

Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Convergys Voice Portal with Avaya Aura® Communication Manager and Avaya Aura® Session Manager via a SIP Trunking Interface - Issue 1.0

Abstract

These Application Notes describe the procedures required for Convergys Voice Portal to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using a SIP trunk.

Avaya SIP, H.323, and digital telephones were used to originate and terminate calls with Userto-User Information to and from the Convergys Voice Portal server. The overall objective of the interoperability compliance testing is to verify proper signaling and call establishment with the Convergys Voice Portal in an Avaya IP Telephony environment.

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

1. Introduction

These Application Notes describe the procedures required for Convergys Voice Portal to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager (SM) using a SIP trunk. Avaya SIP, H.323, and digital telephones were used to originate and terminate calls with User-to-User Information (UUI) to and from the Convergys Voice Portal server. The overall objective of the interoperability compliance testing is to verify proper signaling and call establishment with the Convergys Voice Portal in an Avaya IP environment.

Convergys Voice Portal provides IVR and Messaging functionality via a SIP/VOIP telephony interface. Callers interact with the system via DTMF or Speech input, and may be transferred to agents, as needed.

These Application Notes assume that Communication Manager and Session Manager have already been installed and that basic configuration steps have been performed. Only steps relevant to the configuration used for compliance testing will be described in this document. For further details on configuration steps not covered in this document, consult references [2], [3], and [5].

2. General Test Approach and Test Results

This section describes the testing used to verify the interoperability of Convergys Voice Portal with the Avaya SIP infrastructure (Communication Manager and Session Manager).

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

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing. Avaya SIP, H.323, and digital telephones were used to originate and terminate calls with User-to-User Information (UUI) to and from the Convergys Voice Portal server. The focus of the testing was primarily on verifying the SIP protocol messages between Session Manager and the Convergys Voice Portal server. Additionally, Convergys Voice Portal operations such as routing, DTMF tones, and transfers were tested. The serviceability testing included Communication Manager, Session Manager, and Convergys Voice Portal failure scenarios to verify that Convergys Voice Portal could properly recover from each failure.

2.2. Test Results

Convergys Voice Portal successfully passed compliance testing.

Solution & Interoperability Test Lab Application Notes ©2014 Avaya Inc. All Rights Reserved.

2.3. Support

Technical support for the Convergys Voice Portal can be obtained through the following:

- **Phone:** 800-955-4688
- Web: <u>http://realcare.intervoice.com</u>

3. Reference Configuration

Figure 1 illustrates the configuration used during compliance testing as described in these Application Notes. The configuration comprises of a Session Manager (with its companion System Manager), an Avaya S8300D Server running Communication Manager in an Avaya G450 Media Gateway. The non-SIP phones are supported by Communication Manager running on the S8300D Server and the G450 Media Gateway. The SIP phones register with Session Manager. The Convergys Voice Portal system was built on one physical server using VMware. One virtual machine (VM) was built to run the Convergys Control Center administration and monitoring tool. Two other VMs are built for two separate Convergys Voice Portals (IVRs). This document focuses on the integration to one Convergys Voice Portal (IP address 10.64.21.153).



Figure 1: Convergys Voice Portal interoperating with Communication Manager and Session Manager

MJH; Reviewed:
SPOC 4/11/2014

Solution & Interoperability Test Lab Application Notes ©2014 Avaya Inc. All Rights Reserved.

4. Equipment and Software Validated

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

Equipment/Software	Release/Version			
Avaya S8300D Server with an Avaya	Avaya Aura® Communication Manager 6.3			
G450 Media Gateway	Patch 20850			
HP Proliant DL360 G7	Avaya Aura® Session Manager 6.3 FP2			
Dell TM PowerEdge TM R610 Server	Avaya Aura® System Manager 6.3 SP2			
Avaya 9600 Series IP Deskphones				
• 96x0 (H.323)	Avaya one-X® Deskphone Edition 3.1.5			
• 96x0 (SIP)	Avaya one-X® Deskphone Edition 2.6.9			
• 96x1 (H.323)	Avaya one-X® Deskphone Edition 6.2.2			
• 96x1 (SIP)	Avaya one-X® Deskphone Edition 6.2.1			
Avaya 6210 Analog Phone	-			
Avaya 2420 Digital Phone	-			
Convergys Voice Portal:	6.7.2:			
• CTI Gateway	• 2.0.2			

5. Configure Avaya Aura® Communication Manager

This section describes the Communication Manager configuration required to interoperate with the Session Manager. It focuses on the configuration of the SIP trunk connecting Communication Manager and Session Manager, with the following assumptions:

- Procedures necessary to support SIP and connectivity to Session Manager have been performed as described in references [2], [3], and [5].
- All other components are assumed to be in place and previously configured, including the SIP and ISDN-PRI trunks that connect both sites.

The procedures for configuring Communication Manager include the following areas:

- Verify Communication Manager license (Step 1)
- Administer IP Node Names (Step 2)
- Administer IP network regions (Step 3)
- Administer IP codec set (Step 4)
- Administer SIP signaling group (Step 5)
- Administer SIP trunk group (Steps 6 7)
- Administer route pattern (Step 8)
- Administer AAR analysis for routing calls to Session Manager (Step 9)

The configuration of the Communication Manager was performed using the System Access Terminal (SAT). After the completion of the configuration, perform a **save translation** command to make the changes permanent.

tep	Description						
1.	Communication Manager License Use the display system-parameters customer-options command to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Navigate to Page 2, and verify that there is sufficient remaining capacity for SIP trunks by comparing the Maximum Administered SIP Trunks field value with the corresponding value in the USED column. The license file installed on the system controls the maximum permitted. If there is an insufficient capacity, contact an authorized Avaya sales representative to make the expropriate changes						
	display system-parameters customer-options OPTIONAL FEATURES	Page 2 of 11					
2.	IP PORT CAPACITIES Maximum Administered H.323 Trunks: 4000 Maximum Concurrently Registered IP Stations: 2400 Maximum Concurrently Registered Remote Office Trunks: 4000 Maximum Concurrently Registered IP eCons: 68 Max Concur Registered Unauthenticated H.323 Stations: 100 Maximum Video Capable IP Softphones: 2400 Maximum Video Capable IP Softphones: 2400 Maximum Administered SIP Trunks: 4000 Maximum Number of DS1 Boards with Echo Cancellation: 80 Maximum Number of DS1 Boards with 80 VoIP Channels: 10 Maximum TN2602 Boards with 30 VoIP Channels: 128 Maximum TN2602 Boards with 320 VoIP Channels: 128 Maximum Number of Expanded Meet-me Conference Ports: 300 IP Node Names Use the change node-names ip command to administer a Nan Session Manager. In the configuration used for compliance te SM_21_31 nodes were utilized to administer a SIP trunk betw Manager and Session Manager.	USED 88 6 0 0 0 0 0 0 0 0 0 0 0 0 0	Dr				
	change node-names ip IP NODE NAMES Name IP Address SM_21_31 10.64.20.31 default 0.0.0.0 procr 10.64.21.41	Page 1 of 2					

Step	Description
3.	IP Network Region – Region 1
	This section describes the steps for administering an IP network region in
	Communication Manager for communication between Communication Manager and
	Session Manager All IP endpoints were located in IP network region 1 using the
	parameters described below. Use the change in-network region command to view these
	sattings. The example below shows the values used during compliance testing
	settings. The example below shows the values used during comphance testing.
	• The Authoritative Domain field was configured to match the domain name
	configured on Session Manager (see Section 6, Step 2). In this configuration, the
	domain name is <i>avaya.com</i> . This name appears in the "From" header of SIP
	messages originating from this IP region.
	• A descriptive name was entered for the Name field.
	• IP-IP Direct Audio (Media Shuffling) was enabled to allow audio traffic to be sent
	directly between IP endpoints without using media resources in the Avava Media
	Gateway This was done for both intra-region and inter-region IP-IP Direct Audio
	This is the default setting. Media Shuffling can be further restricted at the trunk level
	an the Signaling Choung form
	on the Signaling Group form.
	• The Codec Set field was set to the IP codec set to be used for calls within this IP
	network region. In this case, IP codec set 1, configured in Step 4, was selected.
	 The default values were used for all other fields.
	change ip-network-region 1 Page 1 of 20
	IP NETWORK REGION
	Region: 1 Location: Authoritative Domain: avava com
	Name: Compliance Testing Stub Network Region: n
	MEDIA PARAMETERS Intra-region IP-IP Direct Audio: yes
	Codec Set: 1 Inter-region IP-IP Direct Audio: yes
	UDP Port Max: 3329
	DIFFSERV/TOS PARAMETERS
	Call Control PHB Value: 46
	Audio PHB Value: 46
	802.1P/O PARAMETERS
	Call Control 802.1p Priority: 6
	Audio 802.1p Priority: 6
	Video 802.1p Priority: 5 AUDIO RESOURCE RESERVATION PARAMETERS
	H.323 Link Bounce Recovery? v
	Idle Traffic Interval (sec): 20
	Keep-Alive Interval (sec): 5
	Keep-Alive Count: 5

Step	Description	
4.	Codecs Use the change ip-codec-set command to verify that C list. The example below shows the value used for comp G.711A and G.729A were also tested but not shown be	G.711MU is contained in the codec pliance testing. Note, codecs elow.
	change ip-codec-set 1 IP Codec Set	Page 1 of 2
	Codec Set: 1 Audio Silence Frames Packet Codec Suppression Per Pkt Size(ms) 1: G.711MU n 2 20 2:	

	Description				
	Signaling Group				
For compliance testing, the signaling group shown below and the associated SIP trunk					
	(administered in Steps 6-7) are used for routing calls to and from the Converges V				
	Portal server via Session Manager. Signaling group 1 was configured using the				
	noremators highlighted holow. All other fields were get as described in reference [
	parameters inginighted below. An other nerds were set as described in reference [2				
	• Group Type was set to sin				
	 Transport Method was get to the As a result Near and Liston Dort and Far. 				
	- Transport Methou was set to us. As a result, Near-enu Listen Fort and Far-				
	Listen Port are automatically set to 5061.				
	Peer Detection Enabled was set to y.				
	• Near-end Node Name was set to <i>procr</i> . Node names are defined in Step 2 abo				
	• Far-end Node Name was set to SM_21_41. This node name maps to the IP ad				
1	of the Session Manager as defined using the change node-names in command				
	of the Session Manager as defined using the change node-names ip command.				
l	• Far-end Network Region was set to 1				
i i	rur enu recevorit region was set to r.				
-					
	• Direct IP-IP Audio Connections was set to y This field must be set to y to en				
	• Direct IP-IP Audio Connections was set to <i>y</i> . This field must be set to <i>y</i> to en				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). Change signaling-group 1 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). Change signaling-group 1 Page 1 of 2 SIGNALING GROUP 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{change signaling-group 1} ^{Page 1 of 2} ^{SIGNALING GROUP} ^{Group Type: sip} 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{change signaling-group 1} ^{Page 1 of 2} ^{SIGNALING GROUP} ^{Group Number: 1} Group Type: sip IMS Enabled? n Transport Method: tls 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y Priority Video? n Enforce SIPS URI for SRTP? y 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y} ^{Priority Video? n} ^{Enforce SIPS URI for SRTP? y}} 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y} ^{Priority Video? n} ^{Enforce SIPS URI for SRTP? y}} Peer Detection Enabled? y Peer Server: SM Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). 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? y Priority Video? n Enforce SIPS URI for SRTP? y Peer Detection Enabled? y Peer Server: SM Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y Priority Video? n Enforce SIPS URI for SRTP? y 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 Near-end Node Name: procr Far-end Node Name: SM 21 31 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y Priority Video? n Enforce SIPS URI for SRTP? y 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 Near-end Node Name: procr Far-end Node Name: SM 21_31 Near-end Listen Port: 5061 Far-end Port: 5061 Fa				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to end Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y Priority Video? n Enforce SIPS URI for SRTP? y 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 Near-end Node Name: procr Far-end Node Name: SM 21_31 Far-end Listen Port: 5061 Far-end Network Region: 1 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y Priority Video? n Enforce SIPS URI for SRTP? y 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 Near-end Node Name: procr Far-end Node Name: SM_21_31 Far-end Listen Port: 5061 Far-end Network Region: 1 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y} ^{Priority Video? n} ^{Enforce SIPS URI for SRTP? y}} 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 Near-end Node Name: procr Far-end Node Name: SM_21_31 Far-end Listen Port: 5061 Far-end Network Region: 1 Far-end Domain: 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to end Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y Priority Video? n Enforce SIPS URI for SRTP? y 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 Near-end Node Name: procr Far-end Node Name: SM 21_31 Far-end Listen Port: 5061 Far-end Network Region: 1 Far-end Domain:				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to end Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). <pre></pre>				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to end Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). <pre>change signaling-group 1</pre> Page 1 of 2 SIGNALING GROUP Group Number: 1 Group Type: sip IMS Enabled? n Transport Method: tls Q-SIP? n IP Video? y Priority Video? n Enforce SIPS URI for SRTP? y 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 Near-end Node Name: procr Far-end Node Name: SM 21_31 Far-end Listen Port: 5061 Far-end Network Region: 1 Far-end Domain: Incoming Dialog Loopbacks: eliminate DTMF over IP: rtp-payload DTMF over IP: rtp-payload Direct IP-IP Audio Connections? y Head to the set to y to end Audio Connections? y Preserver IP: rtp-payload Preserver IP				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to end Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). 				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). ^{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? y ^{Priority Video? n} ^{Enforce SIPS URI for SRTP? y ^{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 ^{Near-end Node Name: procr ^{Far-end Node Name: SM 21 31} ^{Far-end Node Name: SM 21 31} ^{Far-end Network Region: 1} ^{Far-end Domain:} ^{Incoming Dialog Loopbacks: eliminate ^{DTMF over IP: rtp-payload ^{Session Establishment Timer(min): 3 ^{IP} Audio Connections? y ^{Intial IP-IP Direct Media? v ^{Intial IP-IP Direct Media? v}}}}}}}}}}}</sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup></sup>				
	 Direct IP-IP Audio Connections was set to y. This field must be set to y to en Media Shuffling on the trunk level (see Step 3 on IP-IP Direct Audio). 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? y Priority Video? n Enforce SIPS URI for SRTP? y 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 Near-end Node Name: procr Near-end Listen Port: 5061 Far-end Listen Port: 5061 Far-end Network Region: 1 Far-end Domain: Incoming Dialog Loopbacks: eliminate DTMF over IP: rtp-payload Session Establishment Timer(min): 3 Enable Layer 3 Test? y H.323 Station Outgoing Direct Media? n 				

	Description			
	Trunk Group For compliance testing, trunk group 1 was used for the SIP trunk group for routing call to and from the Convergys Voice Portal server via Session Manager. Trunk group 1 w configured using the parameters highlighted below. All other fields were set as descri- in reference [2] .			
	 On Page 1: Group Type field was set to <i>sip</i>. A descriptive name was entered for the Group Name. An available trunk access code (TAC) that was consistent with the existing dial plan was entered in the TAC field. Service Type field was set to <i>tie</i>. Signaling Group was set to the signaling group configured in the previous step. Member Assignment method was set to <i>auto</i>. Signaling Group was set to <i>1</i> (see Step 5). The Number of Members field contained the number of trunks in the SIP trunk group. It determines how many simultaneous SIP calls can be supported by the configuration 			
	 Member Assignment method was set to <i>auto</i>. Signaling Group was set to <i>I</i> (see Step 5). The Number of Members field contained the number of trunks in the SIP trunk group. It determines how many simultaneous SIP calls can be supported by the configuration. 			
	 Member Assignment method was set to <i>auto</i>. Signaling Group was set to <i>1</i> (see Step 5). The Number of Members field contained the number of trunks in the SIP trunk group. It determines how many simultaneous SIP calls can be supported by the configuration. 			

ep	Description
7.	Trunk Group – continued
	 On Page 3: Numbering Format was set to <i>private</i>. This field specifies the format of the calling party number sent to the far-end. UUI Treatment was set to <i>shared</i>. Maximum Size of UUI Contents was set to 128. Default values may be used for all other fields.
	change trunk-group 1 Page 3 of 22 TRUNK FEATURES ACA Assignment? n Measured: none Maintenance Tests? y
	Numbering Format: private UUI Treatment: shared Maximum Size of UUI Contents: 128 Replace Restricted Numbers? n Replace Unavailable Numbers? n
	Modify Tandem Calling Number: no Send UCID? y
	Show ANSWERED BY on Display? y

vep	Description						
8.	Route Pattern						
	Use the change route-pattern command to create a route pat	tern that will route calls to					
	the SIP trunk that connects Communication Manager to Session Manager						
	and Shi trunk that connects Communication Manager to Session Manager.						
	The events helew shows the nexts nettern wood during some	lianas tastina. A descriptiva					
	The example below shows the route pattern used during comp	Shance testing. A descriptive					
	name was entered for the Pattern Name field. The Grp No f	field was set to the trunk					
	group created in Steps 6–7 . The Facility Restriction Level (F	`RL) field was set to a level					
	that allows access to this trunk for all users that require it. Th	e value of 0 is the least					
	restrictive level Numbering Format was set to lov0-nut Th	e default values were used					
	for all other fields	le default values were used					
	for all other fields.						
	change route-pattern 1	Page 1 of 3					
		I UL J					
	Pattern Number: 1 Pattern Name: to	SM_21_31					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n	SM_21_31					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted	SM_21_31 DCS/ IXC					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dats	SM_21_31 DCS/ IXC QSIG Intw					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0	SM_21_31 DCS/ IXC QSIG Intw n user					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2:	SM_21_31 DCS/ IXC QSIG Intw n user n user					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3:	SM_21_31 DCS/ IXC QSIG Intw n user n user n user n user					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3: 4:	SM_21_31 DCS/ IXC QSIG Intw n user n user n user n user n user n user					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3: 4: 5:	SM_21_31 DCS/ IXC QSIG Intw n user n user n user n user n user n user n user n user n user n user					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3: 4: 5: 6:	SM_21_31 DCS/ IXC QSIG Intw n user n user					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3: 4: 5: 6: BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM	SM_21_31 DCS/ IXC QSIG Intw n user n user n user n user n user n user n user n user n user No. Numbering LAR					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3: 4: 5: 6: BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM 0 1 2 M 4 W Request	SM_21_31 DCS/ IXC QSIG Intw n user n user					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3: 4: 5: 6: BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM 0 1 2 M 4 W Request Sub	SM_21_31 DCS/ IXC QSIG Intw n user n user address					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3: 4: 5: 6: BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM 0 1 2 M 4 W Request 1: y y y y y n n rest	SM_21_31 DCS/ IXC QSIG Intw n user n user n user n user n user n user n user n user n user n user ser n user n user No. Numbering LAR lev0-pvt none					
	Pattern Number: 1 Pattern Name: to SCCAN? n Secure SIP? n Grp FRL NPA Pfx Hop Toll No. Inserted No Mrk Lmt List Del Digits Dgts 1: 1 0 2: 3: 4: 5: 6: BCC VALUE TSC CA-TSC ITC BCIE Service/Feature PARM 0 1 2 M 4 W Request 1: y y y y y n n rest 2: y y y y y n n rest	SM_21_31 DCS/ IXC QSIG Intw n user n user n user n user n user n user n user n user n user ser n user n user No. Numbering LAR lev0-pvt none none					

)			Descri	ption				
,]	Routing Calls to Session Manager							
	Automatic Alternate Routing (AAR) was used to route calls to Convergys Voice Port							
	via Session Manager. Tw	o places n	eed to be	change	a to su	pport this rou	ting. F	first, us
	below shows entries previ	ously crea	ted using	the dis	nlav di	ialplan analy	sis co	mmand
	The 3rd entry specifies that	t numbers	s that begi	n with	7 are o	f Call Type a	ar. S	econd.
	the change aar analysis c	ommand t	to create a	n entry	in the	AAR Digit A	nalysi	is Table
	The example below shows	entries p	reviously	created	using	the display as	ar ana	alysis (
	command. The entry specifi	fies that n	umbers th	nat begi	n with	7 and are 5 d	igits lo	ong use
	route pattern 1. Route pat	tern 1 rou	tes calls to	o Sessio	on Man	lager.		
	change dialplan analysis					Pago	1 of	1.2
	Change draipian analysis	DIAL :	PLAN ANALY	SIS TAB	LE	raye		12
			Location:	all		Percent Full	1: 3	
	Dialed Total Call	Diale	d Total	Call	Dial Stri	ed Total Ca	all	
	1 3 dac	00111	g Dengen	1900	0011	ing hengen i	<i>i</i> pc	
	5 5 ext 7 5 aar							
	8 1 fac 9 1 fac							
	* 3 fac							
	display aar analysis 7					Page 2	l of	2
		AAR D	IGIT ANALY Location:	SIS TAB all	LE	Percent Full	1: :	2
	Dialed	Total	Poute	Call	Node	λΝΤ		
	String	Min Max	Pattern	Туре	Num	Reqd		
	7	55	1	aar		n		

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. Session Manager must be administered via System Manager.

The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

The procedures described in this section include configurations in the following areas:

- SIP domain
- Logical/physical Locations where SIP Entities may reside
- **SIP Entities** corresponding to the SIP telephony systems including Communication Manager, the Convergys Voice Portal server, and Session Manager itself
- Entity Links which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies which control call routing between the SIP Entities
- **Dial Patterns** which govern to which SIP Entity a call is routed
- Information corresponding to the **Session Manager** server to be managed by System Manager



Sp Ac Se in	Specify SIP Domain Add the SIP domain for which the communications infrastructure will be authoritative. Select SIP Domains on the left and click the New button (not shown) on the right. Fill in the following:						
	 Name: Enter the domain name specified to be the Authoritative Domain on the IP Network Region form on Communication Manager (see Section 5, Step 3) Type: Select sin 						
	• Notes: D	Descriptive text (optional)					
CI	ick Commit.						
C	AVAYA	Avaya Aura® System	Manager 6.3	Last Logged or Help About Chan	n at November 13, 2 Ige Password Log	013 9:49 g off ac	
		Avaya Aura®System	Manager 6.3	Last Logged or Help About Chan	n at November 13, 2 ige Password Log Routing *	013 9:49 g off ad	
		Avaya Aura [®] System	Manager 6.3	Last Logged or Help About Chan	n at November 13, 2 nge Password Log Routing *	013 9:49 g off ad Hom	
	Routing Domains	Avaya Aura [®] System Home / Elements / Routing / Domains Domain Management	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 Ige Password Log Routing *	013 9:49 g off ac Hom Help	
	Routing Domains Locations Adaptations	Avaya Aura® System Home / Elements / Routing / Domains Domain Management	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 ge Password Log Routing *	013 9:49 g off ac Hon Help	
	Routing Domains Locations Adaptations SIP Entities	Avaya Aura® System Home / Elements / Routing / Domains Domain Management	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 ge Password Log Routing	013 9:49 g off ac Hom Help	
	Routing Domains Locations Adaptations SIP Entities Entity Links	Avaya Aura® System Home / Elements / Routing / Domains Domain Management I Item Refresh	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 ge Password Log Routing *	013 9:4 g off ac Hon Help Enable	
	Routing Domains Locations Adaptations SIP Entities Entity Links Time Ranges	Avaya Aura® System Home / Elements / Routing / Domains Domain Management 1 Item Refresh Name	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 Ige Password Log Routing *	013 9:4 g off ac Hon Help Enable	
	Routing Domains Locations Adaptations SIP Entities Entity Links Time Ranges Routing Policies	Avaya Aura® System Home / Elements / Routing / Domains Domain Management I Item Refresh Name *avaya.com	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 ge Password Log Routing * Filter:	013 9:40 g off ac Hom Help Enable	
	Routing Domains Locations Adaptations SIP Entities Entity Links Time Ranges Routing Policies Dial Patterns	Avaya Aura® System Home / Elements / Routing / Domains Domain Management I Item Refresh Name * avaya.com	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 ge Password Lo Routing * Filter:	1013 9:44 g off ac Hom Help Enable	
	Routing Domains Locations Adaptations SIP Entities Entity Links Time Ranges Routing Policies Dial Patterns Regular Expressions	Avaya Aura® System Home / Elements / Routing / Domains Domain Management 1 Item Refresh Name * avaya.com	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 ge Password Lo Routing * Filter:	g off ac Hon Help Enable	
	Routing Domains Locations Adaptations SIP Entities Entity Links Time Ranges Routing Policies Dial Patterns Regular Expressions Defaults	Avaya Aura® System	Manager 6.3	Last Logged or Help About Chan Commit Cancel	n at November 13, 2 ge Password Lo Routing * Filter:	1013 9:44 g off ad Hom Help	

3.	3. Add Locations				
	Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of routing and bandwidth management. To add a location, select				
	following:	ek on the ritew button (not shown) on the right. This in the			
	Under General:				
	• Name: A dese	criptive name			
	• Notes: Descri	ptive text (optional)			
	The remaining fields under of parameters between Session compliance testing.	<i>General</i> can be filled in to specify bandwidth management Manager and this location. The default values were used for			
	Next, fill in the following:				
	Under Location Pattern:				
	• IP Address Pattern:	An IP address pattern used to logically identify the location			
	• Notes:	Descriptive text (optional)			
	The screen below shows add the Communication Manage	ition of the ".21 and .101 Subnet" Location which includes r, Session Manager, and the Convergys Voice Portal server.			
	Click Commit to save the Lo	ocation definition.			

		Routing
* Routing	Home / Elements / Routing / Locations	
Domains	Location Details	Commit Cancel
Locations		
Adaptations	Call Admission Control has been set to ignore SDP. All calls will be counted using the Defau	lt Audio Bandwidth. Note: If this setting is disabled, you should retu
SIP Entities	See Session Manager -> Session Manager Administration -> Global Settings	
Entity Links	Conoral	
Time Ranges		
Dial Dattorns	* Name: .21 and .101 Subnet	
Dial Patterns	Notes:	
Regular Expressions		
Defaults	Dial Plan Transparency in Survivable Mode	
	Enabled:	
	Listed Directory Number:	
	Accordiated CM SID Entity	
	Associated CM SIP Entity:	
	Overall Managed Bandwidth	
	Overall Manageu balluwiuth	
	Managed Bandwidth Units: Kbit/sec	
	Total Bandwidth:	
	Per-Call Bandwidth Parameters	
	* Default Audio Bandwidth: 80 Kbit/sec	•
	Alarm Threshold	
	Audio Alarm Threshold: 80 - 96	
	* Latency before Audio Alarm Trigger: 5 minutes	
	Location Dattorn	
	Add Remove	
	2 Items Refresh	Filter:
	IP Address Pattern	Notes
	* 10.64.101.*	
	* 10.64.21.*	
	Select : All, None	
		Commit Cancel

Add S A SIP system testing the Av Select	IP Entities Entity must be added for a supported by it using SI a SIP Entity was added raya S8300D Media Serv SIP Entities on the left a	Session Manager and for each SIP-based telephony P trunks. In the configuration used for compliance for the Session Manager itself, the processor Ethernet for er, and the Convergys Voice Portal server. and click on the New button (not shown) on the right. Fill
in the	following:	
Under	General.	
•	Name	A descriptive name
٠	FQDN or IP Address:	FQDN or IP address of the signaling interface for the entity
•	Туре:	"Session Manager" for Session Manager, "CM" for Communication Manager, or "SIP Trunk" for the Convergys Voice Portal server
•	Adaptation:	Leave blank
•	Location:	Select the appropriate Location configured in previous step
•	Time Zone:	Select the proper time zone for this installation
fields i	in the resulting new row Port :	Port number on which the system listens for SIP requests
•	Protocol: Default Domain:	Transport protocol to be used to send SIP requests Select the SIP Domain configured in Step 2 of this section or "ALL"
Defaul Entity	It settings can be used for definition.	r the remaining fields. Click Commit to save the SIP

The following screen shows the addition of Session Manager. Two **Port** entries are added. TLS (well-known port 5061) is used for communication with Communication Manager. TCP (well-known port 5060) is used for communication with the Convergys Voice Portal server.

Also note that the entries under *Entity Links* are populated automatically after the Entity Links are administered (**Step 5** below).

Image Adaptations Adaptations <t< th=""><th>* Routing</th><th>Home / Elements /</th><th>Routing / SIP Entitie</th><th>s</th><th></th><th></th><th>,</th></t<>	* Routing	Home / Elements /	Routing / SIP Entitie	s			,
spretniv peaks General Anapiatom Spretniv peaks General Spretniv peaks Concord peaks Spretniv peak	Domains			-			
Adaptavianie Strientiski Entity Laidi Bried andersis Bried and andersis<	Locations	SIP Entity Details				Commit Cancel	
sintede	Adaptations	General					
initial initinitial initinitial initinitial initinitial initial initial initial	SIP Entities		* Name	: SM_21_31			
The farge is a set of farge is a se	Entity Links		* FQDN or IP Address	: 10.64.21.31			
indir product	Time Ranges		Туре	: Session Manager	2		
Internation Regular propersions Details Lexin::::::::::::::::::::::::::::::::::::	Routing Policies		Notes	:			
Note: Loc.tin: : 10.1 Subret Defaults	Dial Patterns				_		
control Prove:	Defaults		Location	: .21 and .101 Subnet	•		
Ime Zone:	bendits		Outbound Proxy	:	•		
Credential name: SIP Link Monitoring: Safe Anson Credential name: Safe Anson Safe Anson Phruse Safe A			Time Zone	: America/Denver	•		
State State <td< td=""><td></td><td></td><td>Credential name</td><td>:</td><td></td><td></td><td></td></td<>			Credential name	:			
SIP Luk Montoring: We Session Manager Configuration I Entity Links Main Immosi Sip Entity 1 Proteo Posto Posto Particy Partice Posto P		SIP Link Monitori	ng				
Etity Links Add Is terms: Refresh Is term			SIP Link Monitoring	Use Session Manager	Configuration •		
Entity Links Add TS Terms: Refresh IS Terms: Refresh IS TAR 2,2,3 IN TS IN 500 INAL 21,7 IN 500 INAL 24, 7 IN 500 INAL 24 IN 500 INAL 24 IN 100 INAL 24, 7 IN 500 INAL 24 IN 100 INAL 24, 7 IN 500 INAL 24 IN 100 INAL 24, 7 IN 500 INAL 24 IN 100 INAL 24, 7 IN 100 INAL 24, 7 IN 100 INAL 24, 7 IN 100 INAL 24 IN 100 INAL 24, 7 IN 100 INAL 24 IN 100 INAL 24, 7 IN 100 INAL 24 INAL							
15 Item: Refresh FRE: Image: Refresh Image: Refresh Image: Refresh FRE: Im		Add Remove					
SPE Entity 1 Port oco SPE Entity 2 Port Connection Policy Deny Here S SN_21_31 TCS 5060 AuA_21_72 \$0600 Trusted Image: Signal for the si		15 Items Refresh					Filter: E
MAX_21_31 TCP \$566 AAAA_21_72 \$566 Turasted I SM_21_31 TCP \$566 Coverys \$566 Turasted I SM_21_31 TCP \$566 Coverys \$566 Turasted I SM_21_31 TCP \$566 Coverys \$566 Turasted I SM_21_31 TCP \$566 Turasted I I SM_21_31 TCP \$5060 Turasted I I I SM_21_31 TCP \$5060 Turasted I I I I I I Sett: Alternove Iters Iters Iters Iters Iters		SIP Entity 1	Protocol Port	SIP Entity 2	Port	Connection Policy	Deny New Ser
SM21_31 TLS 6001 CM20_27 9 5000 Trusted 9 SM_21_31 TLS 6000 Trusted 9 9 SM_21_31 TLS 6000 Trusted 9 9 Sett: SM_21_31 TLS 6000 Trusted 9 9 Sett: SM_21_31 TLS 6000 Trusted 9 9 Sett: SAL, 20.31 TLS 6000 Trusted 9 9 Sett: AL, None 6000 Trusted 9 9 Add Remove <td></td> <td>SM_21_31 •</td> <td>TCP 💌 * 5060</td> <td>AAM_21_72</td> <td>• * 5060</td> <td>trusted</td> <td></td>		SM_21_31 •	TCP 💌 * 5060	AAM_21_72	• * 5060	trusted	
Immediate		SM_21_31	TLS * * 5061	CM_20_72	* \$5061	trusted •	
Select : Al, None Previous Page 1 of 3 Port TCP Failover port: TS Failover port: Td Remove 4 Items: Refresh Fiter: 5000 1000 5000 1000 5000 1000 5000 1000 3000 1000 3000 1000 </td <td></td> <td>SM_21_31</td> <td>TLS • * 15060</td> <td>ET 21 211</td> <td> \$060 \$063 </td> <td>trusted •</td> <td></td>		SM_21_31	TLS • * 15060	ET 21 211	 \$060 \$063 	trusted •	
Select : All, None < Previous Page 1 of 3		SM_21_31 •	TCP • * 5060	iview	* * 5060	trusted	
4 Items Refresh Filter: Port Protocol Default Domain Notes 5060 TCP * avaya.com *		TCP Failover port:					
Port Protocol Default Domain Notes 5060 TCP avaya.com avayava		4 Items Refresh					Filter: E
S060 TCP avaya.com avayaw.com avaya.com avaya.com avaya.com avaya.com		Port	Protocol	Default Domain	Notes		
5060 UDP avaya.com a		5060	тср 💌	avaya.com 💌			
Sobi TLS avaya.com Sobi TCP avaya.com Select : All, None Select : All, None SIP Responses to an OPTIONS Request Add Remove Filter: 0 Items Refresh Filter: Response Code & Reason Phrase Mark Notes Commit Cancel Cancel		5060	UDP -	avaya.com 💌			
Select : All, None SIP Responses to an OPTIONS Request Add Remove O Items Refresh Filter: Response Code & Reason Phrase Mark Entity Up/Down Notes Commit Cancel		5061	TLS •	avaya.com 💌			
SIP Responses to an OPTIONS Request Add Remove 0 Items Refresh Filter: Response Code & Reason Phrase Mark Entity Up/Down Commit Cancel		Select : All, None					
Add Remove 0 Items Refresh Filter: Response Code & Reason Phrase Mark Entity Up/Down Image: Commit Cancel Commit Cancel		SIP Responses to	an OPTIONS Requ	ıest			
Response Code & Reason Phrase Mark Entity Up/Down Notes Commit Cancel							
Commit Cancel		Add Remove					Filter: F
Commit Cancel		Add Remove 0 Items Refresh	6 Roacon Dhraco			Mark	Filter: E
		Add Remove 0 Items Refresh Response Code	: & Reason Phrase			Mark Entity Up/Down	Filter:

/ Elements / Routing / SIP Entities http Details rral * Name: * FQDN or IP Address: Type: Notes: Adaptation: Location: Time Zone:	CM_21_41 10.64.21.41 CM · .21 and .101 Subnet ·	Commit Cancel	
ntity Details Iral * Name: * FQDN or IP Address: Type: Notes: Adaptation: Location: Time Zone:	CM_21_41 10.64.21.41 CM • .21 and .101 Subnet • America/Denver	Commit	
ral * Name: * FQDN or IP Address: Type: Notes: Adaptation: Location: Time Zone:	CM_21_41 10.64.21.41 CM • .21 and .101 Subnet • America/Denver		
* Name: * FQDN or IP Address: Type: Notes: Adaptation: Location: Time Zone:	CM_21_41 10.64.21.41 CM • .21 and .101 Subnet • America/Denver		
* FQDN or IP Address: Type: Notes: Adaptation: Location: Time Zone:	10.64.21.41		
Type: Notes: Adaptation: Location: Time Zone:	CM .21 and .101 Subnet America/Denver		
rype: Notes: Adaptation: Location: Time Zone:	.21 and .101 Subnet		
Notes: Adaptation: Location: Time Zone:	.21 and .101 Subnet •		
Adaptation: Location: Time Zone:	.21 and .101 Subnet . America/Denver		
Location: Time Zone:	.21 and .101 Subnet		
Location: Time Zone:	America/Denver		
Time Zone:	America/Denver		
override Port & Transport with DNS SRV:			
* SIP Timer B/F (in seconds):	4		
Credential name:			
Call Detail Recording:	both 💌		
Detection			
Loop Detection Mode:	Off		
ink Monitoring			
SIP Link Monitoring:	Use Session Manager Configuration		
Supports Call Admission Control:			
Shared Bandwidth Manager:			
Association:			
Backup Session Manager Bandwidt Association:	h v		
y Links Remove			
m Refresh			Filter
SIP Entity 1 Protocol Port	SIP Entity 2 Port	Connection Policy	Deny New S
SM_21_31 • TLS • * 5061	CM_21_41 * 506	trusted •	
t : All, None			
ailover port:			
ailover port:			
Responses to an OPTIONS Require	est		
			Filter:
ms Refresh		Mark Entity Up/Down	Notes
ms Refresh Response Code & Reason Phrase			
t : aik aik	SM_21_31 TLS * 5061 All, None	SM_21_31 TLS 5061 CM_21_41 V \$066 All, None over port:	SM_21_31 TLS • [S061] CM_21_41 • [S061] trusted • All, None over port:

Solution & Interoperability Test Lab Application Notes ©2014 Avaya Inc. All Rights Reserved.

		ura° System	Manager 6.3		Last Logged on Help About Chang	at November 13, 20 ge Password Log Routing *
[∞] Routing	Home / Elements / Ro	uting / SIP Entities				
Domains	CID Estitu Dataila				Commit Consol	
Locations	SIP Entity Details				Conter	
Adaptations	General					
SIP Entities		* Name:	Convergys			
Entity Links	*	FQDN or IP Address:	10.64.21.153			
Time Ranges		Type:	SIP Trunk v			
Routing Policies		Notes:				
Dial Patterns						
Regular Expressions		Adaptation:		•		
Deraults		Location:	•			
		Time Zone:	America/Denver	-		
	Override Port & Tra	nsport with DNS SRV:				
	* SIP Tir	ner B/F (in seconds):	4			
		Credential name:				
		Call Detail Recording:	egress 🔻			
	SIP Link Monitoring	Loop Detection Mode: SIP Link Monitoring:	Off Use Session Manager Config	uration 💌		
	Supports C	all Admission Control:				
	Shared	Bandwidth Manager:				
	Primary Sessi	on Manager Bandwidth	_			
	Backup Sessie Entity Links Add Remove	Association: on Manager Bandwidth Association:				
	1 Item Refresh					Filter
	SIP Entity 1	Protocol Port	SIP Entity 2	Port	Connection Policy	Denv New 9
	SM_21_31	TCP • * 5060	Convergys	 * 5060 	trusted	
	Select : All, None					
	SIP Responses to a Add Remove	n OPTIONS Reque	est			
	0 Items Refresh					Filter
	Response Code &	Reason Phrase			Mark Entity	Notes

5.	Add Entity Link A SIP trunk betw link. In the config one for Session M the Convergys V	ts ween Session Manager and a telephony system is described by an Entity guration used for compliance testing, two Entity Links were configured; Manager to Communication Manager and one for Session Manager to oice Portal server.
	To add an Entity shown) on the rig in the new row th	Link, select Entity Links on the left and click on the New button (not ght. For the link to Communication Manager, fill in the following fields hat is displayed:
	 Name: SIP Entitive Protocol: Port: SIP Entitive Port: Trusted: Click Commit to the screen below Communication for the screen below to the screen below to	A descriptive name by 1: Select the Session Manager SIP Entity configured in previous step Select "TLS" Port number to which the other system sends SIP requests by 2: Select the Communication Manager SIP Entity configured in previous step Port number on which the other system receives SIP requests Select "trusted" b save the configuration.
	Αναγα	Avaya Aura [®] System Manager 6.3 Last Logged on at November 13, 2013 9:49 AM Help About Change Password Log off admin Routing * Home
	▼ Routing	Home / Elements / Routing / Entity Links
	Domains Locations Adaptations	Help ? Entity Links Commit Cancel
	Entity Links	1 Item Refresh Filter: Enable
	Time Ranges	Name SIP Entity 1 Protocol Port SIP Entity 2 Port Connection New
	Routing Policies	
	Regular Expressions Defaults	Select : All, None
		Commit

AVAYA		Avaya Au	ura® System	Manag	ger 6.3	3	l Help	Last Logged o About Char	n at Novembe nge Password
▼ Routing	I Home	e / Elements / Rou	ting / Entity Links	;					Routing
Domains	Entity	Links					Commit	Cancel	
Locations									
SIP Entities									
Entity Links	1 Iter	em Refresh							F
Time Ranges		Name	SIP Entity 1	Protocol	Port	SIP Entity 2		Port	Connecti Policy
Dial Patterns		* Convergys	* SM_21_31 •	тср 💌	* 5060	* Convergys	•	* 5060	trusted
Regular Expressions	۲.								
Defaults	Selec	ct : All, None							
							Commit	Cancel	
dd Routing I routing polic nust be added	Policy cy shou for rou	ld be creat	ed for eac to Commu	h "Ro nicati	uting on Ma	Destinati anager (fr	ion". A	A rout e Con	ing po ivergy
dd Routing I routing polic nust be added ortal server).	Policy cy shou for rou Likewi ce Port	Ild be creat ating calls t ise, a routin tal server (ted for eac to Commu ng policy r from Com	h "Ro nicati nust b munio	uting on Ma be add cation	Destinati anager (fr ed for ro Manage	ion". A rom th uting c r).	A rout e Con calls to	ing po ivergy o the
dd Routing I routing polic nust be added ortal server). onvergys Voi o add a routin not shown) on	Policy by shou for rou Likewi ice Port ng polic the rig	Ild be creat tting calls t ise, a routin tal server (cy, select F ght. The fo	ted for eac to Commu ng policy r from Com Routing Po llowing sc	h "Ro nicati nust b munic olicies reen i	uting on Ma be add cation s on th s disp	Destinati anager (fr ed for ro Manage he left and layed. Fi	ion". A rom th uting c r). d click ll in th	A rout e Con calls to con the	ing po ivergy o the e Nev owing
dd Routing I routing polic oust be added ortal server). onvergys Voi o add a routin ot shown) on	Policy by shou for rou Likewi ice Port g polic the rig	Ild be creat ting calls t ise, a routin tal server (cy, select F ght. The fo	ted for eac to Commu ng policy r from Com Routing P o llowing sc	h "Ro nicati nust b munic blicies reen i	uting on Ma be add cation s on th s disp	Destinati anager (fr ed for ro Manage te left and layed. Fi	fon". A rom th uting c r). d click ll in th	A rout e Con calls to on the	ing po ivergy o the e Nev owing
dd Routing I routing polici oust be added ortal server). onvergys Voi o add a routin ot shown) on nder <i>General</i> nter a descrip	Policy by shou for rou Likewi ice Port g polic the rig	Ild be creat ting calls t ise, a routin tal server (cy, select F ght. The fo me in Nan	ted for eac to Commu ng policy r from Com Routing P llowing sc ne and opt	h "Ro nicati nust b munic olicies reen i	uting on Ma be add cation s on th s disp text ir	Destination anager (fr ed for ro Manage Manage le left and layed. Fi Notes .	ion". <i>A</i> rom th uting c r). d click ll in th	A rout e Con calls to on the foll	ing po ivergy o the e Nev owing
dd Routing I routing polic oust be added ortal server). onvergys Voi o add a routin ot shown) on inder <i>General</i> nter a descrip	Policy cy shou for rou Likewi ice Port g polic the rig tive nat	Ild be creat ting calls t ise, a routin tal server (cy, select F ght. The fo me in Nan	ted for eac to Commu ng policy r from Com Routing Po llowing sc ne and opt	h "Ro nicati nust b munic olicies reen i	uting on Ma be add cation s on th s disp text ir	Destinati anager (fr ed for ro Manage he left and layed. Fi Notes .	ion". A com th uting c r). d click ll in th	A rout e Con calls to con the foll	ing po ivergy the the Nev owing
dd Routing I routing polic ust be added ortal server). onvergys Voi o add a routin ot shown) on nder <i>General</i> nter a descrip nder <i>SIP Ent</i> ilick Select	Policy by shou for rou Likewi ice Port g polic the rig tive nat	Id be creat tring calls t ise, a routin tal server (cy, select F ght. The fo me in Nan Destination	ted for eac to Commung policy in from Com Routing Policy Illowing sc ne and opt	h "Ro nicati nust b munic olicies reen i ional	uting on Ma be add cation s on th s disp text ir	Destination anager (fr ed for ro Manage the left and layed. Fi Notes .	fon". A rom th uting c r). d click ll in th	A rout e Con calls to on th e foll	ing po vergy o the e Nev owing
dd Routing I routing polic ust be added ortal server). onvergys Voi o add a routin ot shown) on nder <i>General</i> nter a descrip nder <i>SIP Enti</i> lick Select , au	Policy by shou for rou Likewi ice Port g polic the rig tive native tive native tive national	ald be creat ating calls to tal server (cy, select F ght. The fo me in Nam Destination in select the	ed for eac to Commung policy in from Com Routing Policy Illowing sc ne and opt appropria	h "Ro nicati nust b munic olicies reen i ional te SIP	uting on Ma be add cation s on th s disp text ir	Destination anager (fr ed for ro Manage ne left and layed. Fi Notes .	ton". A rom th uting c r). d click ll in th	A rout e Con calls to on the foll routin	ing po vergy o the e Nev owing
dd Routing I routing polic ust be added ortal server). onvergys Voi o add a routin ot shown) on nder <i>General</i> nter a descrip nder <i>SIP Ent</i> i lick Select , an plies.	Policy cy shou for rou Likewi ice Port ng polic the rig tive nat tive nat	Ild be creat ating calls to tal server (cy, select F ght. The fo me in Nan Destination in select the	ted for eac to Commung policy r from Com Routing Pe llowing sc ne and opt appropria	h "Ro nicati nust b municolicies reen i ional te SIP	uting on Ma be add cation s on th s disp text ir P Entit	Destination anager (fr ed for ro Manage he left and layed. Fi Notes .	ton". A rom th uting c r). d click ll in th	A rout e Con calls to on the foll routin	ing po vergy o the e Nev owing
dd Routing I routing polic ust be added ortal server). onvergys Voi o add a routin ot shown) on nder <i>General</i> nter a descrip nder <i>SIP Entr</i> ick Select , an plies.	Policy cy shou for rou Likewi ice Port ng polic the rig tive nat tive nat	Ild be creat ting calls t ise, a routin tal server (cy, select F ght. The fo me in Nan Destination a select the	eed for eac to Commung policy r from Com Routing Policy llowing sc ne and opt	h "Ro nicati nust b munic olicies reen i ional te SIP	uting on Ma be add cation s on th s disp text ir	Destination anager (fr ed for ro Manage he left and layed. Fi Notes .	fon". A rom th uting c r). d click ll in th	A rout e Con calls to on th e foll routin	ing po vergy o the e Nev owing
dd Routing I routing polic ust be added ortal server). onvergys Voi o add a routin ot shown) on nder <i>General</i> nter a descrip nder <i>SIP Entri</i> ick Select , an plies.	Policy by shou for rou Likewi ice Port g polic the rig tive nat tive nat tive nat tive nat	ald be creat ating calls to a routin tal server (cy, select F ght. The for me in Nan Destination a select the	ed for eac to Commu ng policy r from Com Routing P llowing sc ne and opt	h "Ro nicati nust b munic olicies reen i ional	uting on Ma be add cation s on th s disp text ir P Entit	Destination anager (fr ed for ro Manage the left and layed. Fi Notes .	ion". <i>A</i> rom th uting c r). d click ll in th	A rout e Con calls to on th e foll	ing po vergy o the e Nev owing
dd Routing I routing polic ust be added ortal server). onvergys Voi o add a routin ot shown) on nder <i>General</i> nter a descrip nder <i>SIP Entr</i> lick Select , an oplies. nder <i>Time of</i> lick Add , and	Policy by shou for rou Likewi ice Port ag polic the rig tive nat tive nat tive nat tive nat d then Day: I select	ald be creat ating calls to tal server (cy, select F ght. The fo me in Nan Destination a select the	red for eac to Commung policy in from Com Routing Policy Illowing sc ne and opt appropria	h "Ro nicati nust b munic olicies reen i ional te SIP me ra	uting on Ma e add cation s on th s disp text ir P Entit nge.	Destination anager (fr ed for ro Manage te left and layed. Fi Notes .	ion". <i>A</i> rom th uting c r). d click ll in th	A rout e Con calls to on th e foll routin	ing po vergy o the e Nev owing
Id Routing I routing polic ast be added rtal server). onvergys Voi add a routin ot shown) on ader <i>General</i> ater a descrip ader <i>SIP Entr</i> ick Select , an plies. ader <i>Time of</i> ick Add , and	Policy by shou for rou Likewi ice Port ag polic the rig tive nat tive nat	ald be creat ating calls to ase, a routin tal server (cy, select F ght. The fo me in Nam Destination a select the c the defaul	ed for eac to Commung policy in from Com Routing Po llowing sc ne and opt appropria	h "Ro nicati nust b munic olicies reen i ional te SIP me ra	uting on Ma be add cation s on th s disp text ir P Entit nge.	Destination anager (fr ed for ro Manage ne left and layed. Fi Notes .	fon". A rom th uting c r). d click ll in th	A rout e Con calls to on the foll	ing po vergy o the e Nev owing
dd Routing I routing polic ust be added ortal server). onvergys Voi o add a routin ot shown) on der <i>General</i> uter a descrip nder <i>SIP Entr</i> ick Select , an plies. nder <i>Time of</i> ick Add , and	Policy cy shou for rou Likewi ice Port ag polic the rig : tive nan <i>ity as D</i> nd then <i>Day</i> : l select e used f	Id be creat ting calls t ise, a routin tal server (cy, select F ght. The fo me in Nan Destination a select the the defaul for the rem	ed for eac to Commung policy r from Com Routing Pe llowing sc ne and opt appropria	h "Ro nicati nust b municolicies reen i ional te SIP me ra	uting on Ma be add cation s on th s disp text ir P Entit nge. lick C	Destination anager (fr ed for ro Manage he left and layed. Fi Notes . by to which commit to	fon". A rom th uting c r). d click ll in th ch this	A rout e Con calls to on the foll routin	ing pol vergy o the e Nev owing ng pol
Id Routing I routing polic ist be added rtal server). onvergys Voi add a routin ot shown) on ider <i>General</i> ter a descrip ider <i>SIP Entr</i> ick Select , an plies. ider <i>Time of</i> ick Add , and finition. The	Policy cy shou for rou Likewi ice Port ag polic the rig : tive nan <i>ity as D</i> nd then <i>Day</i> : I select e used f follow	Ild be creat ating calls to take, a routin tal server (cy, select F ght. The fo me in Nan Destination a select the the defaul for the rem	ed for eac to Commung policy in from Com Routing Po llowing sc ne and opt appropria appropria	h "Ro nicati nust b municolicies reen i ional te SIP me ra	uting on Ma be add cation s on th s disp text ir P Entit nge. lick C	Destination anager (fr ed for ro Manage ne left and layed. Fi Notes . by to which commit the colicy use	fon". A rom th uting c r). d click ll in th ll in th ch this	A rout e Con calls to on the foll routin	ing pol vergy o the e Nev owing ng pol

Routing	Illome / Ele	omonto / Dou	iting / Pout	ing Pol	icies							Routing
Domains	nome y Ere	cinents / Roo	iting / Kout	ing ron	icies							
Locations	Routing Poli	icy Details								Commit	ancel	
Adaptations												
SID Entition	General											
SIP Entities				Name:	CM_21_4	1]			
Entity Links			D	isabled:								
Time Ranges				Dotrioc:	0							
Routing Policies				ACC ICSI					1			
Dial Patterns				Notes:								
Regular Expressions	-											
Defaults	SIP Entity	y as Destina	ation									
	Select											
	Name		FQDN	or IP Ad	dress					Туре	No	tes
	CM_21_41		10.64.	21.41						СМ		
	Time of D	Day										
	Add	Nove View Ga	aps/Overlaps									
	1 Item Re	fresh										Filter:
	Ran	iking 🔶 Nar	me Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
	0	24/7	7 🗸	1	1	\checkmark	1	\checkmark	1	00:00	23:59	Time Rang
	Dial Patte	erns										
	Add Rem	nove										
	3 Items R	lefresh										Filter:
	Patt	tern 🔺	Min Max	1	Emergen	cy Call	s	IP Domai	n	Originating L	ocation	Note
	5	9	; 5				a١	vaya.com		-ALL-		CM_21
	8	6	j 6				- /	ALL-		-ALL-		
	91	1	.2 12				- /	ALL-		-ALL-		
	Regular E	Expressions	1									
	Add Rem	nove										
	0 Items R	efresh							_			Filter
	Patt	tern		Rank O	order				Den	У	Notes	•
										Commit	incel	

AVAYA	Avaya Au	ura® Syste	m Manager	6.3	Last Help Abo	Logged on at Noven ut Change Passw	nber 13, 2013 9:4 vord Log off a
[™] Routing	Home / Elements / Rou	ting / Routing	Policies				
Domains							Hel
Locations	Routing Policy Details				Commit Car	ncel	
Adaptations	General						
SIP Entities		* Nar	e: Convergys				
Entity Links		Dicabl	ad.				
Time Ranges		* Dotri					
Routing Policies		Keur	es. 0				
Dial Patterns		Not	es:				
Regular Expressions	CID Entity as Destin	tion					
Delauits	SIP Entity as Destina						
	Select						
	Name	FQDN or IF	Address		Туре	Notes	
	Ranking Nar O 24/7 Select : All, None Dial Patterns Add Remove 1 Item Refresh Pattern * Pattern *	Min Max	ue Wed Thu 2 2 2 2 Emergency Ca	Fri Sat Su 2 2 2 2 SIP Domain	n Start Time	End Time 1 23:59 T	Notes ime Range 24/7 Filter: Enab Notes
	7 5	5		avaya.com	-ALL-		
	Select : All, None Regular Expressions Add Remove						
	0 Items Refresh	Ra	nk Order		Deny	Notes	Filter: Enabl
		K			Commit	ncel	

Add I	Dial Patterns	
A Dia	l Pattern is asso	ciated with a Routing Policy to direct calls to a destination based
on dia	led digits.	
To ado showr	d a dial pattern, a) on the right. F	select Dial Patterns on the left and click on the New button (not Fill in the following, as shown in the screens below:
Under	General:	
•	Pattern:	Dialed number or prefix
٠	Min:	Minimum length of dialed number
٠	Max:	Maximum length of dialed number
•	SIP Domain:	SIP domain specified in Step 2 of this section, or ALL.
٠	Notes:	Comment on purpose of dial pattern.
Under Click Locat	<i>Originating Lo</i> Add, and then s ion Name field	<i>ocations and Routing Policies</i> : select the appropriate Location (or "ALL") for Originating and select the appropriate Routing Policy from the list.
Defau	lts can be used t	for the remaining fields. Click Commit to save the Dial Pattern.
Derud		

Routing Home / Elements / Routing / Dial F Domains Locations Locations Dial Pattern Details Adaptations General SIP Entities F Entity Links Emergency Dial Patterns Emergency Regular Expressions Emergency Defaults Originating Locations and Routi Add Remove 1 Item Refresh Originating Location Name + Originating Location Name + Originating Location Name +	Patterns Patterns Patterns * Min: 5 * Max: 5 cy Call: Priority: 1 y Type: Notes: CM_21_ ting Policies	m ∙ 41		Commit) (C	Cancel	ting ×
Domains Dial Pattern Details Locations Dial Pattern Details Adaptations General SIP Entities Fine Ranges Routing Policies Emergency Processions Defaults Emergency Processions Originating Locations and Routing Add Add Remove 1 Item Refresh Originating Location Name (Control Name)	Pattern: 5 * Min: 5 * Max: 5 cy Call: Priority: 1 y Type: Notes: CM_21_ ting Policies	m • 41		Commit) C	Cancel	
Locations Adaptations SIP Entities Entity Links Time Ranges Routing Policies Dial Patterns Regular Expressions Defaults Originating Locations and Route Add Remove I Item Refresh Originating Location Name Originating Location	Pattern: 5 * Min: 5 * Max: 5 cy Call: 1 y Type: Pomain: avaya.co Notes: CM_21_ ting Policies	m ∙ 41				
Addplations General SIP Entities * P. Entity Links * Time Ranges Routing Policies Emergency Dial Patterns Emergency Regular Expressions Emergency Defaults Originating Locations and Roution Add Remove 1 Item Refresh Originating Location Name + Originating Location	Pattern: 5 Min: 5 Max: 5 cy Call: vriority: vrype: Notes: CM_21_ ting Policies	m ▼ 41				
Entity Links Filme Ranges Routing Policies Dial Patterns Regular Expressions Defaults Coriginating Locations and Routi Add Remove I Item Refresh Coriginating Location Name + Or Lo	attern: 5 * Min: 5 * Max: 5 cy Call: 1 priority: 1 y Type: 2000000000000000000000000000000000000	m • 41				
Time Ranges Routing Policies Dial Patterns Regular Expressions Defaults Emergency Picture SIP Defaults Originating Locations and Routing Add Remove 1 Item Refresh Originating Location Name + Originating Location Name + Originating Location Name +	* Min: 5 * Max: 5 cy Call: Priority: 1 y Type: Notes: CM_21 ting Policies]		
Routing Policies Emergency Dal Patterns Emergency Regular Expressions Emergency Defaults Emergency SIP be SIP be Add Remove 1 Item Refresh Originating Location Name + Originating Location Name + Originating Location Name +	* Max: 5 cy Call: Priority: 1 y Type: Notes: CM_21_ ting Policies	m • 41]		
Dial Patterns Emergency Regular Expressions Emergency Defaults Emergency SIP Defaults Originating Locations and Routs Add Remove 1 Item Refresh Originating Location Name + Originating Loc	cy Call:	m ∙ 41				
Regular Expressions Emergency Pi Defaults Emergency Pi Originating Locations and Routi Add Add Remove 1 Item Refresh Originating Location Name + Originating Location Name + Originating Location Name +	Priority: 1 y Type: pomain: avaya.co Notes: CM_21 ting Policies	41				
Deriduits Emergency SIP Dr Originating Locations and Rout Add Remove 1 Item Refresh Originating Location Name + Or Lo	y Type: Domain: avaya.co Notes: CM_21 ting Policies	41				
SIP D: Originating Locations and Rout Add Remove 1 Item Refresh Originating Location Name + Or Lo	Notes: CM_21_	• • • • • • • • • • • • • • • • • • •				
Originating Locations and Rout Add Remove 1 Item Refresh Originating Location Name * Or	Notes: <u>CM_21_</u>	.41				
Originating Location Name - Or						Filter:
- 011 -	originating ocation Notes	Routing Policy Name	Rank F Di	outing Policy isabled	Routing Policy Destination	Routi Policy
Select : All, None Denied Originating Locations						
						Filter
Originating Location					Notes	Theer.
				Commit	Cancel	

Αναγα	Avaya Aura [®] System Manager 6.3	Last Logged on at Novembe Help About Change Passwon
	Home / Elements / Douting / Dial Dattorns	Routin
Routing	Home / Elements / Routing / Dial Patterns	
Domains	Dial Pattern Details	Commit Cancel
Adaptations		
SID Entities	General	
Entity Links	* Pattern: 7	
Time Ranges	* Min: 5	
Routing Policies	* Max: 5	
Dial Patterns	Emergency Call:	
Regular Expressions	Emergency Priority: 1	
Defaults	Emergency Type	
	STP Domain. avaya.com	
	Add Remove 1 Item Refresh Originating Location Name A Originating Location Notes Policy Name	Routing Routing Policy Rank Policy Destination
		Disabled
		contragio
	Select : All, None	
	Denied Originating Locations	
	Add Remove	
	Add Remove	
	Add Remove 0 Items Refresh	

8.	Add Session Manager						
	Adding the Session Manager provides the linkage between System Manager and Session						
	Manager. This configuration procedure should have already been properly executed if						
	the Session Manager used has been set up for other purposes. This configuration step is						
	included here for reference and completeness. To add Session Manager, navitage to						
	Home \rightarrow Elements \rightarrow Session Manager \rightarrow Session Manager Administration. Click						
	New under the "Session Manager Instances" section (not shown), and fill in the fields as						
	described below and shown in the following screen (note that the screen below is for						
	Edit Session Manager since it was already administered):						
	Under Coursel						
	Under General:						
	• SIP Entity Name: Select the name of the SIP Entity created for Session						
	Manager						
	• Description : Any descriptive text						
	Management Access						
	Point Host Name/IP : IP address of the Session Manager management interface.						
	Under Security Module:						
	• Network Mask : Enter the proper network mask for Session Manager.						
	• Default Gateway: Enter the default gateway IP address for Session Manager						
	Accept default settings for the remaining fields.						

ΑνΑγΑ	Avaya	Adia System	nalid	iger 0.5	Help About Change Password	5, 2013 9 Log off
					Session Manager	× Ho
Session Manager	Home / Elements / S	Gession Manager / Se	ssion Ma	nager Administratior	n	н
Dashboard Session Manager	Edit Session N	lanager			Commit Cancel	
Administration	General I Security Modu	le I NIC Bonding I Monitorin	a i CDR i Pe	ersonal Profile Manager (PF	PM) - Connection Settings Event Server	
Communication Profile Editor	Expand All Collapse All			5 (, , ,	
Network Configuration	General 💌					
Device and Location Configuration		SIP Entity Name	SM_21_31			
Application	*Management Acc	ess Point Host Name/IP	10.64.21	.30		
Configuration System Status	*Dire	ct Pouting to Endpoints	Enable	•		
 System Tools 	- Dire	VMware Virtual Machine				
Performance						
	Security Module 🖲)				
		SIP Entity IP Address	10.64.21.3	31		
		*Network Mask	255.255.	.255.0		
		*Default Gateway	10.64.21	1		
		*Call Control PHB	46			
		*QOS Priority	6			
		*Speed & Duplex	Auto	•		
	NIC Bonding 💌					
		Enable	Bonding			
		Driver Monitor	ing Mode	ARP Monitoring		
	ARP Interval (msecs)	100		Link Monitoring Freque	ency (msecs) 100	
	ARP Target IP			Down D	Delay (msecs) 200	
	ARP Target IP			Up D	Delay (msecs) 200	
	Monitoring 💌					
		Enable Monitoring	V			
	*Pro	eactive cycle time (secs)	120			
		*Number of Retries	1			
	CDR 💌		_			
		Enable CDR User	CDR User			
		Password				
		Confirm Password				
	Personal Profile Ma	anager (PPM) - Conne	ction Se	ttings 💌		
	Personal Profile Ma	anager (PPM) - Conne d PPM Client Connection	ection Se	ttings 💌		
	Personal Profile Ma Limiter *Maximum Co	anager (PPM) - Conne d PPM Client Connection nnection per PPM Client	ection Se	ttings •		
	Personal Profile Ma Limiter Maximum Co P	anager (PPM) - Conne I PPM Client Connection nnection per PPM Client PM Packet Rate Limiting Rate Limiting Threshold	ection Se 3 200	ttings ▼]	
	Personal Profile Ma Limiter Maximum Co P *PPM Packet	anager (PPM) - Conne I PPM Client Connection nnection per PPM Client PM Packet Rate Limiting Rate Limiting Threshold	v v 200	ttings *		
	Personal Profile Ma Limiter Maximum Co P *PPM Packet Event Server *	anager (PPM) - Conne I PPM Client Connection nnection per PPM Client PM Packet Rate Limiting Rate Limiting Threshold	In the second se	ttings ∞		
	Personal Profile Mi Limite *Maximum Co P *PPM Packet Event Server * Clear Subscriptio	anager (PPM) - Conne J PPM Client Connection nnection per PPM Client PM Packet Rate Limiting Rate Limiting Threshold n on Notification Failure	ection Se 3 200 No ▼	ttings *		
	Personal Profile Mi Limiter Maximum Co P *PPM Packet Event Server * Clear Subscriptio	anager (PPM) - Conne d PPM Client Connection nnection per PPM Client PM Packet Rate Limiting Rate Limiting Threshold n on Notification Failure	International Sector Se	ttings *	Commit Cancel	
	Personal Profile M: Limiter *Maximum Co P *PPM Packet Event Server * Clear Subscriptio	anager (PPM) - Conne d PPM Client Connection nnection per PPM Client PM Packet Rate Limiting Rate Limiting Threshold n on Notification Failure	In the second se	ttings *	Commit Cancel	

7. Configure Convergys Voice Portal

This section provides steps to configure Convergys Voice Portal. Convergys installs, configures, and customizes the Voice Portal application for end customers. This section describes the initial Voice Portal configuration.

Launch a web browser, enter <u>http://localhost:8070/ccportal/portal</u> in the URL. Log in with the appropriate credentials and click the **Accept** button on the following screen (not shown) to access the **System View** page.

CONVERGYS Outlineing Outling Control Center			A
Welcor	Username Password Erowser Configuration Information		
		100%	T

Select the **Node View** link at the top right.

CONVERGYS Outlinking Outdoing CO	ntrol Center	iviadmin Last Login Time: 8/29/13 2:11:14 Pl Invalid Logon Attempts: About Node View System View Logout Edit Accoun	M 🔺
System View	Preferences Alert		
		Problem Nodes	?
No problems dete	ected.		
8		System View	?
⊞∎		Unassigned	
https://localbost:8074/cc	oortal/portal?homeClicked=true	⊕ 100	
rrcps///ocail03000/4/cq	son tarpor tarmonic clicked = a de		10 11

Click the **Configure** tab on the top left to start configuring Convergys Voice Portal.

CONVERGYS Outlikinking Outdoing	Control Co	enter	АЬ	iviadmin Last Login Tim In Dut <u>Node View</u> <u>System View</u> <u>I</u>	e: 8/29/13 2:11:14 PM valid Logon Attempts: 0 Logout Edit Account
All Servers	Configure	Manage	Advanced Notification Gateway	State Control Engine	
Node View	Monitor	Control			
					All Servers
• 🖲 Problem	n Nodes		Shown: of 0 Sorted by:	Name Sort Order: A	Ascending 💌
• C All Syst	ems Inassigned		Name	Туре	Health
	 Unassigned 	ed			
	View				
•					100%

Expand and navigate to Unassigned \rightarrow vm-ms-153 (10.64.21.153) \rightarrow Media Server 3.5 and above \rightarrow HAL HMP Configuration \rightarrow New Config Description. Click the Edit link next to the New Config Description field.

Control C	enter	Invalid Logen Attem About Node View System View Logout Edit Ac
Configure	Hanage Advanced Hountadon Gateway State Control Engine	
	Configuration O	verview
	Name	Actions
	Global Configurations	
	CCXML-AppServer Mapping	View
Active		Actions
	Configurations	Add
	E.== Unassigned	
	🖻 🚚 vm-ms-153 (10.64.21.153)	Add Product Clone Delete
	🖻 🦲 Media Server 3.5 and above	Add Type Delete
	🕀 🧰 Telephony Core	Add Delete
	E C Supplementary Services	Add Delete
	😑 🦳 HAL HMP Configuration	Add Delete
۰	E Sew Config Description	Edit Copy Delete
	E Call Control Configuration	Add Delete
	CCXML Configuration (*)	Add Delete
	CONTRACT Configuration (*)	Add Delete
	Application Routing (*)	Add Delete
	🗄 🦳 Dynamic Log Configuration (*)	Add Delete
	Description	Add Delete
	🗄 🗒 vm-ms-154 (10.64.21.154)	Add Product Clone Delete
	Submit	Revert

Select **System VOIP Configuration** under the System VOIP Parameters menu on the left. Under **Inbound SIP Header Keys**, add a **User-to-User** key. This will make the contents of the User-to-User header of the SIP INVITE available to Voice Portal applications processing inbound calls.

CONVERGYS outliding Outles	Middmin Last Login Time : Inva About Node View System View Go	1/11/13 3:31:41 PM Id Logon Attempts: 3 Jout Edit Account
HALHMP		
Configurations	10.64.21.153 - System VOIP Configuration (New Config Description)	Back ?
View/Edit Boards DTMF Payload & Fax TDD PDD	Enable Record AGC Enable TCP TCM	
System VOIP Parameters System VOIP Registration System VOIP Configuration Secure SIP & RTP	Type of Service Field	
	Outbound Proxy Address	
	SIP Contact Header IP 10.64 21.163	
	Enable SIP 100 Trying Response Enable Inbound Call Analysis	
	Provisional Response Code	
	Inbound SIP Header Keys	Actions
	To	Delete
	From	Delete
	User-to-User	Delete
		Add
	SIP Security KeyInformation	
	Submit Revert	

Select **DTMF Payload and Fax** under the Configurations menu on the left. Use the **DTMF Detect Scheme** drop down menu to select the appropriate setting (*RFC2833 INBAND* is shown in the example below).

CONVERSIS Outling Outrol Center	ividemin Last Login Time: 1/11/13 3/3141 PM Invalid Login Attempts: 3 About Rede View <mark>System View</mark> Logint Edit Account	
Configurations View/Edit Boards	10.64.21.153 - DTMF Payload & Fax (New Config Description)	Back ?
DTMF Payload & Fax TDD PDD System VOIP Parameters System VOIP Configuration System VOIP Configuration Secure SIP & RTP	DTMF Detect Scheme RFC2833_INBAND DTMF Payload 101 Fax Detect Scheme ReimiteT38 Fax Detect Duration(ms) 150	
	Submit Revert	

Select **View/Edit Boards** under the Configurations menu on the left. Select the **Edit** link for the board.

CONVERSIS Outline Control Center			Iviadmin Last Login Timei 11/11/13 3:3 Invalid Logon Atte About Node View System View Logout Edit A	1:41 PM empts: 3 Account
Configurations View/Edit Boards	10.64.21.153 - View	/Edit Boards (New Config Description	n)	Back ?
DTMF Payload & Fax	Board ID	Act	tions	
TDD	0	r da	Delete	
PDD	0	Eult	Delete	
System VOIP Parameters			Add Board	
System VOIP Registration				
System VOIP Configuration		Submit Revert		
Secure SIP & RTP				

Set the **IP** Address for the board and use the drop down menu to select *SIP* for the **Protocol Name**.

CONVERSIS Control Center HMP Board	iviadmin Last Login Time: 11/11/13 3/33 FM Invide Login Attanpts: 3 About Node View System View Logiont Edit Account	
Configurations	10.64.21.153 - Edit Board - 0	Back ?
Board		
Board VOIP Registration		
Codec	Board ID 0	
Alarm		
	IP Address 10.64.21.153	
	Protocol Name SIP	
	Submit Revert	

Select Codec under the Configurations menu on the left. Click the Add link.

CONVERGIS Control Center HMP Board				iviadmin Last Login Tim I About Node View <mark>System View</mark>	H 11/11/13 3:31:41 PM tvalid Logon Attempts: 3 Logout Edit Account
Configurations		10.64.21.1	53 - Codec - Board O		Back ?
Board VOIP Registration			Codec Information		
Codec	Codec Family	Туре	Frame Size	Frames per Packet	Actions
Alarm	G711Codecs	G711M	30	1	<u>Delete</u>
	G729Codecs	G729-ANNEX-A-B	10	2	Delete
		·			Add
			Submit Revert		

Use the drop down menus to select the appropriate settings for **Codec Family**, **Type**, **Frame Size**, and **Frames per Packet**.

CONVERGYS Outling Outling Outling Outling Outling				iviadmin Last Login Tim T <u>About Node View</u> <u>System View</u>	a: 11/11/13 3:31:41 PM nvalid Logon Attempts: 3 Logout Edit Account
Configurations Board		10.64.21.1	53 - Codec - Board O		Back ?
Board VOIP Registration			Codec Information		
Codec	Codec Family	Туре	Frame Size	Frames per Packet	Actions
Alam	G711Codecs	G711M	30	1	<u>Delete</u>
	G729Codecs	G729-ANNEX-A-B	10	2	<u>Delete</u>
	×	V	×	×	<u>Delete</u>
					Add
		I	Submit Revert		

This completes the administration of the Convergys Voice Portal Server.

8. Verification Steps

The following steps may be used to verify the configuration:

- End-to-end verification: Place a call to the Convergys Voice Portal server. Verify the call is answered and voice prompts are played. Verify the SIP messages using a network protocol analyzer.
- DTMF Tones: Place a call to the Convergys Voice Portal server and select the appropriate prompt to enter DTMF tones. Verify Convergys Voice Portal properly identified each DTMF tone.
- Transfer: Place a call to Convergys Voice Portal and select the appropriate prompt to have the call transferred to an Agent. Verify the call is delivered to an Agent and answer the call. Verify there is a two-way talk path.

9. Conclusion

These Application Notes describe the procedures required to configure Convergys Voice Portal to interoperate with an Avaya SIP infrastructure (Communication Manager and Session Manager). Convergys Voice Portal successfully passed compliance testing.

10. Additional References

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

- [1] Avaya Aura® Communication Manager Feature Description and Implementation, Document 555-245-205, Issue 11, Release 6.3, October 2013.
- [2] *Administering Avaya Aura*® *Communication Manager*, Document 03-300509, Issue 9, Release 6.3, October 2013.
- [3] Avaya Aura® Communication Manager Screen Reference, Document 03-602878, Issue 5, October 2013.
- [4] Deploying Avaya Aura® Session Manager, Issue 1, Release 6.3, October 2013.
- [5] Administering Avaya Aura® Session Manager, Document 03-603324, Issue 3, Release 6.3, October 2013.
- [6] *Maintaining and Troubleshooting Avaya Aura*® *Session Manager*, Document 03-603325, Issue 3, Release 6.3, October 2013.

The following documents were provided by Convergys:

- [7] Media Server 4.0 (VoIP) Installation Guide, Document Number 60001490
- [8] Media Server VoiceXML Browser Technical Reference, Document Number 60001390

©2014 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by \mathbb{R} and TM are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at <u>devconnect@avaya.com</u>.