



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Centurion CARES 14.03 with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1 – Issue 1.0**

### **Abstract**

These Application Notes describe the configuration steps required for the Centurion CARES 14.03 to interoperate with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. Centurion CARES is a contact center solution.

In the compliance testing, Centurion CARES used the SIP trunk interface from Avaya Aura® Session Manager to provide IVR and ACD capabilities.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any 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

These Application Notes describe the configuration steps required for the Centurion CARES 14.03 to interoperate with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. CARES is a contact center solution.

In the compliance testing, CARES used the SIP trunk interface from Session Manager to provide IVR and ACD capabilities.

Agents are administered as station users on Communication Manager and have desktops running the Centurion CARES Client application. The CARES Client application is used by agents to log into CARES, to set proper work modes, and to control subsequent calls.

Inbound ACD calls are routed by Communication Manager and Session Manager over the SIP trunk to CARES, with CARES providing relevant IVR call treatment such as greeting announcement and supporting PSTN callers' use of DTMF digits to navigate IVR menu.

Upon requested by PSTN caller to connect with an agent, CARES determines an available agent for the call, establishes a dedicated audio connection over the SIP trunk with the agent telephone when necessary, and bridges the audio path of the agent connection with the customer connection at CARES. Agents are required to use their desktops to perform all subsequent call controls and ACD related activities. The dedicated audio connections with agents stay in place until agents log out of CARES. All SIP communications on CARES are supported using the Dialogic PowerMedia Host Media Processing (HMP) SIP stack.

CARES also support outbound promotional campaign calls, with outbound calls launched by CARES and with call classifications supported by Dialogic PowerMedia HMP.

## 2. General Test Approach and Test Results

The feature test cases were performed manually. Incoming calls were made from the PSTN to CARES. DTMF input were manually input from callers for proper IVR navigation and menu selection, including request to connect with an agent. Manual call controls from the CARES Client application were exercised to verify features such as answering and transferring of calls.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet connection to CARES.

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.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interfaces between Avaya and Centurion did not include use of any specific encryption features as requested by Centurion.

### 2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included OPTIONS, DTMF, G.711MU, media shuffling, session refresh, ANI, dialing ahead, agent work modes, screen pop, hold/resume, music on hold, mute/unmute, blind/supervised transfer, supervised conference, multiple agents, queuing, internal call, long duration, and outbound campaign calls.

The serviceability testing focused on verifying the ability of CARES to recover from adverse conditions, such as disconnecting and reconnecting the Ethernet connection to CARES.

## 2.2. Test Results

All test cases were executed, and the following were observations on CARES:

- Depending on the customer network, a delay may need to be added to the beginning of the inbound call flow for PSTN callers to hear the entire greeting announcement. See **Section 7.6** for more details.
- By design, the PSTN calling party number does not populate at the conference-to agent desktop.
- In the serviceability scenario, for an agent that had a call that was dropped during an Ethernet outage to the CARES server, the desktop can no longer be used to answer subsequent calls post server recovery and with agent work state toggled between Idle – Pre-work and Pending ACD Call. When this occurs, the workaround is to manually drop the dedicated connection with CARES from the agent telephone, exit from the CARES Client application and log back in.

## 2.3. Support

Technical support on CARES can be obtained through the following:

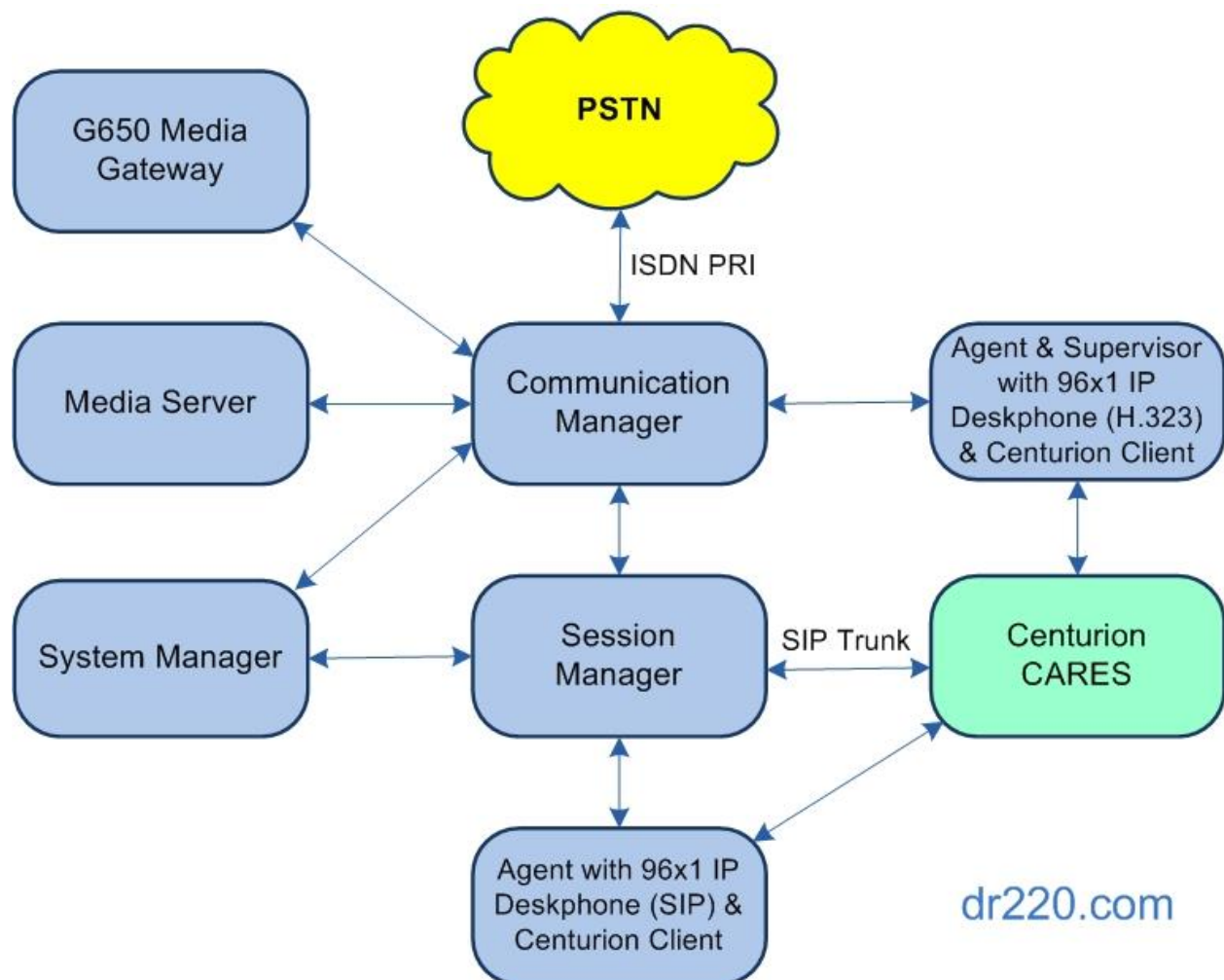
- **Phone:** + 1 (262) 317-5678
- **Email:** [support@centurioncares.com](mailto:support@centurioncares.com)

### 3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**. SIP trunk was used between Session Manager and CARES and the applicable domain name was “dr220.com”.

A 5-digits Uniform Dial Plan (UDP) was used to facilitate routing with CARES. Unique extension ranges were assigned to stations users on Communication Manager (6xxxx) and to the CARES main number (54000).

The configuration of Session Manager is performed via the web interface of System Manager. The detailed administration of basic connectivity between Communication Manager, System Manager, and Session Manager is not the focus of these Application Notes and will not be described.



**Figure 1: Compliance Testing Configuration**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager in Virtual Environment	8.1 (8.1.0.1.1.890.25517)
Avaya G650 Media Gateway	NA
Avaya Aura® Media Server in Virtual Environment	8.0.1.121
Avaya Aura® Session Manager in Virtual Environment	8.1 (8.1.0.0.810007)
Avaya Aura® System Manager in Virtual Environment	8.1 (8.1.0.0.079814)
Avaya 9611G & 9641G IP Deskphone (H.323)	6.8202
Avaya 9641G IP Deskphone (SIP)	7.1.6.1.3
Centurion CARES on Microsoft Windows Server 2016 <ul style="list-style-type: none"><li>• Base Package</li><li>• CARES Server</li><li>• Dialogic PowerMedia HMP</li></ul>	14.03 Standard 14.03.279 14.03.1720 3.0.395
Centurion CARES Client on Microsoft Windows 10	14.03.1739 Pro

## 5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify license
- Administer node names
- Administer codec set
- Administer network region
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- Administer route pattern
- Administer private numbering
- Administer uniform dial plan
- Administer AAR analysis
- Administer PSTN trunk group
- Administer tandem calling party number

In the compliance testing, the Avaya endpoints used encrypted signaling connections with encrypted media. A separate set of codecs set, network region and network region map were created for integration with CARES.

### 5.1. Verify License

Log into the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the “display system-parameters customer-options” command. Navigate to **Page 2** and verify that there is sufficient capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

display system-parameters customer-options		Page 2 of 12
OPTIONAL FEATURES		
IP PORT CAPACITIES	USED	
Maximum Administered H.323 Trunks: 12000	10	
Maximum Concurrently Registered IP Stations: 18000	2	
Maximum Administered Remote Office Trunks: 12000	0	
Max Concurrently Registered Remote Office Stations: 18000	0	
Maximum Concurrently Registered IP eCons: 414	0	
Max Concur Reg Unauthenticated H.323 Stations: 100	0	
Maximum Video Capable Stations: 41000	0	
Maximum Video Capable IP Softphones: 18000	0	
<b>Maximum Administered SIP Trunks: 40000</b>	<b>40</b>	
Max Administered Ad-hoc Video Conferencing Ports: 24000	0	
Max Number of DS1 Boards with Echo Cancellation: 999	0	

## 5.2. Administer Node Names

Use the “change node-names ip” command. Note the **Name** and **IP Address** of the processor or existing C-LAN circuit pack that will be used for connectivity to CARES, in this case “procr” and “10.64.101.236”.

Also note the **Name** and **IP Address** of the Session Manager signaling interface, in this case “sm7-sig” and “10.64.101.238”.

change node-names ip		Page	1 of	2
		IP NODE NAMES		
Name	IP Address			
G430	192.168.200.43			
aes7	10.64.101.239			
clan	10.64.125.32			
default	0.0.0.0			
gateway	10.64.125.1			
medpro	10.64.125.33			
ms7	10.64.101.233			
procr	10.64.101.236			
procr6	::			
sm7-sig	10.64.101.238			

## 5.3. Administer Codec Set

Administer a codec set for integration with CARES. Use the “change ip-codec-set n” command, where “n” is an existing codec set number to use for interoperability.

For **Audio Codec**, enter the pertinent G.711 variant as shown below. Note that G.711 is the only codec type supported by CARES. For **Media Encryption** and **Encrypted SRTCP**, retain the default values of “none” and “enforce-unenc-srtcp” as shown below. Retain the default values for the remaining fields.

change ip-codec-set 4

Page1 of 2

IP MEDIA PARAMETERS

Codec Set: 4

Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1: G.711MU	n	2	20
2:			
3:			
4:			
5:			
6:			
7:			

Media Encryption

Encrypted SRTCP: enforce-unenc-srtcp

1: none



## 5.4. Administer Network Region

Administer a network region for integration with CARES. Use the “change ip-network-region n” command, where “n” is an existing network region number to use for interoperability.

Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Authoritative Domain:** The SIP domain from **Section 3**.
- **Name:** A descriptive name.
- **Codec Set:** The codec set number from **Section 5.3**.

```
change ip-network-region 4                                     Page 1 of 20
IP NETWORK REGION
Region: 4              NR Group: 4
Location:              Authoritative Domain: dr220.com
Name: CARES           Stub Network Region: n
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
Codec Set: 4          Inter-region IP-IP Direct Audio: yes
                      IP Audio Hairpinning? n
UDP Port Min: 2048
UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
```

Navigate to **Page 4**, and specify the same codec set number to be used for calls with network regions used by Avaya endpoints and by the trunk with the PSTN. In the compliance testing, network region “1” was used by the Avaya endpoints and by the trunk with the PSTN.

```
change ip-network-region 4                                     Page 4 of 20

Source Region: 4      Inter Network Region Connection Management  I      M
                                                                G  A  t
dst codec direct  WAN-BW-limits  Video      Intervening  Dyn  A  G  c
rgn  set  WAN  Units  Total Norm  Prio Shr  Regions  CAC  R  L  e
1    4    y    NoLimit
2
3
4    4
5
6
7
8
```

## 5.5. Administer SIP Trunk Group

Use the “add trunk-group n” command, where “n” is an available trunk group number, in this case “54”. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** “tie”

<b>add trunk-group 54</b>		Page 1 of 4	
TRUNK GROUP			
Group Number: 54	<b>Group Type: sip</b>	CDR Reports: y	
<b>Group Name: SIP trunk to CARES</b>	COR: 1	TN: 1	<b>TAC: 1054</b>
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
<b>Service Type: tie</b>	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group:		
	Number of Members: 0		

Navigate to **Page 3**, and enter “private” for **Numbering Format**.

<b>add trunk-group 54</b>		Page 3 of 4	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Suppress # Outpulsing? n	<b>Numbering Format: private</b>	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			

## 5.6. Administer SIP Signaling Group

Use the “add signaling-group n” command, where “n” is an available signaling group number, in this case “54”. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Transport Method:** “tls”
- **Near-end Node Name:** The processor node name from **Section 5.2**.
- **Far-end Node Name:** The Session Manager node name from **Section 5.2**.
- **Near-end Listen Port:** An available port for integration with CARES.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** The network region number from **Section 5.4**.
- **Far-end Domain:** The domain name from **Section 3**.

add signaling-group 54		Page 1 of 2
SIGNALING GROUP		
Group Number: 54	<b>Group Type: sip</b>	
IMS Enabled? n	<b>Transport Method: tls</b>	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: Others	Clustered? n
Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? n		
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? y		
Alert Incoming SIP Crisis Calls? n		
<b>Near-end Node Name: procr</b>		<b>Far-end Node Name: sm7-sig</b>
<b>Near-end Listen Port: 5054</b>		<b>Far-end Listen Port: 5054</b>
		<b>Far-end Network Region: 4</b>
<b>Far-end Domain: dr220.com</b>		
Incoming Dialog Loopbacks: eliminate		Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload		RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3		Direct IP-IP Audio Connections? y
Enable Layer 3 Test? y		IP Audio Hairpinning? n
H.323 Station Outgoing Direct Media? n		Initial IP-IP Direct Media? n
		Alternate Route Timer(sec): 6

## 5.7. Administer SIP Trunk Group Members

Use the “change trunk-group n” command, where “n” is the trunk group number from **Section 5.5**. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.6**.
- **Number of Members:** The desired number of members, in this case “10”.

change trunk-group 54		Page 1 of 4	
TRUNK GROUP			
Group Number: 54	Group Type: sip	CDR Reports: y	
Group Name: SIP Trunk to CARES	COR: 1	TN: 1	TAC: 1054
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: 54			
Number of Members: 10			

## 5.8. Administer Route Pattern

Use the “change route-pattern n” command, where “n” is an existing route pattern number to be used for integration with CARES, in this case “54”. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.5**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.
- **Numbering Format:** “lev0-pvt”

change route-pattern 54		Page 1 of 3	
Pattern Number: 54		Pattern Name: CARES	
SCCAN? n	Secure SIP? N	Used for SIP stations? n	
Grp FRL NPA Pfx Hop Toll No. Inserted		DCS/ IXC	
No Mrk Lmt List Del Digits		QSIG	
Dgts		Intw	
1: 54	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	
1: y y y y y n	n	rest	lev0-pvt none
2: y y y y y n	n	rest	none

## 5.9. Administer Private Numbering

Use the “change private-numbering 0” command, to define the calling party number to send to CARES. Add an entry for the trunk group defined in **Section 5.5**.

In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 54 will result in a 5-digit calling number. The calling party number will be in the SIP “From” header.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len	
5	2			5	Total Administered: 3
5	5			5	Maximum Entries: 540
5	6	54		5	

## 5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 54000 to CARES. Note that other routing methods may be used. Use the “change uniform-dialplan 0” command and add an entry to specify the use of AAR for routing of digits 54000, as shown below.

change uniform-dialplan 0					Page 1 of 2
UNIFORM DIAL PLAN TABLE					
					Percent Full: 0
Matching Pattern	Len	Del	Insert Digits	Node Net Conv Num	
54000	5	0	aar	n	

## 5.11. Administer AAR Analysis

Use the “change aar analysis 0” command and add an entry to specify how to route calls to 54000. In the example shown below, calls with digits 54000 will be routed as an AAR call using route pattern “54” from **Section 5.8**.

change aar analysis 0					Page 1 of 2
AAR DIGIT ANALYSIS TABLE					
Location: all					Percent Full: 2
Dialed String	Total Min	Total Max	Route Pattern	Call Type	Node Num ANI Req'd
54000	5	5	54	aar	n

## 5.12. Administer PSTN Trunk Group

Use the “change trunk-group n” command, where “n” is the existing trunk group number used to reach the PSTN, in this case “13”. Navigate to **Page 3**.

For **Modify Tandem Calling Number**, enter “tandem-cpn-form” to allow modification of calling party number for calls to the PSTN.

change trunk-group 13			Page	3 of 21
TRUNK FEATURES				
ACA Assignment? n	Measured: none	Wideband Support? n		
		Maintenance Tests? y		
	Data Restriction? n	NCA-TSC Trunk Member: 1		
	Send Name: y	Send Calling Number: y		
Used for DCS? n		Send EMU Visitor CPN? n		
Suppress # Outpulsing? n	Format: natl-pub			
Outgoing Channel ID Encoding: preferred	UII IE Treatment: service-provider			
		Replace Restricted Numbers? n		
		Replace Unavailable Numbers? n		
		Send Connected Number: y		
Network Call Redirection: none		Hold/Unhold Notifications? n		
Send UII IE? y	<b>Modify Tandem Calling Number: tandem-cpn-form</b>			
Send UCID? n				
Send Codeset 6/7 LAI IE? y		Dsl Echo Cancellation? n		
Apply Local Ringback? n	US NI Delayed Calling Name Update? n			
Show ANSWERED BY on Display? y	Invoke ID for USNI Calling Name: variable			

## 5.13. Administer Tandem Calling Party Number

Use the “change tandem-calling-party-num” command, to define the calling party number to send to the PSTN for tandem calls from CARES.

By default, CARES sends the agent’s internal extension on CARES as calling party number for manual outbound calls placed by the agent, and sends a blank calling party number for outbound campaign calls launched by CARES.

In the example shown below, all tandem calls to the PSTN will have all calling party number digits deleted and replaced with “3035354000”, which takes care of the blank and 4-digits calling party number (4xxx) sent by CARES. Note that alternatively CARES can be configured to send a specific calling party number for outbound campaign calls, in which case the calling party number modification only needs to be for the CARES internal agent extensions.

change tandem-calling-party-num						Page	1 of 67
CALLING PARTY NUMBER CONVERSION FOR TANDEM CALLS							
		Incoming	Outgoing				
		Number	Trunk				
		Format	Group(s)	Delete	Insert	Outgoing	
Len	CPN					Number	
any	any		13	all	3035354000	Format	
						pub-unk	

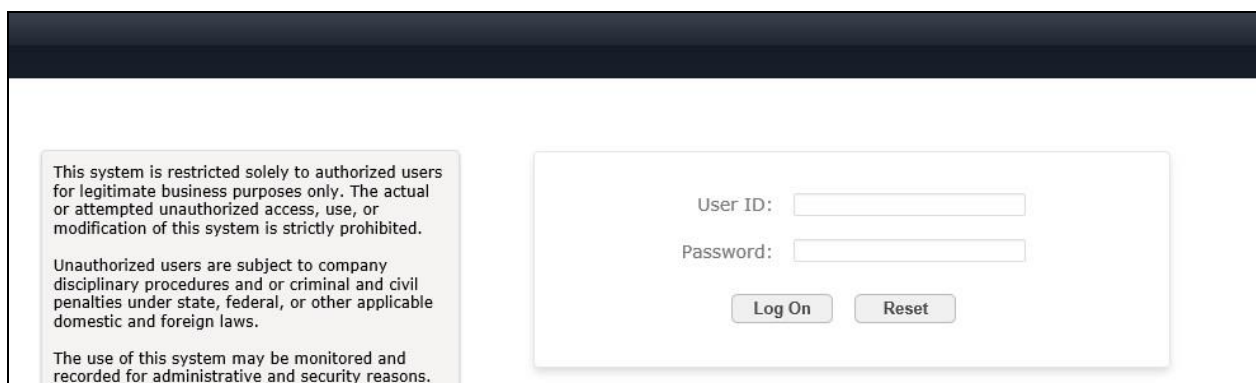
## 6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager, which is performed via the web interface of System Manager. The procedures include the following areas:

- Launch System Manager
- Administer locations
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

### 6.1. Launch System Manager

Access the System Manager web interface by using the URL <https://ip-address> in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

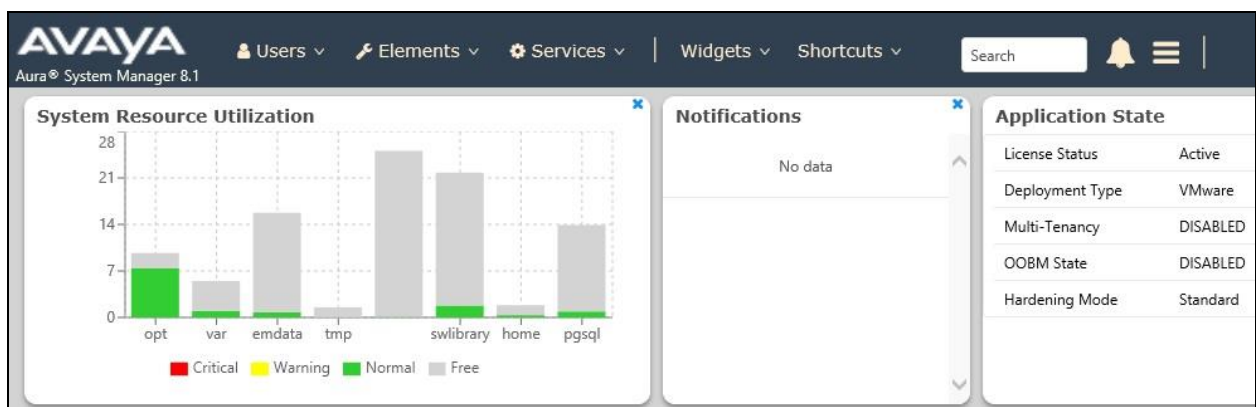
Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons.

User ID:

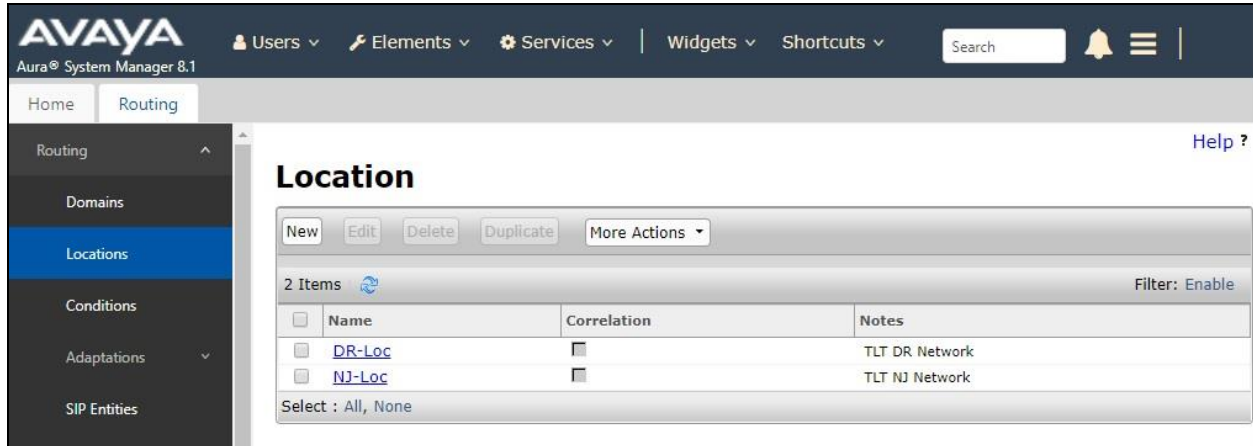
Password:

The screen below is displayed next.

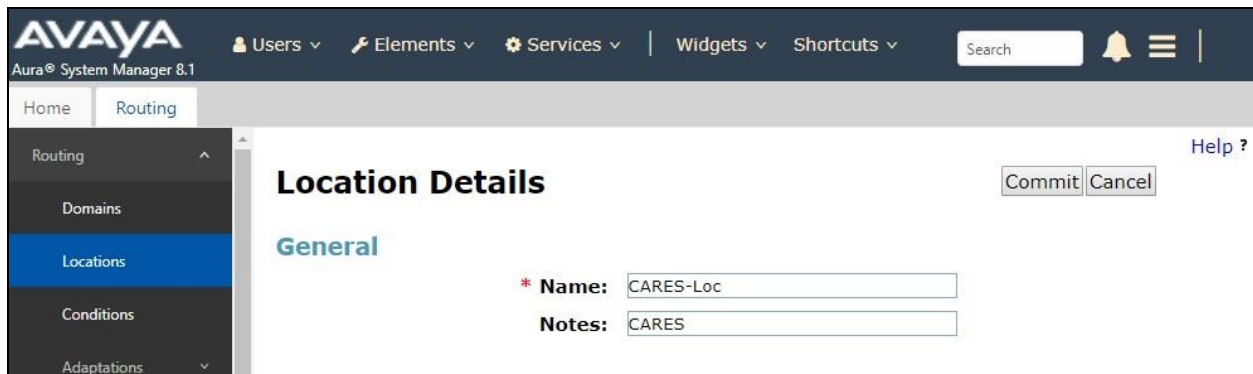


## 6.2. Administer Locations

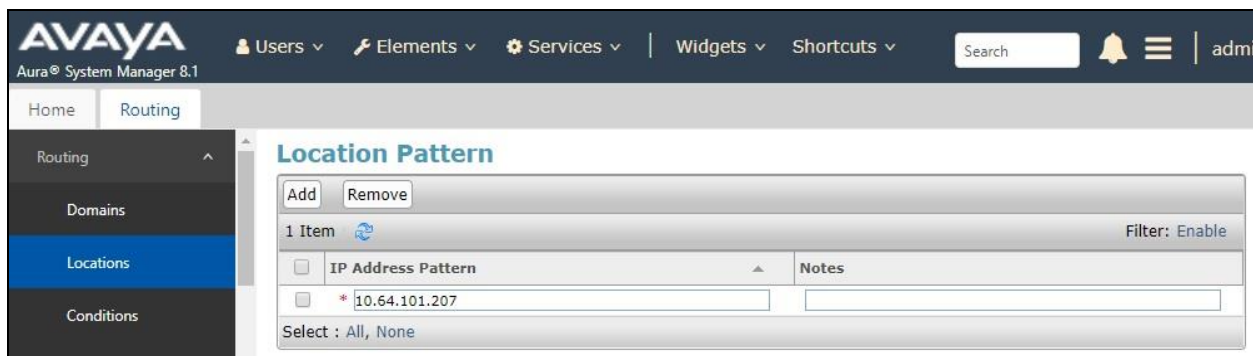
Select **Elements** → **Routing** → **Locations** from the top menu to display the **Location** screen below. Select **New** to add a new location for CARES.



The **Location Details** screen is displayed next. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**.



Scroll down to the **Location Pattern** sub-section and click **Add**. For **IP Address Pattern**, enter the IP address of CARES as shown below. Retain the default values in the remaining fields.





## 6.3. Administer SIP Entities

Add two SIP entities, one for CARES and one for the new SIP trunk with Communication Manager.

### 6.3.1. SIP Entity for CARES

Select **Routing** → **SIP Entities** from the left menu and click **New** in the subsequent screen (not shown) to add a new SIP entity for CARES.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of CARES.
- **Type:** “SIP Trunk”
- **Location:** Select the CARES location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

The screenshot shows the AVAYA Aura System Manager 8.1 interface. The top navigation bar includes 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. The left sidebar shows the 'Routing' menu with 'SIP Entities' selected. The main content area displays the 'SIP Entity Details' form. The 'General' tab is active, showing fields for Name, FQDN or IP Address, Type, Notes, Adaptation, Location, Time Zone, SIP Timer B/F, Minimum TLS Version, Credential name, Securable, and Call Detail Recording. The 'Loop Detection' tab is visible at the bottom.

Field	Value
Name	CARES
FQDN or IP Address	10.64.101.207
Type	SIP Trunk
Notes	
Adaptation	
Location	CARES-Loc
Time Zone	America/New_York
SIP Timer B/F (in seconds)	4
Minimum TLS Version	Use Global Setting
Credential name	
Securable	<input type="checkbox"/>
Call Detail Recording	egress

Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DR-SM”.
- **Protocol:** “UDP”
- **Port:** “5060”
- **SIP Entity 2:** The CARES entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

Note that CARES only support UDP in the current release.

### Entity Links

Override Port & Transport with DNS SRV: ☐

Add Remove

1 Item
Filter: Enable

	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	* SM-CARES	DR-SM	UDP	* 5060	CARES	* 5060	trusted

Select : All, None

### SIP Responses to an OPTIONS Request

Add Remove

0 Items
Filter: Enable

	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--	-------------------------------	---------------------	-------

Commit Cancel

### 6.3.2. SIP Entity for Communication Manager

Select **Routing** → **SIP Entities** from the left menu and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Manager. Note that this SIP entity is used for integration with CARES.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The processor IP address from **Section 5.2**.
- **Type:** “CM”
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.

**AVAYA** Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾ Search 🔍 🔔 ☰

Home Routing

Routing ^

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

**SIP Entity Details** Commit Cancel Help ?

**General**

\* **Name:** DR-CM-5054

\* **FQDN or IP Address:** 10.64.101.236

**Type:** CM ▾

**Notes:**

**Adaptation:** ▾

**Location:** DR-Loc ▾

**Time Zone:** America/New\_York ▾

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

**Minimum TLS Version:** Use Global Setting ▾

**Credential name:**

**Securable:** ☐

**Call Detail Recording:** none ▾

**Loop Detection**

Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DR-SM”.
- **Protocol:** The signaling group transport method from **Section 5.6**.
- **Port:** The signaling group far-end listen port number from **Section 5.6**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group near-end listen port number from **Section 5.6**.
- **Connection Policy:** “trusted”

### Entity Links

Override Port & Transport with DNS ☐ SRV: ☐

Add Remove

1 Item
Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy
<input type="checkbox"/>	* SM-CM-5054	DR-SM	TLS	* 5054	DR-CM-5054	* 5054	trusted

Select : All, None

### SIP Responses to an OPTIONS Request

Add Remove

0 Items
Filter: Enable

<input type="checkbox"/>	Response Code & Reason Phrase	Mark Entity Up/Down	Notes
--------------------------	-------------------------------	---------------------	-------

Commit Cancel

Add two routing policies, one for CARES and one for the new SIP trunk with Communication Manager.

Select **Routing** → **Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for CARES. The **Routing Policy Details** screen is displayed.

In the **SIP Entity as Destination** sub-section, click **Select** and select the CARES entity name from **Section 6.3.1**. The screen below shows the result of the selection.

TLT; Reviewed: SPOC 1/2/2020	Solution & Interoperability Test Lab Application Notes ©2019 Avava Inc. All Rights Reserved.	21 of 37 CARES-SM81
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## 6.4.2. Routing Policy for Communication Manager

Select **Routing** → **Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Manager. The **Routing Policy Details** screen is displayed.

In the **General** sub-section, enter a descriptive **Name**. Enter optional **Notes** and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Manager entity name from **Section 6.3.2**. The screen below shows the result of the selection.

**Routing Policy Details**

**General**

\* **Name:** To-CM-5054

**Disabled:** ☐

\* **Retries:** 0

**Notes:**

**SIP Entity as Destination**

Select

Name	FQDN or IP Address	Type	Notes
DR-CM-5054	10.64.101.236	CM	

## 6.5. Administer Dial Patterns

Add a new dial pattern for CARES and update existing dial patterns for Communication Manager to allow calls from CARES.

### 6.5.1. Dial Pattern for CARES

Select **Routing → Dial Patterns** from the left pane and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach CARES. The **Dial Pattern Details** screen is displayed.

In the **General** sub-section, enter the following values for the specified fields and retain the default values for the remaining fields.

- **Pattern:** The CARES main number from **Section 3**.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching CARES. In the compliance testing, the entry allowed for call origination from Communication Manager resources in location “Lab”. The CARES routing policy from **Section 6.4.1** was selected as shown below.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ Widgets ▾ Shortcuts ▾ Search 🔍 🔔 ☰

Home Routing

Routing  
Domains  
Locations  
Conditions  
Adaptations ▾  
SIP Entities  
Entity Links  
Time Ranges  
Routing Policies  
Dial Patterns  
Dial Patterns  
Origination Dial...

**Dial Pattern Details** Commit Cancel Help ?

**General**

\* Pattern: 54000

\* Min: 5

\* Max: 5

Emergency Call: ☐

SIP Domain: -ALL- ▾

Notes:

**Originating Locations and Routing Policies**

Add Remove

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"/>	DR-Loc	TLT DR Network	To-CARES	0	<input type="checkbox"/>	CARES	

Select : All, None

## 6.5.2. Dial Pattern for Communication Manager

Select **Routing** → **Dial Patterns** from the left pane and click on the applicable dial pattern for Communication Manager in the subsequent screen, in this case dial pattern “6” (not shown). The **Dial Pattern Details** screen is displayed.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new entry as necessary for calls from CARES. In the compliance testing, the new entry allowed for call origination from the CARES location from **Section 6.2** and the Communication Manager routing policy from **Section 6.4.2** were selected as shown below. Retain the default values in the remaining fields.

Repeat this section to make similar changes to applicable Communication Manager dial pattern to reach the PSTN. In the compliance testing, CARES will add the prefix “91” for outbound campaign calls to the PSTN, and therefore the existing dial pattern for “91” was also changed (not shown below).

**AVAYA**  
Aura® System Manager 7.1

Last Logged on at November 12, 2019 2:48 PM  
Go... Log off

Home Routing x

Home / Elements / Routing / Dial Patterns

### Dial Pattern Details

Commit Cancel

#### General

\* Pattern: 6

\* Min: 5

\* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: -ALL-

Notes: To CM7

#### Originating Locations and Routing Policies

Add Remove

2 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
<input type="checkbox"/>	CARES-Loc	CARES	To-CM-5054	0	<input type="checkbox"/>	DR-CM-5054	
<input type="checkbox"/>	Lab		cm15014	0	<input type="checkbox"/>	cm15014	

Select : All, None



## 7. Configure Centurion CARES

This section provides the procedures for configuring CARES. The procedures include the following areas:

- Launch web interface
- Administer trunk
- Administer trunk node
- Administer dial plan
- Administer outbound campaign parameter
- Administer inbound call flow


The detailed administration of the Dialogic resource and of contact center resources such as splits, skills, agents, and outbound campaigns are assumed to be in place and are not covered in these Application Notes.

The configuration of CARES is typically performed by the Centurion implementation team, and the procedural steps are presented in these Application Notes for informational purposes.

### 7.1. Launch Web Interface

Access the CARES web interface by using the URL <https://ip-address:8443> in an Internet browser window, where “ip-address” is the IP address of the CARES server.

The **Quantum SignPost** screen is display. Log in using the appropriate credentials.



The screenshot shows the Quantum SignPost web interface. At the top, there is a dark blue header with a yellow logo on the left, the text "Quantum SignPost" in the center, and "Logged out Log In" on the right. Below the header is a light gray main area. In the center of the main area is a white login box with a blue header labeled "Login". Inside the login box, there are two input fields: "Username:" and "Password:". Below these fields is a "Login" button. At the bottom of the page, there is a footer with a small logo on the left, the text "Voice Platform Administration powered by Centurion Quantum SignPost (14.3.226.RELEASE-12) Copyright 2013-2018, Centurion CARES, Inc." in the center, and the word "Main" on the right.

## 7.2. Administer Trunk

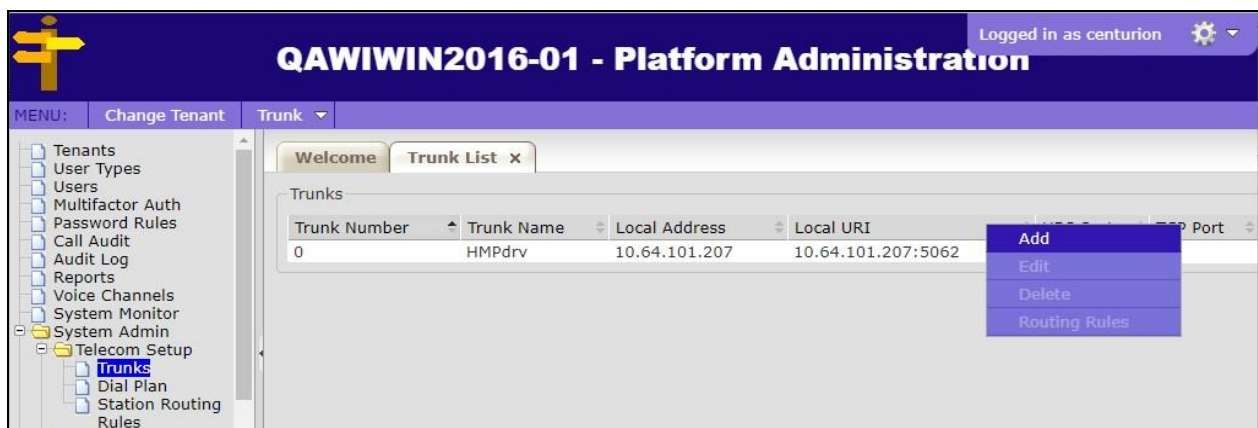
The screen below is displayed. Select **Platform Administration**.



The **Welcome** tab is displayed in the right pane, as shown below.



Select **System Admin** → **Telecom Setup** → **Trunks** in the left pane to display the **Trunk List** tab in the right pane. A list of existing trunks is displayed. Right click in the Trunks area and select **Add** to add a new trunk.



The **Add Trunk** tab is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Trunk Number:** The next available trunk number, in this case “1”.
- **Trunk Name:** A descriptive trunk name.
- **Local Address:** IP address of the CARES server.
- **Local URI:** IP address of the CARES server and protocol port from **Section 6.3.1**.
- **UDP Port:** UDP protocol port from **Section 6.3.1**.

QAWIWIN2016-01 - Platform Administration

Logged in as centurion

MENU: Change Tenant Node

Trunk

Trunk Number: 1

Trunk Name: Avaya

Local Address: 10.64.101.207

Local URI: 10.64.101.207:5060

UDP Port: 5060

TCP Port:

TLS Port:

☐ SRTP

Save Cancel

### 7.3. Administer Trunk Node

The **Trunk List** tab is displayed again. Right click on the newly added trunk from **Section 7.2** and select **Edit**.

QAWIWIN2016-01 - Platform Administration

Logged in as centurion

MENU: Change Tenant Trunk

Trunks

Trunk Number	Trunk Name	Local Address	Local URI	UDP Port	TCP Port
0	HMPdrv	10.64.101.207	10.64.101.207:5062	5062	0
1	Avaya	10.64.101.207	10.64.101.207:5060	5060	0

Context Menu:

- Add
- Edit
- Delete
- Routing Rules

The **Edit Trunk** tab is displayed (content not shown below). Right click in the **Trunk Nodes** area and select **Add** (not shown) to display the **Add Node** tab. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Node Number:** “0”
- **Node Name:** A descriptive node name.
- **Address:** IP address of Session Manager signaling interface from **Section 5.2**.
- **Port:** The transport port number from **Section 6.3.1**.
- **URI:** IP address of Session Manager signaling interface and protocol port.

The screenshot shows the 'Add Node' tab in the 'Edit Trunk 1' window. The left sidebar shows the navigation tree with 'System Admin' > 'Telecom Setup' > 'Trunks' selected. The main area contains the following fields:

Trunk Number	1	Node Number	0
Node Name	AvayaNode	Address	10.64.101.238
Port	5060	URI	10.64.101.238:5060
User Id		Password	
Realm		Auth Id	

Buttons: Save, Cancel

## 7.4. Administer Dial Plan

Select **System Admin** → **Telecom Setup** → **Dial Plan** from the left pane to display the **Dial Plan** tab. Create entries for inbound calls to the CARES main number 54000, outbound calls to agent station extensions 6xxxx on Communication Manager, and outbound calls to the PSTN with applicable network prefix. Three dial plan entries were created in the compliance testing as shown below.

The screenshot shows the 'Dial Plan' tab in the 'Edit Trunk 1' window. The left sidebar shows the navigation tree with 'System Admin' > 'Telecom Setup' > 'Dial Plan' selected. The main area contains the following search and results section:

**SipDialPlan Search**

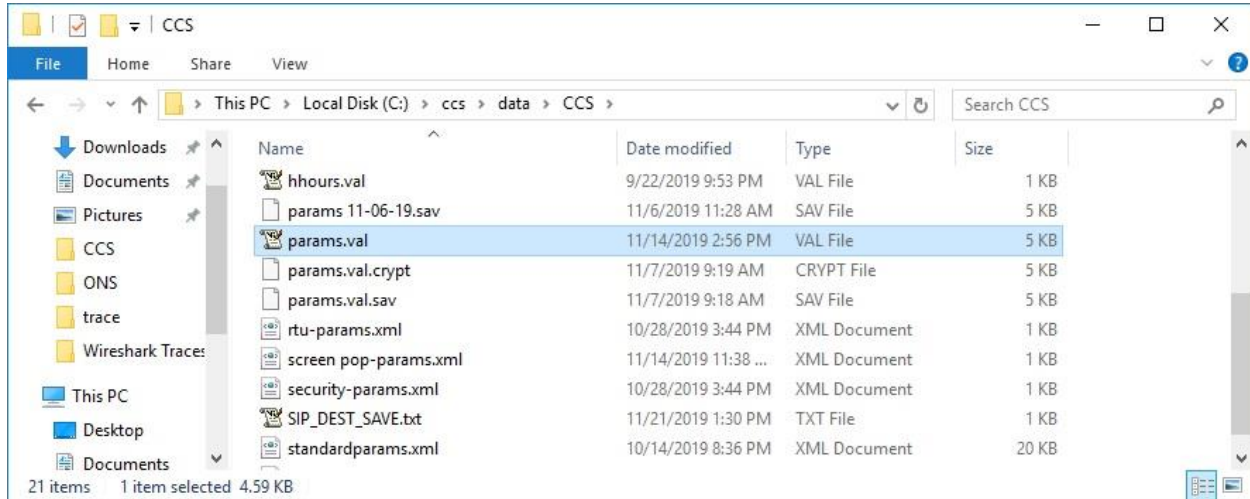
Trunk/Hub Number		Number Pattern	
Description		Search	

**Search Results**

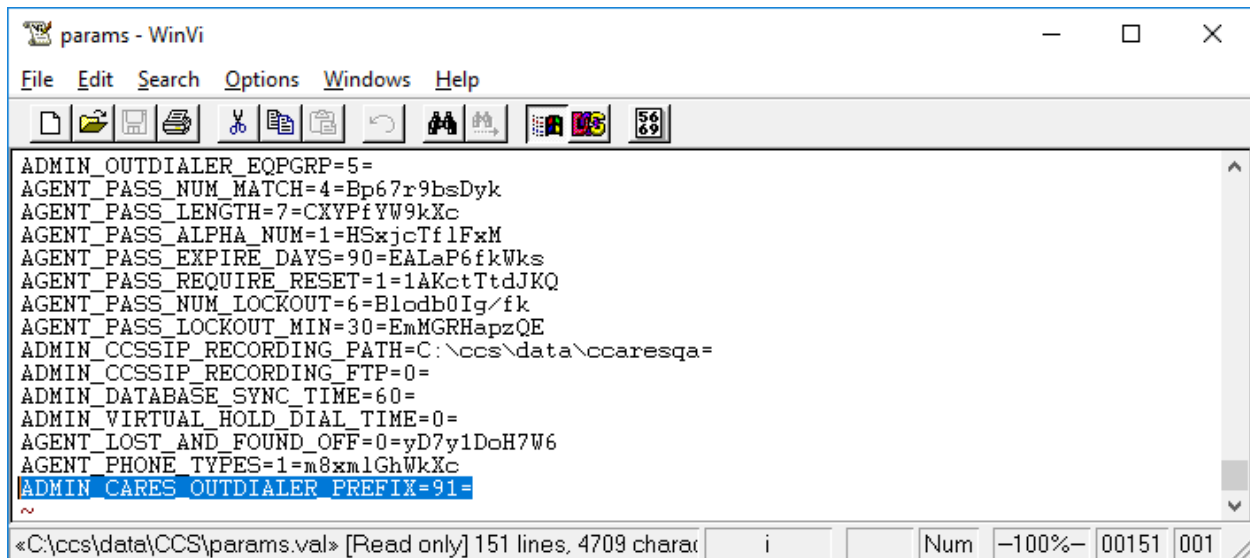
Trunk/Hub Number	Number Pattern	Description
0	54000	Inbound to Dialogic
1	6xxxx	Outbound to CM
1	91xxxxxxxxxx	Outbound to PSTN

## 7.5. Administer Outbound Campaign Parameter

From the CARES server, navigate to the **C:\ccs\data\CCS** directory to locate the **params.val** file shown below.



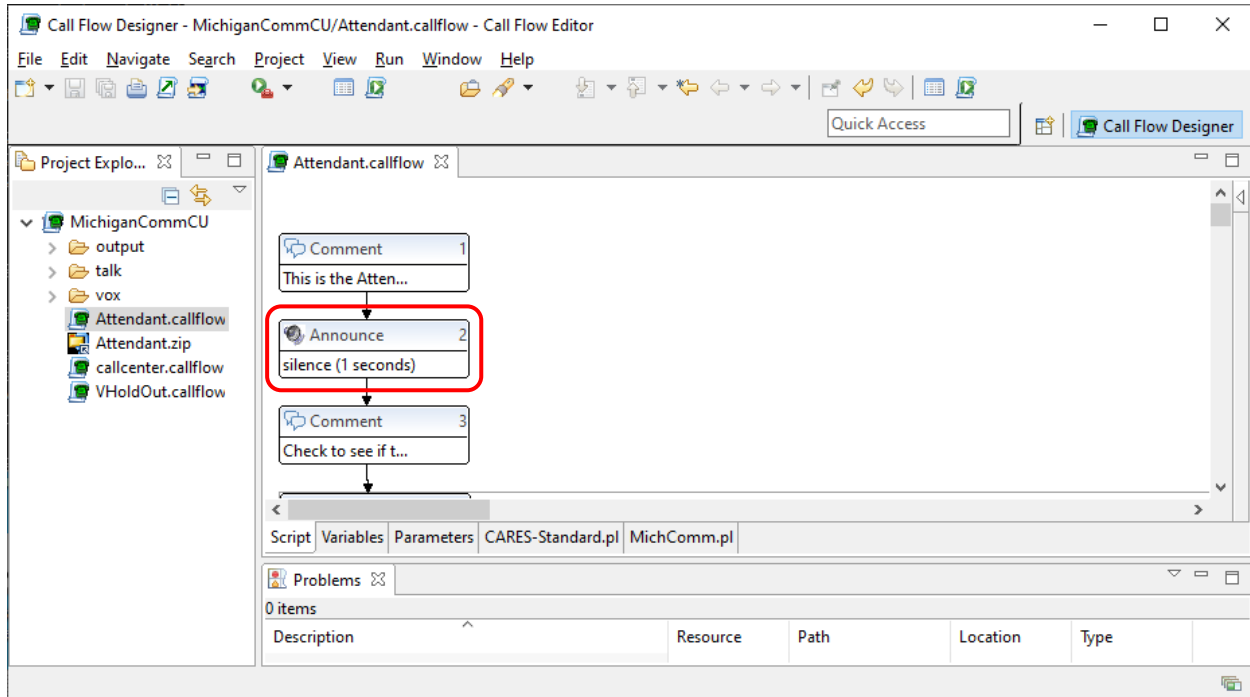
Open the **params** file with the Notepad application. Scroll down to the bottom of the file, and set **ADMIN\_CARES\_OUTDIALER\_PREFIX** to the network dialing prefix to reach the PSTN, in this case “91” as shown below.



## 7.6. Administer Inbound Call Flow

Depending on the customer network, a delay may need to be added to the inbound call flow so that the greeting announcement played to PSTN callers can be heard in its entirety. Update the inbound call flow if necessary.

In the compliance testing, a one-second delay was added to the beginning of the inbound call flow as shown below.



## 8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and CARES.

### 8.1. Verify Avaya Aura® Communication Manager

On Communication Manager, verify status of the SIP trunk group by using the “status trunk n” command, where “n” is the trunk group number administered in **Section 5.5**. Verify that all ports are in the “in-service/idle” state as shown below.

```
status trunk 54

                                TRUNK GROUP STATUS

Member      Port      Service State      Mtce Connected Ports
                                Busy

0054/001 T00113    in-service/idle    no
0054/002 T00114    in-service/idle    no
0054/003 T00115    in-service/idle    no
0054/004 T00116    in-service/idle    no
0054/005 T00160    in-service/idle    no
0054/006 T00161    in-service/idle    no
0054/007 T00162    in-service/idle    no
0054/008 T00163    in-service/idle    no
0054/009 T00164    in-service/idle    no
0054/010 T00165    in-service/idle    no
```

Verify status of the SIP signaling group by using the “status signaling-group n” command, where “n” is the signaling group number administered in **Section 5.6**. Verify that the **Group State** is “in-service” as shown below.

```
status signaling-group 54

                                STATUS SIGNALING GROUP

      Group ID: 54
      Group Type: sip

      Group State: in-service
```



## 8.2. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements** → **Session Manager** to display the **Session Manager Dashboard** screen (not shown). Select **Session Manager** → **System Status** → **SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen. Click on the CARES entity name from **Section 6.3.1**.

**SIP Entity Link Monitoring Status Summary**

This page provides a summary of Session Manager SIP entity link monitoring status.

**SIP Entities Status for All Monitoring Session Manager Instances**

Run Monitor As of 11:14 AM

1 Item Filter: Enable

Session Manager	Type	Monitored Entities					
		Down	Partially Up	Up	Not Monitored	Deny	Total
<a href="#">DR-SM</a>	Core	5	0	6	0	0	11

Select : All, None

**All Monitored SIP Entities**

Run Monitor

11 Items Filter: Enable

SIP Entity Name
<a href="#">CARES</a>

The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are “UP” as shown below.

**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.

**Status Details for the selected Session Manager:**

**All Entity Links to SIP Entity: CARES**

Summary View

1 Item Filter: Enable

Session Manager Name	IP Address Family	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<a href="#">DR-SM</a>	IPv4	10.64.101.207	5060	UDP	FALSE	UP	200 OK	UP

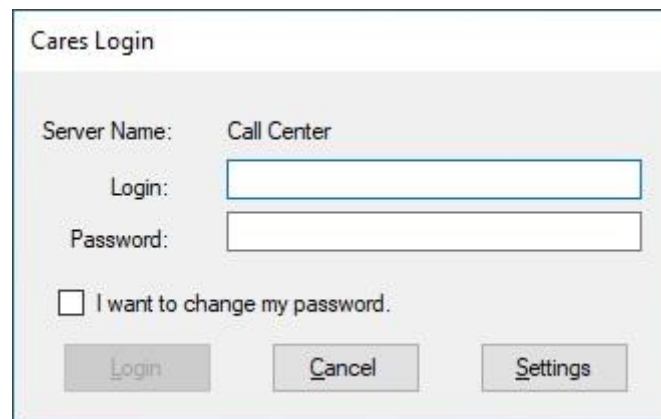


### 8.3. Verify Centurion CARES

From the agent PC, double-click on the CARES Client shortcut shown below, which was created as part of CARES Client installation.

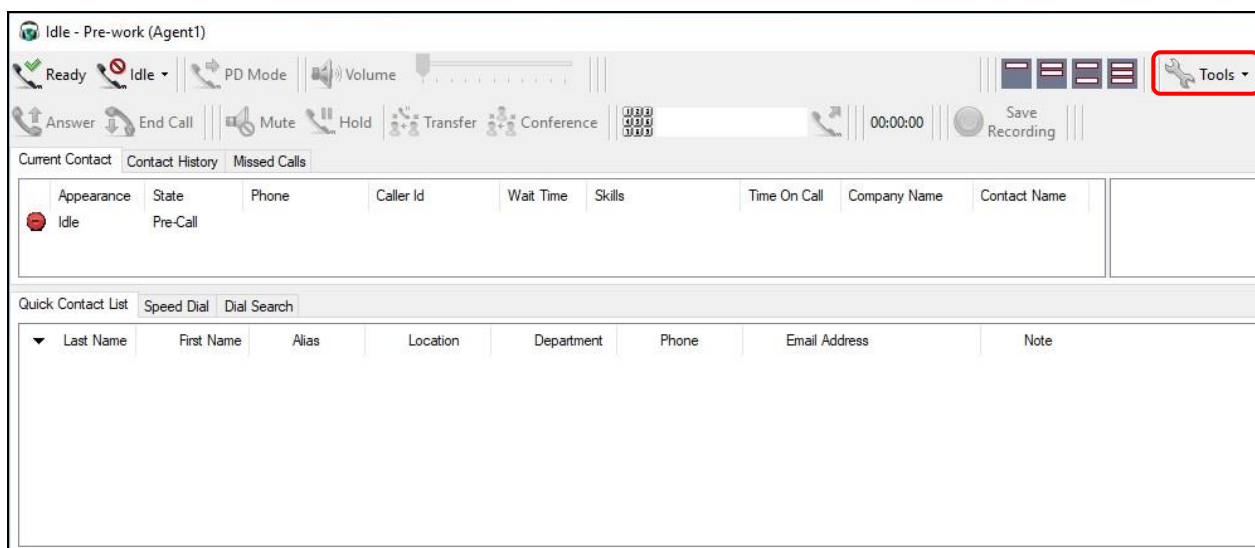


The **Cares Login** screen is displayed. Log in using the appropriate agent credentials.

A dialog box titled "Cares Login". It contains the following fields and controls:

- Server Name: Call Center
- Login: [text input field]
- Password: [password input field]
- ☐ I want to change my password.
- Buttons: Login, Cancel, Settings

The screen below is displayed next. Upon initial log in, select **Tools → Flex Settings → Add New Number**.

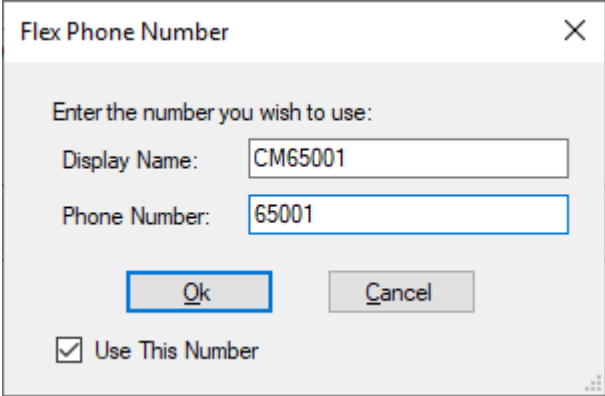
The main interface of the CARES Client application. It features a top toolbar with various call control icons (Ready, Idle, PD Mode, Volume, Answer, End Call, Mute, Hold, Transfer, Conference, etc.) and a "Tools" button highlighted with a red box. Below the toolbar is a "Current Contact" section with a table showing contact details. At the bottom is a "Quick Contact List" section with a table for searching and listing contacts.

Appearance	State	Phone	Caller Id	Wait Time	Skills	Time On Call	Company Name	Contact Name
Idle	Pre-Call							

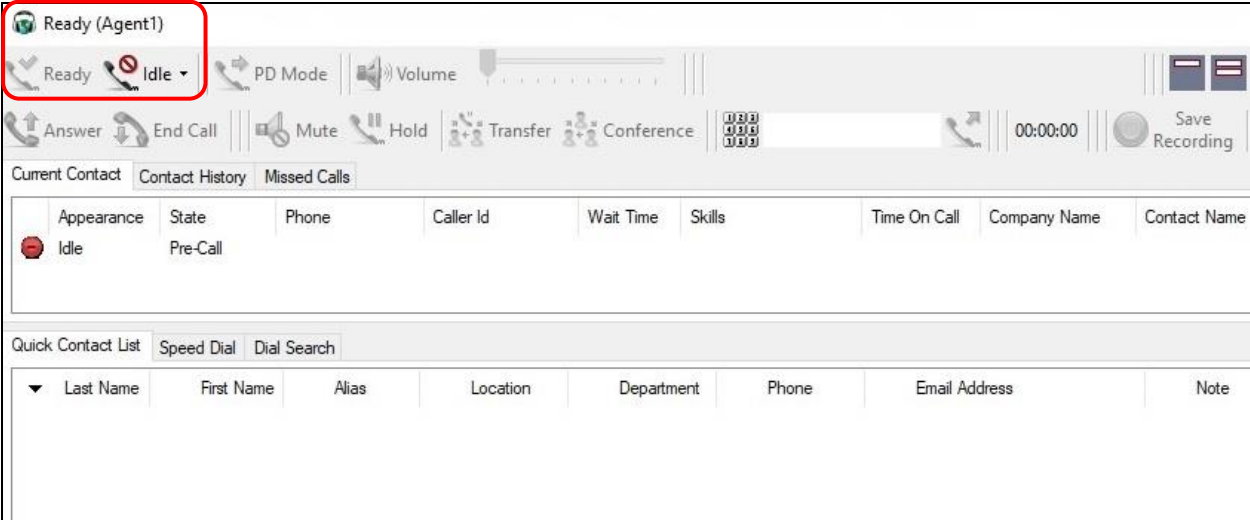
Last Name	First Name	Alias	Location	Department	Phone	Email Address	Note
-----------	------------	-------	----------	------------	-------	---------------	------

The **Flex Phone Number** pop-up box is displayed. For **Display Name**, enter a descriptive name. For **Phone Number**, enter the pertinent station user extension on Communication Manager, in this case “65001”. Check **Use This Number**.



A dialog box titled "Flex Phone Number" with a close button (X) in the top right corner. The text "Enter the number you wish to use:" is displayed. Below this, there are two input fields: "Display Name:" with the value "CM65001" and "Phone Number:" with the value "65001". At the bottom, there are two buttons: "Ok" and "Cancel". Below the buttons, there is a checkbox labeled "Use This Number" which is checked.

Select **Ready** from the screen below and verify that the screen is updated to reflect **Ready**.



A screenshot of an agent's interface. At the top left, there is a status indicator "Ready (Agent1)" with a green icon, which is highlighted by a red box. Below this, there are several icons and buttons: "Ready" (green), "Idle" (red), "PD Mode", "Volume", "Answer", "End Call", "Mute", "Hold", "Transfer", "Conference", "000 000 000", "00:00:00", and "Save Recording". Below these are tabs for "Current Contact", "Contact History", and "Missed Calls". The "Current Contact" tab is active, showing a table with columns: Appearance, State, Phone, Caller Id, Wait Time, Skills, Time On Call, Company Name, and Contact Name. The first row shows "Idle" under Appearance, "Pre-Call" under State, and empty cells for the other columns. Below this are tabs for "Quick Contact List", "Speed Dial", and "Dial Search". The "Quick Contact List" tab is active, showing a table with columns: Last Name, First Name, Alias, Location, Department, Phone, Email Address, and Note. The table is currently empty.

Establish an incoming trunk call from PSTN with CARES. Verify that the calling party hears the appropriate IVR greeting.

Enter the DTMF digit to select the option associated with connection to an agent. Verify that CARES place a call to the available agent if not already connected, and that the agent desktop is updated to reflect a ringing ACD call along with the PSTN calling party number as shown below. Select **Answer**.

ACD Ringing (Agent1)

Ready Idle PD Mode Volume

Answer End Call Mute Hold Transfer Conference 00:00:05 Save Recording

Current Contact Contact History Missed Calls

Appearance	State	Phone	Caller Id	Wait Time	Skills	Time On Call	Company Name	Contact Name
ACD Call	Ringing	19089532103				00:00:05		

Quick Contact List Speed Dial Dial Search

Agent2

Verify that the agent is connected to the PSTN caller with two-way talk path, and that the agent desktop is updated to reflect the call in the **Connected** state.

ACD Connected (Agent1)

Ready Idle PD Mode Volume

Answer End Call Mute Hold Transfer Conference 00:00:15 Save Recording

Current Contact Contact History Missed Calls

Appearance	State	Phone	Caller Id	Wait Time	Skills	Time On Call	Company Name	Contact Name
ACD Call	Connected	19089532103				00:00:15		

Quick Contact List Speed Dial Dial Search

Agent2

## 9. Conclusion

These Application Notes describe the configuration steps required for Centurion CARES 14.03 to successfully interoperate with Avaya Aura® Communication Manager 8.1 and Avaya Aura® Session Manager 8.1. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

## 10. Additional References

This section references the product documentation relevant to these Application Notes.

1. *Administering Avaya Aura® Communication Manager*, Release 8.1.x, Issue 3, August 2019, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Release 8.1, Issue 1, June 2019, available at <http://support.avaya.com>.
3. *CARES Client*, Version 14.3, 6/12/19, available at <ftp://caresdl:download@ftp.centurioncares.com/pub/Documentation>.

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