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.


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# 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.


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## 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


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# 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.

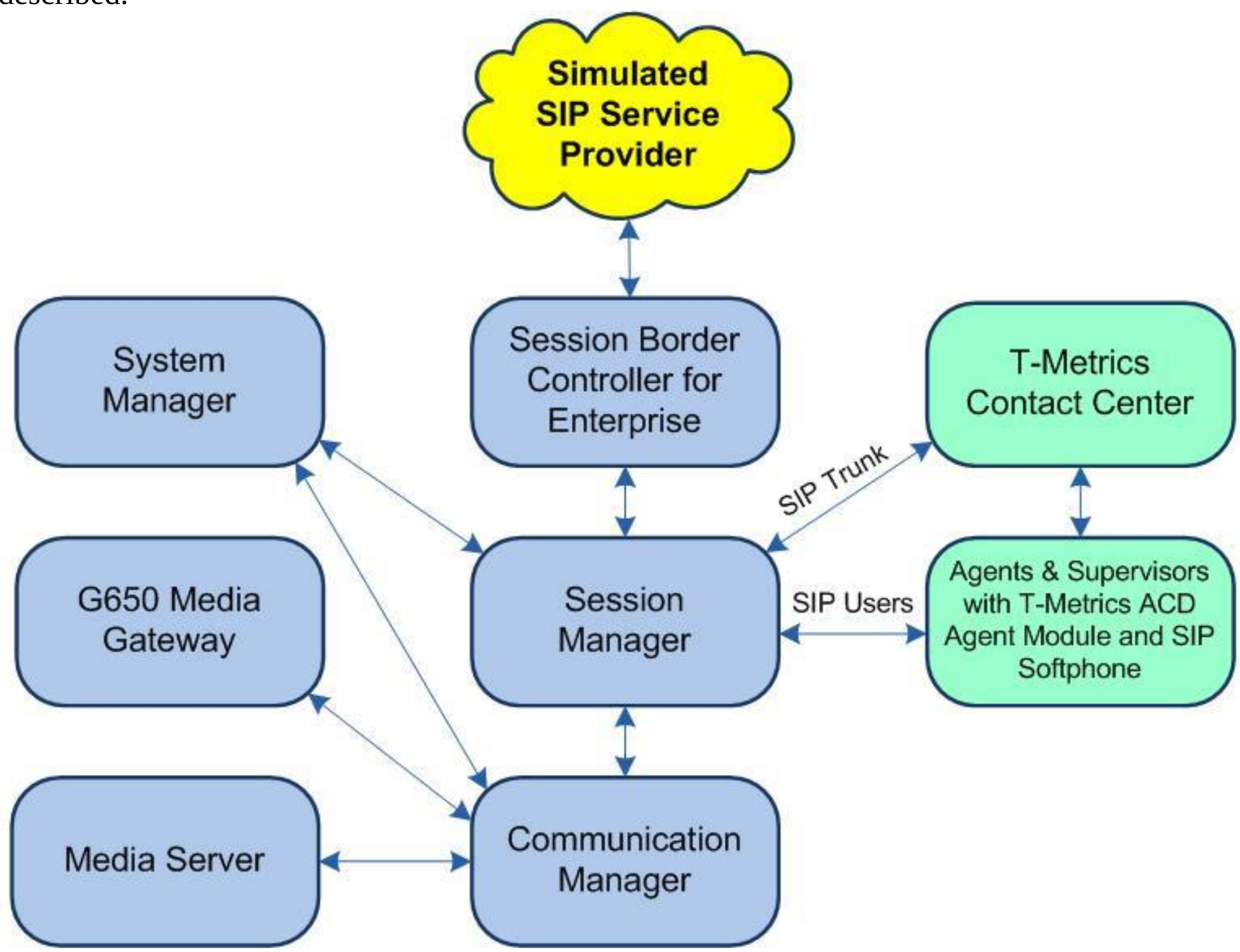


**Figure 1: Compliance Testing Configuration**


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## 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<br>• TMI ACD Controller Module<br>• TMI DigiSIP Module<br>• TMI Event Server Module | Standard<br>22 Jan 21<br>21 Jan 21<br>19 Nov 20 |
| T-Metrics ACD Agent Module & T-Metrics SIP Softphone (TMI_SIP) on Windows 10 Pro | a15 Jan 21<br>21.2.23.1 (2021.10.20.3) |


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# 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.


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## 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
```


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## 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
```


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## 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
```


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## 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
   UDP Port Min: 2048                       IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

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
   UDP Port Min: 2048                       IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```


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## 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
 Dial Access? n                                  Night Service:
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
                                                UUI 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
```


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## 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
                                               Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                    RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload              Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                       IP Audio Hairpinning? n
         Enable Layer 3 Test? y                      Initial IP-IP Direct Media? n
H.323 Station Outgoing Direct Media? n               Alternate Route Timer(sec): 6
```


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## 5.8. Administer SIP Trunk Group Members

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.

- **Signaling Group:** The signaling group number from **Section 5.7**.
- **Number of Members:** The desired number of members, in this case "10".

```
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
```

## 5.9. Administer Route Pattern

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.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.6**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.
- **Numbering Format:** "lev0-pvt"

```
change route-pattern 53                                            Page   1 of   4
                    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
```


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## 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          Total      Route     Call   Node  ANI
          String         Min  Max   Pattern    Type   Num   Reqd
   53                    5    5      53        aar          n
```


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## 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.

```
change trunk-group 212                                         Page   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


                                Modify Tandem Calling Number: tandem-cpn-form
               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**".

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


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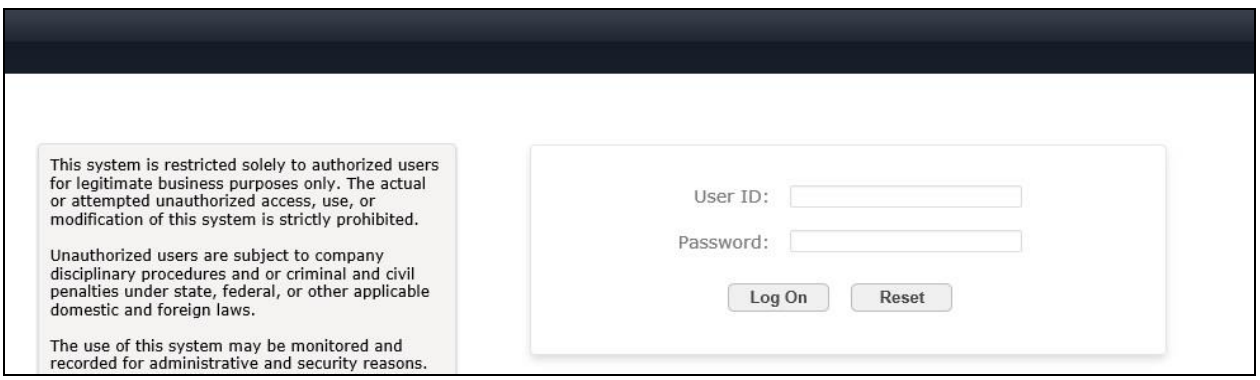
# 6. Configure Avaya Aura® Session Manager

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:

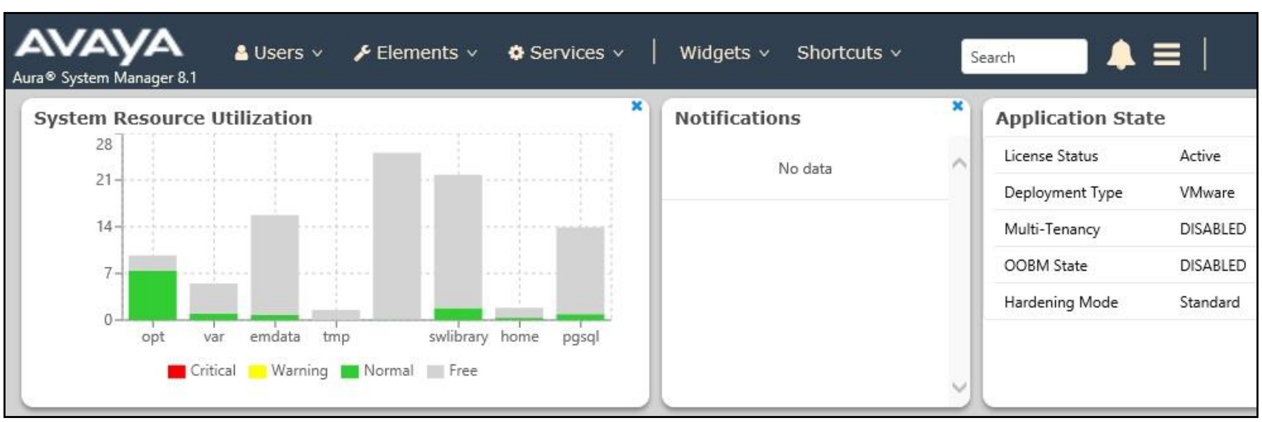
- Launch System Manager
- Administer locations
- Administer SIP entities
- Administer routing policies
- Administer dial patterns
- Administer SIP users
- Administer Session Manager entity

## 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.





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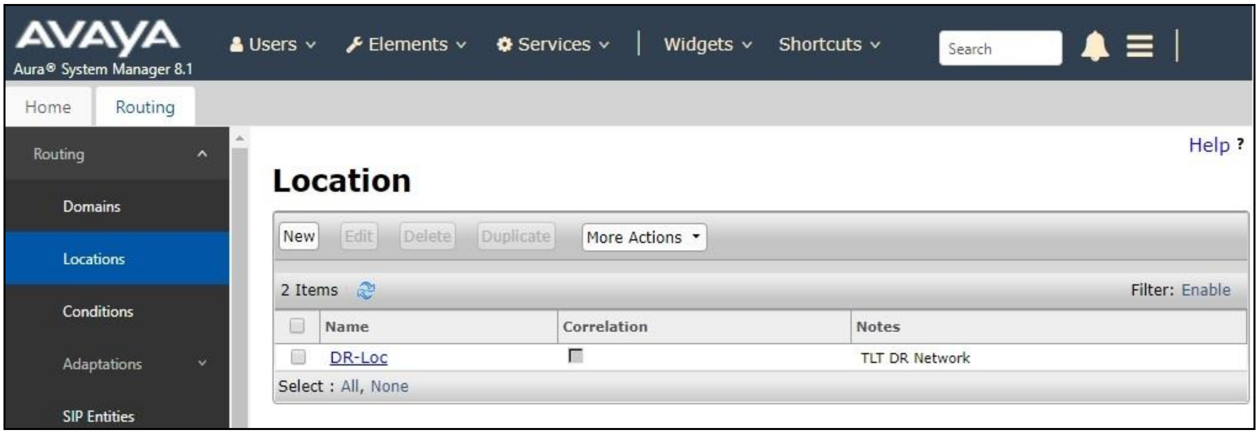
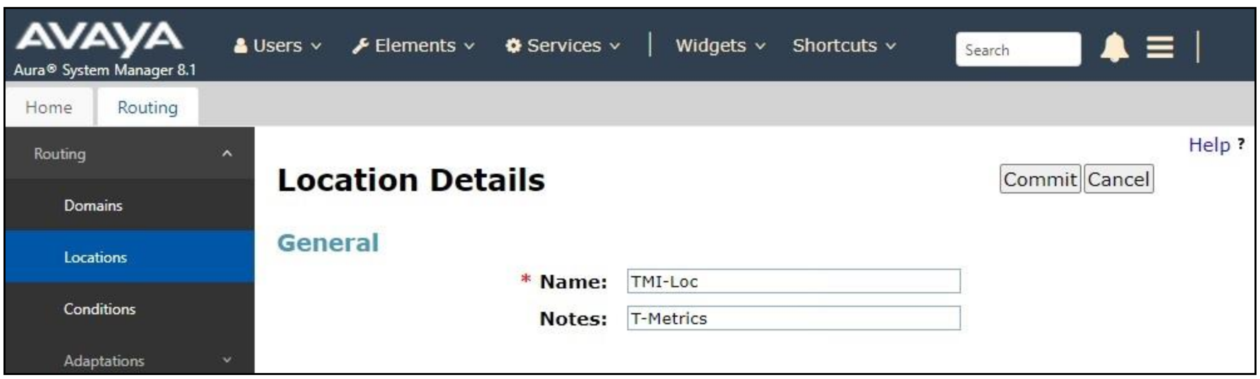
## 6.2. Administer Locations

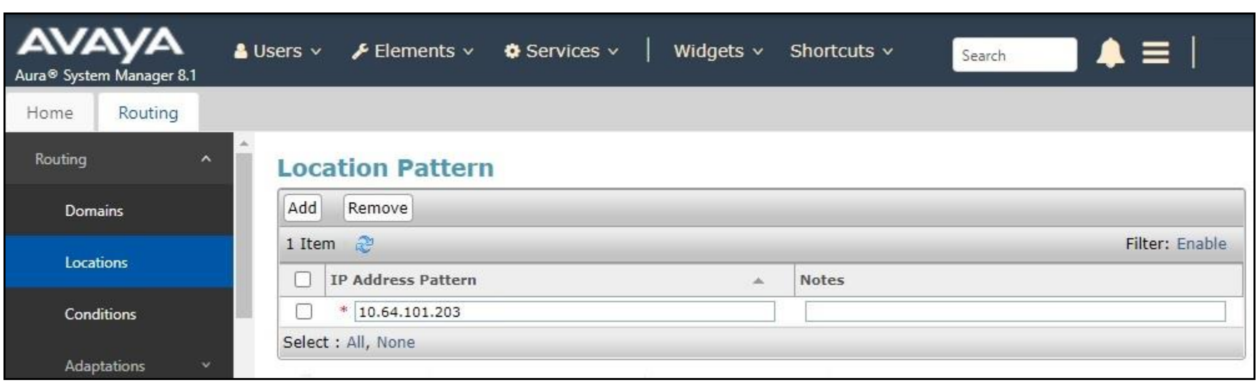
Select **Elements → Routing → Locations** from the top menu to display the **Location** screen below. Select **New** to add a new location for Contact Center.




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.





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## 6.3. Administer SIP Entities

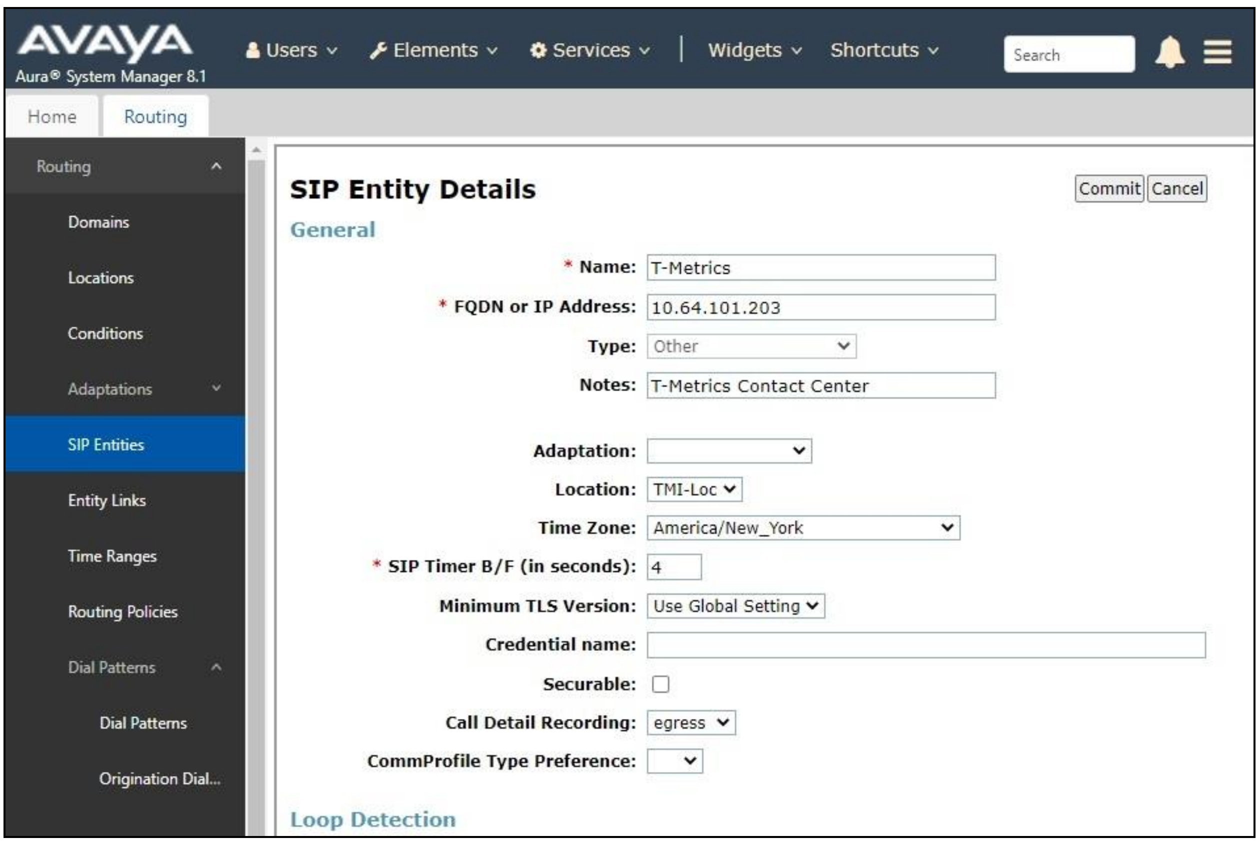
Add two SIP entities, one for Contact Center and one for the new SIP trunk with Communication Manager.

### 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.



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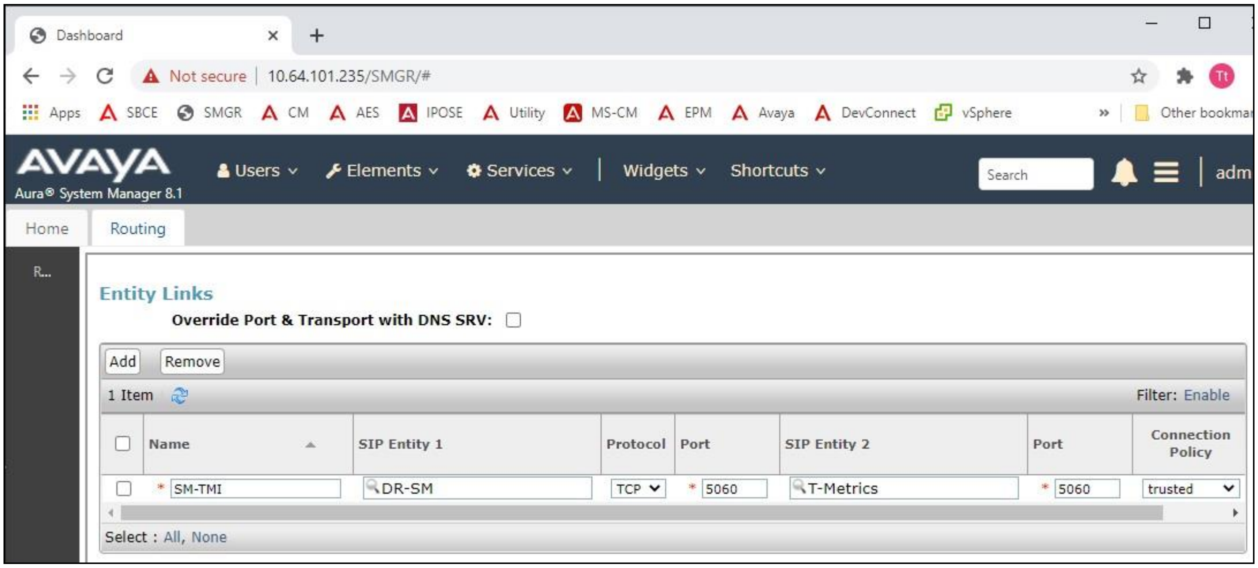
Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **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"





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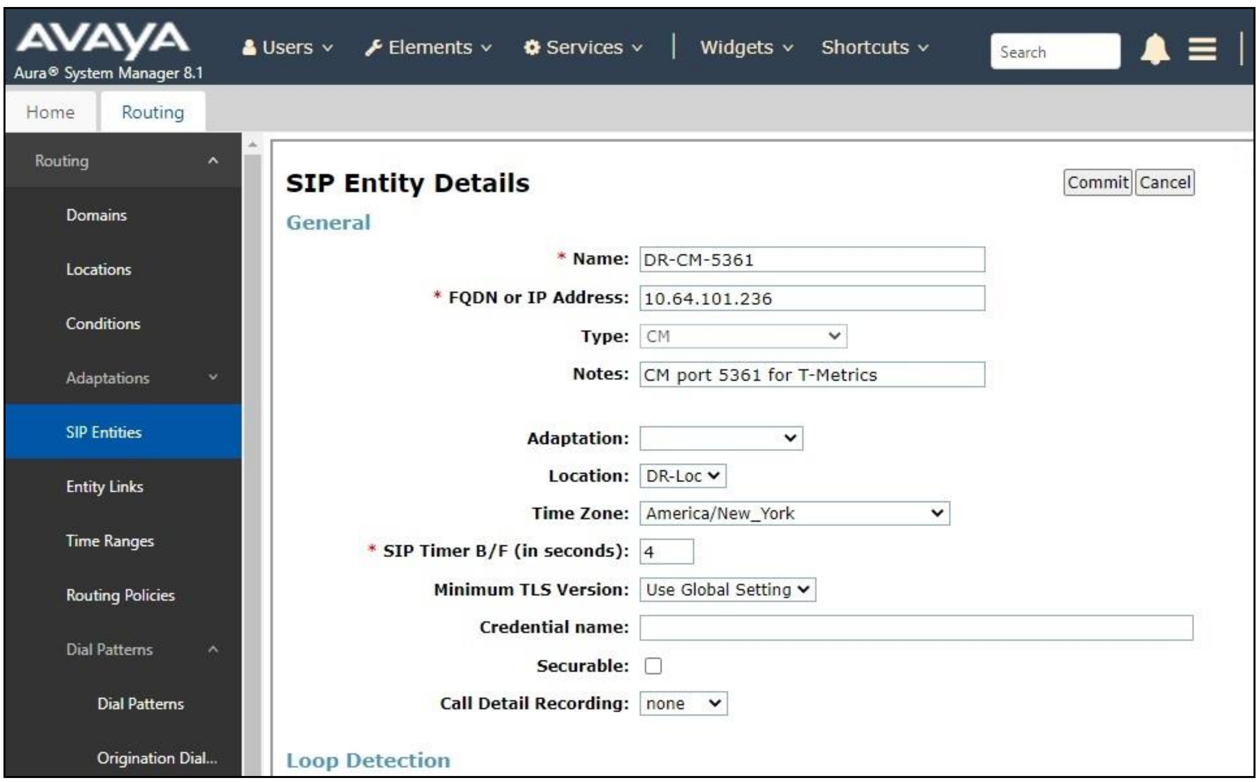
### 6.3.2. SIP Entity for Communication Manager

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.





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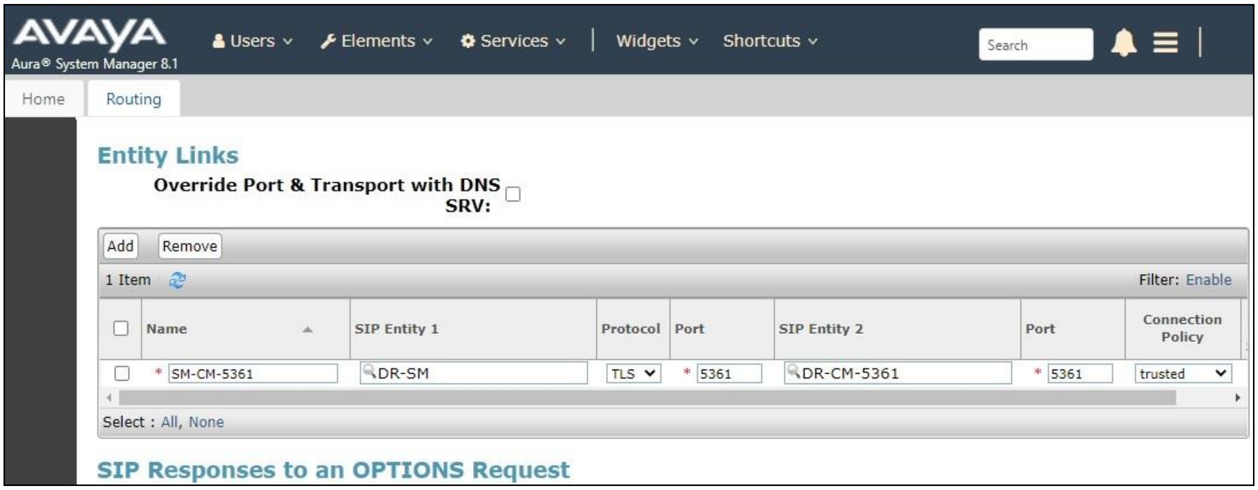
Scroll down to the **Entity Links** sub-section and click **Add** to add an entity link. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **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"





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## 6.4. Administer Routing Policies

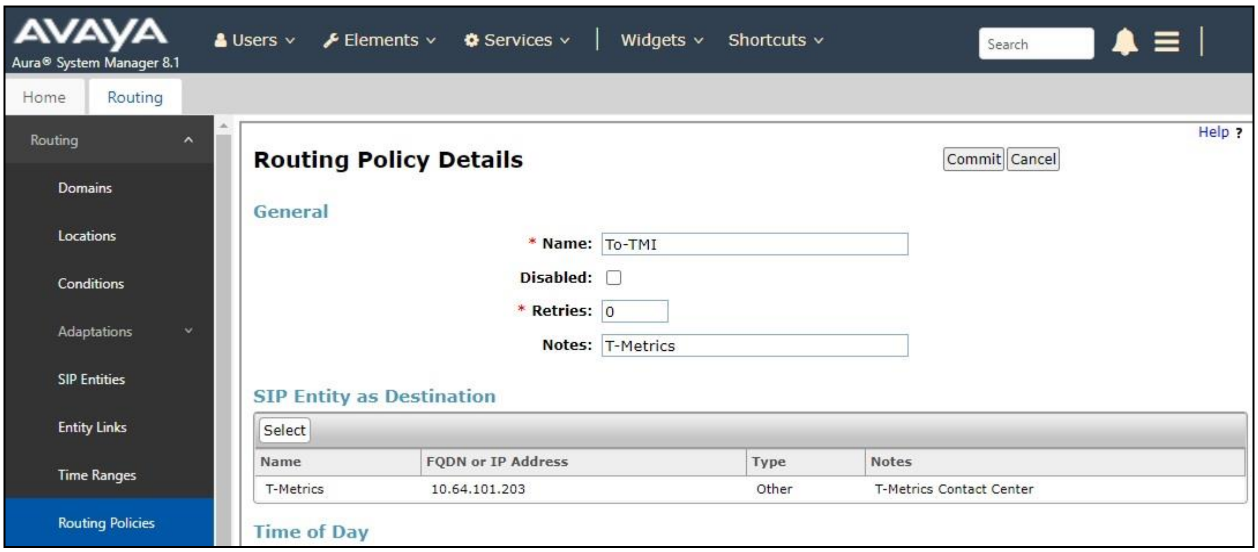
Add two routing policies, one for Contact Center and one for the new SIP trunk with Communication Manager.

### 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.



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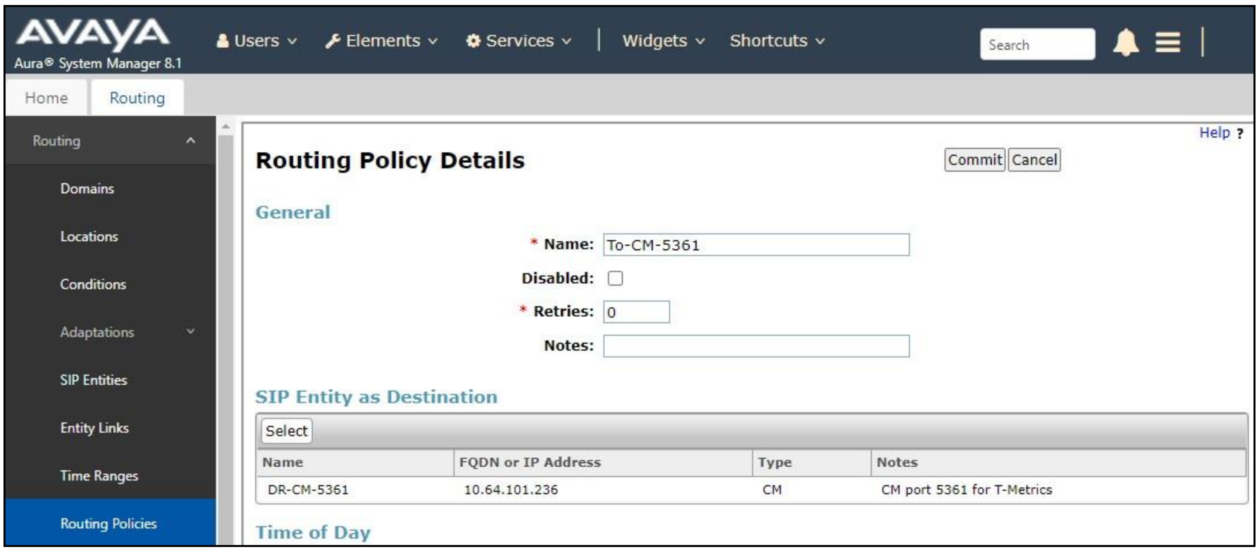
### 6.4.2. Routing Policy for Communication Manager

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.





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## 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.

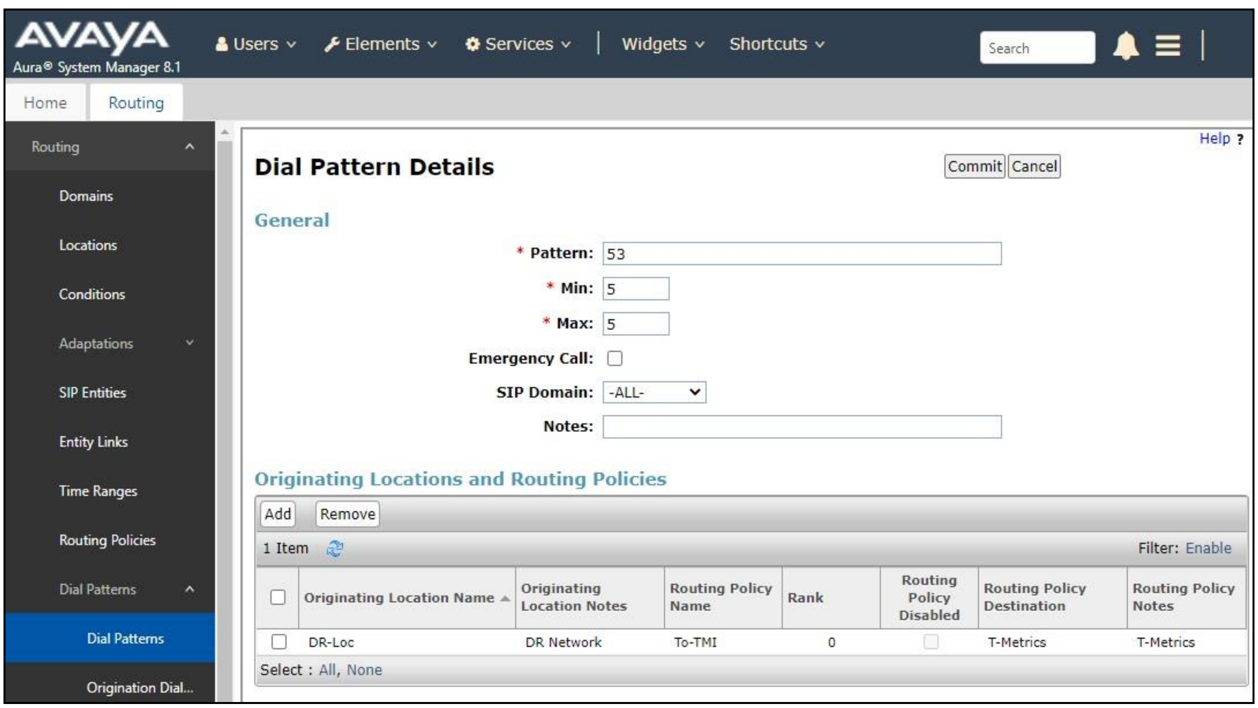
### 6.5.1. Dial Pattern for Contact Center

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.





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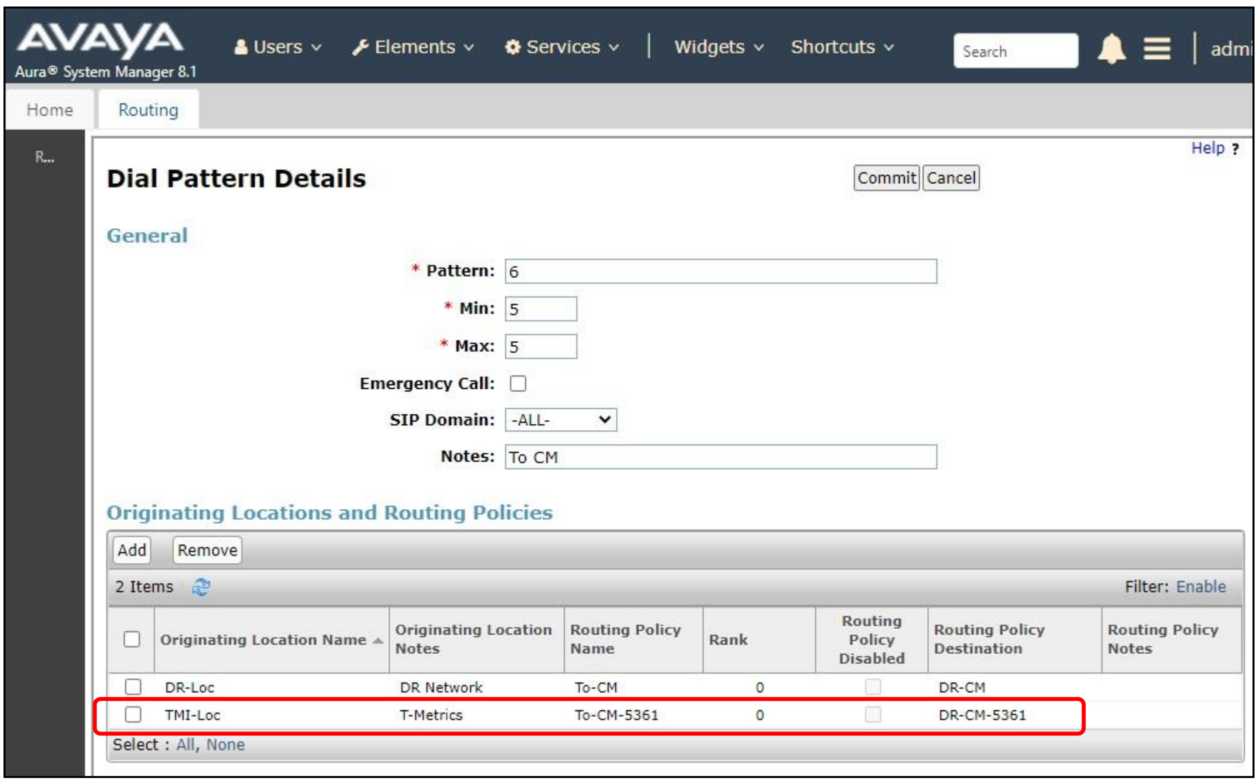
### 6.5.2. Dial Pattern for Communication Manager

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 Contact Center. In the compliance testing, the new entry allowed for call origination from the Contact Center 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, 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).





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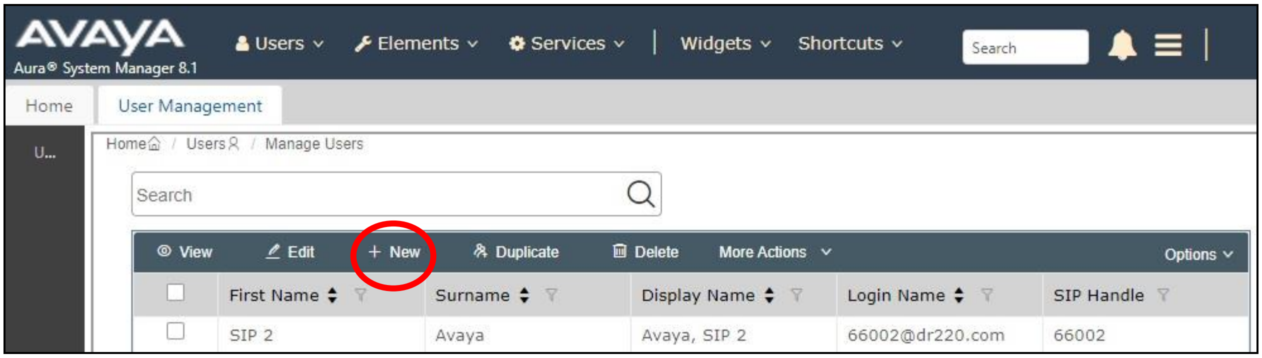
## 6.6. Administer SIP Users

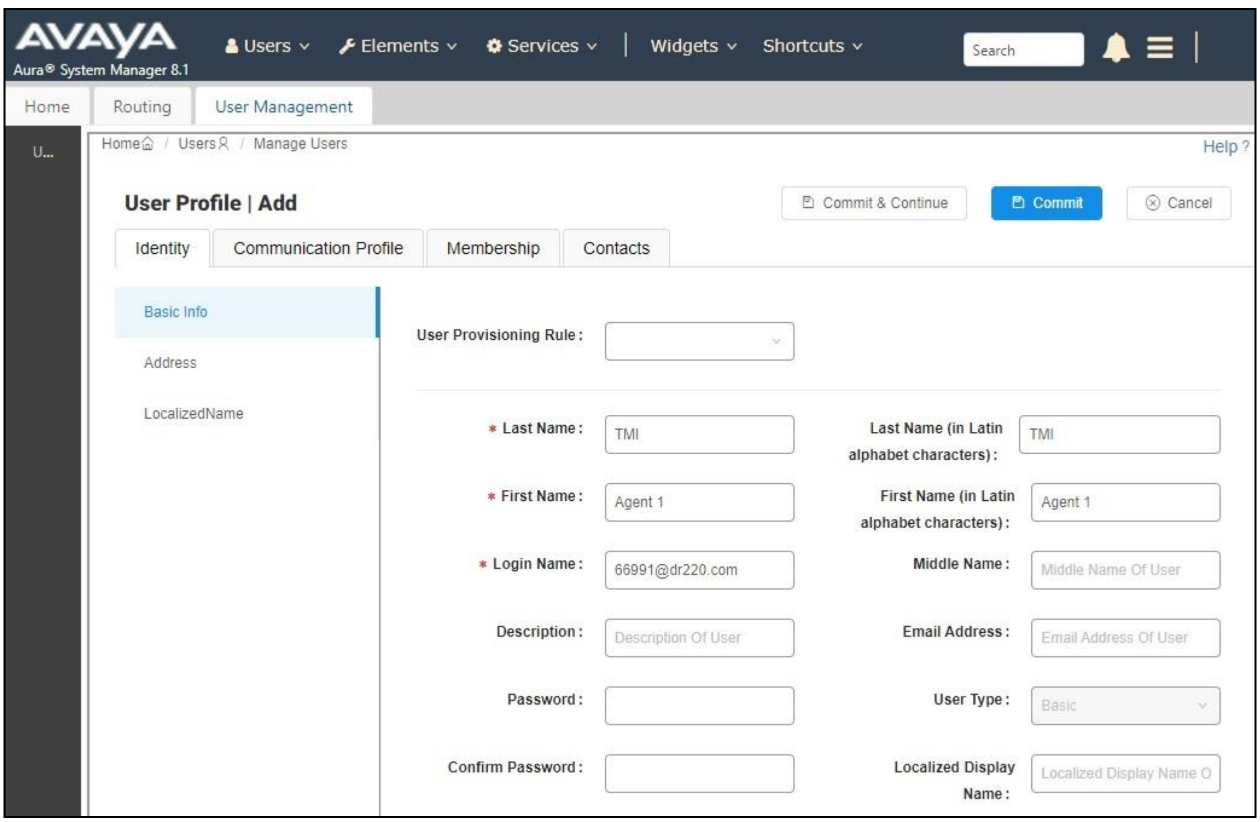
Create a SIP user for each Contact Center agent and supervisor. Select **Users → User Management → Manage Users** from the top menu to display the screen below. Click **New** to add a SIP user.




### 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.





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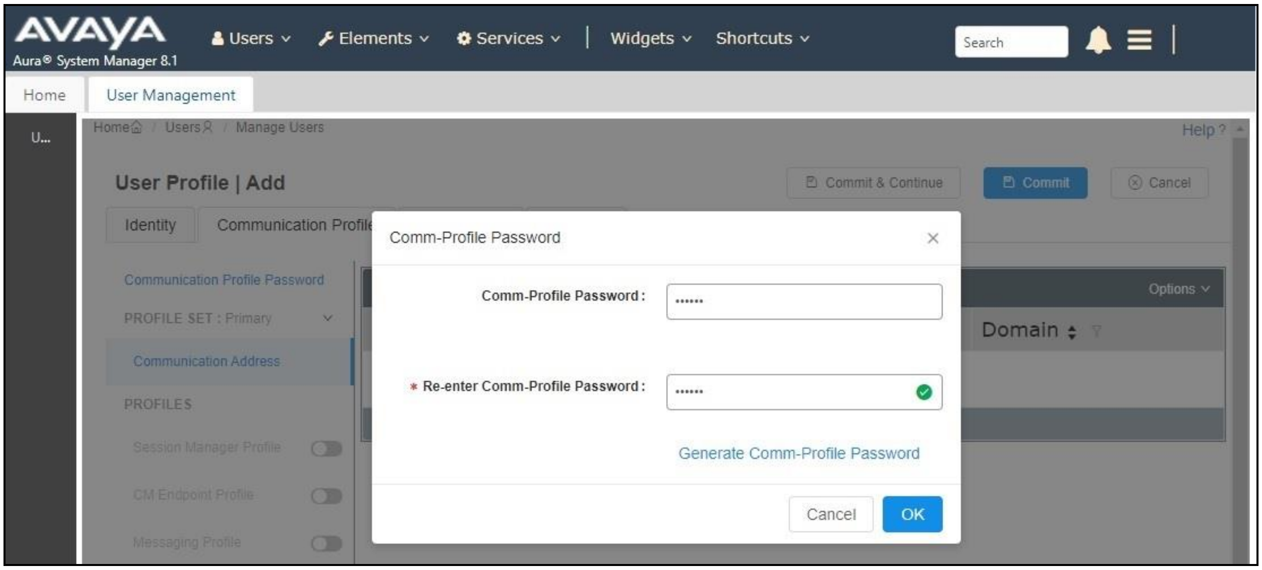
### 6.6.2. Communication Profile

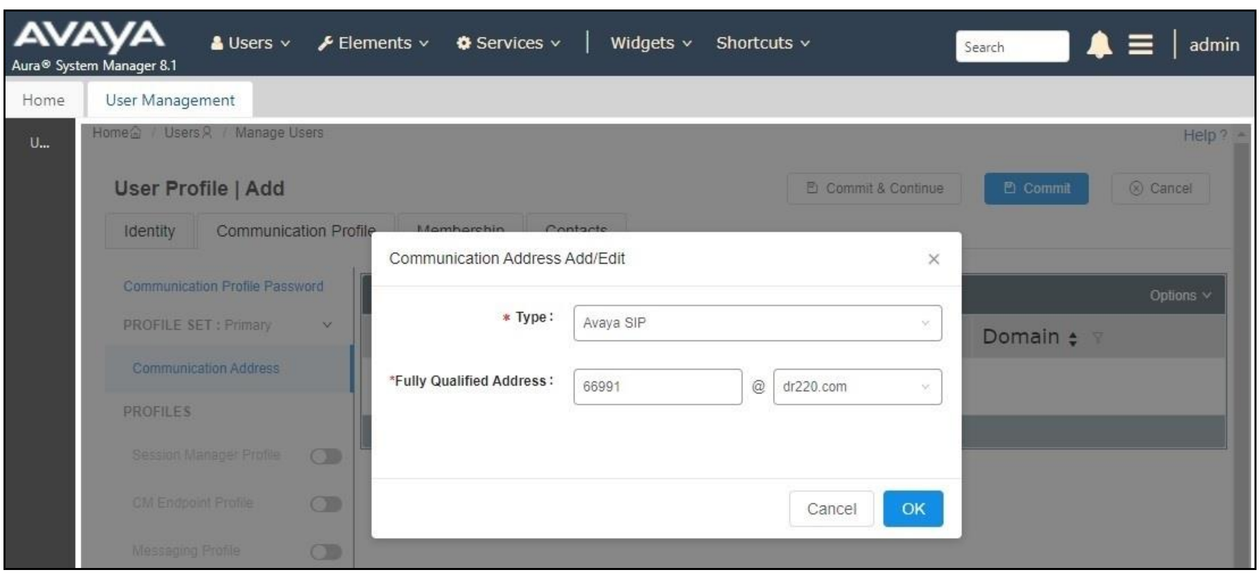
Select the **Communication Profile** tab, followed by **Communication Profile Password** to display the **Comm-Profile Password** pop-up box.

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**.





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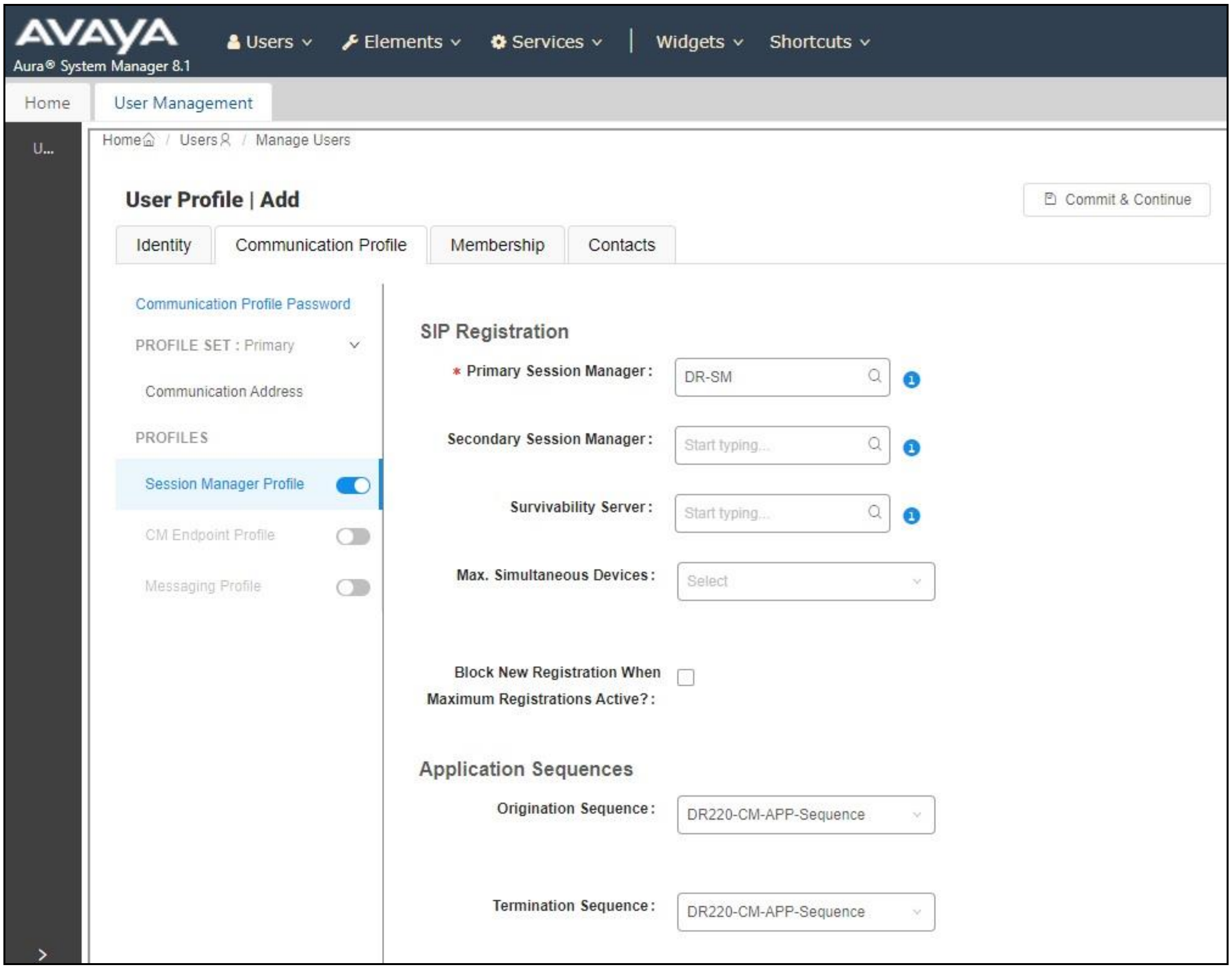
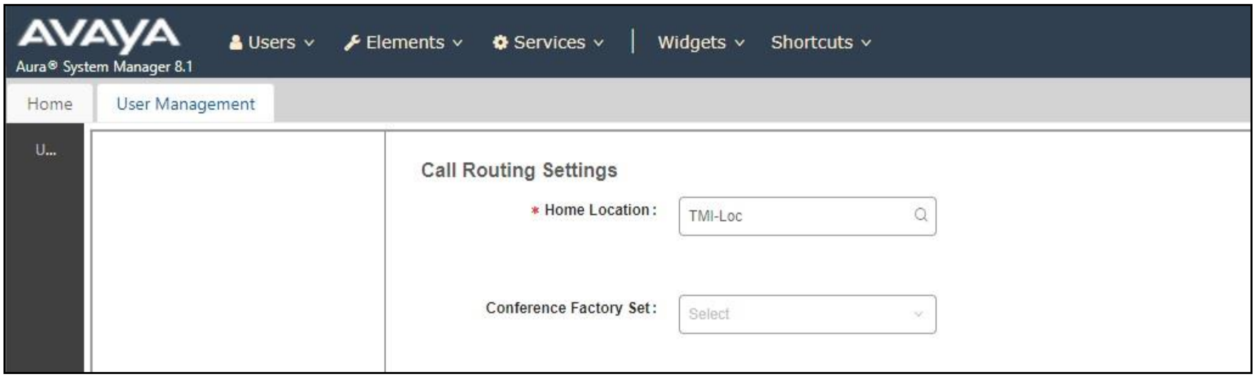
Select **Session Manager Profile** from left.  For **Primary Session Manager**, **Origination Sequence** and **Termination Sequence**, select values that correspond to the applicable Session Manager and Communication Manager as shown below.




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







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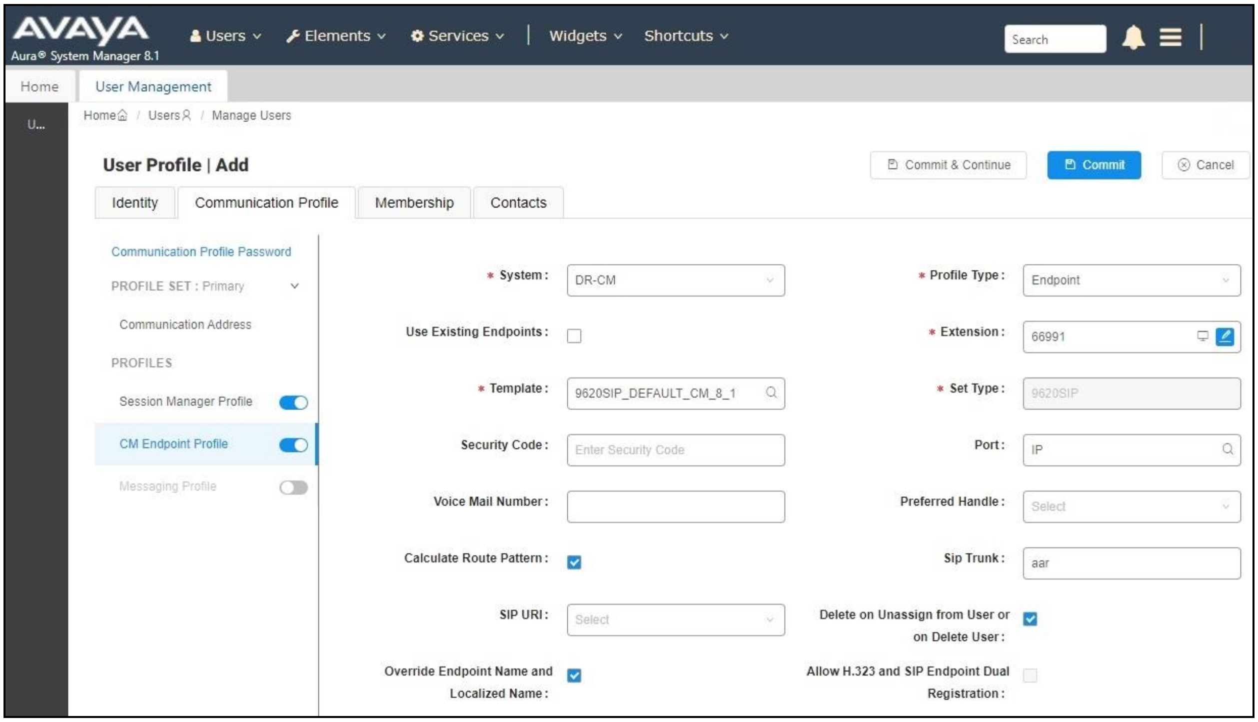
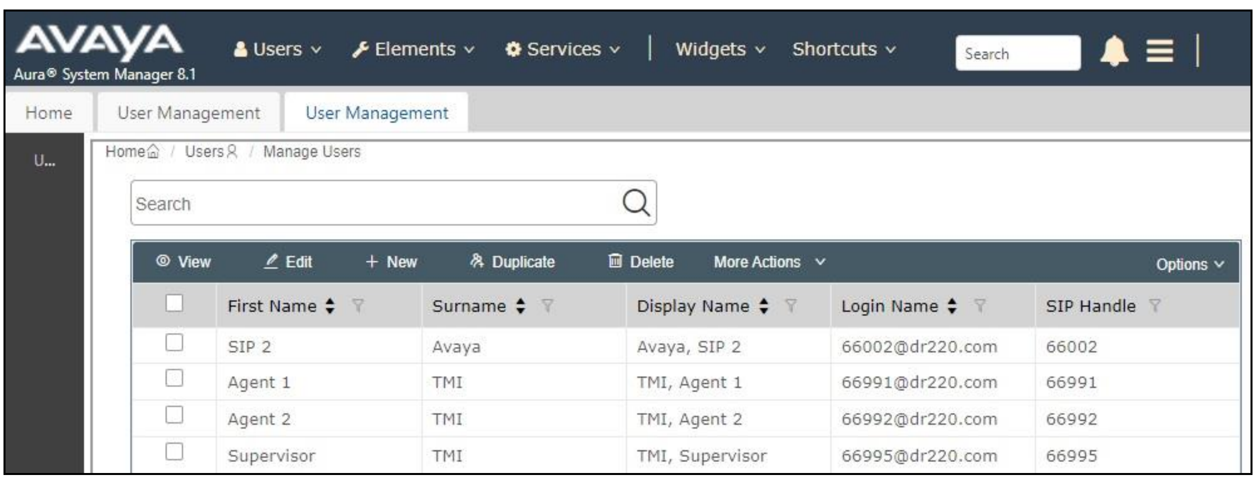
Select **CM Endpoint Profile** from left. For **System**, select value that corresponds to applicable Communication Manager. For **Template**, select "**9620SIP_DEFAULT_CM_8_1**". For **Extension**, enter the SIP user extension from **Section 6.6.1**. Retain the default values in the remaining fields.



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.





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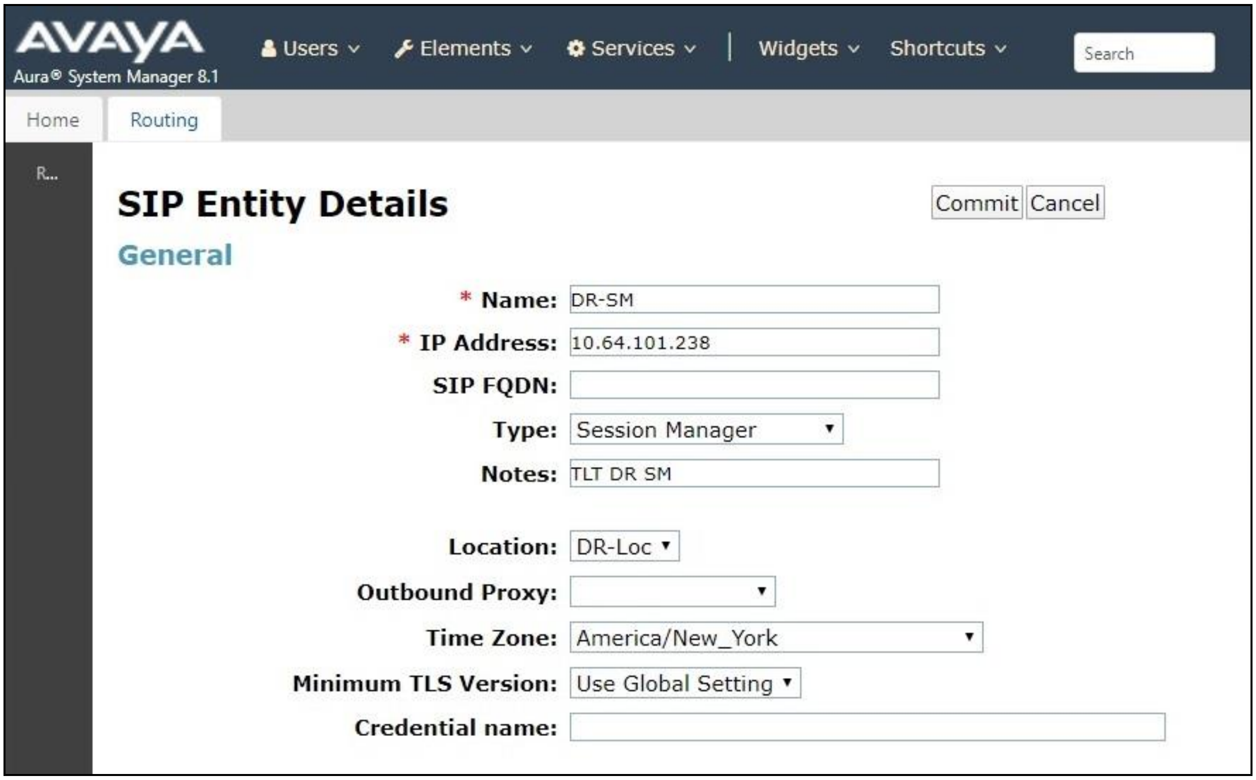
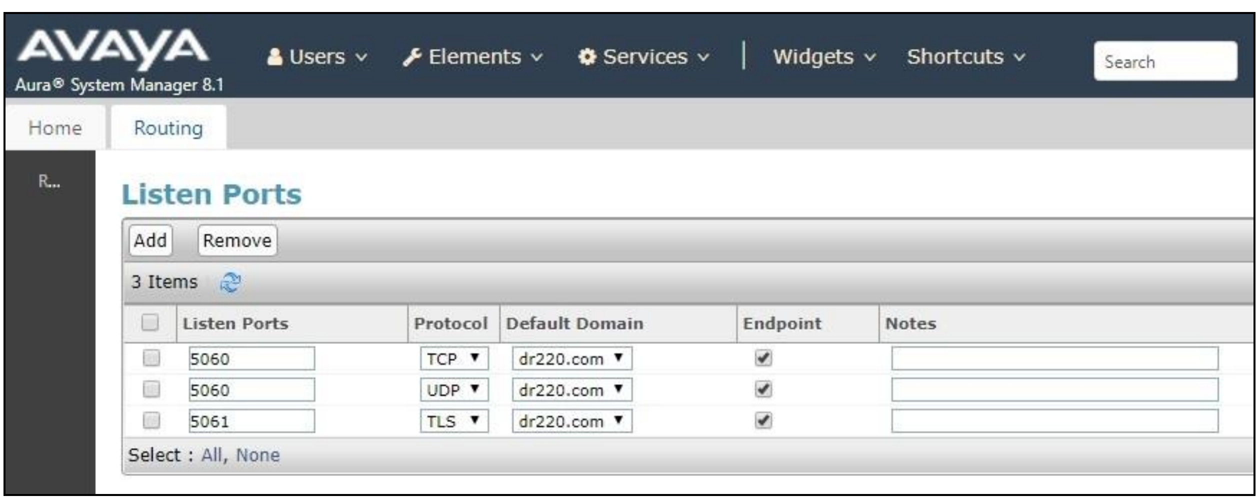
## 6.7. Administer Session Manager Entity

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.




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 thise case "**TCP**". Also verify that the corresponding **Endpoint** column is checked, as shown beow.



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# 7. Configure T-Metrics Contact Center

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

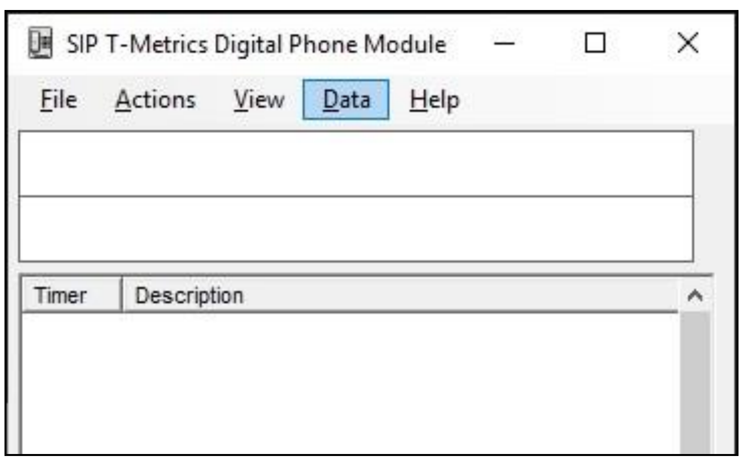
- Administer Digital Phone Module
- Administer ACD Controller
- Administer ACD Agent Module
- Administer SIP Softphone

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.





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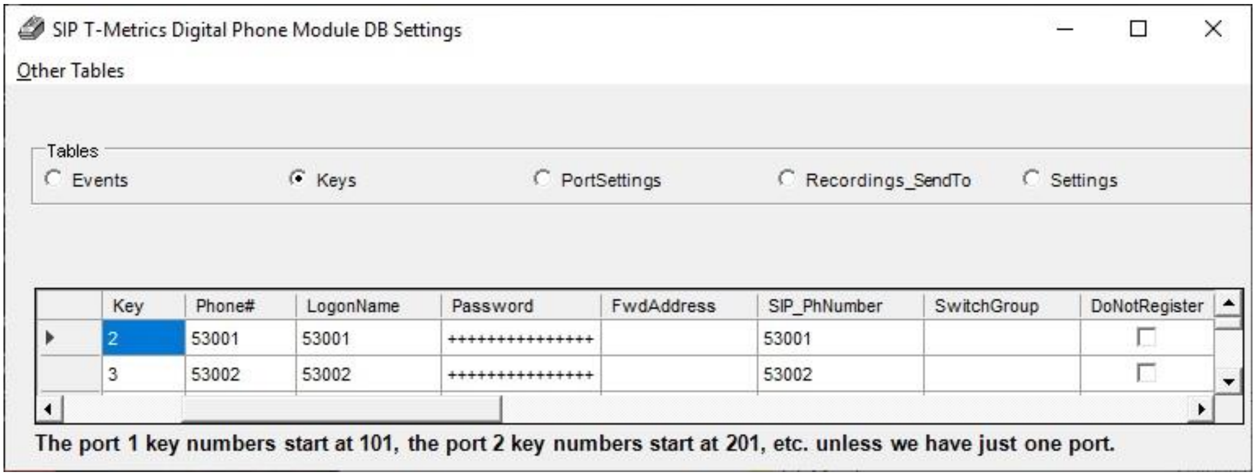
### 7.1.1. Keys

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

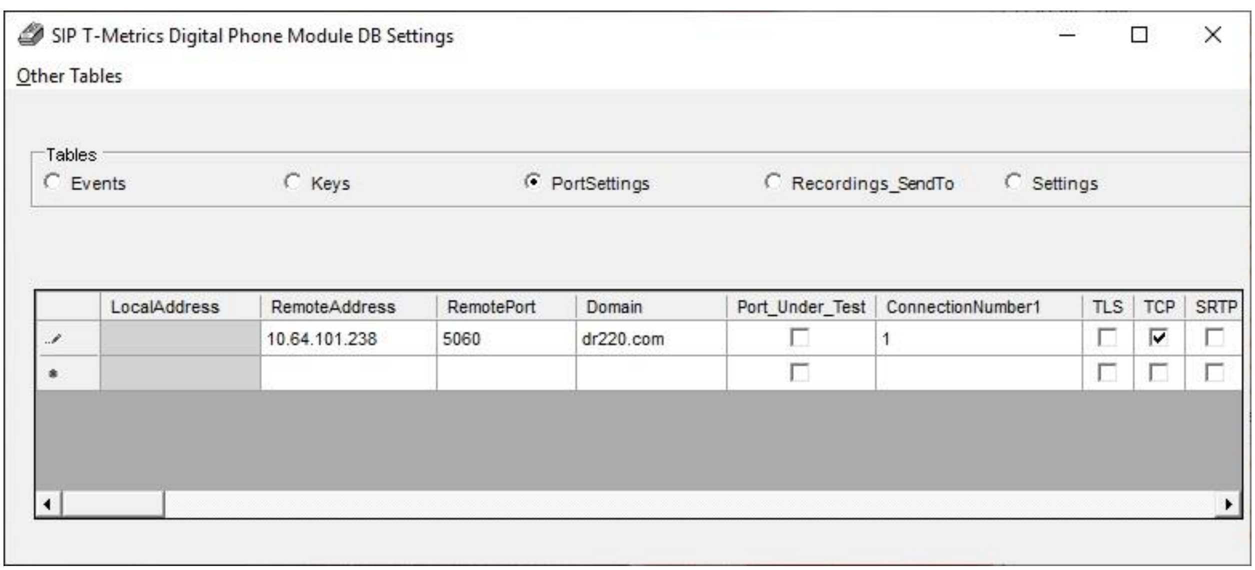
Create an entry for each dialed number that can be routed to Contact Center and associated with a pre-configured IVR script. In the compliance testing, two IVR scripts were pre-configured and therefore two entries were created with numbers "**53001**" and '**53002**" as shown below.




### 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**.

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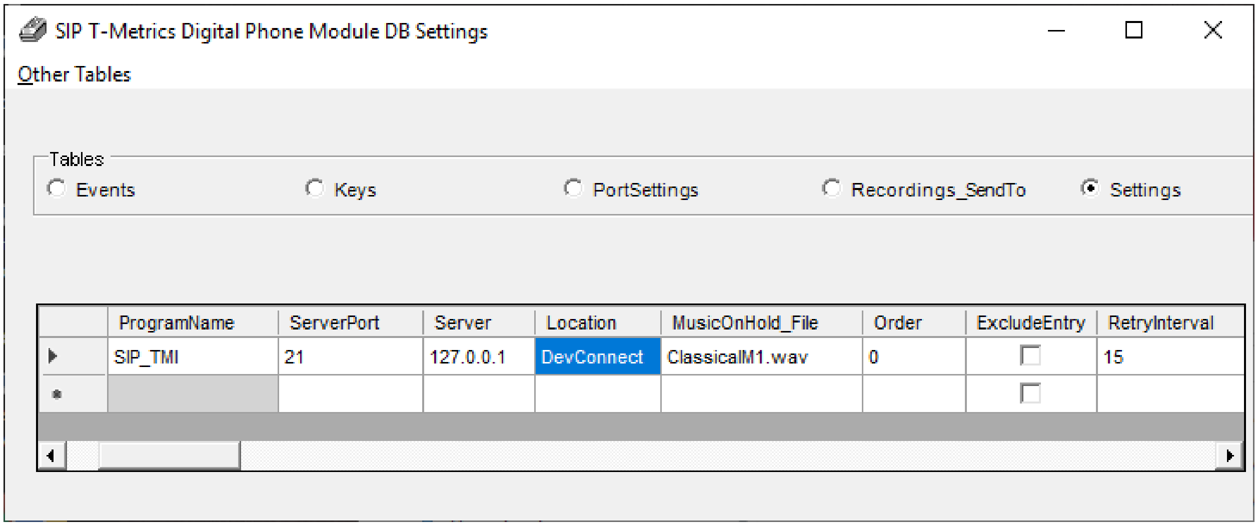
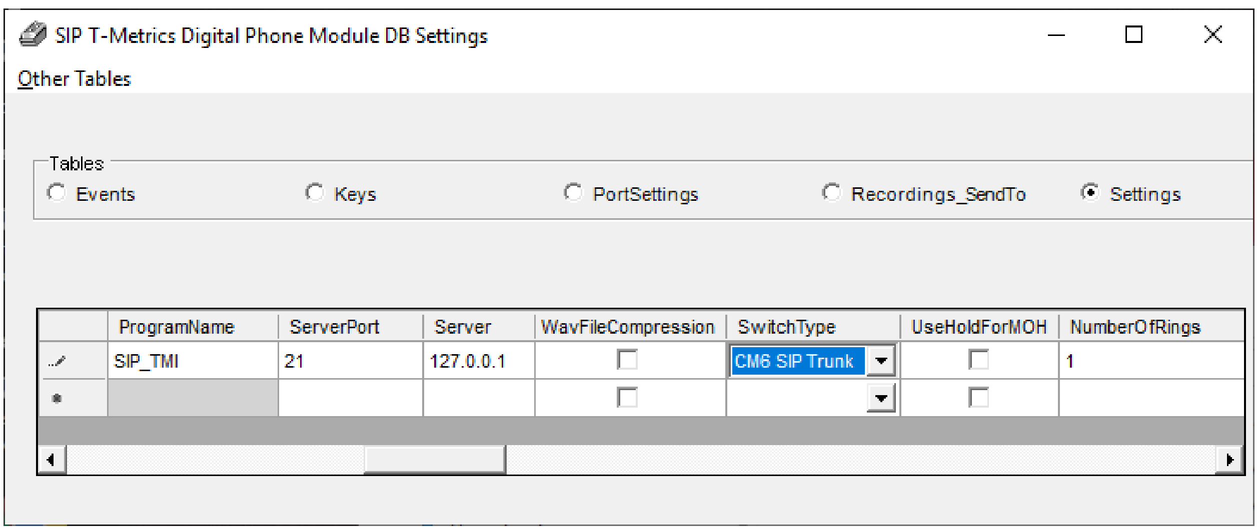
### 7.1.3. Settings

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




Navigate to **SwitchType** and select "**CM6 SIP Trunk**".





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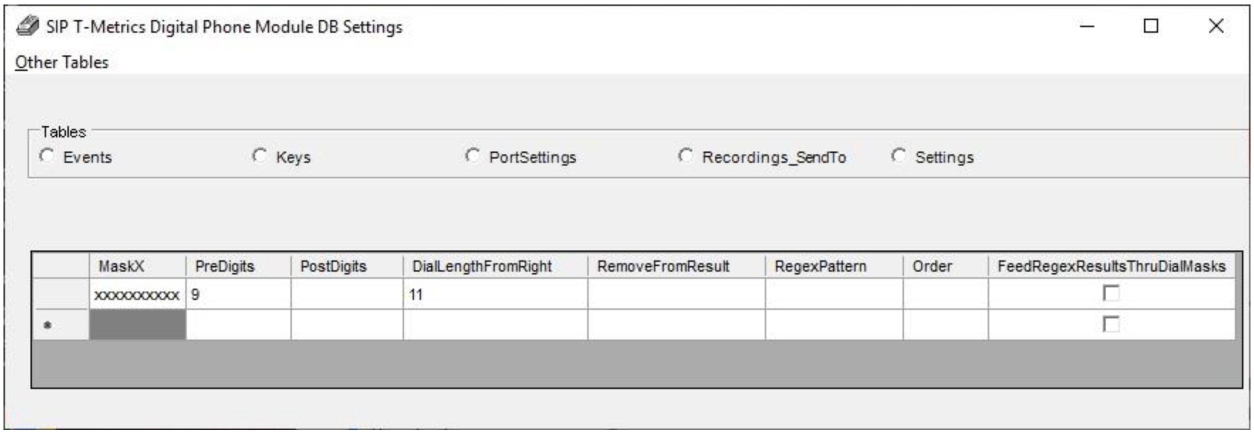
### 7.1.4. Dialing Mask

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.





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## 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.

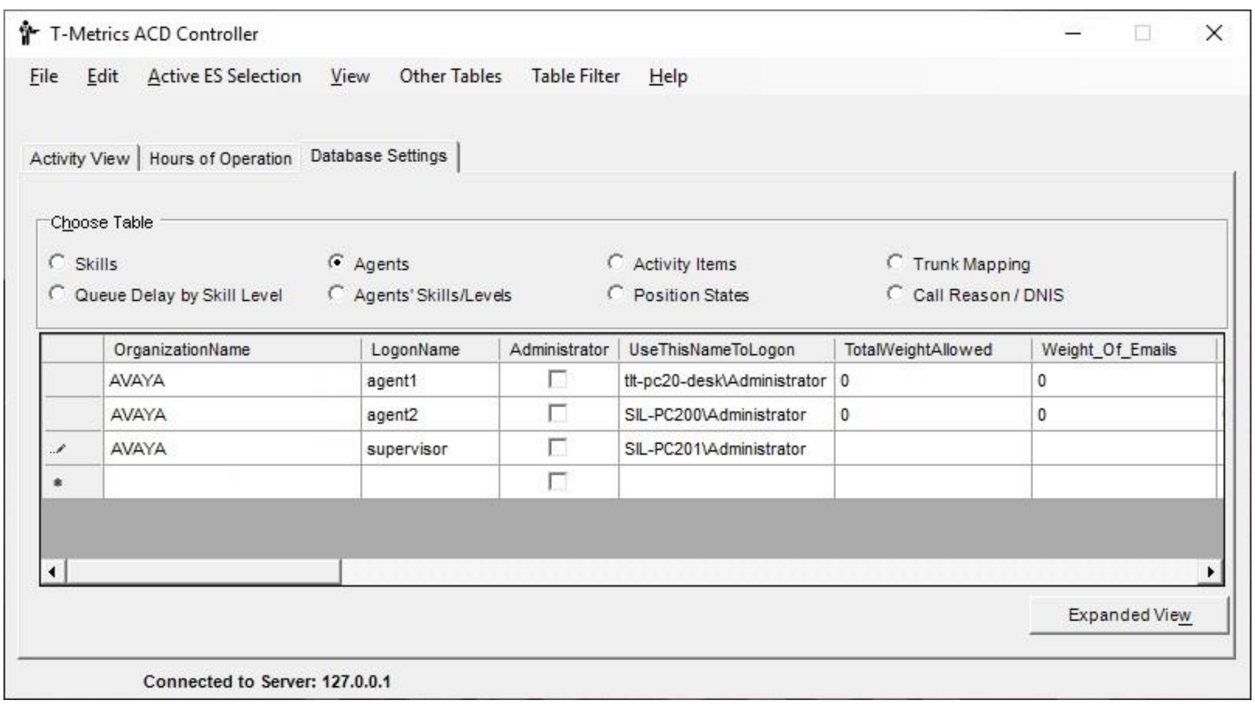

### 7.2.1. Agents

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 login 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.





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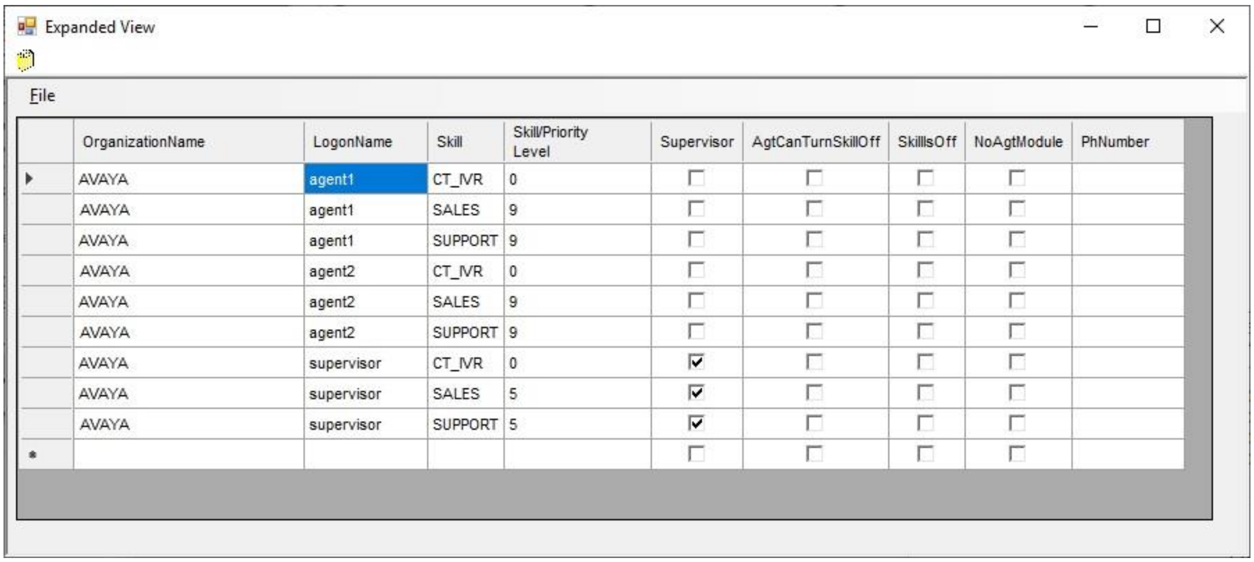
### 7.2.2. Agent Skills

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.

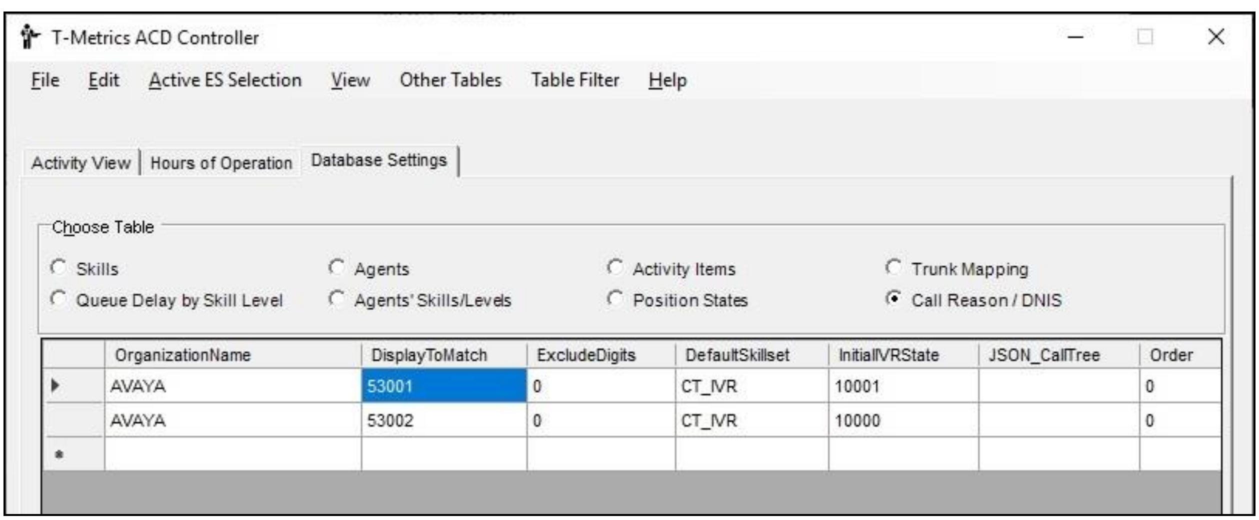



| | OrganizationName | LogonName | Skill | Skill/Priority Level | Supervisor | AgtCanTurnSkillOff | SkillsOff | NoAgtModule | PhNumber |
|---|---|---|---|---|---|---|---|---|---|
| ▸ | AVAYA | agent1 | CT_IVR | 0 | ☐ | ☐ | ☐ | ☐ | |
| | AVAYA | agent1 | SALES | 9 | ☐ | ☐ | ☐ | ☐ | |
| | AVAYA | agent1 | SUPPORT | 9 | ☐ | ☐ | ☐ | ☐ | |
| | AVAYA | agent2 | CT_IVR | 0 | ☐ | ☐ | ☐ | ☐ | |
| | AVAYA | agent2 | SALES | 9 | ☐ | ☐ | ☐ | ☐ | |
| | AVAYA | agent2 | SUPPORT | 9 | ☐ | ☐ | ☐ | ☐ | |
| | AVAYA | supervisor | CT_IVR | 0 | ☑ | ☐ | ☐ | ☐ | |
| | AVAYA | supervisor | SALES | 5 | ☑ | ☐ | ☐ | ☐ | |
| | AVAYA | supervisor | SUPPORT | 5 | ☑ | ☐ | ☐ | ☐ | |
| * | | | | | ☐ | ☐ | ☐ | ☐ | |

### 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.




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



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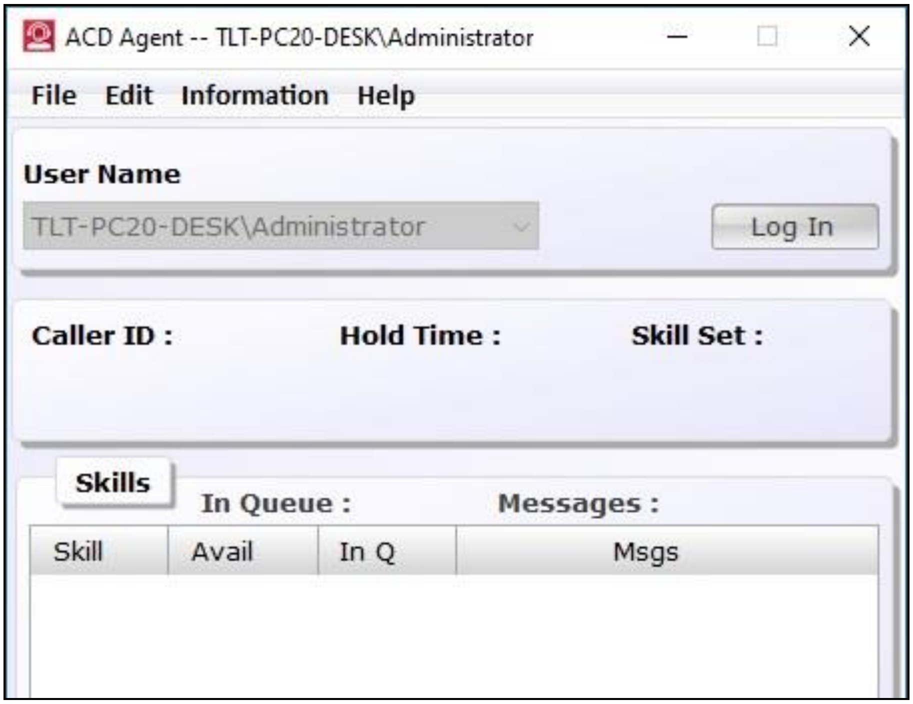
## 7.3. Administer ACD Agent Module

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.





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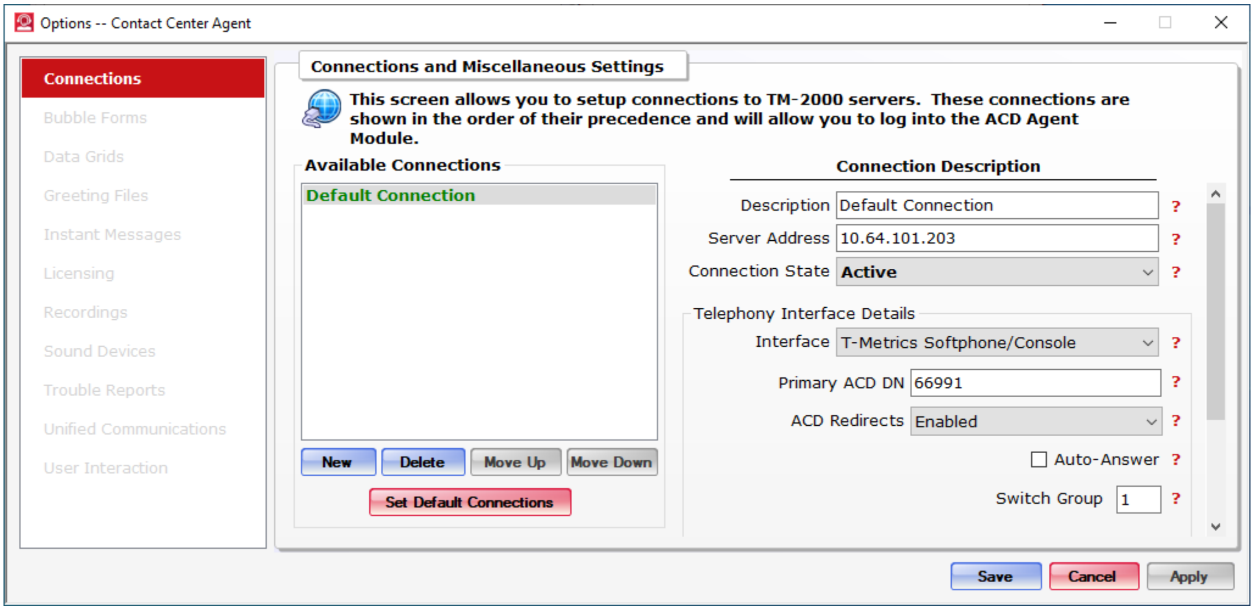
The **Options – Contact Center Agent** screen is displayed. Enter the following values for the specified fields and retain default values for the remaining fields.

- **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**.





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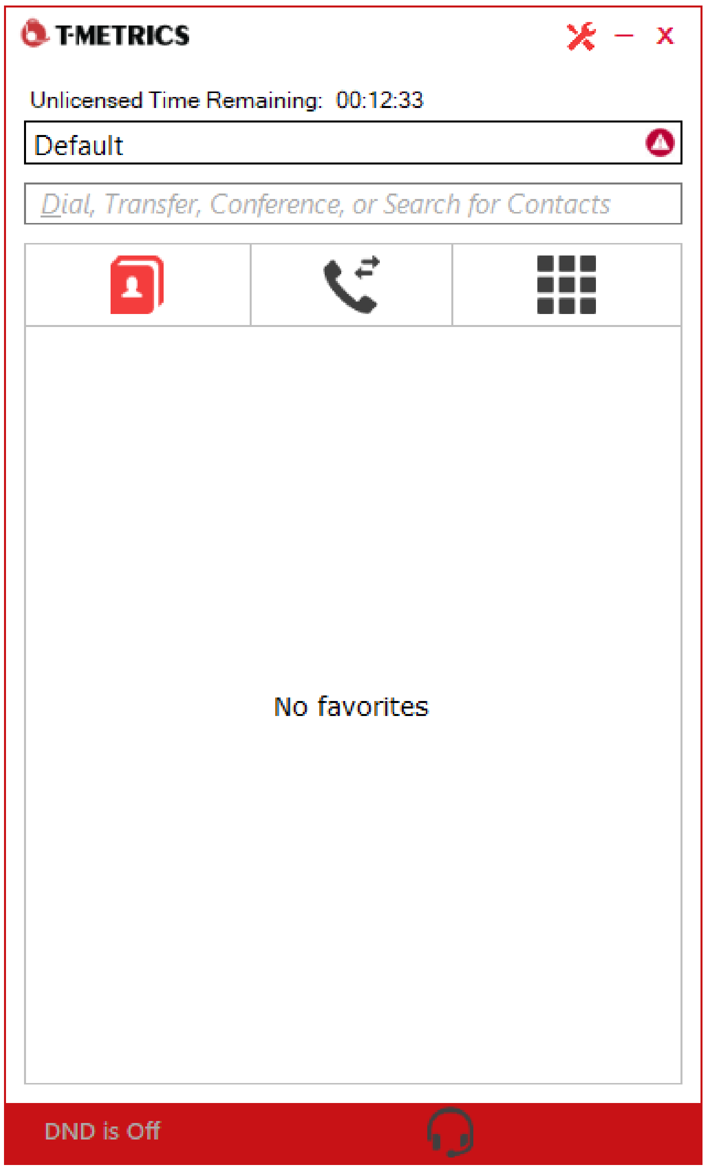
## 7.4. Administer SIP Softphone

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.





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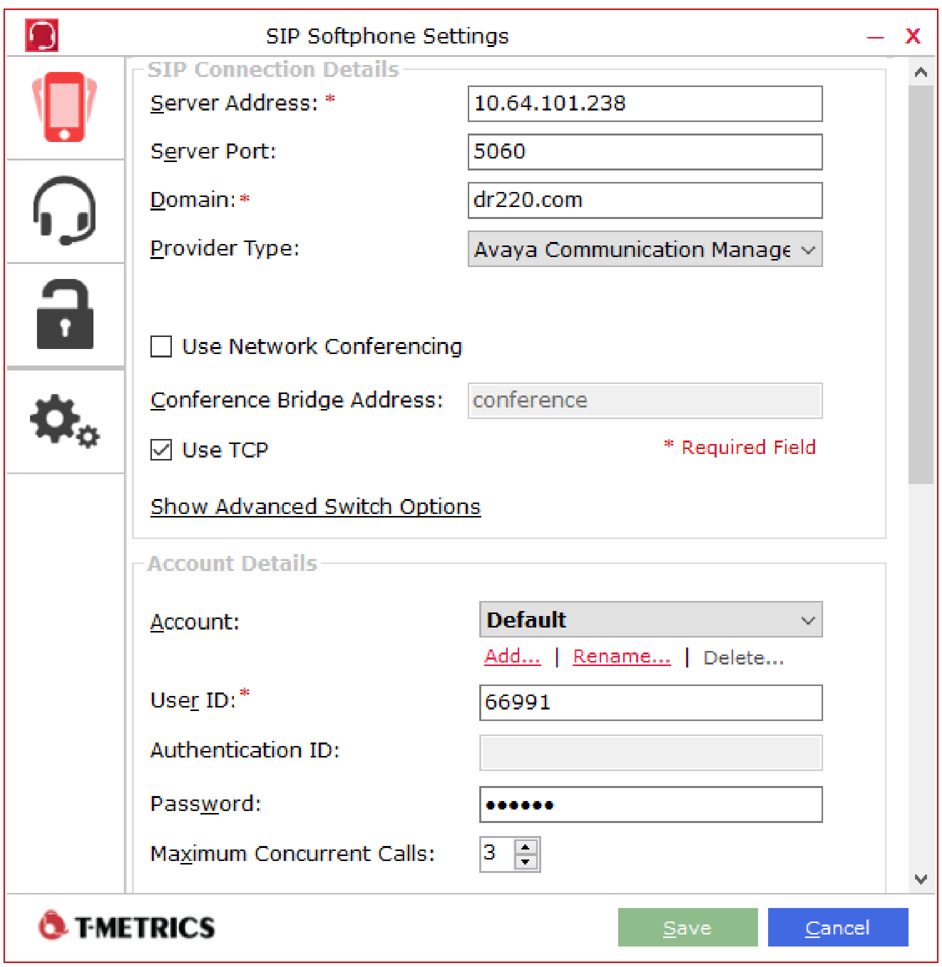
The **SIP Softphone Settings** screen is displayed.  Enter the following values for the specified fields and retain default values for the remaining fields.

- **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.



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# 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
```


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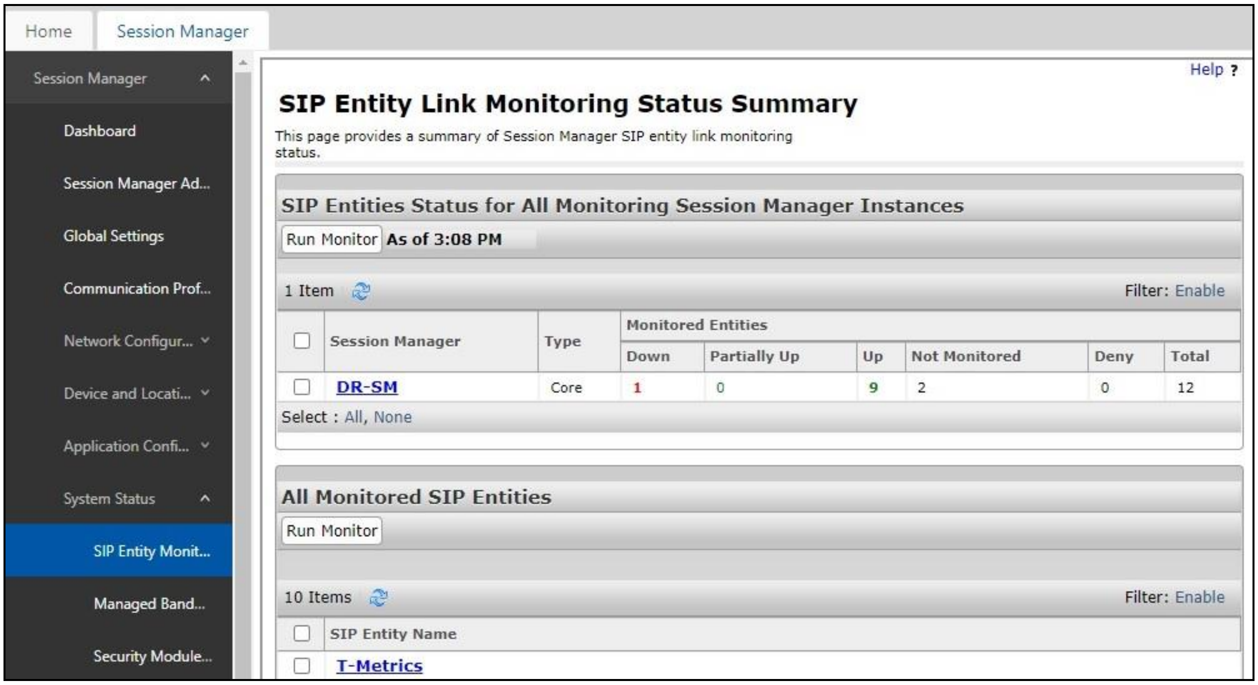
## 8.2. Verify Avaya Aura® Session Manager

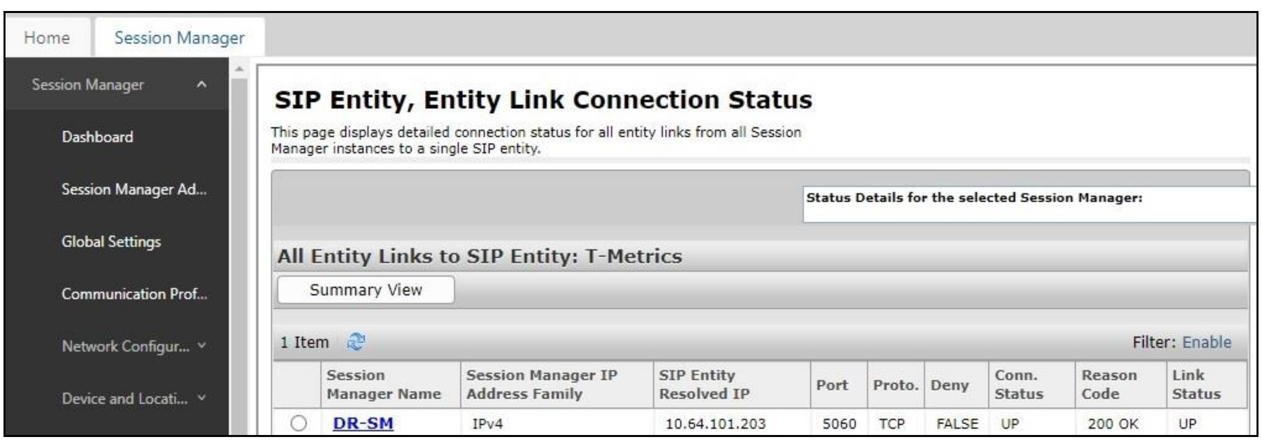
From the System Manager home page (not shown), select **Elements → Session Manager** to display the **Session Manager Dashboard** screen (not shown).

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





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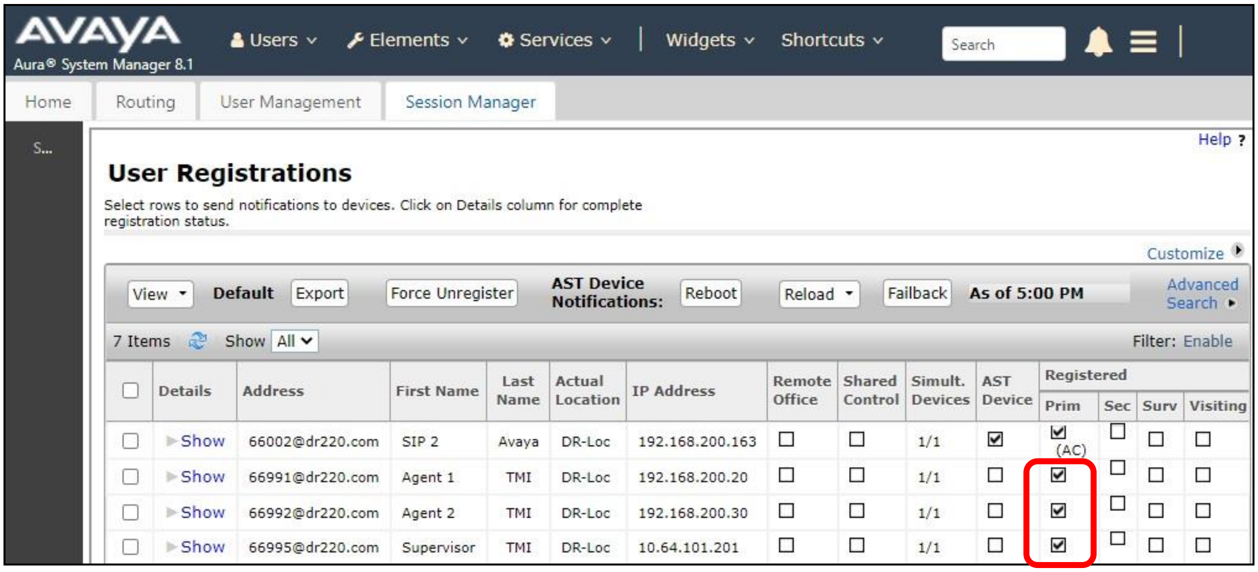
### 8.2.2. User Integration

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.





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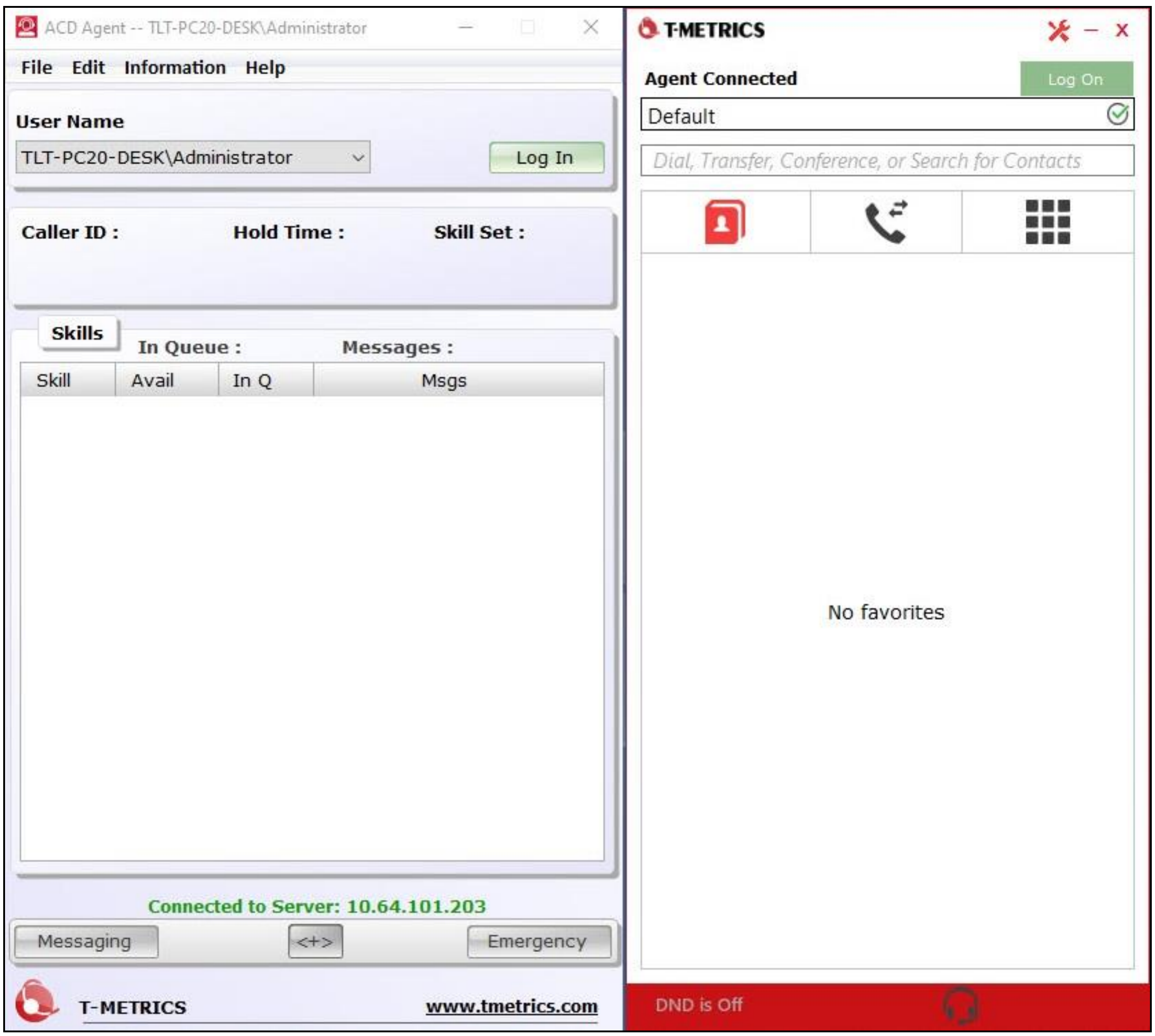
## 8.3. Verify T-Metrics Contact Center

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**.





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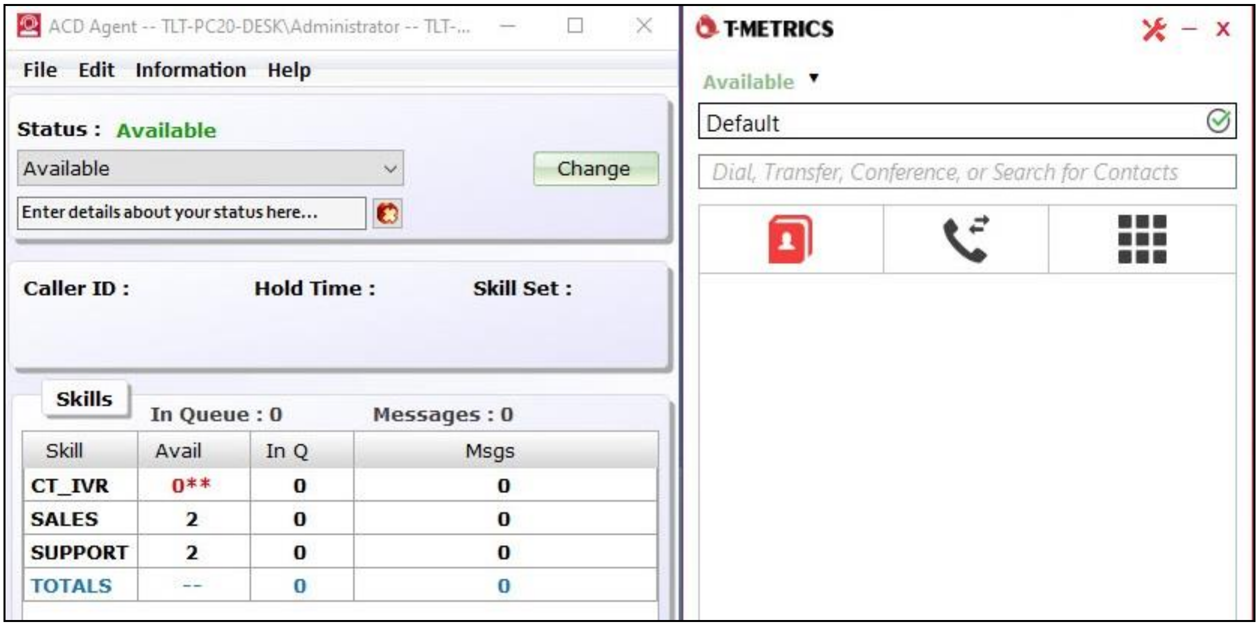
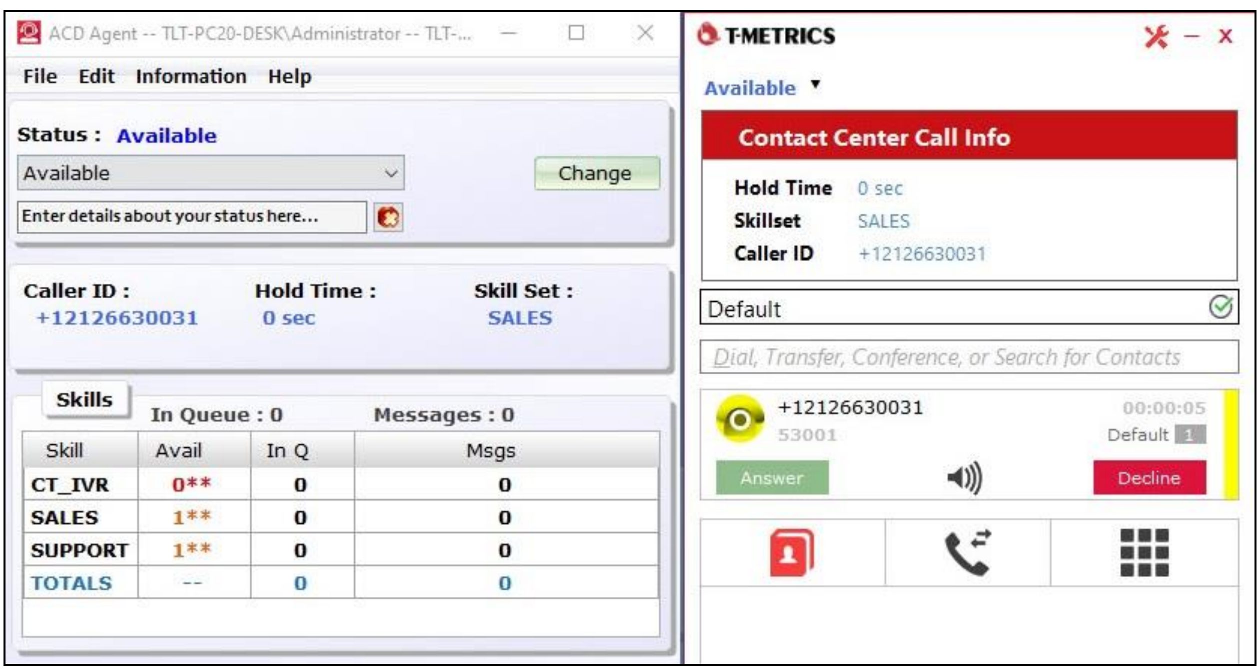
Verify that both the **ACD Agent** and **T-METRICS** screens are updated to reflect the **Available** status in green, as shown below.




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.





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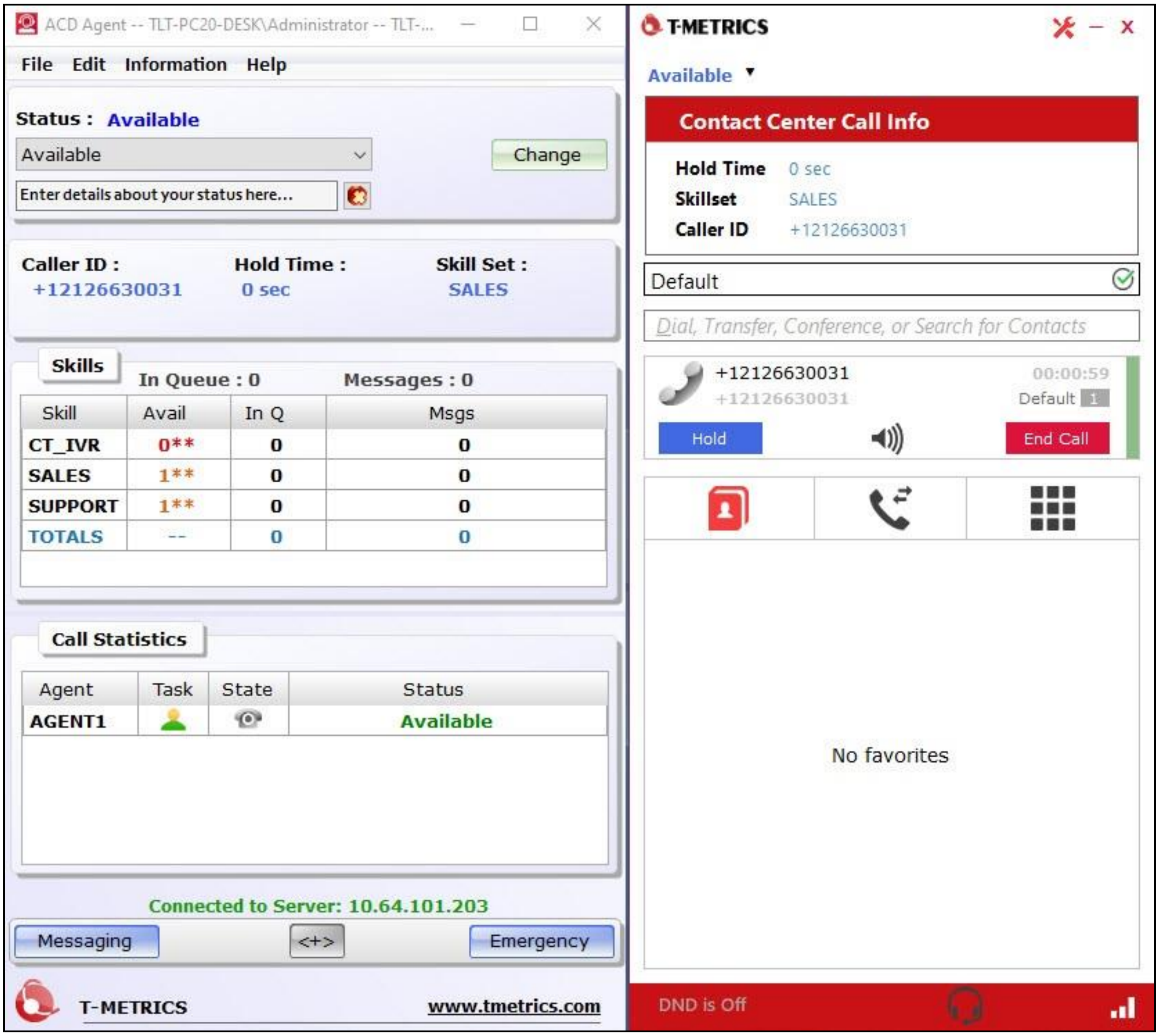
Verify that the **T-METRICS** screen is updated to reflect a connected call, and that the agent headset is connected to the PSTN caller with two-way talk paths.



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# 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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