**DevConnect Program**

# Application Notes for Enghouse Trio Version 9 to interoperate with Avaya Aura® Communication Manager Release 10.1, Avaya Aura® Session Manager Release 10.1 – Issue 1.0

## Abstract

These Application Notes describe the configuration steps required for Enghouse Trio to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as 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 Avaya DevConnect Program.

# 1. Introduction

These Application Notes outline the steps necessary to configure Trio from Enghouse Interactive AB to interoperate with Avaya Aura® Communication Manager (Communication Manager) and Avaya Aura® Session Manager (Session Manager). Trio is a client/server-based application running on Windows Server operating systems. Trio provides users with an attendant answering position for Communication Manager, as well as a call referral function that provides spoken information about the status of the extension called. The Trio Attendant client provides a view of contacts, schedules, and communication tasks and was installed on the same server as the Trio server but can be installed on a separate platform if required. Trio connects to the Communication Manager using a SIP trunk via the Session Manager.

# 2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using Communication Manager. The Trio server ccommunicates with the Communication Manager using a SIP trunk through the Session Manager. See **Figure 1** for a network diagram. A Dial plan was configured on the Communication Manager to route calls to Trio. Calls placed to the Trio server automatically places a call to the telephone the Attendant is using for answering purposes. When the attendant answers the call, the Trio server bridges the two calls. When the attendant extends the call to another telephone, Trio server performs a SIP Refer method, and the caller and the called user are now directly connected.

It is possible to have multiple Trio attendant positions on a Communication Manager system. A variety of Avaya telephones were installed and configured on the Communication Manager.

**Note:** During compliance testing an Avaya SIP and H.323 endpoints were used as the attendant's telephones.

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 Trio did not include use of any specific encryption features as requested by Enghouse.

## 2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing. The serviceability testing introduced failure scenarios to see if Trio could resume after a link failure with Communication Manager. The testing included:

- Incoming internal and external calls
- Outgoing internal and external calls
- Supervised and unsupervised transfer with answer
- Directing calls from busy extensions and extensions that do not answer
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions
- Serviceability

## 2.2. Test Results

The tests were all functional in nature and performance testing was not included. All test cases passed successfully.

## 2.3. Support

For technical support for Enghouse Interactive AB products, please use the following web link. http://www.trio.com/web/Support.aspx

Enghouse Interactive AB can also be contacted as follows.

- **Phone**: +46 (0)8 457 30 00
- **Fax**: +46 (0)8 31 87 00
- **E-mail**: triosupport@enghouse.com

# 3. Reference Configuration

**Figure** 1 illustrates the network topology used during compliance testing. The Avaya solution consists of a Communication Manager, which has a SIP Trunk connection to the Trio server via the Session Manager. Avaya H.323 and SIP stations were used as the Trio Attendant telephones during compliance testing. SIP and H.323 stations were configured on the Communication Manager to generate outbound/inbound calls to/from the PSTN. The simulated enterprise voice has SIP trunk to PSTN through Avaya Session Border Controller.

**Note:** The Trio Attendant (client) was installed on the same server as the Trio Server but can be installed on a separate workstation if required.

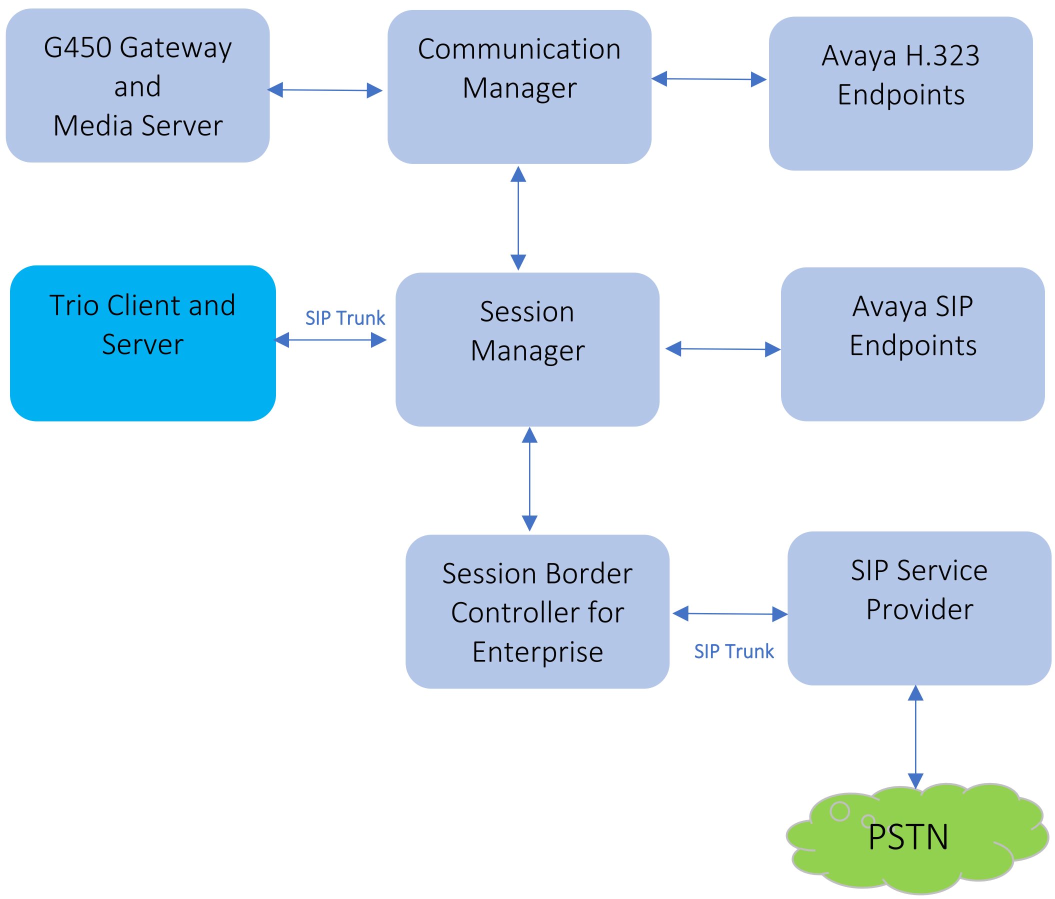



**Figure 1: Test Configuration Diagram**

# 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 | 10.1.2.0 FP2<br>01.0.974.0-27783 |
| Avaya G450 Media Gateway | FW 42.18.0 |
| Avaya Aura® Media Server | 10.1.0.125 |
| Avaya Aura® System Manager | 10.1.2.0 Feature Pack 2<br>10.1.2.0.0715476 |
| Avaya Aura® Session Manager | 10.1.2.0 Feature Pack 2<br>10.1.0.02.1012016 |
| Avaya Session Border Controller | 10.1.1.0-35-21872 |
| Avaya 96x1 Series IP Deskphones | 6.8.5.4.10 (H.323) |
| Avaya 96x1 Series IP Deskphones | 7.1.15.2.1 (SIP) |
| Avaya J100 Series SIP Deskphones | 4.1.0.0.9 |
| Avaya Workplace Client for Windows | 3.32.0.75 |
| Enghouse Trio | 9.2 |

# 5. Configure Avaya Aura® Communication Manager

Configuration and verification operations on the Communication Manager illustrated in this section were all performed using Avaya Site Administrator Emulation Mode. The information provided in this section describes the configuration of the Communication Manager for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 10**.

It is implied a working system is already in place. The configuration operations described in this section can be summarized as follows: (Note: During Compliance Testing all inputs not highlighted in Bold were left as Default)

- Verify License
- Administer System Parameters Features
- Administer IP Node Names
- Administer SIP Signalling group
- Administer SIP Trunk Group
- Administer IP Network Region
- Administer IP Codec Set
- Administer Route Pattern
- Administer Private Numbering
- Administer Dialing Plan

## 5.1. Verify License

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

```
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      8
         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     4
               Maximum Video Capable IP Softphones:  18000    11
                   Maximum Administered SIP Trunks: 40000     30
  Max Administered Ad-hoc Video Conferencing Ports: 24000      0
   Max Number of DS1 Boards with Echo Cancellation: 999    0



        (NOTE: You must logoff & login to effect the permission changes.)
```

## 5.2. Administer System Parameter Features

During compliance testing Trio suggested that the Station Call Transfer Recall Timer was set to be 20 seconds. Use the "change system-parameters features" command to change the **Station Call Transfer Recall Timer** on **Page 6**.

```
change system-parameters features                               Page   6 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
         Public Network Trunks on Conference Call: 5             Auto Start? n
    Conference Parties with Public Network Trunks: 6              Auto Hold? y
 Conference Parties without Public Network Trunks: 6         Attendant Tone? y
         Night Service Disconnect Timer (seconds): 180        Bridging Tone? n
                 Short Interdigit Timer (seconds): 3        Conference Tone? n
              Unanswered DID Call Timer (seconds):           Intrusion Tone? n
              Line Intercept Tone Timer (seconds): 30    Mode Code Interface? n
                 Long Hold Recall Timer (seconds): 0
                      Reset Shift Timer (seconds): 0
     Station Call Transfer Recall Timer (seconds): 20       Recall from VDN? n
           Trunk Alerting Tone Interval (seconds): 15
                              DID Busy Treatment: tone
               Allow AAR/ARS Access from DID/DIOD? n
                 Allow ANI Restriction on AAR/ARS? n
Use Trunk COR for Outgoing Trunk Disconnect/Alert? n
                  7405ND Numeric Terminal Display? n                  7434ND? y
DISTINCTIVE AUDIBLE ALERTING
             Internal: 1   External: 2   Priority: 3
                        Attendant Originated Calls: external
    DTMF Tone Feedback Signal to VRU - Connection:          Disconnection:
```

Enable **Create Universal Call ID (UCID)**, which is located on **Page 5**. For **UCID Network Node ID**, enter an available node ID.

```
change system-parameters features                               Page   5 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
SYSTEM PRINTER PARAMETERS
  Endpoint:               Lines Per Page: 60

SYSTEM-WIDE PARAMETERS
                                     Switch Name: cm10
            Emergency Extension Forwarding (min): 10
          Enable Inter-Gateway Alternate Routing? n
Enable Dial Plan Transparency in Survivable Mode? n
                              COR to Use for DPT: station
                EC500 Routing in Survivable Mode: dpt-then-ec500
MALICIOUS CALL TRACE PARAMETERS
               Apply MCT Warning Tone? n   MCT Voice Recorder Trunk Group:
      Delay Sending RELease (seconds): 0  Notification using Crisis Alert? n
SEND ALL CALLS OPTIONS
     Send All Calls Applies to: station    Auto Inspect on Send All Calls? n
    Send All Calls on Ringing Bridge Leaves Call Ringing on Other Bridges? n
              Preserve previous AUX Work button states after deactivation? n
UNIVERSAL CALL ID
     Create Universal Call ID (UCID)? y    UCID Network Node ID: 1
     Copy UCID for Station Conference/Transfer? y
```

Navigate to **Page 13** and enable **Send UCID to ASAI**. This parameter allows for the universal call ID to be sent to Trio.

```
change system-parameters features                               Page  13 of  19
                        FEATURE-RELATED SYSTEM PARAMETERS
 CALL CENTER MISCELLANEOUS
            Callr-info Display Timer (sec): 10
                          Clear Callr-info: next-call
         Allow Ringer-off with Auto-Answer? n

     Reporting for PC Non-Predictive Calls? n

             Agent/Caller Disconnect Tones? n
Interruptible Aux Notification Timer (sec): 3
   Zip Tone Burst for Callmaster Endpoints: double



  ASAI
                   Copy ASAI UUI During Conference/Transfer? y
               Call Classification After Answer Supervision? y
                                          Send UCID to ASAI? y
                 For ASAI Send DTMF Tone to Call Originator? y
         Send Connect Event to ASAI For Announcement Answer? y
  Prefer H.323 Over SIP For Dual-Reg Station 3PCC Make Call? n
```

## 5.3. Administer IP Node Names

Use the "change node-names ip" command (not shown) and add an entry for Session Manager. In this case, **SM10** and **10.33.1.42** are entered as **Name** and **IP Address**. Note the **procr** and **10.33.1.43** entry, which is the node **Name** and **IP Address** for the processor board. These values will be used later to configure the SIP trunk to Session Manager in **Section 0**.

```
change node-names ip                                              Page   1 of   2
                                  IP NODE NAMES
    Name              IP Address
SM10                10.33.1.42
default             0.0.0.0
lsp                 10.33.1.7
procr               10.33.1.43
( 16 of 18   administered node-names were displayed )
Use 'list node-names' command to see all the administered node-names
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name
```

Navigate to **Page 3** and enter "private" for **Numbering Format**.

```
change trunk-group 1                                              Page   3 of   5
TRUNK FEATURES
          ACA Assignment? n            Measured: both
                                                        Maintenance Tests? y



   Suppress # Outpulsing? n  Numbering Format: private
                                                UUI Treatment: shared
                                              Maximum Size of UUI Contents: 128
                                                 Replace Restricted Numbers? y
                                                Replace Unavailable Numbers? y

                                Modify Tandem Calling Number: no
               Send UCID? n


 Show ANSWERED BY on Display? y
```

## 5.4. Administer SIP Signalling Group

Use the "add signaling-group n" command, where "n" is an available signalling group number, in this case "1". 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:** An existing C-LAN node name or "procr" from **Section 0**.
- **Far-end Node Name:** The existing node name for Session Manager from **Section 0**.
- **Near-end Listen Port:** An available port for integration with Session Manager.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Session Manager.
- **Far-end Domain:** Leave this field empty as CM accepts any incoming call.
- **Direct IP-IP Audio Connections?:** "y".

**Note**: if the **Far-end domain** field is set to a specific domain and incoming call from the Trio server has a different domain such as having IP address in the URI, the incoming call is rejected.

```
change signaling-group 1                                        Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? n
     IP Video? n                                  Enforce SIPS URI for SRTP? n
  Peer Detection Enabled? n  Peer Server: SM                      Clustered? n
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                   Far-end Node Name: SM10
 Near-end Listen Port: 5061                  Far-end Listen Port: 5061
                                          Far-end Network Region: 1

Far-end Domain:
                                              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? y
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

## 5.5. Administer SIP Trunk Group

Use the "add trunk-group n" command, where "n" is an available trunk group number, in this case "1". 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".
- **Signaling Group:** Set it to the signaling group number 1 as defined in **Section 5.4**.
- **Number of Members:** Enter a number of SIP trunk, in this case 10 SIP trunk members used.

```
change trunk-group 1                                            Page   1 of   5
                                TRUNK GROUP

Group Number: 1                    Group Type: sip          CDR Reports: y
  Group Name: Private Trunk                COR: 1        TN: 1        TAC: #01
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 10
```

## 5.6. Administer IP Network Region

Use the "change ip-network-region n" command, where "n" is the existing far-end network region number used by the SIP signalling group from **Section 5.4**. For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter "yes" for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with Trio.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1       NR Group: 1
Location: 1       Authoritative Domain: avayalab.com
    Name: Loc-1                     Stub Network Region: n
MEDIA PARAMETERS                    Intra-region IP-IP Direct Audio: yes
      Codec Set: 1                  Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                          IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
```

## 5.7. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.6**. Update the audio codec types in the **Audio Codec** fields as necessary. Configure the codec as shown below.

```
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                     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: none
 3:
 4:
```

## 5.8. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Trio, in this case "1". Enter the following values for the specified fields and retain the default values for the remaining fields.

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

```
change route-pattern 1                                          Page   1 of   4
                    Pattern Number: 1      Pattern Name: SIP-TLS-To-SM
    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: 1    0                                                              n   user
 2:                                                                     n   user
 3:                                                                     n   user
 4:                                                                     n   user
 5:                                                                     n   user
 6:                                                                     n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev0-pvt  next
 2: y y y y y n  n            rest                                         next
```

## 5.9. Administer Private Numbering

Use the “change private-numbering 0” command, to define the calling party number to send to Trio. Add an entry for the trunk group defined in **Section 5.5**. In the example shown below, all calls originating from a 4-digit extension beginning with “3” and routed to trunk group all trunks will result in a 4-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
 4  3                                               4  Total Administered: 15
```

## 5.10. Administer Dialing Plan

Use the “change dialplan analysis” command to add a dialing entry “52” with 4-digit length and AAR call type used to route calls to the Trio server.

```
change dialplan analysis                                            Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all          Percent Full: 6

    Dialed   Total  Call      Dialed   Total  Call      Dialed   Total  Call
    String   Length Type      String   Length Type      String   Length Type
 0             1  udp      40           4  aar      78            5  aar
 0             3  fac      411          3  udp      8             1  fac
 1             4  ext      43           4  aar      9             1  fac
 1            11  udp      44           4  udp      *             3  dac
 52            4  aar      441         12  udp      #             3  dac
```

Use the “change aar analysis 0” command and add an entry in the AAR table to specify how to route calls to 52xx. In the example shown below, calls with digits 52 will be routed as an AAR call using route pattern “1” from **Section 5.8**.

```
change aar analysis 51                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location: all          Percent Full: 0

          Dialed           Total     Route     Call   Node  ANI
          String          Min  Max  Pattern    Type   Num   Reqd
     52                    4    4     1        aar          n
```

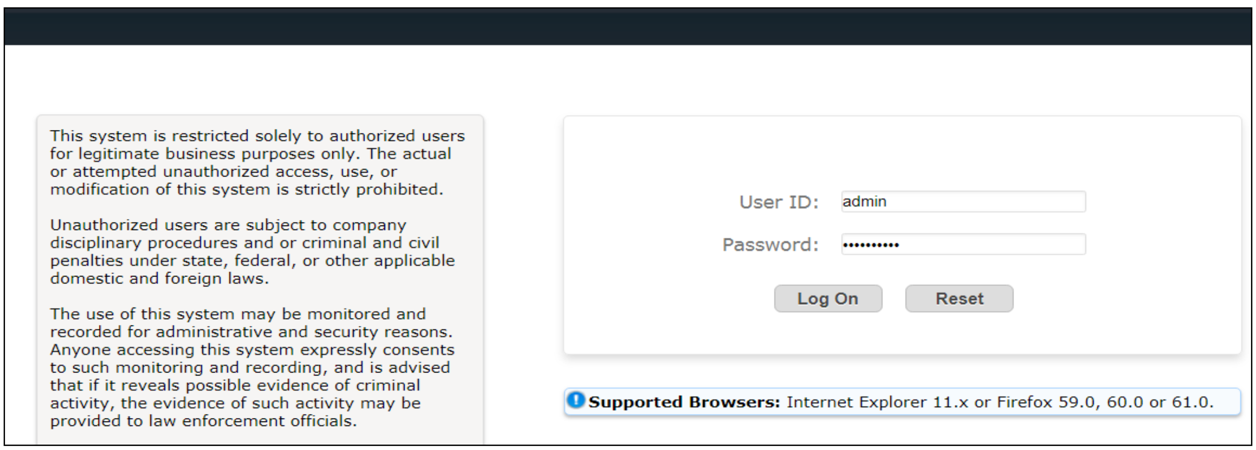
# 6. Configure Avaya Aura® Session Manager

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

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

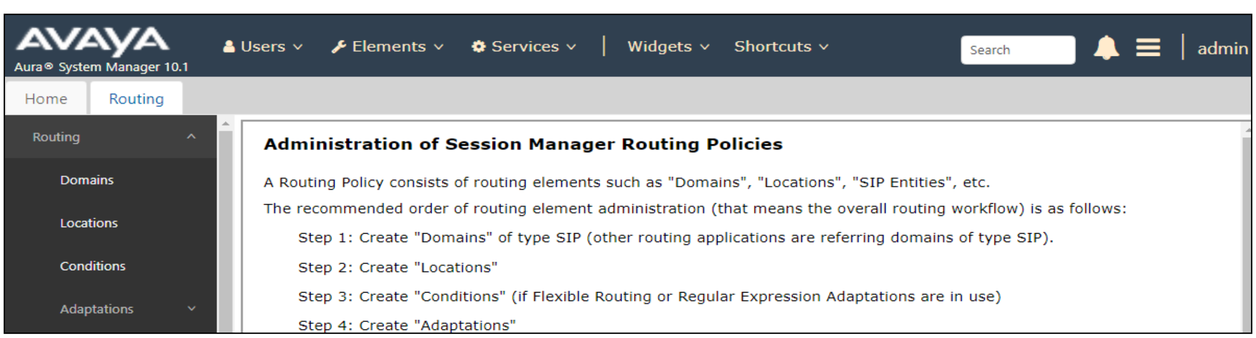
## 6.1. Launch System Manager

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

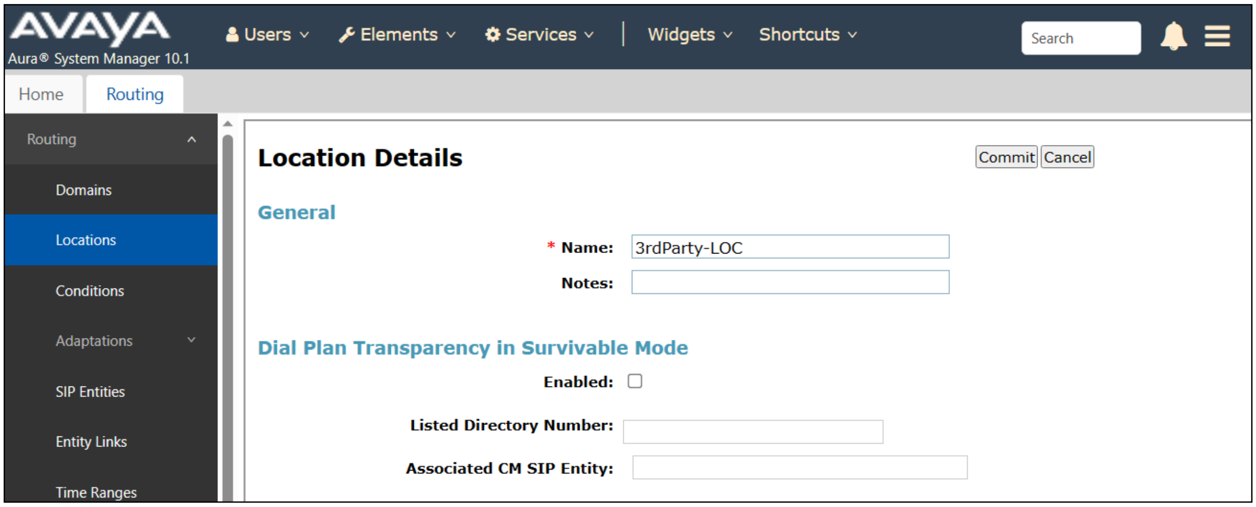



## 6.2. Administer Locations

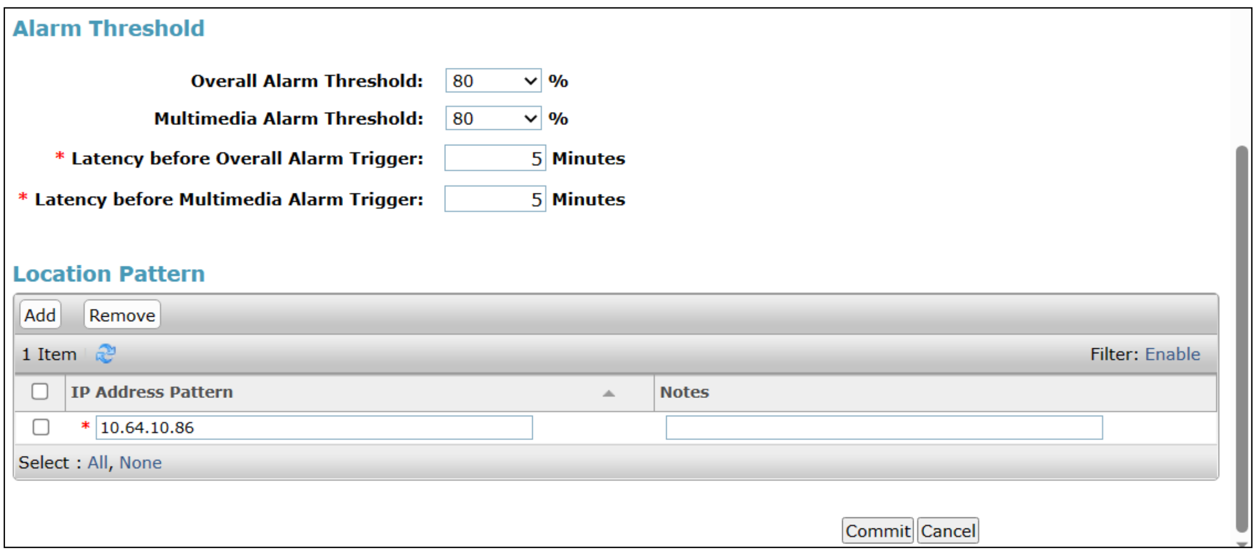
In the subsequent screen (not shown), select **Elements → Routing** to display the **Administration of Session Manager Routing Policies** screen below. Select **Routing → Locations** from the left pane and click **New** in the subsequent screen (not shown) to add a new location for Trio.



The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.




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



## 6.3. Administer Adaption

Session Manager can be configured to use Adaptation Modules to modify the From header of the incoming INVITE message sent from Session Manager to the Trio server. See the observation noted in **Section 2.2** for more information.

During compliance test, to make the call from and to Communication Manager via Session Manager, Adaptation to translate IP address into domain name is used for Trio SIP entity. Below are the steps that were used during compliance testing to create the needed adaptation. Select **Adaptations** on the left panel menu and then click on the **New** button in the main window (not shown).

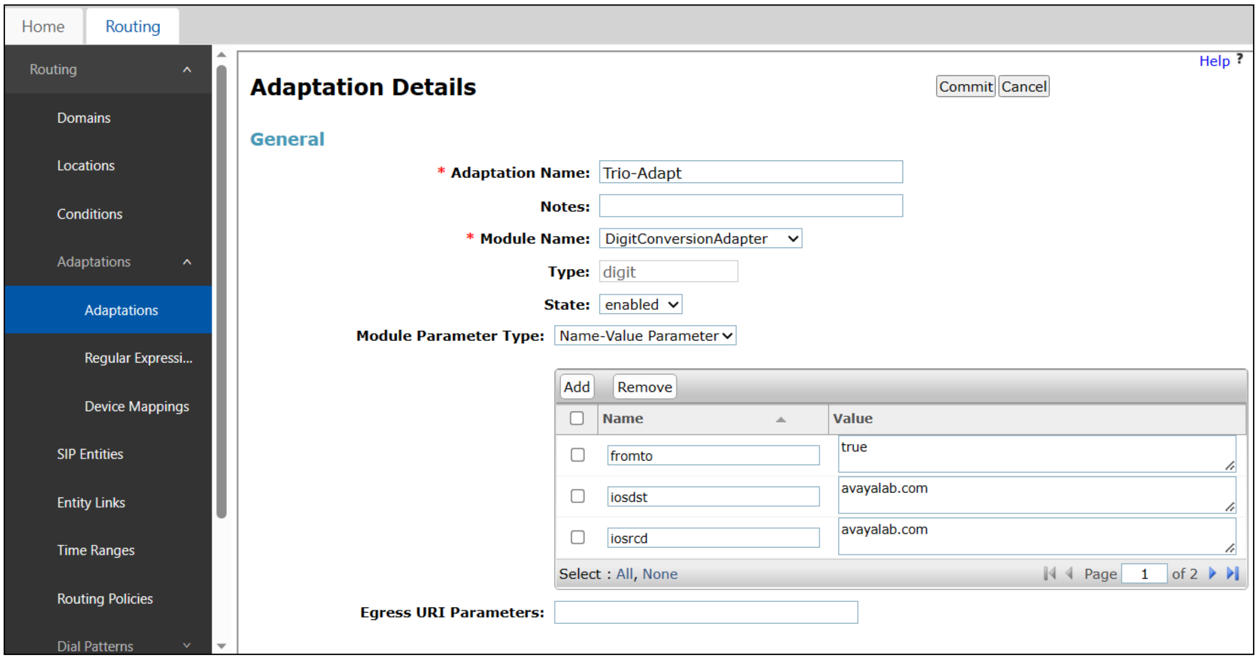
Enter the following for the Trio Adaptation.

- **Adaptation Name:** An informative name (e.g., Trio-Adapt).
- **Module Name:** Select "DigitConversionAdapter".
- **Module Parameter Type:** Select "Name-Value Parameter".

Click **Add** to add a new row for the following values as shown below table:

| Name | Value |
|---|---|
| fromto | true |
| iodstd | Enter the domain name of system, e.g.: **avayalab.com** |
| iosrcd | Enter the domain name of system, e.g.: **avayalab.com** |
| odstd | Enter IP address of Trio SIP Server, e.g.: **10.64.10.86** |
| osrcd | Enter IP address of Session Manager Server, e.g.: **10.33.1.42** |

Once the correct information is entered click the **Commit** button. Below is the screenshot showing the adaptation created for Trio. Select next to see the 2nd page to view rest of the adaptations.



## 6.4. Administer SIP Entities

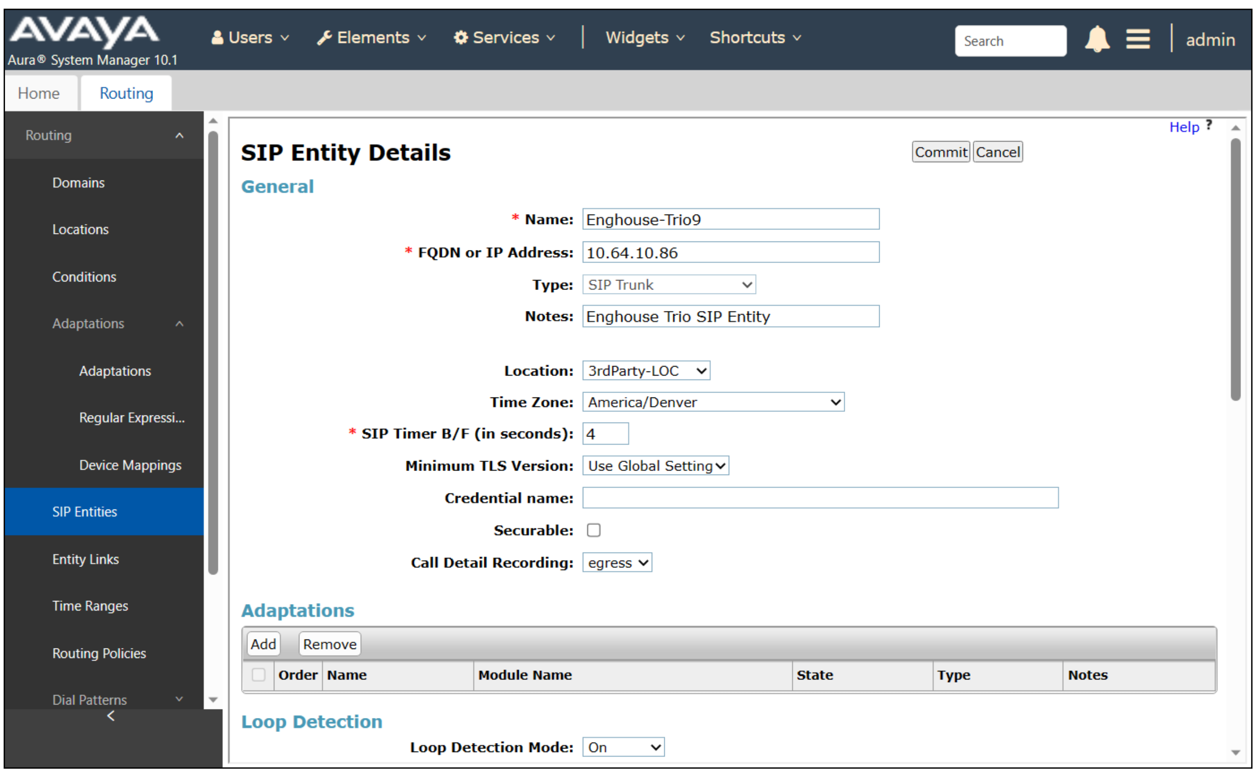
In the SIP entity, two SIP entities were added, one for Trio and another one for Communication Manager.

### 6.4.1. SIP Entity for Trio

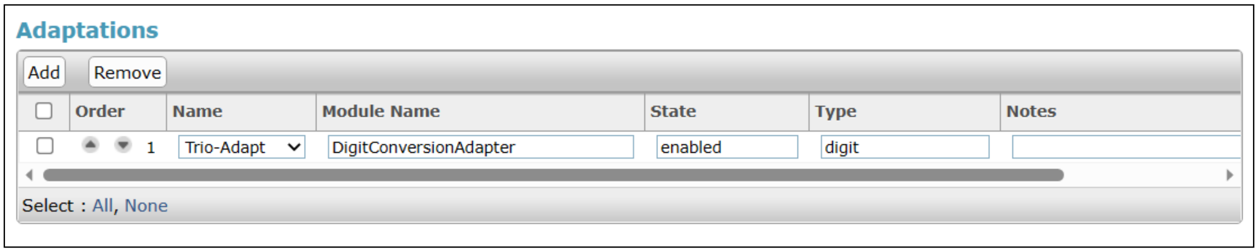
Select **Routing → SIP Entities** from the left pane and click **New** in the subsequent screen (not shown) to add a new SIP entity for Trio.

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 Trio server.
- **Type:** "SIP trunk"
- **Notes:** Any desired notes.
- **Location:** Select the location name as defined from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.



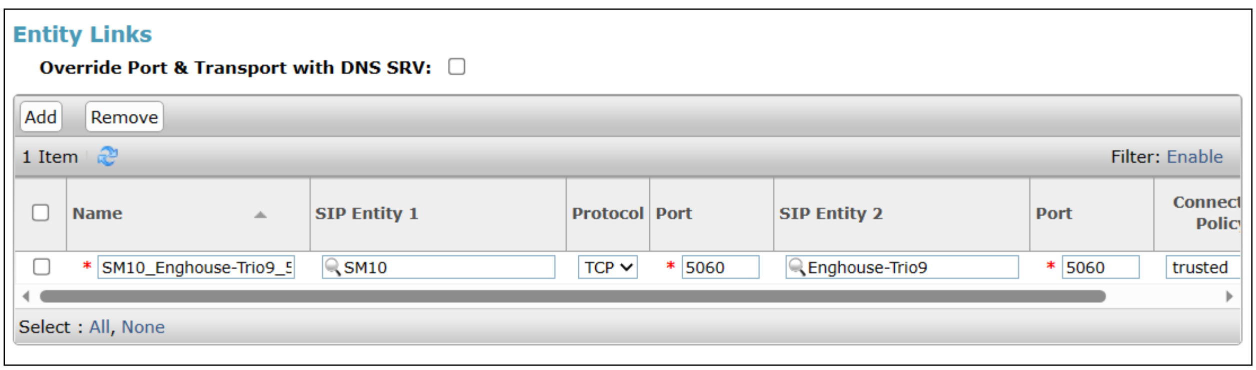
In the **Adaptations** section, select **Add** to add the Trio adaptation as configured in **Section 6.3**.




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 "SM10".
- **Protocol:** "TCP"
- **Port:** "5060"
- **SIP Entity 2:** The Trio entity name from this section.
- **Port:** "5060"
- **Connection Policy:** "trusted"

Note that Trio can support UDP and TCP, and the compliance testing used the TCP protocol.

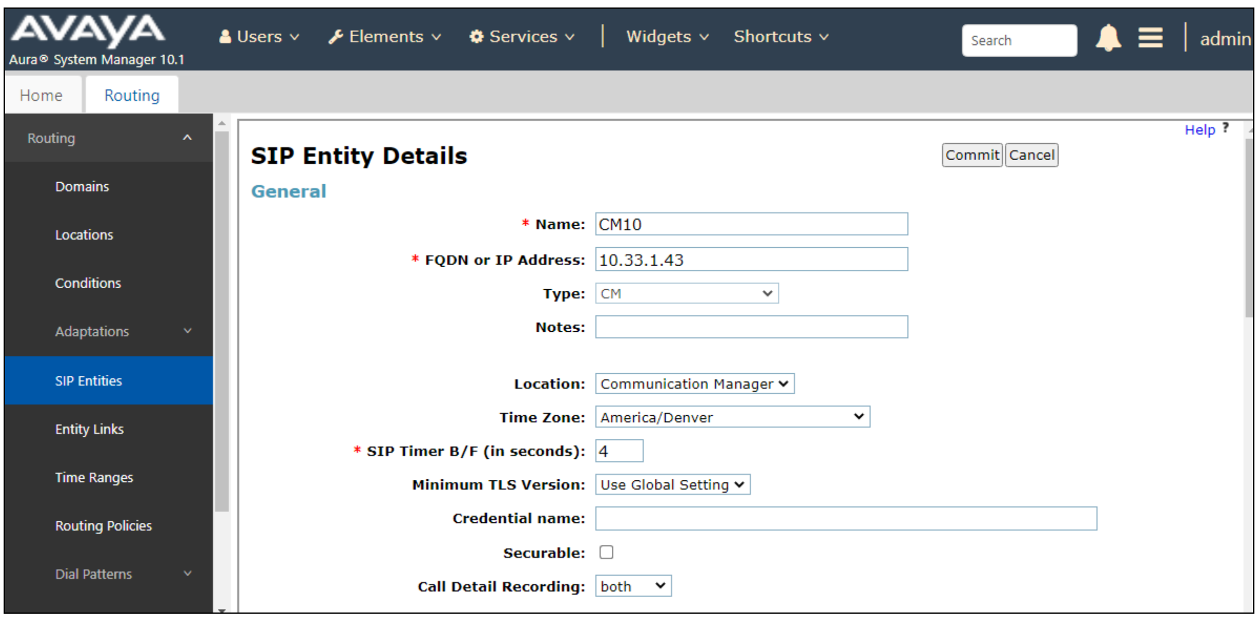


### 6.4.2. SIP Entity for Communication Manager

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

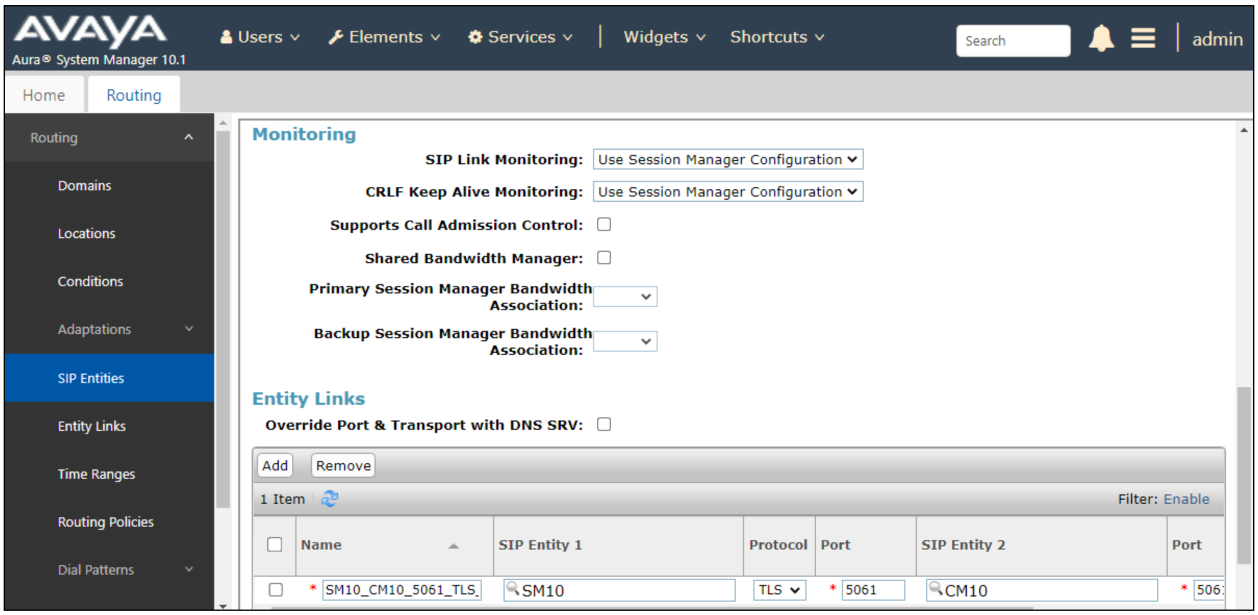
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing CLAN or the processor interface.
- **Type:** "CM"
- **Notes:** Any desired notes.
- **Location:** Select the applicable location for Communication Manager.
- **Time Zone:** Select the applicable time zone.



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 "SM10".
- **Protocol:** Select TLS protocol.
- **Port:** Enter the TLS port 5061.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** Enter the TLS port 5061.
- **Connection Policy:** "trusted"



## 6.5. Administer Routing Policies

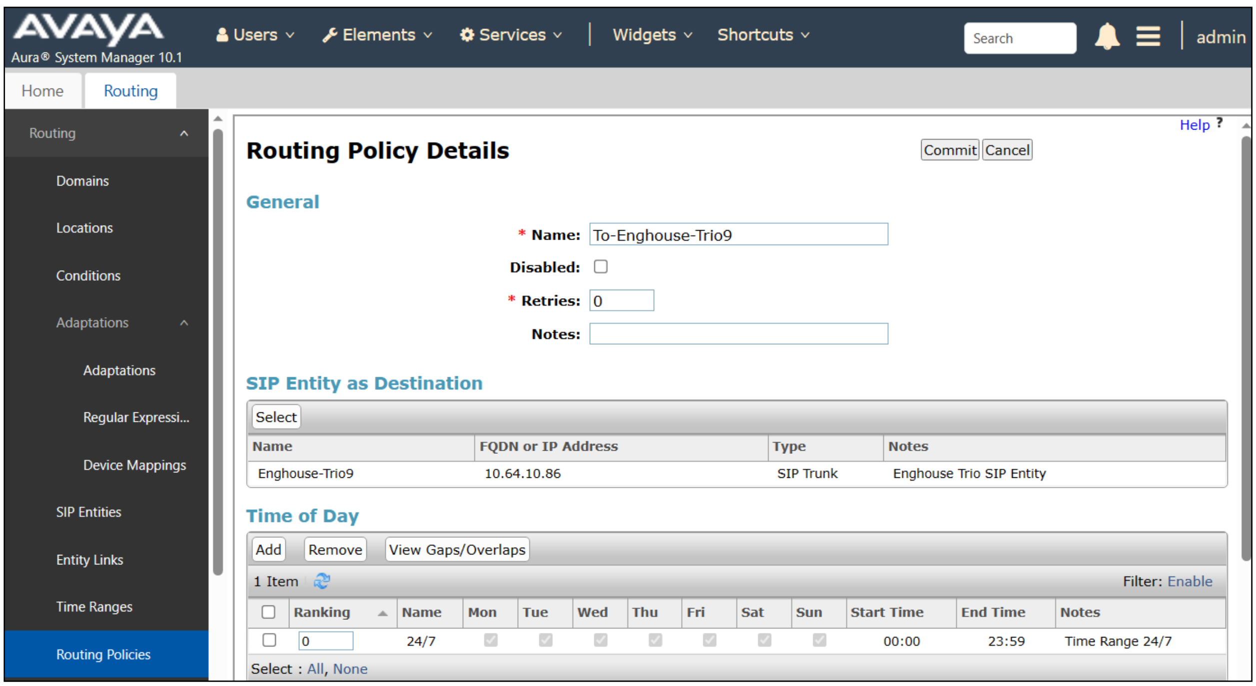
Add two new routing policies, one for Trio and one for the new SIP trunks with Communication Manager.

### 6.5.1. Routing Policy for Trio

Select **Routing → Routing Policies** from the left pane and click **New** in the subsequent screen (not shown) to add a new routing policy for Trio.

The **Routing Policy Details** screen is displayed. In the **General** 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 **Enghouse-Trio9** SIP entity name from **Section 6.4.1**. The screen below shows the result of the selection.

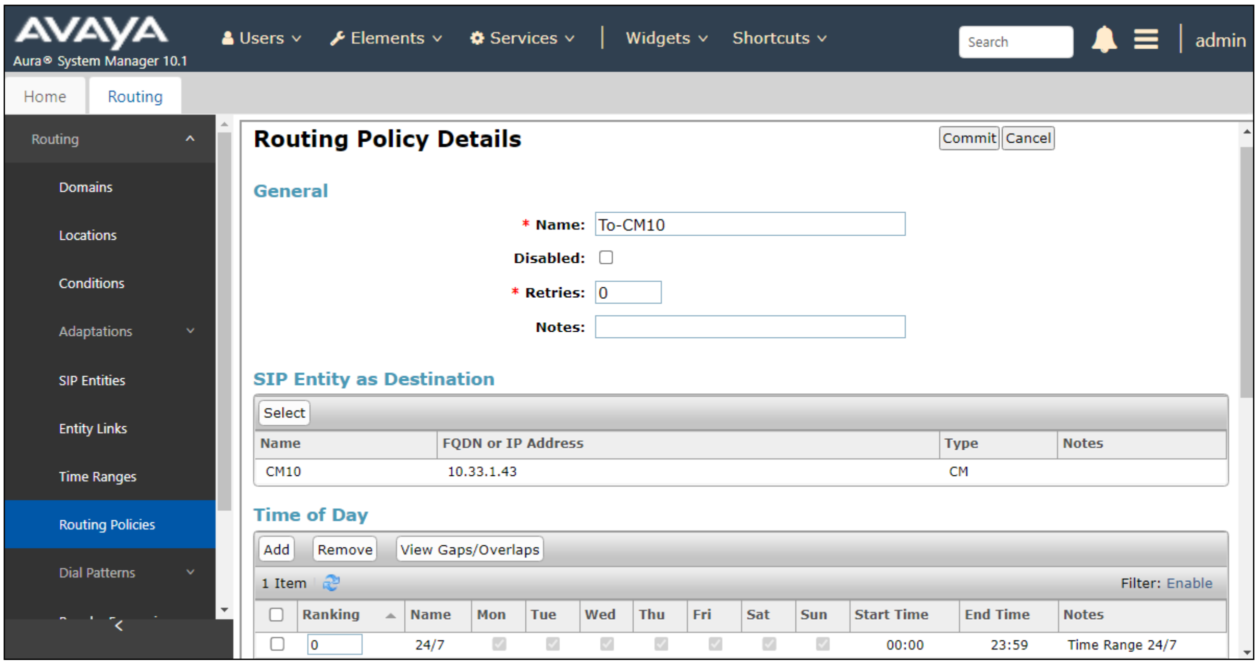


## 6.5.2. Routing Policy for Communication Manager

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** 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 6.4.2**. The screen below shows the result of the selection.



## 6.6. Administer Dial Patterns

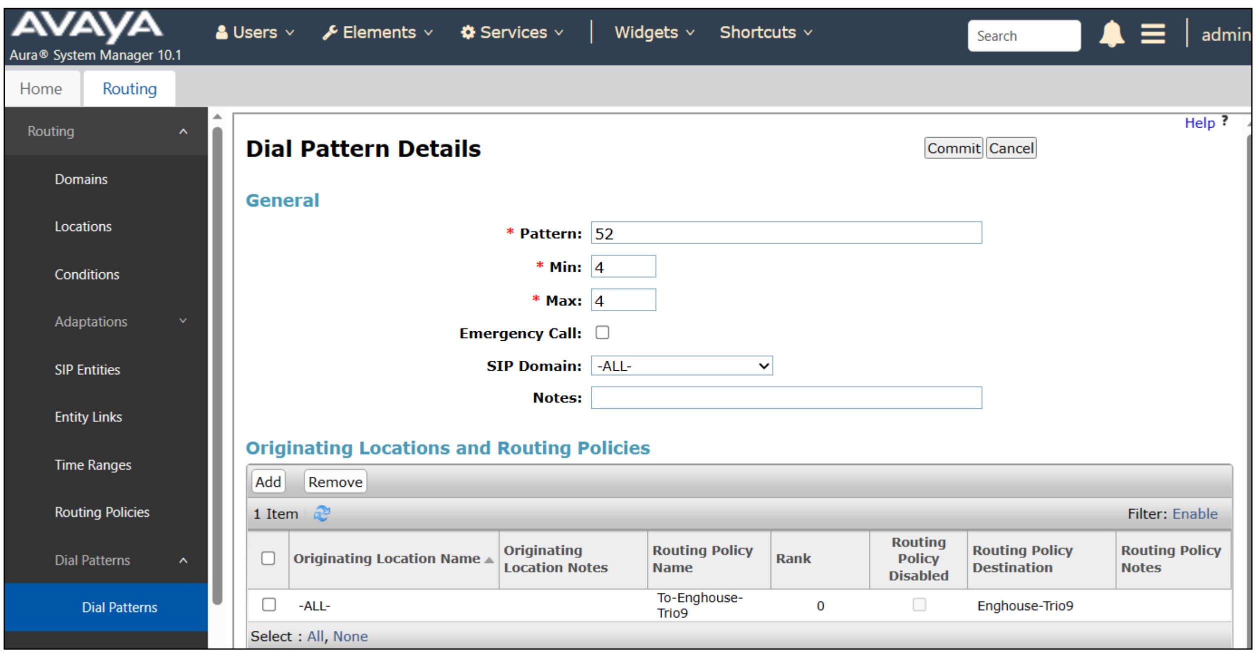
Add a new dial pattern for Trio and update existing dial patterns for Communication Manager.

### 6.6.1. Dial Pattern for Trio

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 the Trio server. 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:** A dial pattern to match, in this case "52".
- **Min:** The minimum number of digits to match, in this case "4" was used.
- **Max:** The maximum number of digits to match, in this case "4" was used.
- **SIP Domain:** Select the applicable domain, in this case "All" selected.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching the Trio server. In the compliance testing, the entry allowed for call originations from Communication Manager endpoint in locations "All". The routing policy **To-Enghouse-Trio9** from **Section 6.5.1** were selected as shown below.

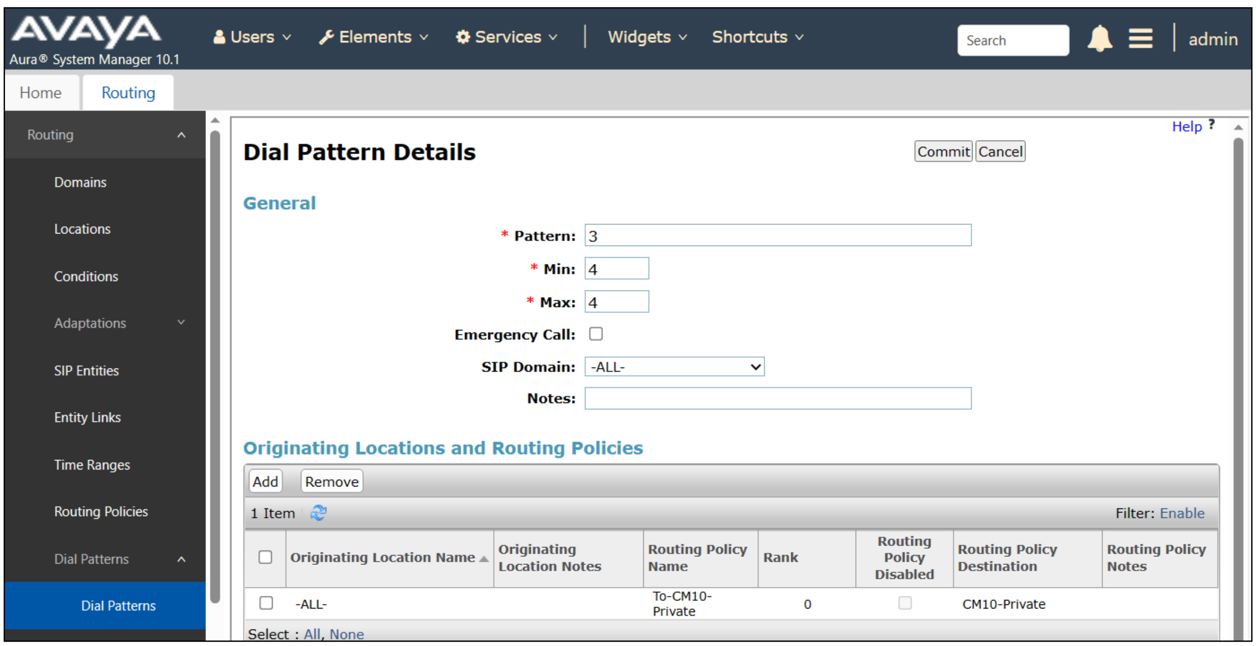


## 6.6.2. Dial Pattern for Communication Manager

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 Communication Manager. 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:** A dial pattern to match, in this case "3".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** Select the applicable domain, in this case "All".

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Communication Manager. In the compliance testing, the entry allowed for call originations from Communication Manager endpoint in locations "All". The Communication Manager routing policy from **Section 6.5.2** was selected as shown below.



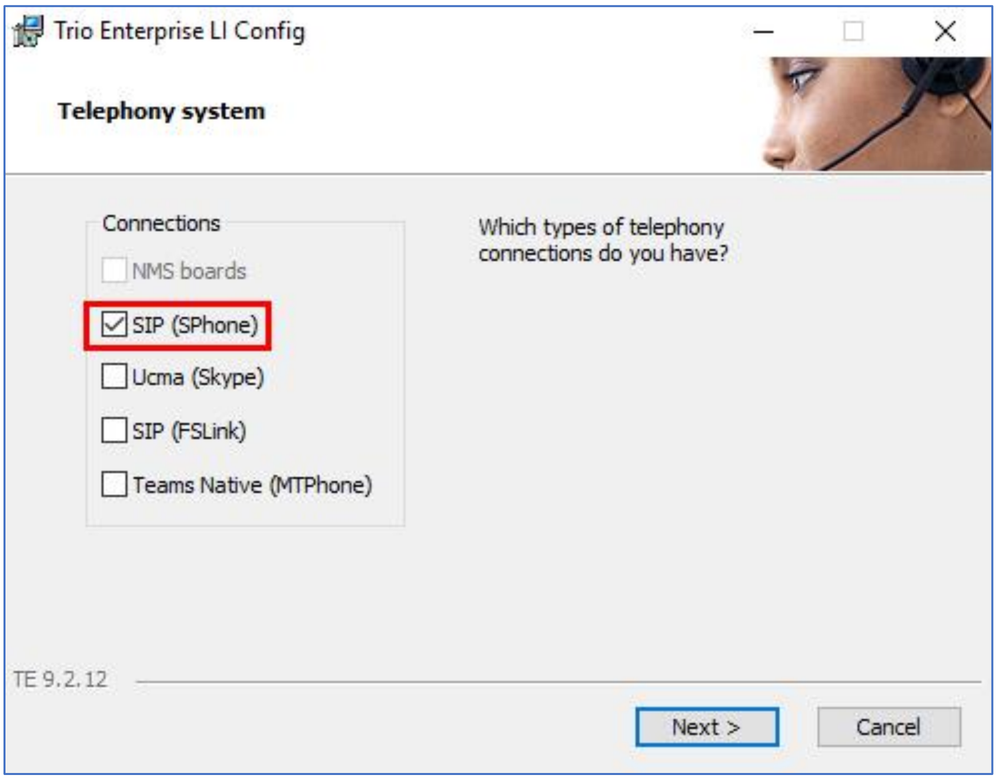
# 7. Configure Enghouse Trio

This section shows how to configure Trio to successfully connect to Session Manager. The installation of the Trio software is assumed to be completed and the Trio services are up.

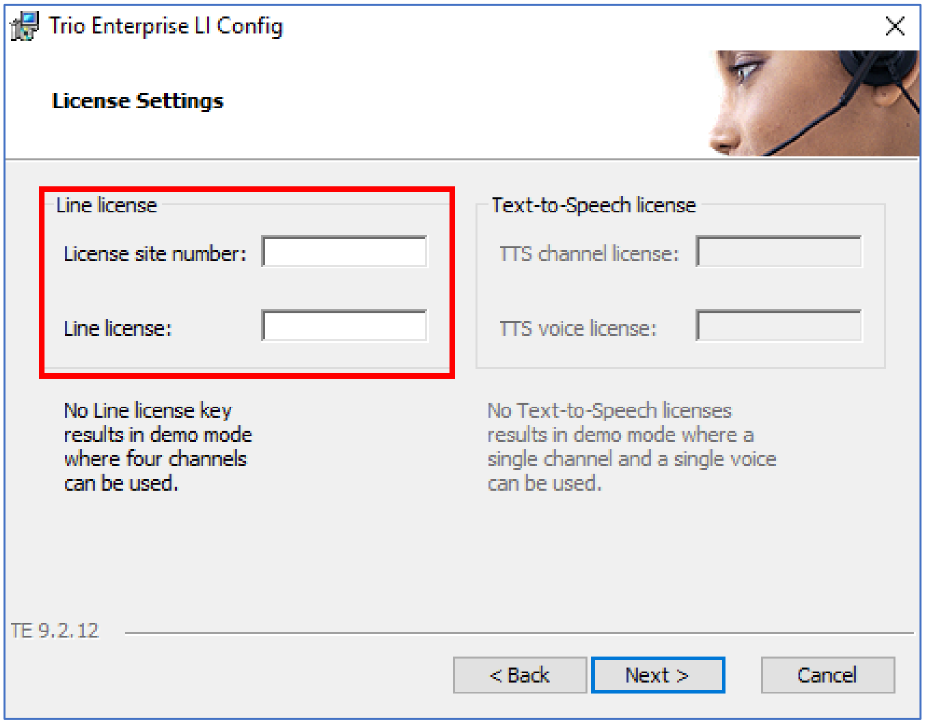
Trio Enterprise connects as a SIP Line Trunk to IP Office. This section shows how to configure Trio Enterprise to successfully connect to IP Office using SIP trunk. The installation of the Trio Enterprise software is assumed to be completed and the Trio services are up and running. The steps to configure a SIP Trunk are as follows.

Launch the 'TeleVoice Config' shortcut, 

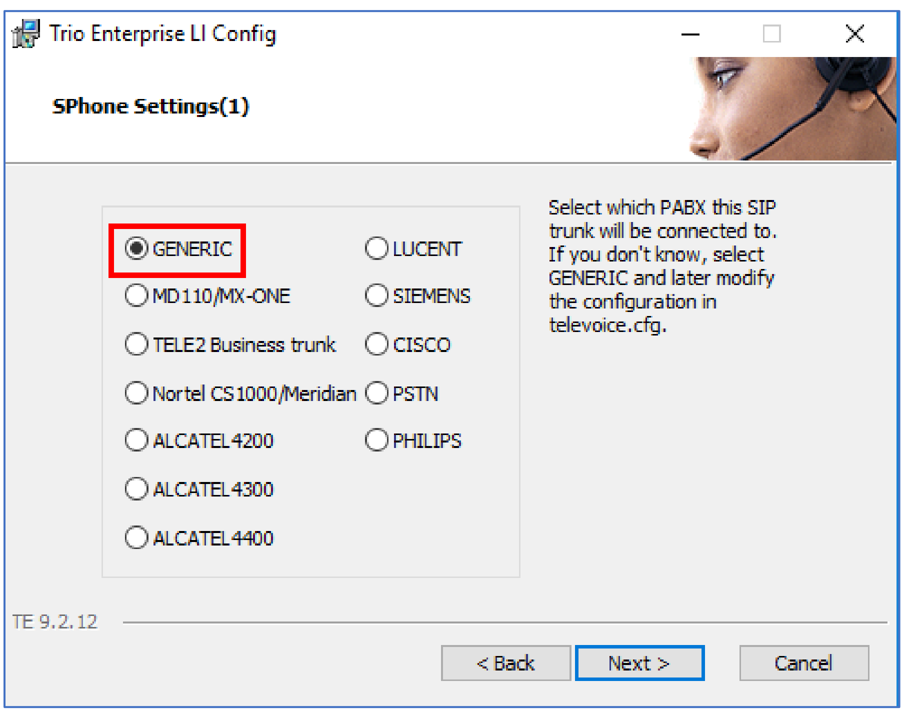

When the configuration window opens, check the **SIP** check box followed by the **Next** button.



In the subsequent window, enter the **License site number** and **Line license** as supplied directly by Enghouse Interactive AB or the Trio Enterprise reseller. Click on the **Next** button to continue.



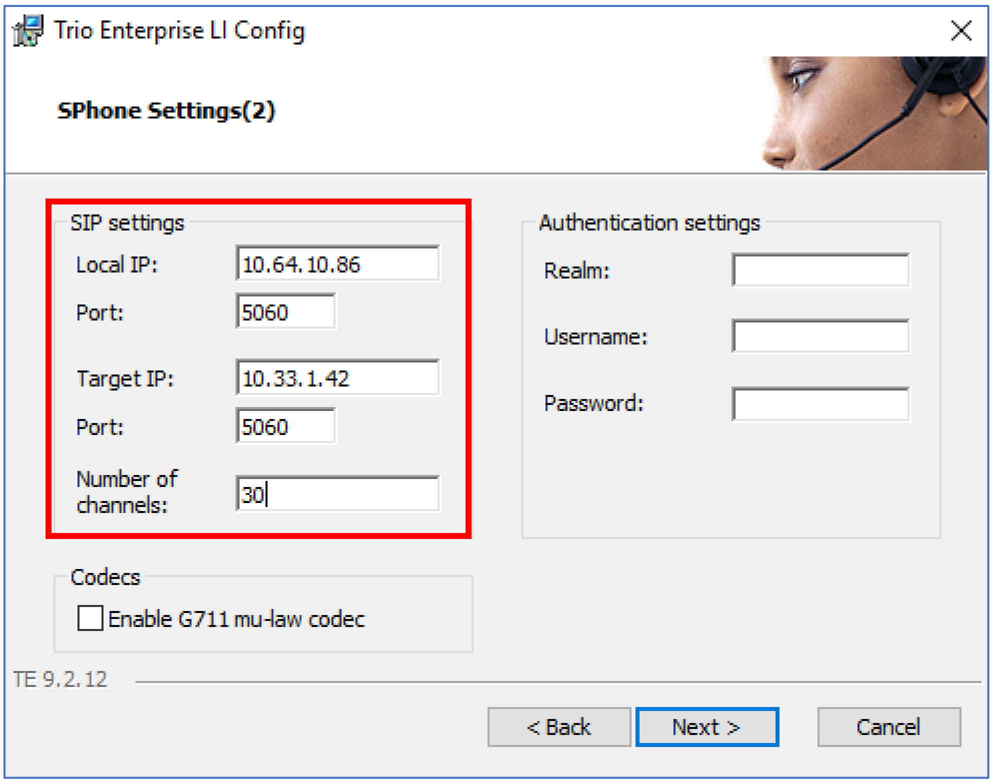

In the subsequent window, click on the GENERIC radio button followed by the **Next** button to continue.



In the subsequent window enter the following settings:

- **Local IP**: Enter the local IP address of the Trio Enterprise server.
- **Port**: Enter the SIP Port "5060".
- **Target IP**: Enter the IP address of the Session Manager.
- **Port**: Enter the SIP Port "5060".
- **Number of channels**: Enter "30" as the number of channels.

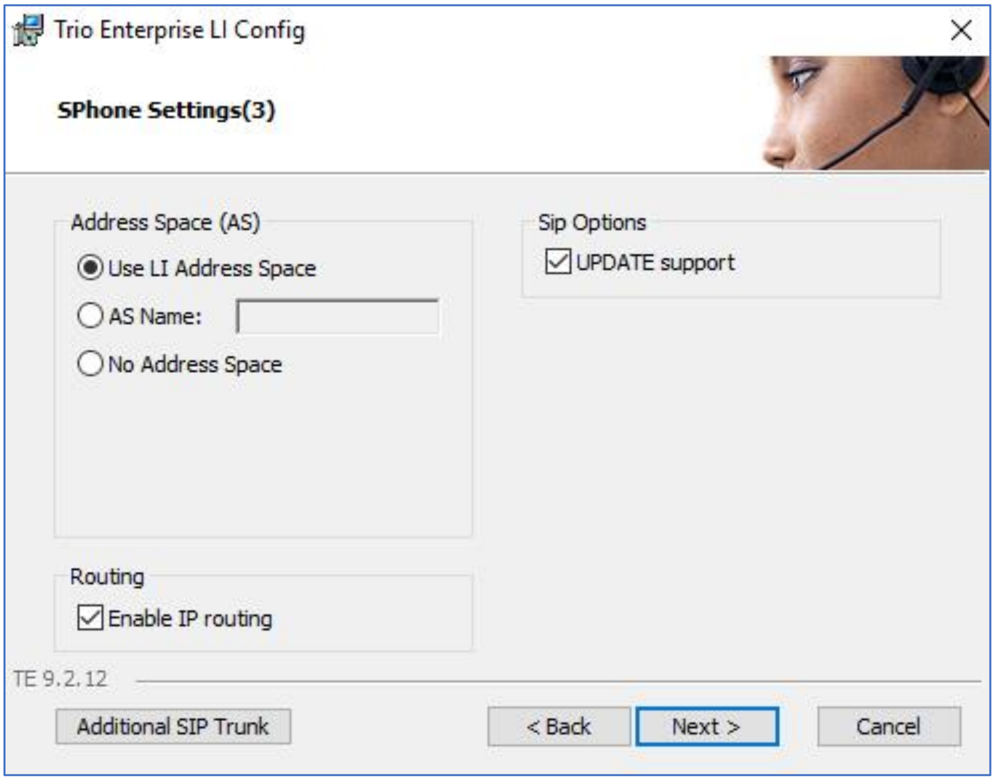
Click on the **Next** button to continue.



In the subsequent window enter the following settings:

- **Use LI Address Space**: Click on the radio button.
- **Enable IP routing**: Check the box.
- **UPDATE support**: Check the box.

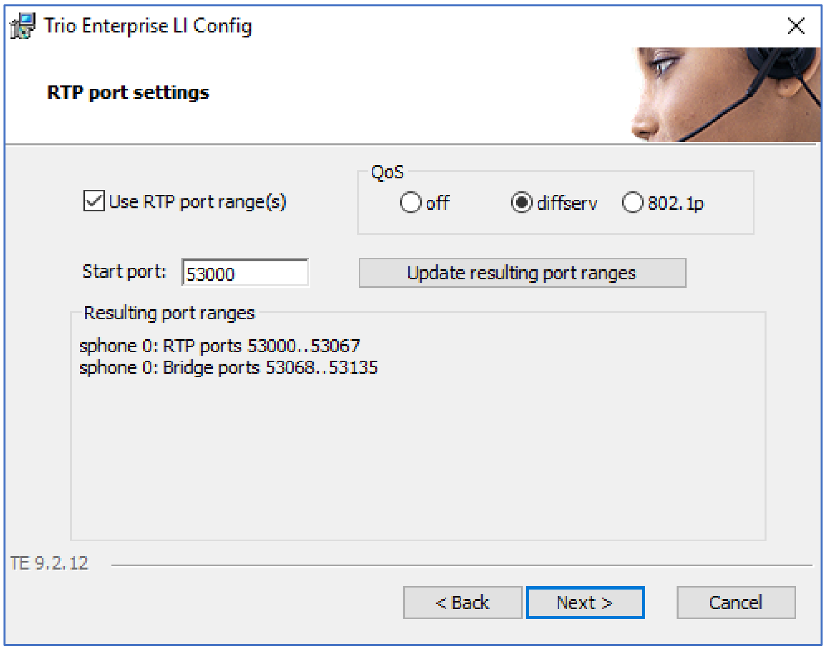
Click on the **Next** button to continue.



In the subsequent window enter the following settings:

- **Use RPT port range(s):** Check the box.
- **diffserv:** Click on the radio button.
- **Start port:** Enter "53000".

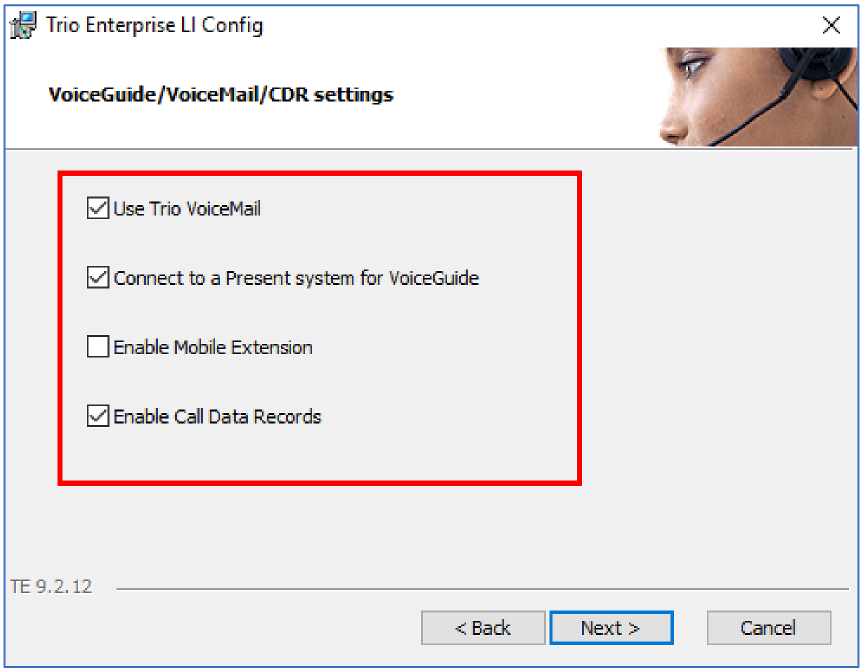
Click on the **Next** button to continue.



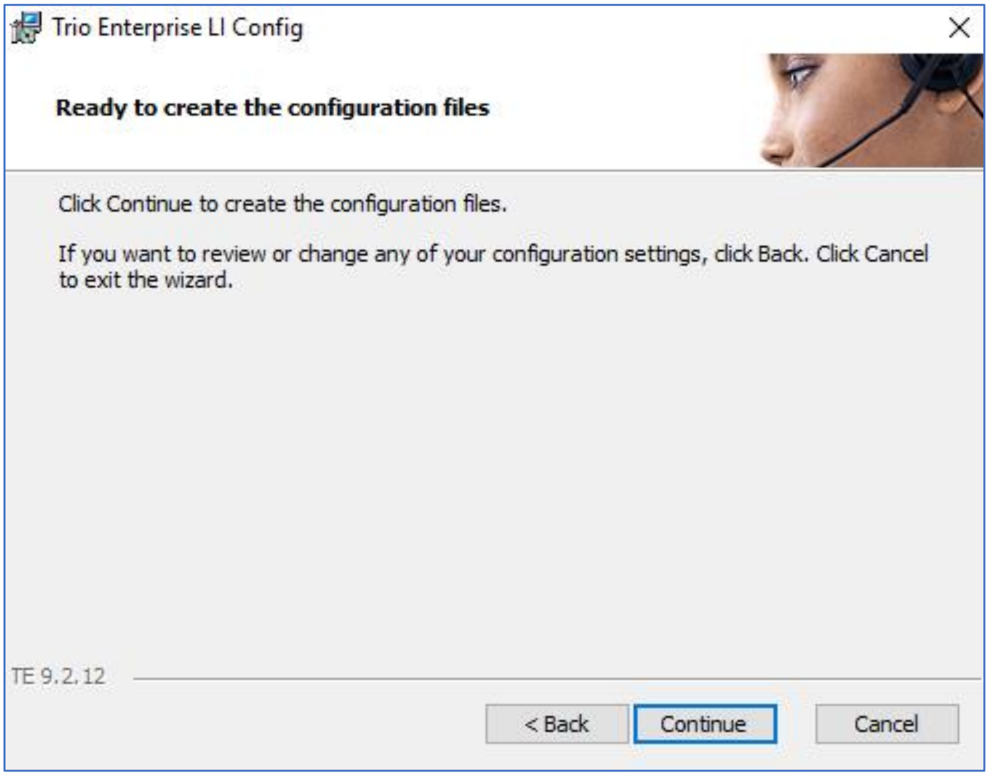
In the subsequent window enter the following settings:

- **Use Trio VoiceMail:** Check the box.
- **Connect to a Present system for VoiceGuide:** Check the box.
- **Enable Call Data Records:** Check the box.

Click on the **Next** button to continue.



In the subsequent window shown below, click on **Continue** button to complete the configuration for Trio.



# 8. Verification Steps

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

## 8.1. Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the "status trunk n" command, where "n" is the trunk group number administered in **Section** Error! Reference source not found.. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 1

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0001/001 T00001   in-service/idle    no
0001/002 T00002   in-service/idle    no
0001/003 T00003   in-service/idle    no
0001/004 T00004   in-service/idle    no
0001/005 T00005   in-service/idle    no
0001/006 T00006   in-service/idle    no
0001/007 T00007   in-service/idle    no
0001/008 T00008   in-service/idle    no
0001/009 T00009   in-service/idle    no
0001/010 T00010   in-service/idle    no
0001/011 T00011   in-service/idle    no
0001/012 T00012   in-service/idle    no
0001/013 T00013   in-service/idle    no
0001/014 T00014   in-service/idle    no
```

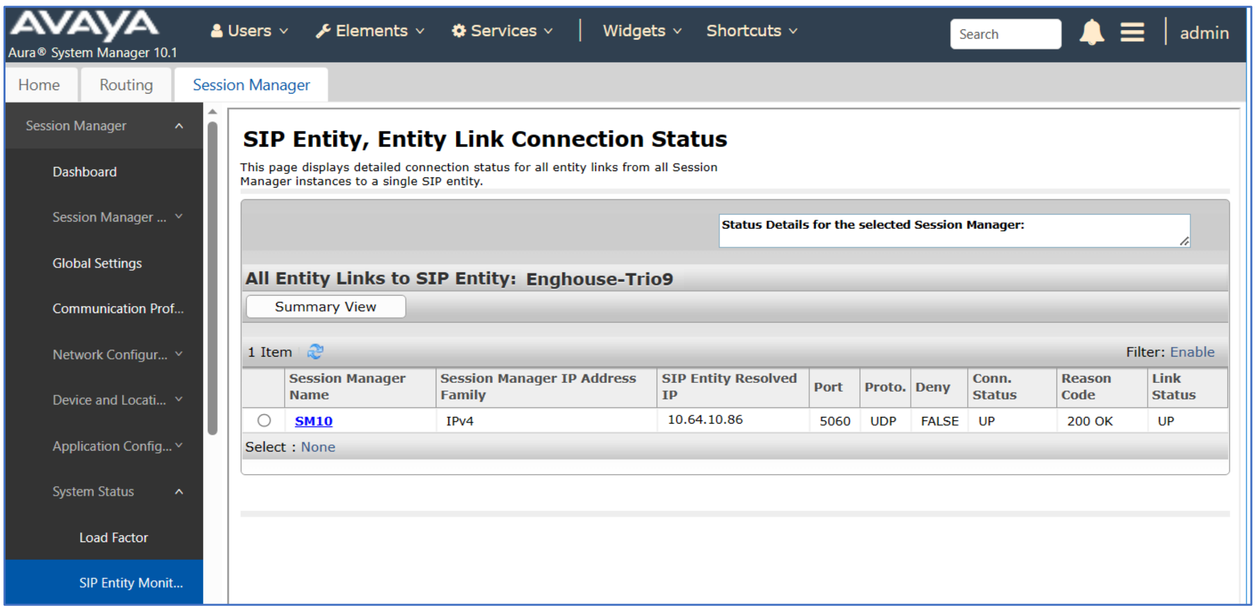
Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 5.4**. Verify that the **Group State** is "in-service", as shown below.

```
status signaling-group 1
                         STATUS SIGNALING GROUP

        Group ID: 1
      Group Type: sip

     Group State: in-service
```

## 8.2. Verify Session Manager

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

Select **Session Manager → System Status → SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen (not shown). Click the **Enghouse-Trio9** entity name. The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that the **Conn Status** and **Link Status** are "UP", as shown below.




## 8.3. Verify Trio Attendant

Trio calls the enterprise station to make it as the attendant and ready to receive incoming call, PSTN user places a call to Trio. Trio bridges the call to the attendant and verify the call is established between the Trio attendant and PSTN user.

# 9. Conclusion

These Application Notes describe the procedures required to configure Trio from Enghouse Interactive AB to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunks. All feature functionality test cases described in **Section** Error! Reference source not found. were passed with the observations noted in **Section** Error! Reference source not found..

# 10. Additional References

This section references the Avaya documentation that are relevant to these Application Notes. Product documentation for Avaya Aura® Session Manager, including the following, is available at: http://support.avaya.com/

[1] Administering Avaya Aura® Session Manager, Document 03-300509, Issue 10, Release 10.1, February 2023
[2] Administering Avaya Aura® System Manager, Issue 9.0, Release 10.1, February 2023
[3] Administering Avaya Aura® Communication Manager, Document 03-300509, Issue 10, Release 10.1, February 2023
[4] Avaya Aura® Communication Manager Feature Description and Implementation, Document 555-245-205, Issue 9.0, Release 10.1, February 2023

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