: eliminate		RFC 3389 Comfort Noise? n		
DTMF over IP: rtp-payload		Direct IP-IP Audio Connections? y		
Session Establishment Timer(min): 3		IP Audio Hairpinning? n		
Enable Layer 3 Test? y		Initial IP-IP Direct Media? n		
H.323 Station Outgoing Direct Media? n		Alternate Route Timer(sec): 6		

Configure the Trunk Group form as shown below. This trunk group is used for calls to and from novamail. Enter a descriptive name in the Group Name field. Set the Group Type field to sip. Enter a TAC code compatible with the Communication Manager dial plan. Set the Service Type field to tie. Specify the signaling group associated with this trunk group in the Signaling Group field, and specify the Number of Members supported by this SIP trunk group. Accept the default values for the remaining fields.

change trunk-group 1		Page 1 of 21	
TRUNK GROUP			
Group Number: 1	Group Type: sip	CDR Reports: r	
Group Name: SIPTRK	COR: 1	TN: 1	TAC: *801
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
		Member Assignment Method: auto	
		Signaling Group: 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 NEC to prevent unnecessary SIP messages during call setup. Session refresh is used throughout the duration of the call, to check the other side has not gone away, for the compliance test a value of **600** was used.

change 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			
Disconnect Supervision - In? y Out? y			
XOIP Treatment: auto Delay Call Setup When Accessed Via IGAR? n			

Settings on **Page 3** can be left as default. However the **Numbering Format** in the example below is set to **private**.

change trunk-group 1	Page 3 of 21
TRUNK FEATURES	
ACA Assignment? n	Measured: none
	Maintenance Tests? y
Suppress # Outpulsing? n	Numbering Format: private
	UUI Treatment: service-provider
	Replace Restricted Numbers? n
	Replace Unavailable Numbers? n
	Hold/Unhold Notifications? y
	Modify Tandem Calling Number: no
Show ANSWERED BY on Display? y	

Settings on **Page 4** are as follows.

change trunk-group 1	Page 4 of 21
PROTOCOL VARIATIONS	
	Mark Users as Phone? y
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n	
Send Transferring Party Information? y	
Network Call Redirection? y	
Build Refer-To URI of REFER From Contact For NCR? n	
Send Diversion Header? n	
Support Request History? y	
Telephone Event Payload Type: 120	
Convert 180 to 183 for Early Media? n	
Always Use re-INVITE for Display Updates? n	
Identity for Calling Party Display: P-Asserted-Identity	
Block Sending Calling Party Location in INVITE? n	
Accept Redirect to Blank User Destination? n	
Enable Q-SIP? n	
Interworking of ISDN Clearing with In-Band Tones: keep-channel-active	
Request URI Contents: may-have-extra-digits	

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