



Avaya Solution & Interoperability Test Lab

Application Notes for T-Metrics Contact Center with Avaya Aura® Session Manager 8.1 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for T-Metrics Contact Center to interoperate with Avaya Aura® Session Manager 8.1 and Avaya Aura® Communication Manager 8.1.

T-Metrics Contact Center is a multi-channel contact center solution that can handle voice, email, web chat, video, social media, and SMS contacts. The compliance testing focused on the voice integration with Avaya Aura® Session Manager using the SIP trunk and user interfaces.

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 T-Metrics Contact Center to interoperate with Avaya Aura® Session Manager 8.1 and Avaya Aura® Communication Manager 8.1.

Contact Center is a multi-channel contact center solution that can handle voice, email, web chat, video, social media, and SMS contacts. The compliance testing focused on the voice integration with Session Manager using the SIP trunk and user interfaces.

The Contact Center solution consisted of the Contact Center server, along with T-Metrics ACD Agent Module and T-Metrics SIP Softphone client applications running on the agent and supervisor desktops.

The Contact Center server integrated with Session Manager via SIP trunk, and the SIP Softphone on each agent and supervisor desktop integrated with Session Manager as a SIP user.

The Contact Center server consisted of the TMI ACD Controller Module, TMI DigiSIP Module and TMI Event Server Module components. The TMI DigiSIP Module is the component responsible for SIP trunk integration with Session Manager.

Incoming calls from PSTN were routed over SIP trunk to the Contact Center server. The Contact Center server invoked the IVR script associated with the routed number to play greeting announcement, collect DTMF for menu navigation, and use of SIP REFER to transfer calls to available agents.

Each agent and supervisor desktop has the ACD Agent Module and SIP Softphone client applications. The ACD Agent Module application is used to log into Contact Center to set agent status with ACD functionality provided by Contact Center. The SIP Softphone application is used to register with Session Manager as a SIP user for media termination and handling of calls such as answer and drop.

The conference feature was accomplished by the SIP Softphone application via local bridge of talk paths for active calls at the agent desktop. The supervisor monitor feature was accomplished by the ACD Agent Module application with proprietary implementation that does not involve Session Manager.

2. General Test Approach and Test Results

The feature test cases were performed manually. Upon launch of the SIP Softphone client application, the application automatically registered as a SIP user with Session Manager.

Incoming calls were placed manually from PSTN to Contact Center. Manual call controls from the SIP Softphone client application were exercised to verify features such as answering and transferring of calls.

The serviceability test cases were performed manually such as disconnecting/reconnecting the Ethernet connection to the Contact Center server and/or client.

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 interface between Avaya systems and Contact Center did not include use of any specific encryption features as requested by T-Metrics.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing focused on verifying the following on Contact Center:

- Proper handling of SIP exchanges including registration, DTMF, OPTIONS, G.711MU, codec negotiation, and user registration.
- Proper handling of call scenarios including screen pop, answer, decline, hold/resume, mute/unmute, drop, abort, blind/supervised transfer, supervised conference, non-ACD call, queuing, outgoing call, multiple calls, multiple agents, long duration, local do not disturb setting at the softphone, and recording of basic calls.

The serviceability testing focused on verifying the ability of Contact Center to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to the Contact Center server and client.

2.2. Test Results

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

- SIP Softphone does not support media shuffling and therefore the related parameters need to be disabled on the associated network region on Communication Manager. See **Section 5.5** for configuration details.
- In the conference scenario, after the PSTN party drops from the 3-way conference, the agent SIP Softphone screens were not updated and still reflected connection with PSTN instead of with the remaining party.
- For an internal call between two SIP Softphones, the called SIP Softphone reflected Unknown as the calling party name.

2.3. Support

Technical support on Contact Center can be obtained through the following:

- **Phone:** +1 (704) 525-5551 opt 2
- **Email:** support@tmetrics.com
- **Web :** <http://service.tmetrics.com/servicedesk/customer/user/login>

3. Reference Configuration

The configuration used for the compliance testing is shown in **Figure 1**. SIP trunk was used between the Contact Center server and Session Manager, and SIP user was used between each SIP Softphone client application and Session Manager. The applicable domain name was “dr220.com”.

A five-digits Uniform Dial Plan (UDP) was used to facilitate routing with Contact Center. Unique extensions were assigned to users on Communication Manager (6xxxx) and to Contact Center (53xxx).

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.

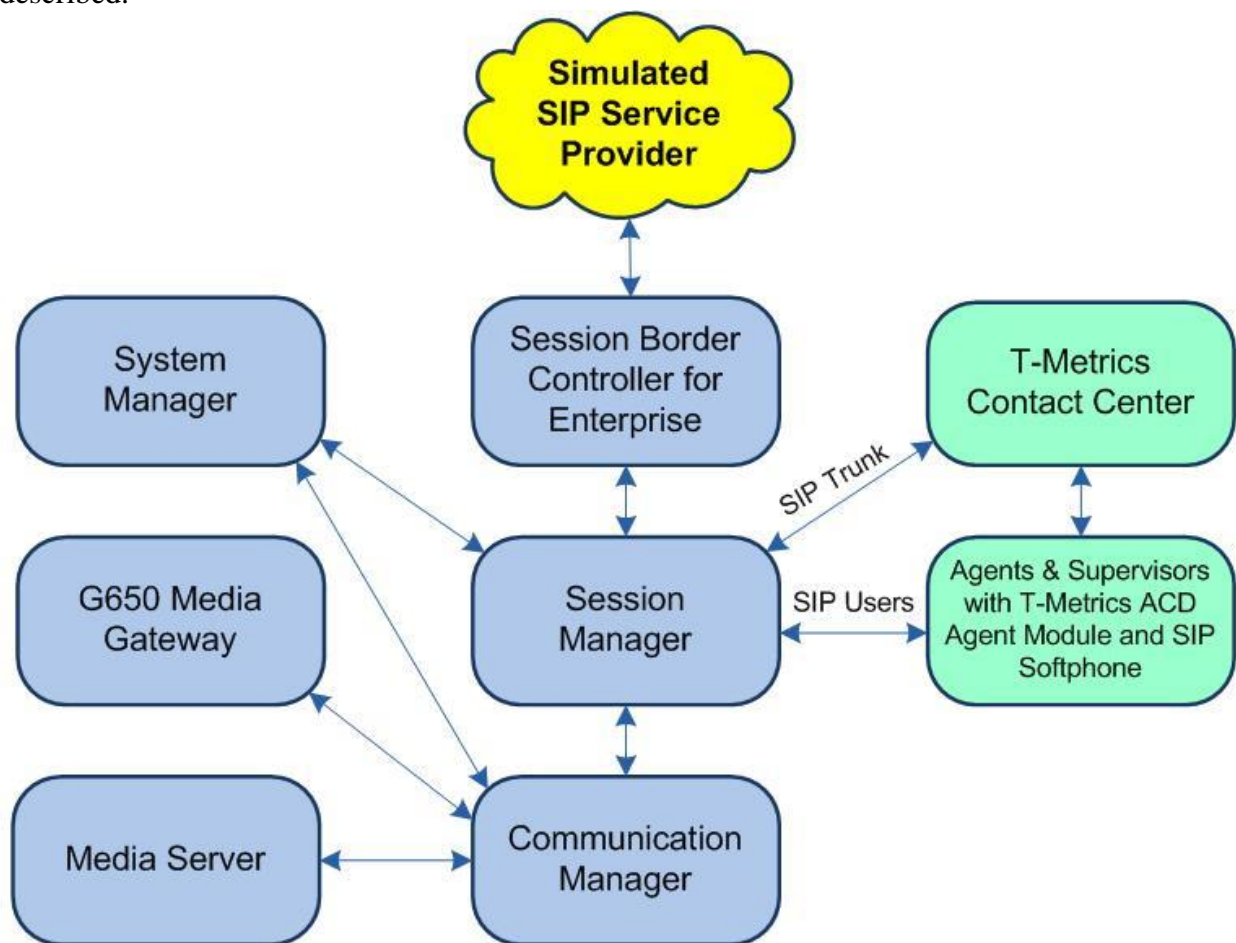


Figure 1: Compliance Testing Configuration

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	8.1.3 (8.1.3.0.1.890.26685)
Avaya G650 Media Gateway	NA
Avaya Aura® Media Server in Virtual Environment	8.0.2.138
Avaya Aura® Session Manager in Virtual Environment	8.1.3 (8.1.3.0.813014)
Avaya Aura® System Manager in Virtual Environment	8.1.3 (8.1.3.0.1012091)
Avaya Session Border Controller for Enterprise in Virtual Environment	8.1.1 (8.1.1.0-19390)
T-Metrics Contact Center on Windows Server 2019 <ul style="list-style-type: none">TMI ACD Controller ModuleTMI DigiSIP ModuleTMI Event Server Module	Standard 22 Jan 21 21 Jan 21 19 Nov 20
T-Metrics ACD Agent Module & T-Metrics SIP Softphone (TMI_SIP) on Windows 10 Pro	a15 Jan 21 21.2.23.1 (2021.10.20.3)

5. Configure Avaya Aura® Communication Manager

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

- Verify license
- Administer system parameters features
- 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, a separate set of trunk group, signaling group, network region, and codec set was used for SIP trunk integration with Contact Center server.

As for SIP user integration with SIP Softphone, an existing set of trunk group, signaling group, network region, and codec set was used, and the only configuration needed was on the network region and codec set.

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 to verify that there is sufficient capacity for SIP users by comparing the **Maximum Off-PBX Telephones - OPS** field value with the corresponding value in the **USED** column.

display system-parameters customer-options		Page 1 of 12
OPTIONAL FEATURES		
G3 Version: V18	Software Package: Enterprise	
Location: 2	System ID (SID): 1	
Platform: 28	Module ID (MID): 1	
		USED
Platform Maximum Ports:	81000	204
Maximum Stations:	41000	20
Maximum XMOBILE Stations:	41000	0
Maximum Off-PBX Telephones - EC500:	41000	0
Maximum Off-PBX Telephones - OPS:	41000	4

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 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
Maximum Administered SIP Trunks:	40000	40

5.2. Administer System Parameters Features

Use the “**change system-parameters features**” command to allow for trunk-to-trunk transfers.

For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to “**all**” to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class Of Restriction or Class Of Service levels. See reference [1] for more details.

```
change system-parameters features                               Page 1 of 19
      FEATURE-RELATED SYSTEM PARAMETERS
      Self Station Display Enabled? n
      Trunk-to-Trunk Transfer: all
      Automatic Callback with Called Party Queuing? n
      Automatic Callback - No Answer Timeout Interval (rings): 3
      Call Park Timeout Interval (minutes): 10
      Off-Premises Tone Detect Timeout Interval (seconds): 20
      AAR/ARS Dial Tone Required? y
```

5.3. Administer Node Names

Use the “**display node-names ip**” command. Note the **Name** and **IP Address** of the processor or existing C-LAN circuit pack that will be used for SIP trunk integration with Contact Center, 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**”.

```
display 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
```

5.4. Administer Codec Set

Administer a codec set for SIP trunk integration with Contact Center server and a codec set for SIP user integration with SIP Softphone.

Use the “**change ip-codec-set n**” command, where “**n**” is an existing codec set number to be used for SIP trunk integration with Contact Center. For **Audio Codec**, enter the pertinent G.711 variant as shown below. Note that Contact Center only supports the G.711 codec variant. 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 3                                     Page 1 of 2

                                IP MEDIA PARAMETERS

Codec Set: 3

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

Media Encryption                                Encrypted SRTCP: enforce-unenc-srtcp
1: none
```

Use the “**change ip-codec-set n**” command, where “**n**” is an existing codec set number used with SIP users. For **Audio Codec**, make certain that a G.711 variant is configured. For **Media Encryption**, make certain that “**none**” is included. For **Encrypted SRTCP**, make certain that “**best-effort**” or “**enforce-unenc-srtcp**” is configured.

```
change ip-codec-set 1                                     Page 1 of 2

                                IP MEDIA PARAMETERS

Codec Set: 1

Audio      Silence      Frames      Packet
Codec      Suppression   Per Pkt    Size(ms)
1: G.711MU      n           2          20
2: G.729        n           2          20
3:
4:
5:
6:
7:

Media Encryption                                Encrypted SRTCP: best-effort
1: 1-srtp-aescm128-hmac80
2: aes
3: none
```

5.5. Administer Network Region

Administer a network region for SIP trunk integration with Contact Center and a network region for SIP user integration with SIP Softphone.

Use the “**change ip-network-region n**” command, where “**n**” is an existing network region number to be used for SIP trunk integration with Contact Center.

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 5.4**.

```
change ip-network-region 3                                     Page 1 of 20
                                                                IP NETWORK REGION
Region: 3              NR Group: 3
Location:              Authoritative Domain: dr220.com
Name: T-Metrics        Stub Network Region: n
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: yes
Codec Set: 3           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
```

Use the “**change ip-network-region n**” command, where “**n**” is an existing network region used with SIP users.

Enter “**no**” for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, which is required by the SIP Softphone client application.

```
change ip-network-region 1                                     Page 1 of 20
                                                                IP NETWORK REGION
Region: 1              NR Group: 1
Location:              Authoritative Domain: dr220.com
Name: main             Stub Network Region: n
MEDIA PARAMETERS      Intra-region IP-IP Direct Audio: no
Codec Set: 1           Inter-region IP-IP Direct Audio: no
                      IP Audio Hairpinning? y
UDP Port Min: 2048
UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
Call Control PHB Value: 46
Audio PHB Value: 46
Video PHB Value: 26
```

5.6. Administer SIP Trunk Group

Use the “**add trunk-group n**” command, where “**n**” is an available trunk group number, in this case “**53**”. 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”

add trunk-group 53		Page 1 of 4	
TRUNK GROUP			
Group Number: 53	Group Type: sip	CDR Reports: y	
Group Name: T-Metrics CC	COR: 1	TN: 1	TAC: 1053
Direction: two-way	Outgoing Display? n	Night Service:	
Dial Access? n			
Queue Length: 0			
Service Type: tie	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group:		
	Number of Members: 0		

Navigate to **Page 3** and enter “**private**” for **Numbering Format**. Retain the default values for the remaining fields.

add trunk-group 53		Page 3 of 4	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Suppress # Outpulsing? n	Numbering Format: private		
	UII Treatment: service-provider		

Navigate to **Page 4** and enter “**101**” for **Telephone Event Payload Type**. Retain the default values for the remaining fields.

add trunk-group next		Page 4 of 4	
PROTOCOL VARIATIONS			
Mark Users as Phone? n			
Prepend '+' to Calling/Alerting/Diverting/Connected Number? n			
Send Transferring Party Information? n			
Network Call Redirection? n			
Send Diversion Header? n			
Support Request History? y			
Telephone Event Payload Type: 101			

5.7. Administer SIP Signaling Group

Use the “**add signaling-group n**” command, where “**n**” is an available signaling group number, in this case “**53**”. 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 5.3**.
- **Far-end Node Name:** The Session Manager node name from **Section 5.3**.
- **Near-end Listen Port:** An available port for integration with Contact Center.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** The network region number from **Section 5.5**.
- **Far-end Domain:** The domain name from **Section 3**.

add signaling-group 53		Page 1 of 2
SIGNALING GROUP		
Group Number: 53	Group Type: sip	
IMS Enabled? n	Transport Method: tls	
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		
Near-end Node Name: procr	Far-end Node Name: sm7-sig	
Near-end Listen Port: 5361	Far-end Listen Port: 5361	
	Far-end Network Region: 3	
Far-end Domain: dr220.com		
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	

Use the “**change trunk-group n**” command, where “**n**” is the trunk group number from **Section 5.6**. Enter the following values for the specified fields and retain the default values for the remaining fields.

- change trunk-group 53** Page 1 of 4

TRUNK GROUP

Group Number: 53 Group Type: sip CDR Reports: y

 Group Name: T-Metrics CC COR: 1 TN: 1 TAC: 1053

 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: 53

Number of Members: 10

Use the “**change route-pattern n**” command, where “**n**” is an existing route pattern number to be used for SIP trunk integration with Contact Center, in this case “**53**”. Enter the following values for the specified fields and retain the default values for the remaining fields.

- ```

change route-pattern 53
Pattern Number: 53 Pattern Name: T-Metrics
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: 53 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

```

## 5.10. Administer Private Numbering

Use the “**change private-numbering 0**” command, to define the calling party number to send to Contact Center. Add an entry for the trunk group defined in **Section 5.6**.

In the example shown below, all calls originating from a 5-digit extension beginning with **6** and routed to trunk group **53** 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                        | Ext  | Trk    | Private | Total |                       |
| Len                        | Code | Grp(s) | Prefix  | Len   |                       |
| 5                          | 6    | 66     |         | 5     | Total Administered: 1 |
| 5                          | 6    | 53     |         | 5     | Maximum Entries: 540  |

## 5.11. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 53xxx to Contact Center. 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 53xxx, as shown below.

|                           |     |     |        |          |                 |
|---------------------------|-----|-----|--------|----------|-----------------|
| change uniform-dialplan 0 |     |     |        |          | Page 1 of 2     |
| UNIFORM DIAL PLAN TABLE   |     |     |        |          |                 |
|                           |     |     |        |          | Percent Full: 0 |
| Matching                  |     |     | Insert |          | Node            |
| Pattern                   | Len | Del | Digits | Net Conv | Num             |
| 53                        | 5   | 0   |        | aar n    |                 |

## 5.12. Administer AAR Analysis

Use the “**change aar analysis 0**” command and add an entry to specify how to route calls to Contact Center at 53xxx. In the example shown below, calls with digits 53xxx will be routed as an AAR call using route pattern “**53**” from **Section 5.9**.

|                          |        |       |       |      |                 |
|--------------------------|--------|-------|-------|------|-----------------|
| change aar analysis 0    |        |       |       |      | Page 1 of 2     |
| AAR DIGIT ANALYSIS TABLE |        |       |       |      |                 |
| Location: all            |        |       |       |      | Percent Full: 1 |
|                          | Dialed |       |       |      |                 |
|                          | String | Total | Route | Call | Node            |
|                          |        | Min   | Max   | Type | Num             |
| 53                       |        | 5     | 5     | 53   | aar n           |

### 5.13. 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 “**212**”. 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.

|                                                      |                |                      |
|------------------------------------------------------|----------------|----------------------|
| <b>change trunk-group 212</b>                        |                | <b>Page</b> 3 of 5   |
| TRUNK FEATURES                                       |                |                      |
| ACA Assignment? n                                    | Measured: none | Maintenance Tests? y |
| Suppress # Outpulsing? n    Numbering Format: public |                |                      |
| UUI IE Treatment: shared                             |                |                      |
| Maximum Size of UUI Contents? 128                    |                |                      |
| Replace Restricted Numbers? n                        |                |                      |
| Replace Unavailable Numbers? n                       |                |                      |
| <b>Modify Tandem Calling Number: tandem-cpn-form</b> |                |                      |
| Send UCID? y                                         |                |                      |
| Show ANSWERED BY on Display? y                       |                |                      |

### 5.14. 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 Contact Center.

In the example shown below, all calls originating from 53xxx and routed to the PSTN trunk group in **Section 5.13** will result in a 10-digit calling number. For **Outgoing Number Format**, use an applicable format, in this case “**pub-unk**”.

|                                                     |        |          |          |        |        |                     |
|-----------------------------------------------------|--------|----------|----------|--------|--------|---------------------|
| <b>change tandem-calling-party-num</b>              |        |          |          |        |        | <b>Page</b> 1 of 67 |
| CALLING PARTY NUMBER CONVERSION<br>FOR TANDEM CALLS |        |          |          |        |        |                     |
| CPN                                                 |        | Incoming | Outgoing |        |        | Outgoing            |
| Len                                                 | Prefix | Number   | Trunk    |        |        | Number              |
| 5                                                   | 53     | Format   | Group(s) | Delete | Insert | Format              |
|                                                     |        |          | 212      |        | 30353  | <b>pub-unk</b>      |



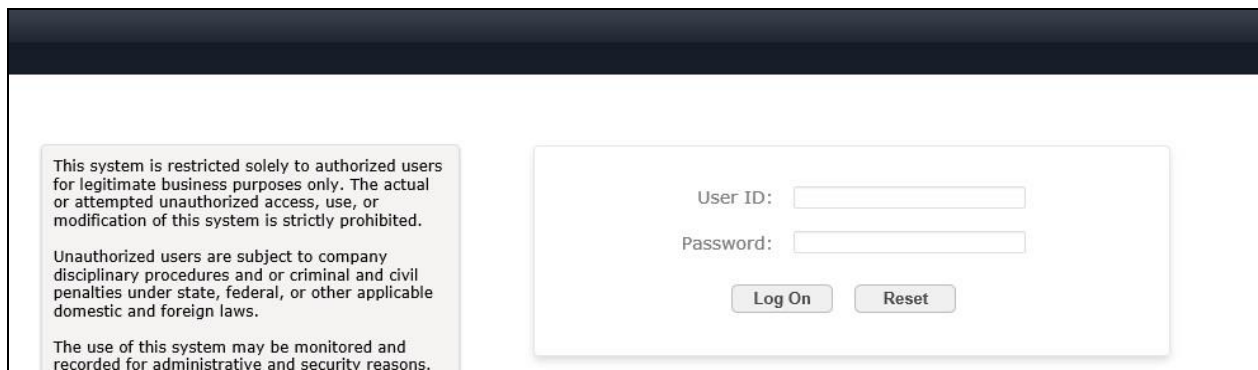
## 6. Configure Avaya Aura® Session Manager

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:

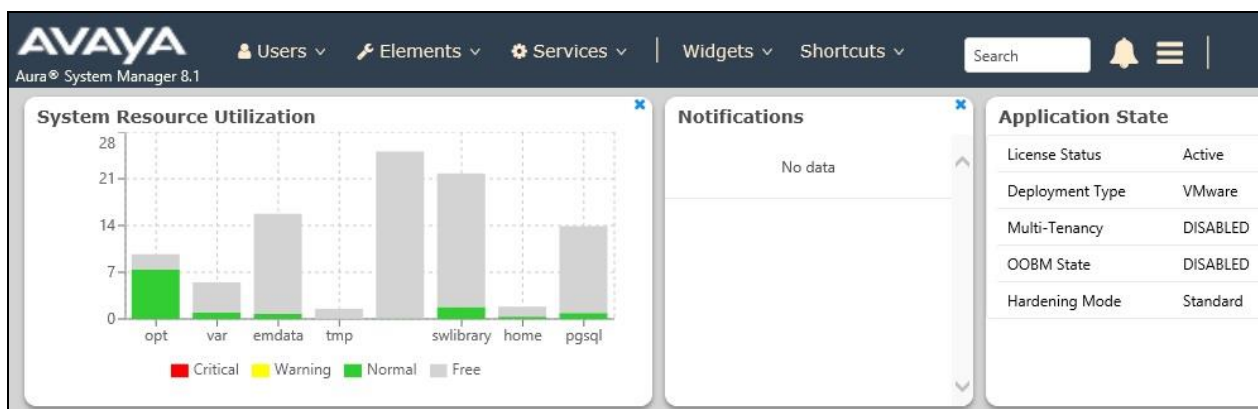
- Launch System Manager
- Administer locations
- Administer SIP entities
- Administer routing policies
- Administer dial patterns
- Administer SIP users
- Administer Session Manager entity

### 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.

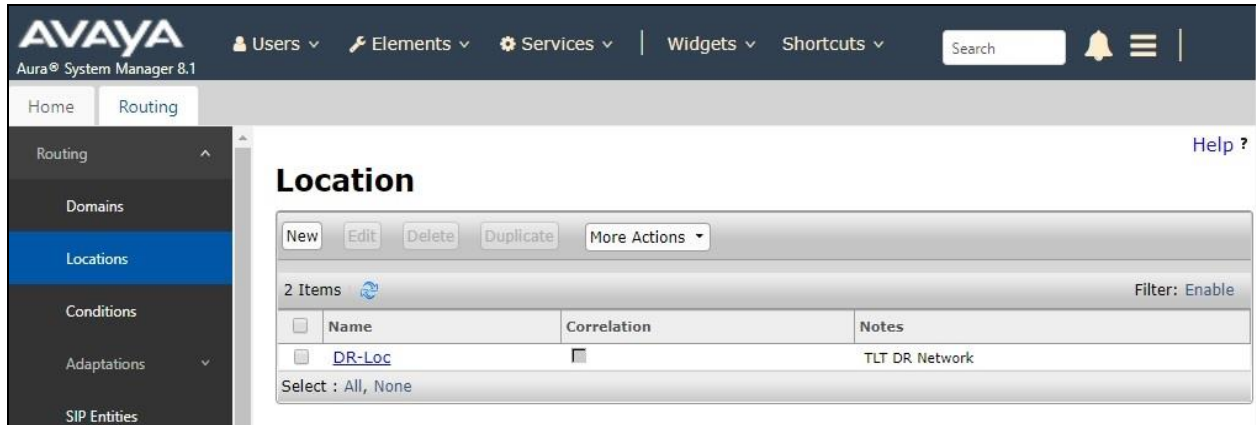


The screen below is displayed next.

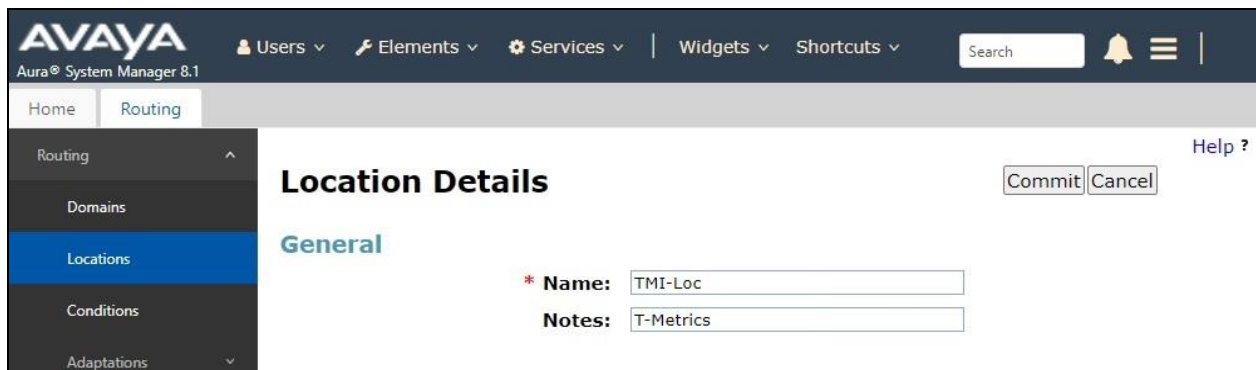


## 6.2. Administer Locations

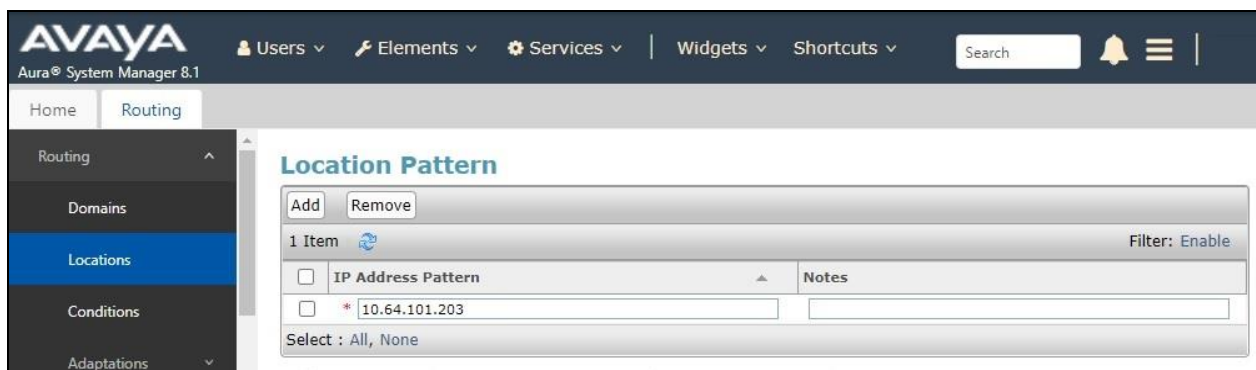
Select **Elements** → **Routing** → **Locations** from the top menu to display the **Location** screen below. Select **New** to add a new location for Contact Center.



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 the Contact Center server as shown below. Retain the default values in the remaining fields.



## 6.3. Administer SIP Entities

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

### 6.3.1. SIP Entity for Contact Center

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

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 the Contact Center server.
- **Type:** “Other”
- **Notes:** Any desired notes.
- **Location:** Select the Contact Center location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

The screenshot displays the Avaya Aura System Manager 8.1 web interface. The top navigation bar includes the Avaya logo, the text 'Aura® System Manager 8.1', and several menu items: 'Users', 'Elements', 'Services', 'Widgets', and 'Shortcuts'. A search bar and notification icons are also present. The left sidebar shows a tree view with 'Routing' selected, and 'SIP Entities' highlighted under the 'Routing' category. The main content area is titled 'SIP Entity Details' and contains a 'General' tab. The form fields are as follows:

- Name:** T-Metrics
- FQDN or IP Address:** 10.64.101.203
- Type:** Other (dropdown)
- Notes:** T-Metrics Contact Center
- Adaptation:** (empty dropdown)
- Location:** TMI-Loc (dropdown)
- Time Zone:** America/New\_York (dropdown)
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting (dropdown)
- Credential name:** (empty text field)
- Securable:** ☐
- Call Detail Recording:** egress (dropdown)
- CommProfile Type Preference:** (empty dropdown)

At the bottom of the form, there is a 'Loop Detection' section which is currently collapsed.

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:** “TCP”
- **Port:** “5060”
- **SIP Entity 2:** The Contact Center entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

Dashboard x +

Not secure | 10.64.101.235/SMGR/#

Apps SBCE SMGR CM AES IPOSE Utility MS-CM EPM Avaya DevConnect vSphere Other bookmarks

**AVAYA** Users Elements Services Widgets Shortcuts Search adm

Aura® System Manager 8.1

Home Routing

R...

**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-TMI | DR-SM        | TCP      | * 5060 | T-Metrics    | * 5060 | trusted           |

Select : All, None

### 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 SIP trunk integration with Contact Center.

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 or C-LAN circuit pack IP address from **Section 5.3**.
- **Type:** “CM”
- **Notes:** Any desired notes.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.

The screenshot shows the Avaya Aura System Manager 8.1 interface. The left sidebar contains a navigation menu with the following items: Routing, Domains, Locations, Conditions, Adaptations, SIP Entities (selected), Entity Links, Time Ranges, Routing Policies, Dial Patterns, and Origination Dial... The main content area displays the 'SIP Entity Details' form. The form has a 'General' tab selected and a 'Loop Detection' tab at the bottom. The form fields are as follows:

- Name:** DR-CM-5361
- FQDN or IP Address:** 10.64.101.236
- Type:** CM
- Notes:** CM port 5361 for T-Metrics
- Adaptation:** (empty dropdown)
- Location:** DR-Loc
- Time Zone:** America/New\_York
- SIP Timer B/F (in seconds):** 4
- Minimum TLS Version:** Use Global Setting
- Credential name:** (empty text field)
- Securable:** ☐
- Call Detail Recording:** none

The form also includes 'Commit' and 'Cancel' buttons at the top right.

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 5.7**.
- **Port:** The signaling group far-end listen port number from **Section 5.7**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signaling group near-end listen port number from **Section 5.7**.
- **Connection Policy:** “trusted”

**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 |
|--------------|--------------|----------|--------|--------------|--------|-------------------|
| * SM-CM-5361 | DR-SM        | TLS      | * 5361 | DR-CM-5361   | * 5361 | trusted           |

Select : All, None

**SIP Responses to an OPTIONS Request**

## 6.4. Administer Routing Policies

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

### 6.4.1. Routing Policy for Contact Center

Select **Routing** → **Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for Contact Center. 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 Contact Center entity name from **Section 6.3.1**. The screen below shows the result of the selection.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Search [ ] [ ] [ ]

Home Routing

Routing Policies

**Routing Policy Details** [Commit] [Cancel] [Help ?]

**General**

\* Name: To-TMI

Disabled: ☐

\* Retries: 0

Notes: T-Metrics

**SIP Entity as Destination**

Select

| Name      | FQDN or IP Address | Type  | Notes                    |
|-----------|--------------------|-------|--------------------------|
| T-Metrics | 10.64.101.203      | Other | T-Metrics Contact Center |

**Time of Day**

### 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.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Search [ ] [ ] [ ]

Home Routing

Routing Policy Details [Commit] [Cancel] [Help ?]

**General**

\* Name: To-CM-5361

Disabled: ☐

\* Retries: 0

Notes: [ ]

**SIP Entity as Destination**

Select

| Name       | FQDN or IP Address | Type | Notes                      |
|------------|--------------------|------|----------------------------|
| DR-CM-5361 | 10.64.101.236      | CM   | CM port 5361 for T-Metrics |

**Time of Day**



## 6.5. Administer Dial Patterns

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

### 6.5.1. Dial Pattern for Contact Center

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 Contact Center. 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 Contact Center extensions pattern 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 Contact Center. In the compliance testing, the policy allowed for call origination from Communication Manager location “**DR-Loc**”. The Contact Center routing policy from **Section 6.4.1** was selected as shown below.

The screenshot displays the Avaya Aura System Manager 8.1 interface. The left navigation pane shows the 'Routing' section expanded, with 'Dial Patterns' selected. The main content area is titled 'Dial Pattern Details' and includes 'Commit' and 'Cancel' buttons. The 'General' section contains the following fields:

- Pattern:** 53
- Min:** 5
- Max:** 5
- Emergency Call:** ☐
- SIP Domain:** -ALL- (dropdown)
- Notes:** (text area)

The 'Originating Locations and Routing Policies' section features an 'Add' button and a table with one item:

| Originating Location Name       | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled  | Routing Policy Destination | Routing Policy Notes |
|---------------------------------|----------------------------|---------------------|------|--------------------------|----------------------------|----------------------|
| <input type="checkbox"/> DR-Loc | DR Network                 | To-TMI              | 0    | <input type="checkbox"/> | T-Metrics                  | T-Metrics            |

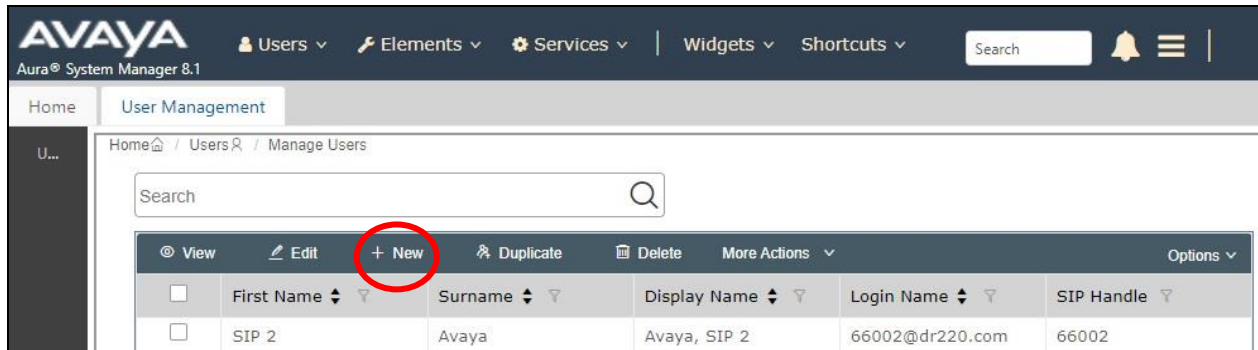
Below the table, there is a 'Select : All, None' option.

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.

Repeat this section to make similar changes to applicable Communication Manager dial pattern to reach the PSTN. In the compliance testing, Contact Center will add the prefix “9” for outbound calls to the PSTN, and therefore the existing dial pattern for “9” was also changed (not shown below).

## 6.6. Administer SIP Users

Create a SIP user for each Contact Center agent and supervisor. Select **Users** → **User Management** → **Manage Users** from the top menu to display the screen below. Click **New** to add a SIP user.



### 6.6.1. Identity

The **User Profile | Add** screen is displayed. Enter desired **Last Name** and **First Name**.

For **Login Name**, enter “n@x”, where “n” is the desired user extension and “x” is the applicable domain name from **Section 3**. Retain the default values in the remaining fields.

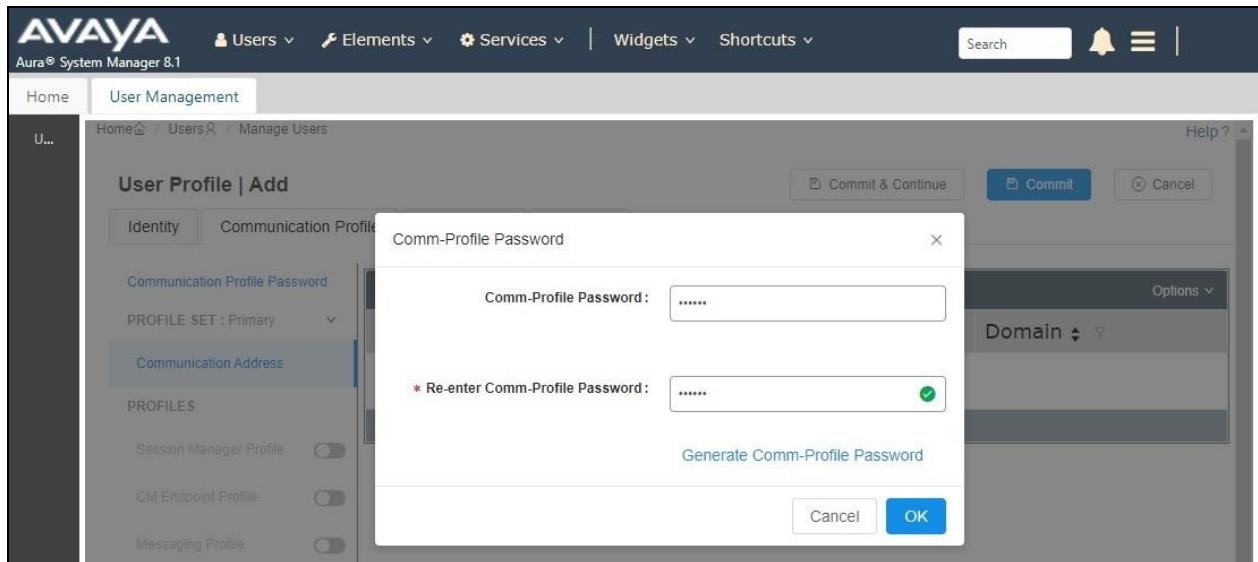
The screenshot shows the 'User Profile | Add' screen in the Avaya Aura System Manager 8.1 interface. The 'Basic Info' tab is selected. The form fields are as follows:

| Field                                     | Value                    |
|-------------------------------------------|--------------------------|
| User Provisioning Rule                    |                          |
| Last Name                                 | TMI                      |
| Last Name (in Latin alphabet characters)  | TMI                      |
| First Name                                | Agent 1                  |
| First Name (in Latin alphabet characters) | Agent 1                  |
| Login Name                                | 66991@dr220.com          |
| Middle Name                               | Middle Name Of User      |
| Description                               | Description Of User      |
| Email Address                             | Email Address Of User    |
| Password                                  |                          |
| User Type                                 | Basic                    |
| Confirm Password                          |                          |
| Localized Display Name                    | Localized Display Name O |

### 6.6.2. Communication Profile

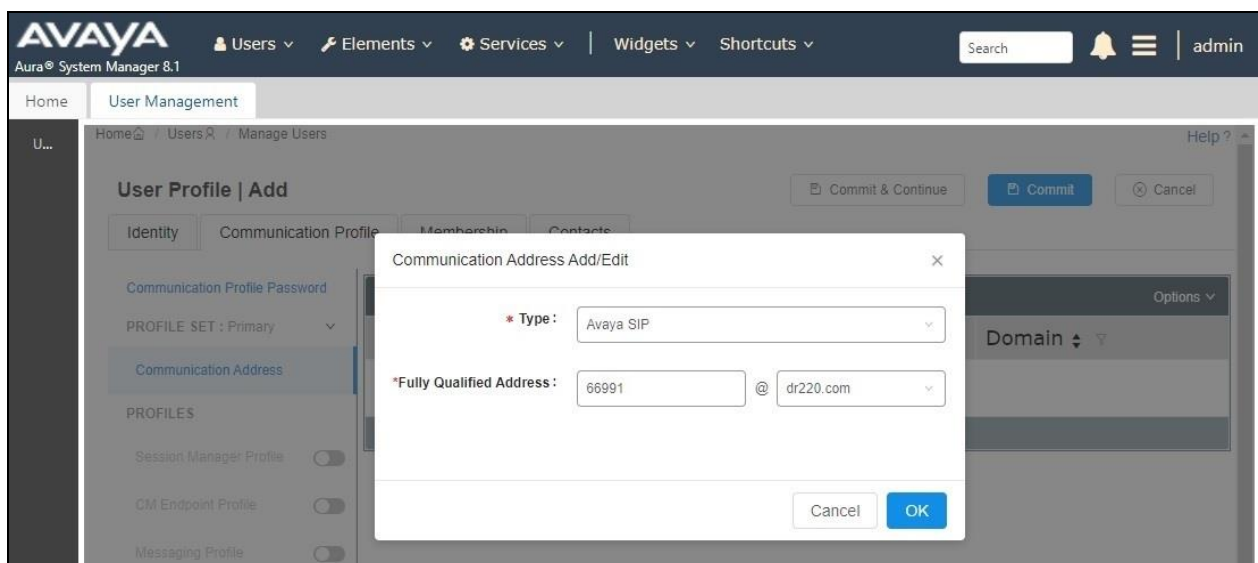
Select the **Communication Profile** tab, followed by **Communication Profile Password** to display the **Comm-Profile Password** pop-up box.

For **Communication-Profile Password** and **Re-enter Comm-Profile Password**, enter the desired password for the SIP user to use for registration.



Select **Communication Address** from the left, followed by **New** to display the **Communication Address Add/Edit** pop-up box.

For **Type**, select “**Avaya SIP**”. For **Fully Qualified Address**, enter and select the SIP user extension and domain name to match the login name from **Section 6.6.1**.



Select **Session Manager Profile** from left. For **Primary Session Manager, Origination Sequence** and **Termination Sequence**, select values that correspond to the applicable Session Manager and Communication Manager as shown below.

**AVAYA** Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Home User Management

Home / Users / Manage Users

**User Profile | Add** Commit & Continue

Identity Communication Profile Membership Contacts

Communication Profile Password

PROFILE SET : Primary ▾

Communication Address

PROFILES

Session Manager Profile ☒

CM Endpoint Profile ☐

Messaging Profile ☐

**SIP Registration**

\* Primary Session Manager : DR-SM

Secondary Session Manager : Start typing...

Survivability Server : Start typing...

Max. Simultaneous Devices : Select

Block New Registration When Maximum Registrations Active? : ☐

**Application Sequences**

Origination Sequence : DR220-CM-APP-Sequence

Termination Sequence : DR220-CM-APP-Sequence

Scroll down to **Home Location** and select the Contact Center location from **Section 6.2**. Retain the default values in the remaining fields.

**AVAYA** Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾

Home User Management

**Call Routing Settings**

\* Home Location : TMI-Loc

Conference Factory Set : Select

Select **CM Endpoint Profile** from left. For **System**, select value that corresponds to applicable Communication Manager. For **Template**, select “9620SIP\_DEFAULT\_CM\_8\_1”. For **Extension**, enter the SIP user extension from **Section 6.6.1**. Retain the default values in the remaining fields.

**Avaya Aura System Manager 8.1**

Home / User Management / Manage Users

**User Profile | Add**

Identity | **Communication Profile** | Membership | Contacts

Communication Profile Password

PROFILE SET : Primary

Communication Address

PROFILES

Session Manager Profile ☐

**CM Endpoint Profile** ☒

Messaging Profile ☐

\* System: DR-CM

\* Profile Type: Endpoint

Use Existing Endpoints: ☐

\* Extension: 66991

\* Template: 9620SIP\_DEFAULT\_CM\_8\_1

\* Set Type: 9620SIP

Security Code: Enter Security Code

Port: IP

Voice Mail Number:

Preferred Handle: Select

Calculate Route Pattern: ☒

SIP Trunk: aar

SIP URI: Select

Delete on Unassign from User or on Delete User: ☒

Override Endpoint Name and Localized Name: ☒

Allow H.323 and SIP Endpoint Dual Registration: ☐

Commit & Continue | **Commit** | Cancel

Repeat **Section 6.6** to add the desired number of SIP users for agents and supervisors. In the compliance testing, three SIP users were created for use by two agents and one supervisor as shown below.

**Avaya Aura System Manager 8.1**

Home / User Management / User Management

Search

| View                     | Edit       | New          | Duplicate       | Delete          | More Actions | Options |
|--------------------------|------------|--------------|-----------------|-----------------|--------------|---------|
| First Name               | Surname    | Display Name | Login Name      | SIP Handle      |              |         |
| <input type="checkbox"/> | SIP 2      | Avaya        | Avaya, SIP 2    | 66002@dr220.com | 66002        |         |
| <input type="checkbox"/> | Agent 1    | TMI          | TMI, Agent 1    | 66991@dr220.com | 66991        |         |
| <input type="checkbox"/> | Agent 2    | TMI          | TMI, Agent 2    | 66992@dr220.com | 66992        |         |
| <input type="checkbox"/> | Supervisor | TMI          | TMI, Supervisor | 66995@dr220.com | 66995        |         |



## 6.7. Administer Session Manager Entity

Select **Elements** → **Routing** → **SIP Entities** from the top menu to display the **Routing** tab, followed by the applicable SIP entity for Session Manager from the left pane (not shown), in this case “DR-SM”. The **SIP Entity Details** screen is displayed.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾ Search

Home Routing

R...

### SIP Entity Details

Commit Cancel

**General**

\* **Name:** DR-SM

\* **IP Address:** 10.64.101.238

**SIP FQDN:**

**Type:** Session Manager ▾

**Notes:** TLT DR SM

**Location:** DR-Loc ▾

**Outbound Proxy:** ▾

**Time Zone:** America/New\_York ▾

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

**Credential name:**

Scroll down to **Listen Ports** sub-section and verify that the transport protocol to be used by Contact Center agents and supervisors is specified in the list, in this case “**TCP**”. Also verify that the corresponding **Endpoint** column is checked, as shown below.

**AVAYA**  
Aura® System Manager 8.1

Users ▾ Elements ▾ Services ▾ | Widgets ▾ Shortcuts ▾ Search

Home Routing

R...

### Listen Ports

Add Remove

3 Items

| <input type="checkbox"/> | Listen Ports | Protocol | Default Domain | Endpoint                            | Notes |
|--------------------------|--------------|----------|----------------|-------------------------------------|-------|
| <input type="checkbox"/> | 5060         | TCP ▾    | dr220.com ▾    | <input checked="" type="checkbox"/> |       |
| <input type="checkbox"/> | 5060         | UDP ▾    | dr220.com ▾    | <input checked="" type="checkbox"/> |       |
| <input type="checkbox"/> | 5061         | TLS ▾    | dr220.com ▾    | <input checked="" type="checkbox"/> |       |

Select : All, None

## 7. Configure T-Metrics Contact Center

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

- Administer Digital Phone Module
- Administer ACD Controller
- Administer ACD Agent Module
- Administer SIP Softphone

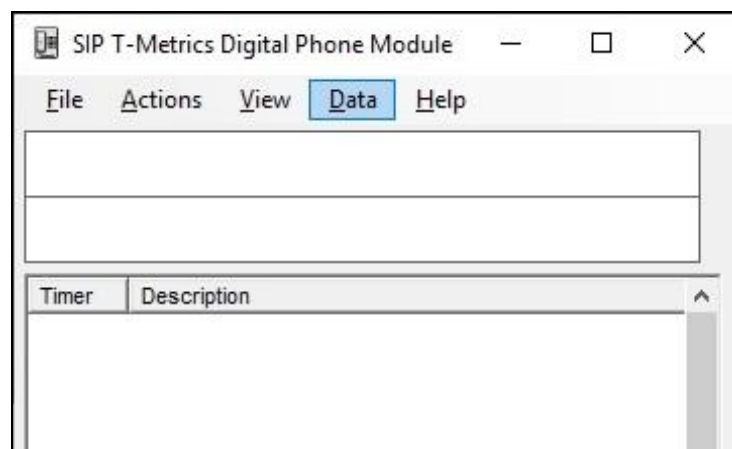
The configuration of Contact Center is performed by T-Metrics installers. Screenshots of integration related configuration are shown in these Application Notes for information purposes only.

### 7.1. Administer Digital Phone Module

From the Contact Center server, double-click on the **Digital Phone Module – SIP** shortcut icon shown below, which was created as part of Contact Center installation.



The **SIP T-Metrics Digital Phone** screen is displayed. Select **Data** from the top menu.

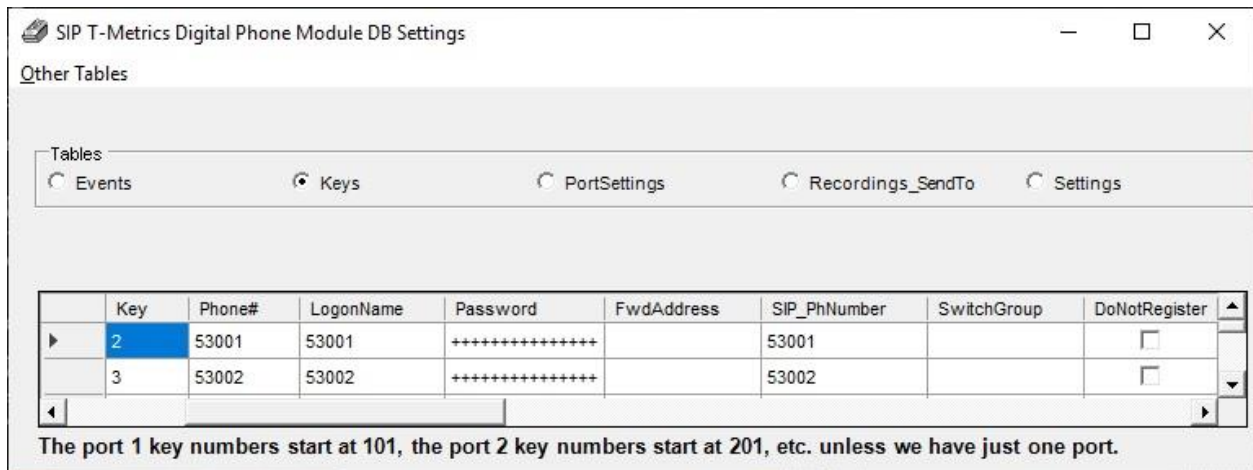




### 7.1.1. Keys

The **SIP T-Metrics Digital Phone Module DB Settings** screen is displayed. Select **Keys**.

Create an entry for each dialed number that can be routed to Contact Center and associated with a pre-configured IVR script. In the compliance testing, two IVR scripts were pre-configured and therefore two entries were created with numbers “**53001**” and “**53002**” as shown below.



The screenshot shows the 'SIP T-Metrics Digital Phone Module DB Settings' window. Under the 'Other Tables' section, the 'Keys' table is selected. The table has the following columns: Key, Phone#, LogonName, Password, FwdAddress, SIP\_PhNumber, SwitchGroup, and DoNotRegister. There are two entries in the table:

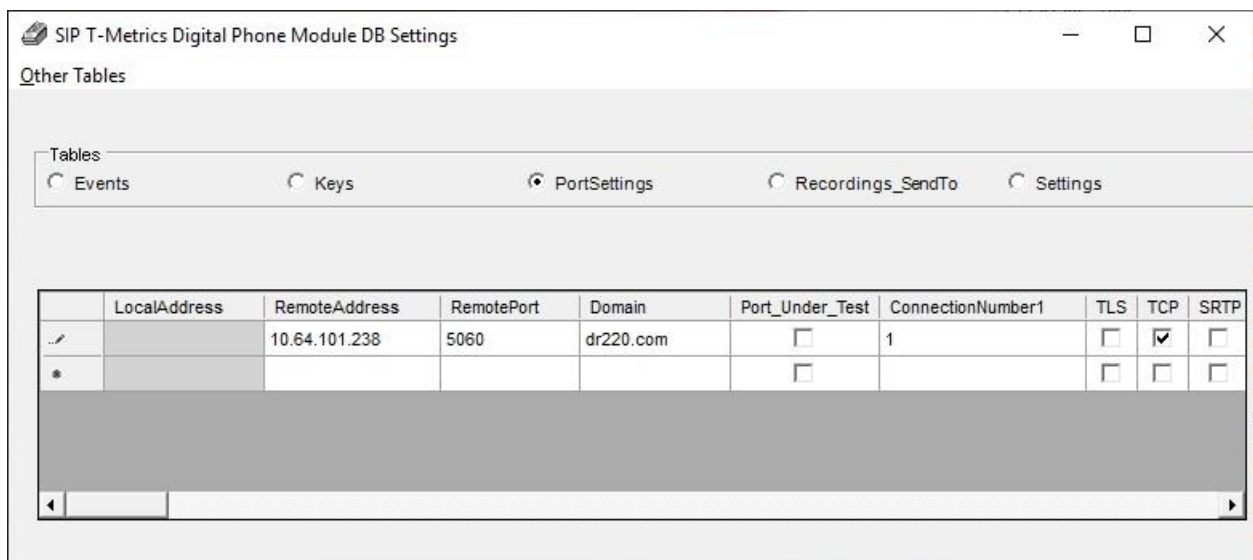
| Key | Phone# | LogonName | Password | FwdAddress | SIP_PhNumber | SwitchGroup | DoNotRegister            |
|-----|--------|-----------|----------|------------|--------------|-------------|--------------------------|
| 2   | 53001  | 53001     | +++++    |            | 53001        |             | <input type="checkbox"/> |
| 3   | 53002  | 53002     | +++++    |            | 53002        |             | <input type="checkbox"/> |

Below the table, a note states: "The port 1 key numbers start at 101, the port 2 key numbers start at 201, etc. unless we have just one port."

### 7.1.2. Port Settings

Select **PortSettings** to display the existing entry and update the specified fields as follows:

- **RemoteAddress:** IP address of Session Manager signaling interface from **Section 5.3**.
- **RemotePort:** The Contact Center SIP entity port from **Section 6.3.1**.
- **Domain:** The SIP domain name from **Section 3**.
- **TCP:** Check this field to match SIP entity protocol from **Section 6.3.1**.

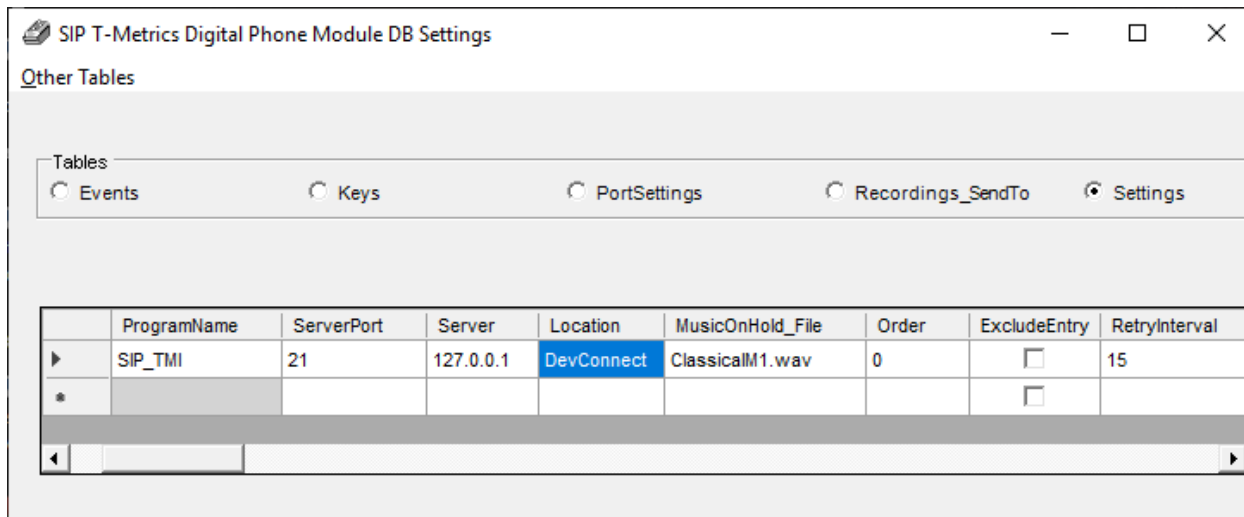


The screenshot shows the 'SIP T-Metrics Digital Phone Module DB Settings' window. Under the 'Other Tables' section, the 'PortSettings' table is selected. The table has the following columns: LocalAddress, RemoteAddress, RemotePort, Domain, Port\_Under\_Test, ConnectionNumber1, TLS, TCP, and SRTP. There are two entries in the table:

| LocalAddress | RemoteAddress | RemotePort | Domain    | Port_Under_Test          | ConnectionNumber1 | TLS                      | TCP                                 | SRTP                     |
|--------------|---------------|------------|-----------|--------------------------|-------------------|--------------------------|-------------------------------------|--------------------------|
|              | 10.64.101.238 | 5060       | dr220.com | <input type="checkbox"/> | 1                 | <input type="checkbox"/> | <input checked="" type="checkbox"/> | <input type="checkbox"/> |
|              |               |            |           | <input type="checkbox"/> |                   | <input type="checkbox"/> | <input type="checkbox"/>            | <input type="checkbox"/> |

### 7.1.3. Settings

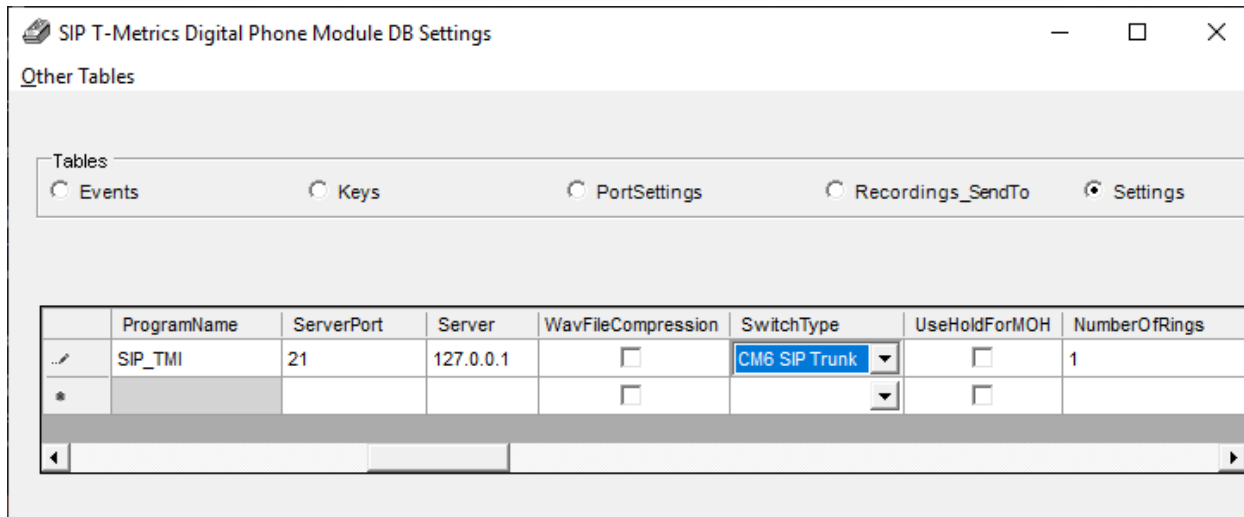
Select **Settings** to display the existing entry. Navigate to the **Location** parameter and enter a desired value, in this case “**DevConnect**”.



The screenshot shows a window titled "SIP T-Metrics Digital Phone Module DB Settings". Under the "Other Tables" section, the "Settings" table is selected. The table has the following columns: ProgramName, ServerPort, Server, Location, MusicOnHold\_File, Order, ExcludeEntry, and RetryInterval. The first row shows ProgramName: SIP\_TMI, ServerPort: 21, Server: 127.0.0.1, Location: DevConnect, MusicOnHold\_File: ClassicalM1.wav, Order: 0, ExcludeEntry: ☐, and RetryInterval: 15. A second row is partially visible with a greyed-out ProgramName and an ExcludeEntry checkbox.

|   | ProgramName | ServerPort | Server    | Location   | MusicOnHold_File | Order | ExcludeEntry             | RetryInterval |
|---|-------------|------------|-----------|------------|------------------|-------|--------------------------|---------------|
| ▶ | SIP_TMI     | 21         | 127.0.0.1 | DevConnect | ClassicalM1.wav  | 0     | <input type="checkbox"/> | 15            |
| * |             |            |           |            |                  |       | <input type="checkbox"/> |               |

Navigate to **SwitchType** and select “**CM6 SIP Trunk**”.



The screenshot shows the same window as before, but the "SwitchType" column is now visible. The first row shows ProgramName: SIP\_TMI, ServerPort: 21, Server: 127.0.0.1, WavFileCompression: ☐, SwitchType: CM6 SIP Trunk (selected from a dropdown), UseHoldForMOH: ☐, and NumberOfRings: 1. A second row is partially visible with a greyed-out ProgramName, WavFileCompression: ☐, and UseHoldForMOH: ☐.

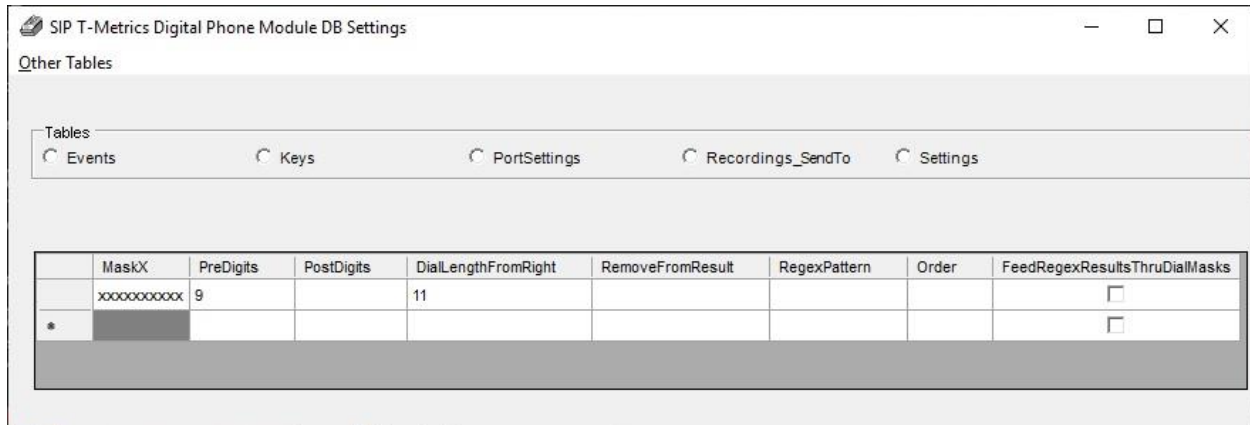
|   | ProgramName | ServerPort | Server    | WavFileCompression       | SwitchType    | UseHoldForMOH            | NumberOfRings |
|---|-------------|------------|-----------|--------------------------|---------------|--------------------------|---------------|
| ✎ | SIP_TMI     | 21         | 127.0.0.1 | <input type="checkbox"/> | CM6 SIP Trunk | <input type="checkbox"/> | 1             |
| * |             |            |           | <input type="checkbox"/> |               | <input type="checkbox"/> |               |

### 7.1.4. Dialing Mask

Select **Other Tables** → **Dialing Mask** from the top menu to display the screen below.

Create an entry to match the dialing pattern for outbound calls to the PSTN for the customer network. In the compliance testing, outbound calls were preceded with the digit “9” and the created entry is shown below.

This setting is used by the Contact Center server for IVR script transfer of calls to the PSTN.



The screenshot shows a web application window titled "SIP T-Metrics Digital Phone Module DB Settings". The "Other Tables" tab is selected. Below the tab, there are radio buttons for "Events", "Keys", "PortSettings", "Recordings\_SendTo", and "Settings". The "Keys" radio button is selected. Below the radio buttons is a table with the following columns: MaskX, PreDigits, PostDigits, DialLengthFromRight, RemoveFromResult, RegexPattern, Order, and FeedRegexResultsThruDialMasks. The table contains two rows. The first row has MaskX "xxxxxxxx", PreDigits "9", PostDigits "", DialLengthFromRight "11", RemoveFromResult "", RegexPattern "", Order "", and FeedRegexResultsThruDialMasks checkbox checked. The second row has MaskX "\*", PreDigits "", PostDigits "", DialLengthFromRight "", RemoveFromResult "", RegexPattern "", Order "", and FeedRegexResultsThruDialMasks checkbox checked.

| MaskX    | PreDigits | PostDigits | DialLengthFromRight | RemoveFromResult | RegexPattern | Order | FeedRegexResultsThruDialMasks       |
|----------|-----------|------------|---------------------|------------------|--------------|-------|-------------------------------------|
| xxxxxxxx | 9         |            | 11                  |                  |              |       | <input checked="" type="checkbox"/> |
| *        |           |            |                     |                  |              |       | <input checked="" type="checkbox"/> |

## 7.2. Administer ACD Controller

From the Contact Center server, double-click on the **ACD Controller Module** shortcut icon shown below, which was created as part of Contact Center installation.



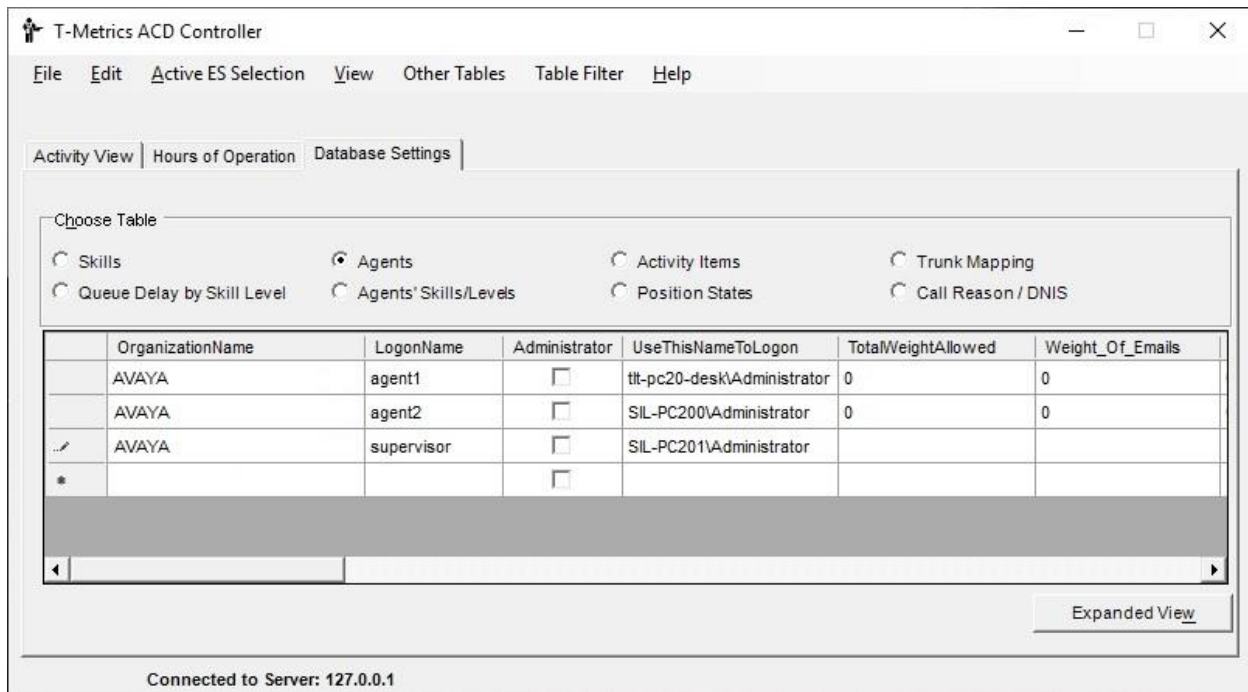
### 7.2.1. Agents

The **T-Metrics ACD Controller** screen is displayed. Select the **Database Settings** tab followed by **Agents** under **Choose Table**.

Create an entry for each agent and supervisor with desired **LogonName** and pertinent computer name and logon for **UseThisNameToLogon**.

Configure **Softphone** and **Audio Capture** (not shown) licenses for both agents and supervisors. In addition, configure **Supervisor Module** (not shown) license for supervisors.

In the compliance testing, two agents and one supervisor were created as shown below.

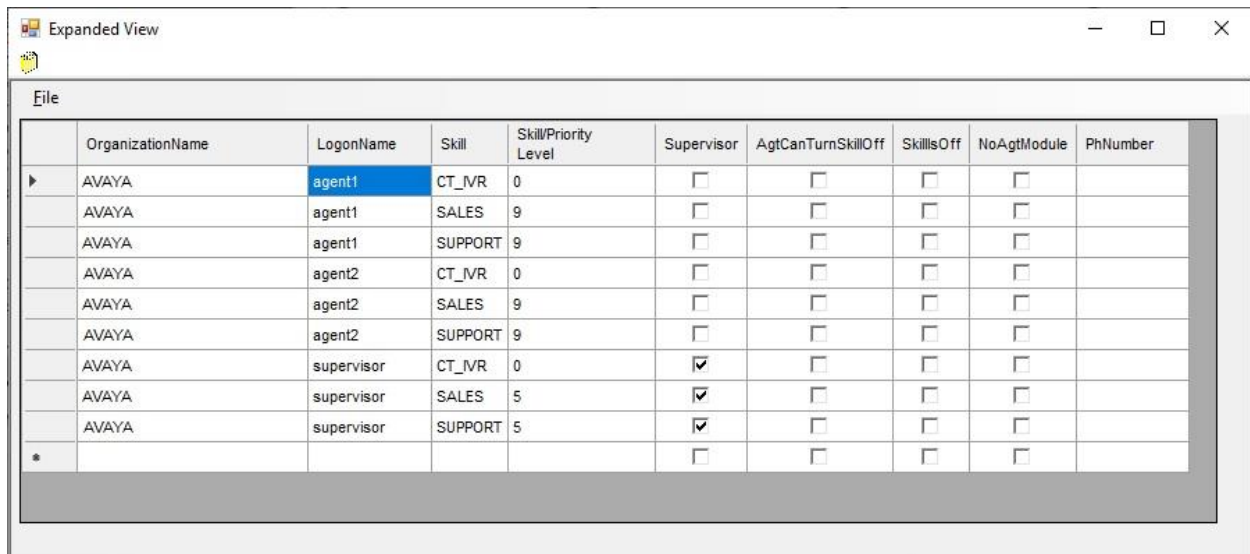
The screenshot shows the T-Metrics ACD Controller application window. The title bar reads "T-Metrics ACD Controller". The menu bar includes "File", "Edit", "Active ES Selection", "View", "Other Tables", "Table Filter", and "Help". The "Database Settings" tab is selected, showing a "Choose Table" section with radio buttons for "Skills", "Agents" (selected), "Activity Items", "Trunk Mapping", "Queue Delay by Skill Level", "Agents' Skills/Levels", "Position States", and "Call Reason / DNIS". Below this is a table with columns: "OrganizationName", "LogonName", "Administrator", "UseThisNameToLogon", "TotalWeightAllowed", and "Weight\_Of\_Emails". The table contains three rows: "AVAYA" with "agent1" and "tlt-pc20-desk\Administrator", "AVAYA" with "agent2" and "SIL-PC200\Administrator", and "AVAYA" with "supervisor" and "SIL-PC201\Administrator". A fourth row with an asterisk is also visible. The status bar at the bottom indicates "Connected to Server: 127.0.0.1".

|   | OrganizationName | LogonName  | Administrator            | UseThisNameToLogon          | TotalWeightAllowed | Weight_Of_Emails |
|---|------------------|------------|--------------------------|-----------------------------|--------------------|------------------|
|   | AVAYA            | agent1     | <input type="checkbox"/> | tlt-pc20-desk\Administrator | 0                  | 0                |
|   | AVAYA            | agent2     | <input type="checkbox"/> | SIL-PC200\Administrator     | 0                  | 0                |
|   | AVAYA            | supervisor | <input type="checkbox"/> | SIL-PC201\Administrator     |                    |                  |
| * |                  |            | <input type="checkbox"/> |                             |                    |                  |

### 7.2.2. Agent Skills

Select **Agents' Skills/Levels** (not shown) under **Choose Table**. Create an entry for each pertinent skill for each agent and supervisor.

In the compliance testing, nine entries were created to associate three pre-configured skills with each agent and supervisor as shown below.



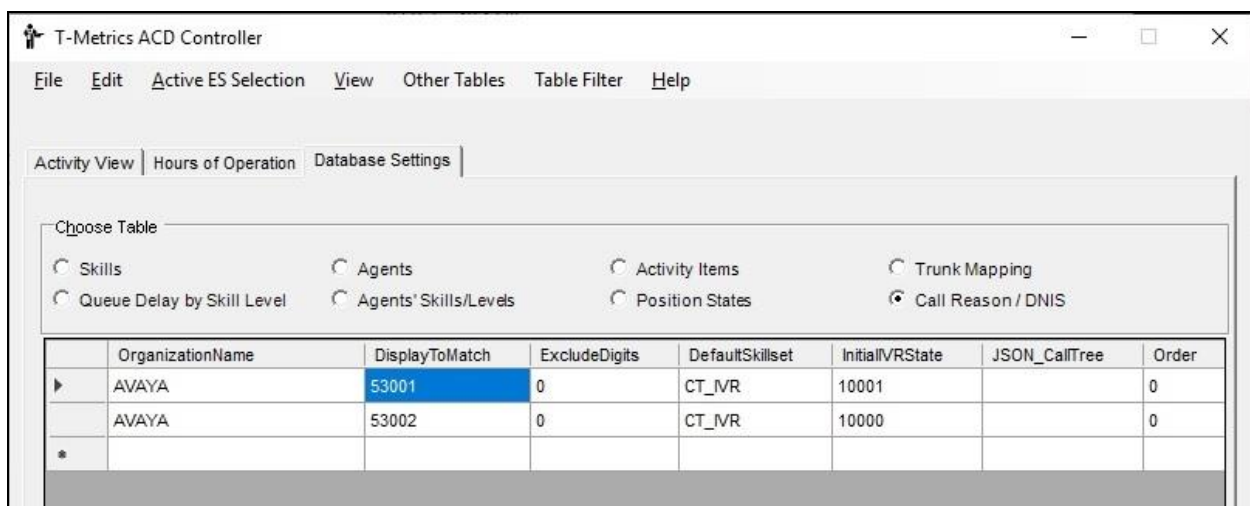
The screenshot shows a window titled "Expanded View" with a menu bar (File) and a table of Agent Skills. The table has columns: OrganizationName, LogonName, Skill, Skill/Priority Level, Supervisor, AgtCanTurnSkillOff, SkillsOff, NoAgtModule, and PhNumber. There are 10 rows of data.

|   | OrganizationName | LogonName  | Skill   | Skill/Priority Level | Supervisor                          | AgtCanTurnSkillOff       | SkillsOff                | NoAgtModule              | PhNumber |
|---|------------------|------------|---------|----------------------|-------------------------------------|--------------------------|--------------------------|--------------------------|----------|
| ▶ | AVAYA            | agent1     | CT_IVR  | 0                    | <input type="checkbox"/>            | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
|   | AVAYA            | agent1     | SALES   | 9                    | <input type="checkbox"/>            | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
|   | AVAYA            | agent1     | SUPPORT | 9                    | <input type="checkbox"/>            | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
|   | AVAYA            | agent2     | CT_IVR  | 0                    | <input type="checkbox"/>            | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
|   | AVAYA            | agent2     | SALES   | 9                    | <input type="checkbox"/>            | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
|   | AVAYA            | agent2     | SUPPORT | 9                    | <input type="checkbox"/>            | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
|   | AVAYA            | supervisor | CT_IVR  | 0                    | <input checked="" type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
|   | AVAYA            | supervisor | SALES   | 5                    | <input checked="" type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
|   | AVAYA            | supervisor | SUPPORT | 5                    | <input checked="" type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |
| * |                  |            |         |                      | <input type="checkbox"/>            | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |          |

### 7.2.3. Call Reason / DNIS

Select **Call Reason / DNIS** under **Choose Table**. Create an entry for each key extension number from **Section 7.1.1**.

In the compliance testing, two entries were created as shown below.



The screenshot shows the "T-Metrics ACD Controller" window. The "Choose Table" dialog is open, showing radio buttons for "Skills", "Agents", "Activity Items", "Trunk Mapping", "Queue Delay by Skill Level", "Agents' Skills/Levels", "Position States", and "Call Reason / DNIS". The "Call Reason / DNIS" option is selected. Below the dialog is a table of Call Reason / DNIS entries.

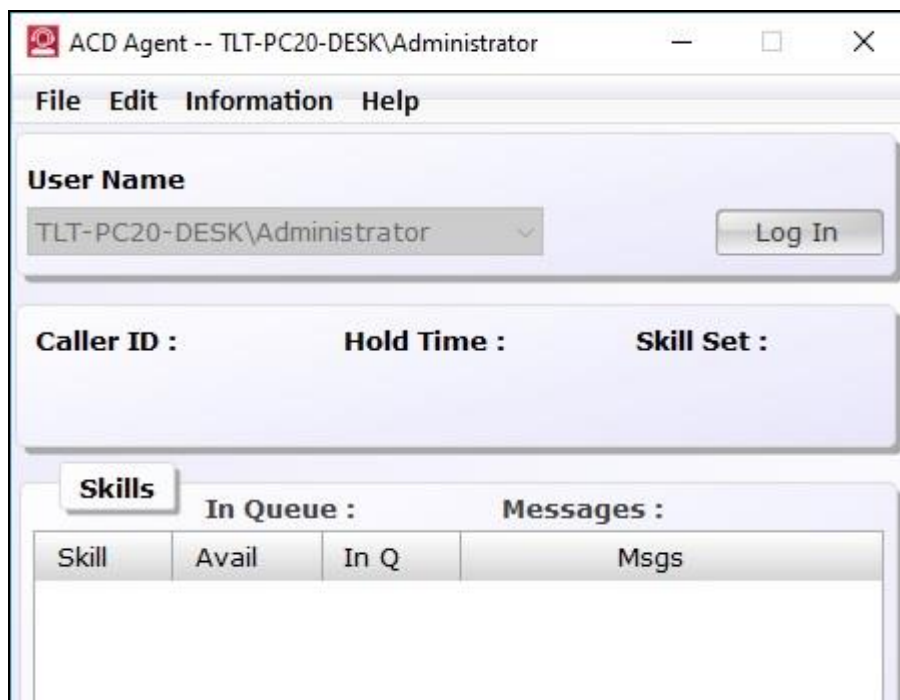
|   | OrganizationName | DisplayToMatch | ExcludeDigits | DefaultSkillset | InitialIVRState | JSON_CallTree | Order |
|---|------------------|----------------|---------------|-----------------|-----------------|---------------|-------|
| ▶ | AVAYA            | 53001          | 0             | CT_IVR          | 10001           |               | 0     |
|   | AVAYA            | 53002          | 0             | CT_IVR          | 10000           |               | 0     |
| * |                  |                |               |                 |                 |               |       |

### 7.3. Administer ACD Agent Module

From an agent or supervisor desktop, double-click on the **ACD Agent Module** shortcut icon shown below, which was created as part of ACD Agent Module installation.



The **ACD Agent** screen is displayed. Select **Edit → Settings** from the top menu.

A screenshot of the ACD Agent application window. The title bar reads "ACD Agent -- TLT-PC20-DESK\Administrator". The menu bar includes "File", "Edit", "Information", and "Help". The main interface has a "User Name" section with a dropdown menu showing "TLT-PC20-DESK\Administrator" and a "Log In" button. Below this are fields for "Caller ID :", "Hold Time :", and "Skill Set :". At the bottom, there is a "Skills" tab and a table with columns "Skill", "Avail", "In Q", and "Msgs".

| Skill | Avail | In Q | Msgs |
|-------|-------|------|------|
|       |       |      |      |

The **Options – Contact Center Agent** screen is displayed. Enter the following values for the specified fields and retain default values for the remaining fields.

- **Server Address:** IP address of the Contact Center server.
- **Interface:** “T-Metrics Softphone/Console”
- **Primary ACD DN:** The pertinent agent or supervisor user extension from **Section 6.6**.

Options -- Contact Center Agent

**Connections**

- Bubble Forms
- Data Grids
- Greeting Files
- Instant Messages
- Licensing
- Recordings
- Sound Devices
- Trouble Reports
- Unified Communications
- User Interaction

**Connections and Miscellaneous Settings**

This screen allows you to setup connections to TM-2000 servers. These connections are shown in the order of their precedence and will allow you to log into the ACD Agent Module.

**Available Connections**

- Default Connection

New Delete Move Up Move Down

Set Default Connections

**Connection Description**

Description: Default Connection ?

Server Address: 10.64.101.203 ?

Connection State: Active ?

**Telephony Interface Details**

Interface: T-Metrics Softphone/Console ?

Primary ACD DN: 66991 ?

ACD Redirects: Enabled ?

☐ Auto-Answer ?

Switch Group: 1 ?

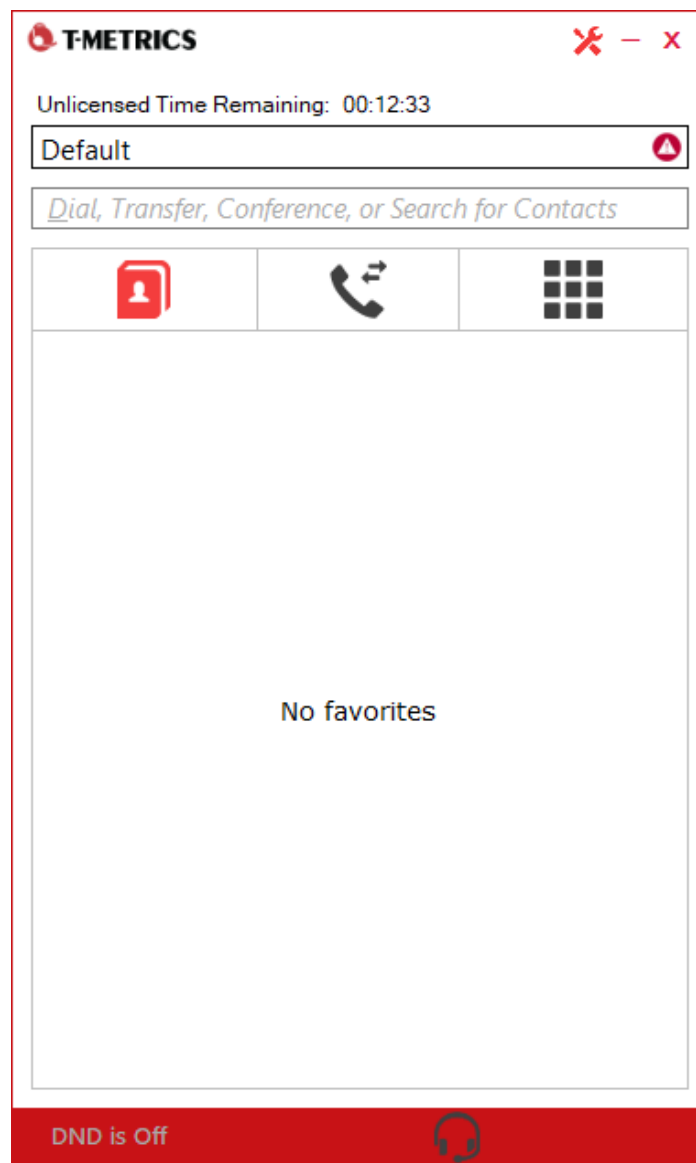
Save Cancel Apply

## 7.4. Administer SIP Softphone

From the same agent or supervisor desktop, double-click on the **SIP Softphone** shortcut icon shown below, which was created as part of SIP Softphone installation.



The **T-METRICS** screen below is displayed. Select the **Settings** icon.





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

- **Server Address:** IP address of Session Manager signaling interface from **Section 5.3**.
- **Domain:** The domain name from **Section 3**.
- **Provider Type:** “Avaya Communication Manager”
- **Use TCP:** Check this field.
- **User\_ID:** The pertinent SIP user credentials from **Section 6.6**.
- **Password:** The pertinent SIP user credentials from **Section 6.6**.

Make certain **Use Network Conferencing** is unchecked. When unchecked, the SIP Softphone will accomplish the conference feature by local bridging of audio paths with other parties. Upon SIP Softphone user leaving the conference, SIP REFER will be used by the SIP Softphone to connect the other parties to each other.

For **Maximum Concurrent Call**, set to the same number of call appearances on the corresponding SIP user in Session Manager. Note that the default number of call appearances for each SIP user on Session Manager is three.

**SIP Softphone Settings**

**SIP Connection Details**

Server Address: \* 10.64.101.238

Server Port: 5060

Domain: \* dr220.com

Provider Type: Avaya Communication Manage ▾

☐ Use Network Conferencing

Conference Bridge Address: conference

☒ Use TCP \* Required Field

[Show Advanced Switch Options](#)

**Account Details**

Account: Default ▾

[Add...](#) | [Rename...](#) | [Delete...](#)

User ID: \* 66991

Authentication ID:

Password: •••••

Maximum Concurrent Calls: 3

**T-METRICS** Save Cancel

## 8. Verification Steps

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

### 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 5.6**. Verify that all ports are in the “**in-service/idle**” state as shown below.

```
status trunk 53

 TRUNK GROUP STATUS

Member Port Service State Mtce Connected Ports
 Busy

0053/0001 T00087 in-service/idle no
0053/0002 T00113 in-service/idle no
0053/0003 T00114 in-service/idle no
0053/0004 T00115 in-service/idle no
0053/0005 T00155 in-service/idle no
0053/0006 T00160 in-service/idle no
0053/0007 T00161 in-service/idle no
0053/0008 T00162 in-service/idle no
0053/0009 T00163 in-service/idle no
0053/0010 T00164 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 5.7**. Verify that the **Group State** is “**in-service**” as shown below.

```
status signaling-group 53

 STATUS SIGNALING GROUP

 Group ID: 53
 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).

### 8.2.1. Trunk Integration

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 Contact Center entity name from **Section 6.3.1**.

The screenshot shows the 'SIP Entity Link Monitoring Status Summary' page. The left navigation pane is open, showing 'SIP Entity Monit...' selected. The main content area has a title bar with 'SIP Entity Link Monitoring Status Summary' and a 'Help ?' link. Below the title bar is a description: 'This page provides a summary of Session Manager SIP entity link monitoring status.' The main content area is divided into two sections. The first section is titled 'SIP Entities Status for All Monitoring Session Manager Instances' and contains a 'Run Monitor' button and a timestamp 'As of 3:08 PM'. Below this is a table with 1 item. The table has columns for 'Session Manager', 'Type', and 'Monitored Entities'. The 'Monitored Entities' section has sub-columns: 'Down', 'Partially Up', 'Up', 'Not Monitored', 'Deny', and 'Total'. The data row shows 'DR-SM' as the Session Manager, 'Core' as the Type, and the following values for monitored entities: Down: 1, Partially Up: 0, Up: 9, Not Monitored: 2, Deny: 0, Total: 12. Below the table is a 'Select : All, None' dropdown. The second section is titled 'All Monitored SIP Entities' and contains a 'Run Monitor' button. Below this is a table with 10 items. The table has columns for 'SIP Entity Name' and 'T-Metrics'.

| Session Manager | Type | Monitored Entities |              |    |               |      |       |
|-----------------|------|--------------------|--------------|----|---------------|------|-------|
|                 |      | Down               | Partially Up | Up | Not Monitored | Deny | Total |
| DR-SM           | Core | 1                  | 0            | 9  | 2             | 0    | 12    |

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

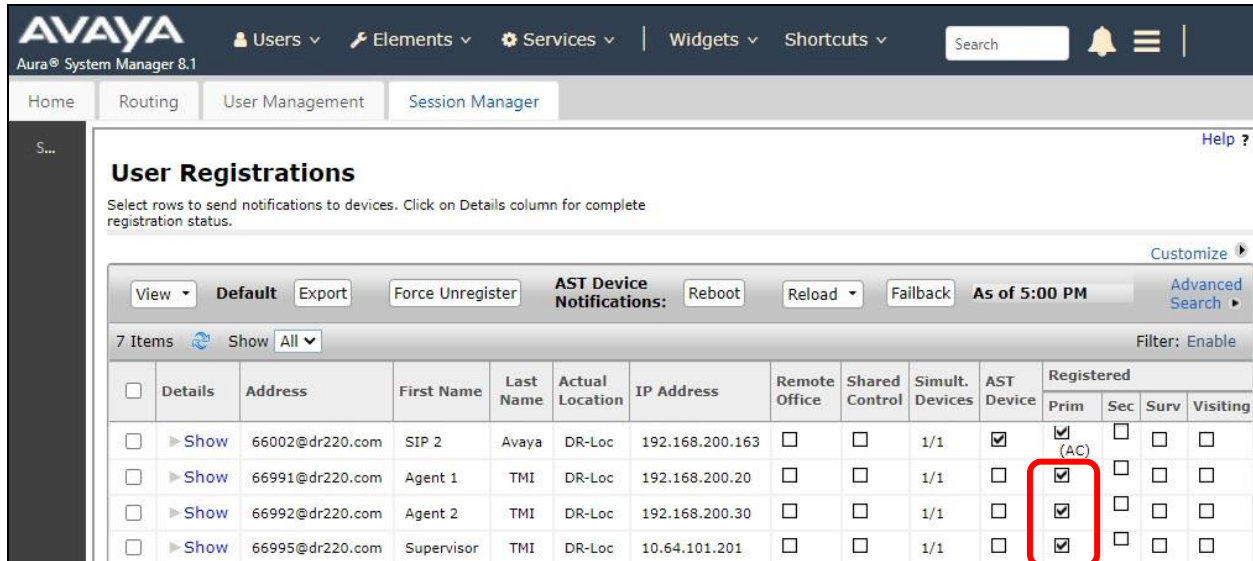
The screenshot shows the 'SIP Entity, Entity Link Connection Status' page. The left navigation pane is open, showing 'SIP Entity Monit...' selected. The main content area has a title bar with 'SIP Entity, Entity Link Connection Status' and a 'Help ?' link. Below the title bar is a description: 'This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.' The main content area is divided into two sections. The first section is titled 'Status Details for the selected Session Manager:' and contains a 'Summary View' button. Below this is a table with 1 item. The table has columns: 'Session Manager Name', 'Session Manager IP Address Family', 'SIP Entity Resolved IP', 'Port', 'Proto.', 'Deny', 'Conn. Status', 'Reason Code', and 'Link Status'. The data row shows 'DR-SM' as the Session Manager Name, 'IPv4' as the Session Manager IP Address Family, '10.64.101.203' as the SIP Entity Resolved IP, '5060' as the Port, 'TCP' as the Proto., 'FALSE' as the Deny, 'UP' as the Conn. Status, '200 OK' as the Reason Code, and 'UP' as the Link Status.

| Session Manager Name | Session Manager IP Address Family | SIP Entity Resolved IP | Port | Proto. | Deny  | Conn. Status | Reason Code | Link Status |
|----------------------|-----------------------------------|------------------------|------|--------|-------|--------------|-------------|-------------|
| DR-SM                | IPv4                              | 10.64.101.203          | 5060 | TCP    | FALSE | UP           | 200 OK      | UP          |

## 8.2.2. User Integration

Select **Elements** → **Session Manager** → **System Status** → **User Registrations** (not shown) to display the **User Registrations** screen.

Verify that all agents and supervisors with launched ACD Agent Module on their desktops are registered with Session Manager, as shown below with a check in the **Registered Prim** column.



**AVAYA** Aura® System Manager 8.1

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

Home Routing User Management Session Manager

S...

### User Registrations

Select rows to send notifications to devices. Click on Details column for complete registration status.

Customize ▾

View ▾ Default Export Force Unregister AST Device Notifications: Reboot Reload ▾ Failback As of 5:00 PM Advanced Search ▾

7 Items Show All ▾ Filter: Enable

|                          | Details | Address         | First Name | Last Name | Actual Location | IP Address      | Remote Office            | Shared Control           | Simult. Devices | AST Device                          | Registered                               |                          |                          |                          |
|--------------------------|---------|-----------------|------------|-----------|-----------------|-----------------|--------------------------|--------------------------|-----------------|-------------------------------------|------------------------------------------|--------------------------|--------------------------|--------------------------|
|                          |         |                 |            |           |                 |                 |                          |                          |                 |                                     | Prim                                     | Sec                      | Surv                     | Visiting                 |
| <input type="checkbox"/> | ► Show  | 66002@dr220.com | SIP 2      | Avaya     | DR-Loc          | 192.168.200.163 | <input type="checkbox"/> | <input type="checkbox"/> | 1/1             | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> (AC) | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | ► Show  | 66991@dr220.com | Agent 1    | TMI       | DR-Loc          | 192.168.200.20  | <input type="checkbox"/> | <input type="checkbox"/> | 1/1             | <input type="checkbox"/>            | <input checked="" type="checkbox"/>      | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | ► Show  | 66992@dr220.com | Agent 2    | TMI       | DR-Loc          | 192.168.200.30  | <input type="checkbox"/> | <input type="checkbox"/> | 1/1             | <input type="checkbox"/>            | <input checked="" type="checkbox"/>      | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |
| <input type="checkbox"/> | ► Show  | 66995@dr220.com | Supervisor | TMI       | DR-Loc          | 10.64.101.201   | <input type="checkbox"/> | <input type="checkbox"/> | 1/1             | <input type="checkbox"/>            | <input checked="" type="checkbox"/>      | <input type="checkbox"/> | <input type="checkbox"/> | <input type="checkbox"/> |

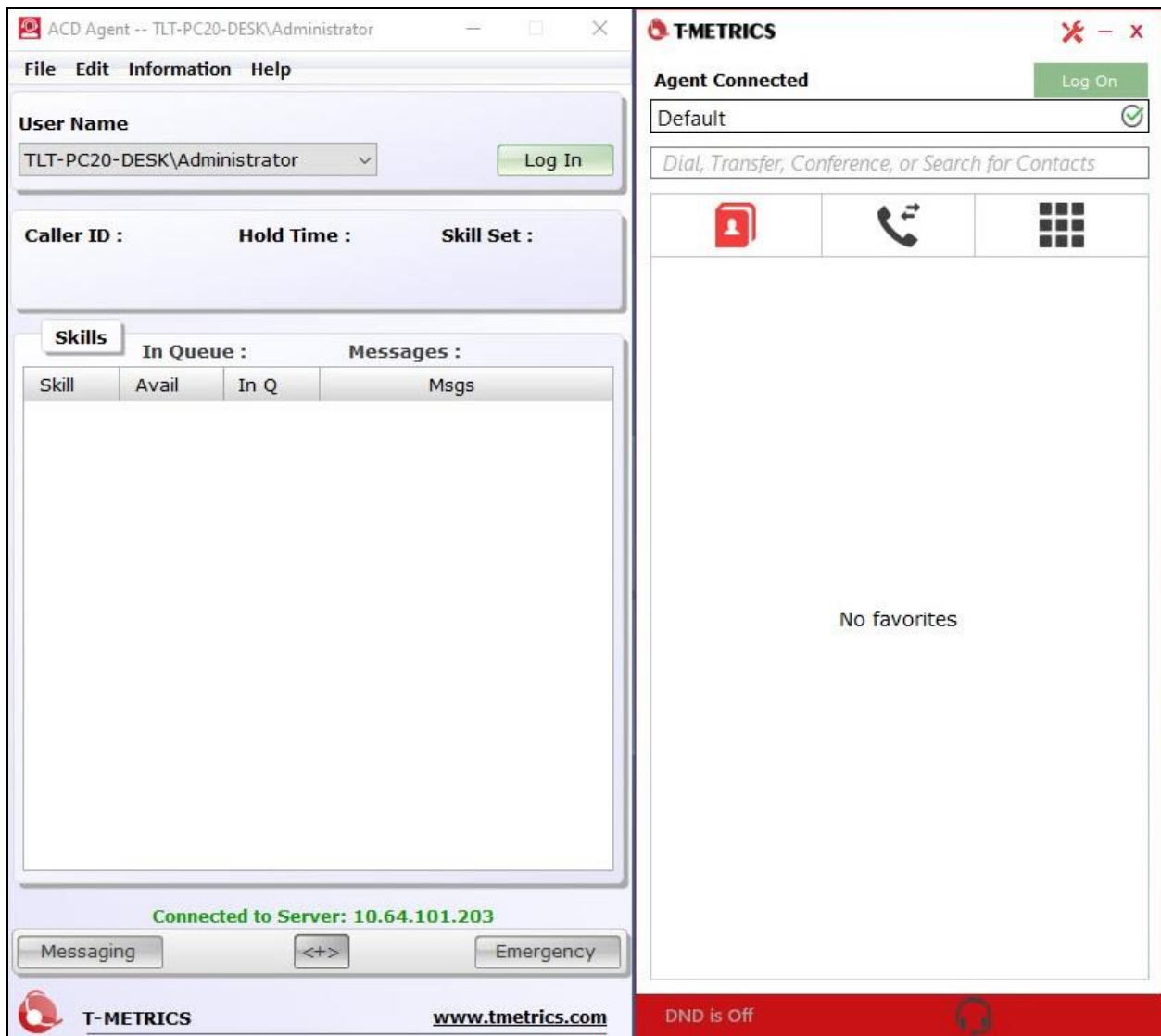
### 8.3. Verify T-Metrics Contact Center

From an agent desktop, follow the procedure in **Section 7.3** to launch the **ACD Agent Module** application, and the **SIP Softphone** application will be launched automatically as shown below.

On the **ACD Agent** screen, verify that the bottom of the screen shows **Connected to Server** as shown below.

On the **T-METRICS** screen, verify that the top of the screen shows **Agent Connected** along with a green checkmark next to **Default**, which is indication that the softphone has registered with Session Manager.

From the **ACD Agent** screen, click **Log In**.



Verify that both the **ACD Agent** and **T-METRICS** screens are updated to reflect the **Available** status in green, as shown below.

The screenshot shows two side-by-side windows. The left window is titled 'ACD Agent -- TLT-PC20-DESK\Administrator -- TLT-...' and has a menu bar with 'File', 'Edit', 'Information', and 'Help'. The status is 'Available' in green. Below the status is a dropdown menu set to 'Available' and a 'Change' button. There is a text field 'Enter details about your status here...' with a small icon. Below this are fields for 'Caller ID', 'Hold Time', and 'Skill Set'. At the bottom is a 'Skills' section with a table showing 'In Queue' and 'Messages' counts for various skills.

| Skill   | Avail | In Q | Msgs |
|---------|-------|------|------|
| CT_IVR  | 0**   | 0    | 0    |
| SALES   | 2     | 0    | 0    |
| SUPPORT | 2     | 0    | 0    |
| TOTALS  | --    | 0    | 0    |

The right window is titled 'T-METRICS' and has a status 'Available' in green. Below the status is a dropdown menu set to 'Default' and a green checkmark icon. There is a text field 'Dial, Transfer, Conference, or Search for Contacts'. Below this are three icons: a red person icon, a black telephone icon, and a black grid icon.

Make a call from the PSTN to a number that gets routed to the Contact Center and is associated with an IVR script. Verify that the PSTN caller hears the greeting from the IVR script and can use DTMF to select the narrated option for transfer to Sales.

Verify that both screens below are updated to reflect an incoming call with “**SALES**” as **Skill Set** along with calling party number in **Caller ID**. Click **Answer** on the **T-METRICS** screen.

The screenshot shows two side-by-side windows. The left window is titled 'ACD Agent -- TLT-PC20-DESK\Administrator -- TLT-...' and has a menu bar with 'File', 'Edit', 'Information', and 'Help'. The status is 'Available' in blue. Below the status is a dropdown menu set to 'Available' and a 'Change' button. There is a text field 'Enter details about your status here...' with a small icon. Below this are fields for 'Caller ID', 'Hold Time', and 'Skill Set'. At the bottom is a 'Skills' section with a table showing 'In Queue' and 'Messages' counts for various skills.

| Skill   | Avail | In Q | Msgs |
|---------|-------|------|------|
| CT_IVR  | 0**   | 0    | 0    |
| SALES   | 1**   | 0    | 0    |
| SUPPORT | 1**   | 0    | 0    |
| TOTALS  | --    | 0    | 0    |

The right window is titled 'T-METRICS' and has a status 'Available' in blue. Below the status is a red banner 'Contact Center Call Info' with the following information: 'Hold Time 0 sec', 'Skillset SALES', and 'Caller ID +12126630031'. Below the banner is a dropdown menu set to 'Default' and a green checkmark icon. There is a text field 'Dial, Transfer, Conference, or Search for Contacts'. Below this is a call information section with a yellow circle icon, the number '+12126630031', and the extension '53001'. There is a green 'Answer' button and a red 'Decline' button. Below this are three icons: a red person icon, a black telephone icon, and a black grid icon.

Verify that the **T-METRICS** screen is updated to reflect a connected call, and that the agent headset is connected to the PSTN caller with two-way talk paths.

The screenshot displays the T-METRICS ACD Agent interface. The left pane shows the agent's status as 'Available' with a 'Change' button. Below this, caller information is displayed: Caller ID: +12126630031, Hold Time: 0 sec, and Skill Set: SALES. A 'Skills' table shows the agent's availability for different skill sets. The 'Call Statistics' table shows the agent's current task and state. The right pane shows 'Contact Center Call Info' with the same caller information. Below this, a 'Default' button is visible. The bottom of the interface shows a 'Connected to Server: 10.64.101.203' status, 'Messaging' and 'Emergency' buttons, and a 'DND is Off' indicator.

**Status : Available**

Available

Enter details about your status here...

**Caller ID :** +12126630031 **Hold Time :** 0 sec **Skill Set :** SALES

**Skills**

| Skill   | Avail | In Q | Msgs |
|---------|-------|------|------|
| CT_IVR  | 0**   | 0    | 0    |
| SALES   | 1**   | 0    | 0    |
| SUPPORT | 1**   | 0    | 0    |
| TOTALS  | --    | 0    | 0    |

**Call Statistics**

| Agent  | Task | State | Status    |
|--------|------|-------|-----------|
| AGENT1 |      |       | Available |

Connected to Server: 10.64.101.203

Messaging

**T-METRICS** [www.tmetrics.com](http://www.tmetrics.com)

**Available**

**Contact Center Call Info**

**Hold Time** 0 sec  
**Skillset** SALES  
**Caller ID** +12126630031

Default ☒

Dial, Transfer, Conference, or Search for Contacts

+12126630031 00:00:59  
+12126630031 Default 1

Hold End Call

No favorites

DND is Off



## 9. Conclusion

These Application Notes describe the configuration steps required for T-Metrics Contact Center to successfully interoperate with Avaya Aura® Session Manager 8.1 and Avaya Aura® Communication 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 8, November 2020, available at <http://support.avaya.com>.
2. *Administering Avaya Aura® Session Manager*, Release 8.1.x, Issue 8, February 2021, available at <http://support.avaya.com>.
3. *Agent Manual*, available at <http://portal.tmetrics.com/OnlineSupport.aspx>.
4. *SIP Softphone User Manual*, available at <http://portal.tmetrics.com/OnlineSupport.aspx>.
5. *Avaya CM/SM SIP Programming Guide – ACD Trunks*, available at <http://portal.tmetrics.com/OnlineSupport.aspx>.
6. *Administrator Manual*, available at <http://portal.tmetrics.com/OnlineSupport.aspx>.



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