Avaya Solution & Interoperability Test Lab

Application Notes for Trio Enterprise to interoperate with Avaya Aura® Communication Manager, Avaya Aura® Session Manager and Avaya Aura® Application Enablement Services - Issue 1.0

Abstract

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

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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

1. Introduction

These Application Notes outline the steps necessary to configure Trio Enterprise from Enghouse Interactive AB to interoperate with Avaya Aura® Communication Manager (Communication Manager), Avaya Aura® Session Manager (Session Manager) and Avaya Aura® Application Enablement Services (Application Enablement Services). Trio Enterprise is a client/server-based application running on Windows Server operating systems. Trio Enterprise 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, it also includes its own built-in voice mail called Trio VoiceMail. The Trio Enterprise 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 Enterprise connects to the Communication Manager using a SIP trunk via the Session Manager. A TSAPI connection on Application Enablement Services enables the Trio Enterprise Absence integration. Trio Enterprise is supplied with all prerequisite software.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using Communication Manager. The Trio Enterprise server Communicates 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 Enterprise. Calls placed to the Trio Enterprise server automatically places a call to the telephone the Attendant is using for answering purposes. When the attendant answers the call the Trio Enterprise server bridges the two calls. When the attendant extends the call to another telephone, Trio Enterprise 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 digital station was used as the attendant's telephone.

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 Enterprise did not include use of any specific encryption features as requested by Enghouse Interactive AB.

2.1. Interoperability Compliance Testing

The interoperability compliance testing included feature and serviceability testing. The serviceability testing introduced failure scenarios to see if Trio Enterprise could resume after a link failure with Communication Manager/Application Enablement Services. 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
- Absence detection
- Message Waiting Indicator

2.2. Test Results

Tests were performed to ensure full interoperability between Trio Enterprise and Avaya Communication Manager. The tests were all functional in nature and performance testing was not included. All test cases passed successfully with the following observation,

- The Codec Set List in Communication Manager cannot have both G.711MU and G.711A together. In case they are present together, then the Codec priority must be configured on the Trio Enterprise Server. Failing to do so, will cause Avaya SIP and digital stations to not hear early media (any pre-recorded audio) from the attendant. Refer to **Sections 5.8** and **8.3**.

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 Enterprise server via the Session Manager. TSAPI is configured on the Trio Enterprise server which enables the Trio Enterprise to interact with telephone on the Communication Manager to act as the Attendant telephone via the Application Enablement Services. An Avaya digital station was used as the Trio Enterprise Attendant telephone during compliance testing. SIP and H.323 stations were configured on the Communication Manager to generate outbound/inbound calls to/from the PSTN. A PRI/T1 trunk on Media Gateway G450 was configured to connect to the simulated PSTN.

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

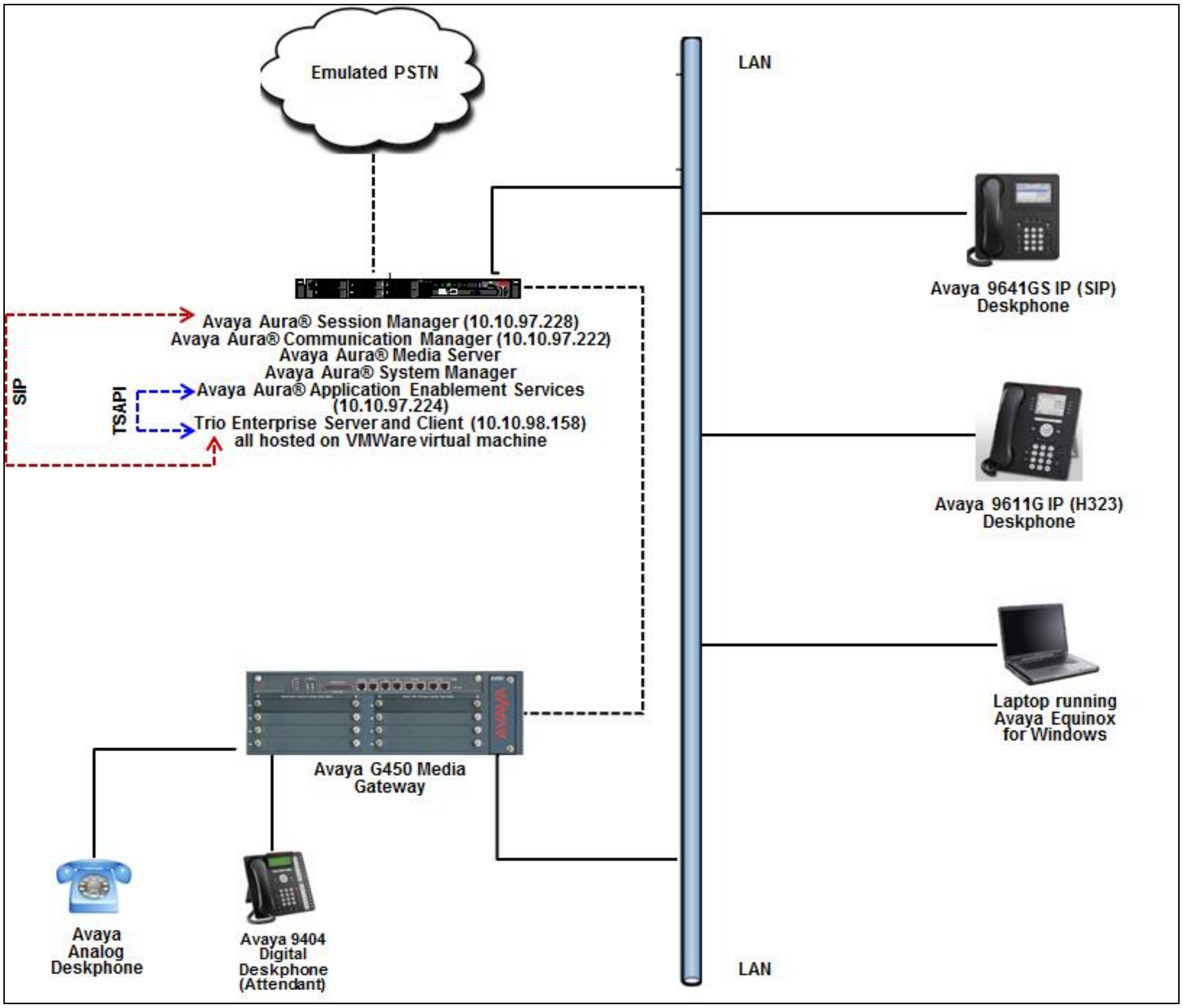


Figure 1: Avaya and Trio Enterprise Reference Configuration

4. Equipment and Software Validated

The following equipment and version were used in the reference configuration described above:

Equipment/Software	Release/Version
Avaya Aura® Communication Manager running on virtualized environment	08.0.0.0.822
Avaya Aura® Application Enablement Services running on virtualized environment	8.0.0.0.0.6-0
Avaya Aura® Session Manager running on virtualized environment	8.0.0.0.800035
Avaya Aura® System Manager	8.0.0.0.931077
Avaya Aura® Media Server	8.0.0.117
Avaya G450 Media Gateway	40.10.0/1
Avaya IP Deskphones - 9641GS (SIP) - 9611G (H.323)	 7.1.3.0.8 6.6604
Avaya Equinox for Windows	3.4.0.152.46-ACW-INTEGRATIONNEXUS1
Avaya 9404 Digital Telephone	R18
Avaya Analog Telephone	N/A
Trio Enterprise Server and Client running on Microsoft Windows 2012 R2 Server TSAPI Client for Windows	7.1 8.0.0-38

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 11**.

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 trunk group
- Administer SIP signalling group
- Administer SIP Trunk Group Members
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer dial plan
- Administer uniform dial plan
- Administer AAR analysis
- Configure interface to Application Enablement Services
- Create a CTI Link to the Application Enablement Services
- Configure Absence diversion

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.

Verify that the **Computer Telephony Adjunct Links** customer option is set to "y" on **Page 4**. If this option is not set to "y", then contact the Avaya sales team or business partner for a proper license file.

```
display system-parameters customer-options                      Page   2 of  12
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 20
           Maximum Concurrently Registered IP Stations: 18000 3
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 128   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 36000 3
                   Maximum Video Capable IP Softphones: 18000 3
                       Maximum Administered SIP Trunks: 12000 58
  Maximum Administered Ad-hoc Video Conferencing Ports: 12000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0


display system-parameters customer-options                      Page   4 of  12
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? y          Audible Message Waiting? y
        Access Security Gateway (ASG)? n              Authorization Codes? y
        Analog Trunk Incoming Call ID? y                       CAS Branch? n
 A/D Grp/Sys List Dialing Start at 01? y                         CAS Main? n
Answer Supervision by Call Classifier? y               Change COR by FAC? n
                                 ARS? y  Computer Telephony Adjunct Links? y
                ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? y
          ARS/AAR Dialing without FAC? y                     DCS (Basic)? y
          ASAI Link Core Capabilities? y                DCS Call Coverage? y
          ASAI Link Plus Capabilities? y               DCS with Rerouting? y
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? y
              ATM WAN Spare Processor? n                          DS1 MSP? y
                                ATMS? y             DS1 Echo Cancellation? y
                  Attendant Vectoring? y
```

5.2. Administer System Parameter Features

During compliance testing Trio Enterprise 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? n
 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? n
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:
            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
SEND ALL CALLS OPTIONS
     Send All Calls Applies to: station    Auto Inspect on Send All Calls? 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 Enterprise.

```
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? n
               Call Classification After Answer Supervision? n
                                          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, **SM-VM** and **10.10.97.228** are entered as **Name** and **IP Address**. Note the **procr** and **10.10.97.222** 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 5.5**. The node **Name** and **IP Address** for Application Enablement Services is **10.10.97.224**, which will be used later in the Application Enablement Services configuration as shown in **Section 5.14**.

```
change node-names ip
                                 IP NODE NAMES
    Name              IP Address
SM-VM               10.10.97.228
procr               10.10.97.222
devvmaes            10.10.97.224
```

5.4. 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".

```
add trunk-group 1                                                     Page   1 of  22
                                TRUNK GROUP

Group Number: 1                   Group Type: sip           CDR Reports: y
  Group Name: Trunk to SM on VM             COR: 1        TN: 1        TAC: #001
   Direction: two-way        Outgoing Display? y
 Dial Access? n                                   Night Service:
Queue Length: 0
Service Type: tie                    Auth Code? n
                                              Member Assignment Method: auto
                                                       Signaling Group: 1
                                                     Number of Members: 24
```

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

```
add trunk-group 1                                                      Page   3 of  22
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y



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

                                                  Hold/Unhold Notifications? y
                                Modify Tandem Calling Number: no
```

5.5. 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 5.3**.
- **Far-end Node Name:** The existing node name for Session Manager from **Section 5.3**.
- **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:** The applicable domain name for the network.
- **Direct IP-IP Audio Connections?:** "y".
- **Initial IP-IP Direct Media?:** "y". This can be set to "n" or "y".

```
display 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? y
  Peer Detection Enabled? y  Peer Server: SM
 Prepend '+' to Outgoing Calling/Alerting/Diverting/Connected Public Numbers? y
Remove '+' from Incoming Called/Calling/Alerting/Diverting/Connected Numbers? n
Alert Incoming SIP Crisis Calls? n
   Near-end Node Name: procr                    Far-end Node Name: SM-VM
 Near-end Listen Port: 5061                   Far-end Listen Port: 5061
                                           Far-end Network Region: 1

Far-end Domain: bvwdev.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? y
         Enable Layer 3 Test? y                     Initial IP-IP Direct Media? y
H.323 Station Outgoing Direct Media? n              Alternate Route Timer(sec): 6
```

5.6. Administer SIP Trunk Group Members

Use the "change trunk-group n" command, where "n" is the trunk group number from **Section 5.4**. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **Signalling Group:** The signalling group number from **Section 5.5**.
- **Number of Members:** The desired number of members, in this case "24".

```
change trunk-group 1                                            Page   1 of  22
                                TRUNK GROUP

Group Number: 1                   Group Type: sip          CDR Reports: y
  Group Name: Trunk to SM on VM          COR: 1        TN: 1        TAC: #001
   Direction: two-way        Outgoing Display? y
 Dial Access? n                                 Night Service:
Queue Length: 0
Service Type: tie                  Auth Code? n
                                            Member Assignment Method: auto
                                                     Signaling Group: 1
                                                   Number of Members: 24
```

5.7. 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 5.5**.

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 Enterprise.

```
change ip-network-region 1                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location:          Authoritative Domain: bvwdev.com
    Name: Region1                 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
```

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

```
change ip-network-region 1                                          Page   4 of   20

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

5.8. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.7**. Update the audio codec types in the **Audio Codec** fields as necessary. As per the observation noted in **Section 2.2** only configure either G.711MU or G.711A. The codec shown below was used in the compliance testing since Trio Enterprise had made the codec priority changes to accommodate both G.711MU and G.711A.

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

                          IP CODEC SET
    Codec Set: 1

    Audio        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  Size(ms)
 1: G.711MU           n          2        20
 2:
Media Encryption                  Encrypted SRTCP: enforce-unenc-srtcp
 1: 1-srtp-aescm128-hmac80
 2: 2-srtp-aescm128-hmac32
 3: none
```

5.9. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Trio Enterprise, 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 5.4**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1      Pattern Name: To SM on VM
    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                     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  none
```

5.10. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to Trio Enterprise. Add an entry for the trunk group defined in **Section 5.4**. In the example shown below, all calls originating from a 5-digit extension beginning with "56" and routed to trunk group "1" 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  56                1                           5     Total Administered: 4
                                                         Maximum Entries: 540
```

5.11. Administer Dial Plan

This section provides a sample dial plan used for routing calls with dialed digits 71xxx to Trio Enterprise. Use the "change dialplan analysis 0" command and add an entry to specify the use of digits pattern "71", as shown below.

```
display dialplan analysis                                       Page   1 of  12
                             DIAL PLAN ANALYSIS TABLE
                                   Location: all           Percent Full: 2

    Dialed    Total  Call      Dialed    Total  Call      Dialed    Total  Call
    String    Length Type      String    Length Type      String    Length Type
   1            4    ext
   71           5    udp
```

5.12. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialled digits 71xxx to Trio Enterprise. 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 71xxx, 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
 71               5   0                   aar  n
```

5.13.Administer AAR Analysis

Use the "change aar analysis 0" command and add an entry to specify how to route calls to 71xxx. In the example shown below, calls with digits 71xxx will be routed as an AAR call using route pattern "1" from **Section 5.9**.

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

          Dialed           Total     Route    Call   Node  ANI
          String          Min  Max  Pattern   Type   Num   Reqd
        71                  5    5      1         aar          n
```

5.14. Configure Interface to Avaya Aura® Application Enablement Services

To configure the Application Enablement Services link, use the "change ip-services" command and enter the following in **Page 1**:

- **Type:** Enter "AESVCS"
- **Enabled:** Enter "y"
- **Local Node:** Enter "procr"
- **Port:** Enter "8765"

```
change ip-services                                                Page   1 of   4

                                   IP SERVICES
 Service      Enabled      Local        Local        Remote       Remote
  Type                     Node         Port         Node         Port
AESVCS          y       procr            8765
```

Navigate to **Page 4** and enter the following:

- **Server ID:** Enter "1"
- **AE Services:** Enter "devvmaes" (The node created in **Section 5.3**. Also note that the name entered in this field should be matched with the host name of Application Enablement Services server)
- **Password:** Enter a password. This password will be used in **Section 6.3**to enable the Application Enablement Services to communicate with the Communication Manager.
- **Enabled:** Enter "y"

```
change ip-services                                                Page   4 of   4
                            AE Services Administration

   Server ID    AE Services        Password           Enabled    Status
                   Server
      1:       devvmaes          *                     y         in use
```

5.15. Create a CTI Link to the Aura® Application Enablement Services

A CTI Link needs to be created to enable the Communication Manager to interoperate with the Application Enablement Services. Use the "add cti-link next" command (Note, during compliance testing cti link 1 was added) and enter the following:

- **Extension:** Enter any unused Extension (i.e. 56000).
- **Type:** Enter "ADJ-IP".
- **Name:** Enter a descriptive name (i.e. DevvmAES)

```
add cti-link 1                                                Page   1 of   3
                                    CTI LINK
 CTI Link: 1
Extension: 56000
     Type: ADJ-IP
                                                                          COR: 1
     Name: DevvmAES
```

5.16. Configure Absence diversion

A VDN extension followed by a reason code (list of reason code 1 to 9 is managed on Trio Enterprise) and # can be dialed by users to initiate a diversion for specific reasons. An absence diversion can be cancelled by dialing the VDN extension followed by # #. The following steps are needed to configure Absence diversions:

- Configure VDN 1
- Configure Vector 1
- Configure VDN 2
- Configure Vector 2

5.16.1. Configure VDN 1

During compliance testing VDN 56007 was used. Use the “add vdn x” command, (where x is the VDN) and configure the following:

- **Name*:** Enter an informative name (i.e. Phone diversion).
- **Destination:** Enter “Vector Number 7”.

```
add vdn 56007                                                  Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 56007
                                 Name*: Phone Diversion
                           Destination: Vector Number        7
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none     Report Adjunct Calls as ACD*? n


        VDN of Origin Annc. Extension*:
```

5.16.2. Configure Vector 7

Configure the Vector that was used as the **Vector Number** in **Section 5.16.1**. Use the "add vector 7" command, and configure the following:

- **Name:** Enter an informative name (i.e. Phone diversion).
- **Line 01:** Enter "wait-time 1 secs hearing silence".
- **Line 02:** Enter "collect 9 digits after announcement none for none".
- **Line 03:** Enter "route-to number 56008 with cov n if unconditionally".

In this example, using monitored phone dial 56007 + reason code + #, call is routed to 56008 which will trigger Trio Enterprise to set the phone absence with appropriate reason announcement.

```
add vector 7                                                  Page   1 of   6
                                  CALL VECTOR

    Number: 7               Name: Phone Diversion
Multimedia? n      Attendant Vectoring? n     Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y    BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    1   secs hearing silence
02 collect      9    digits after announcement none     for none
03 route-to     number 56008            with cov n if unconditionally
04
```

5.16.3. Configure VDN 2

Configure a VDN using the "route-to number" as used in **Section 5.16.2**. This VDN is used for activating referrals from the phone set. Use the "add vdn 56008" command, and configure the following:

- **Name*:** Enter an informative name (i.e. diversion).
- **Destination:** Enter "Vector Number 8".

```
display vdn 56008                                                 Page   1 of   3
                            VECTOR DIRECTORY NUMBER

                             Extension: 56008
                                 Name*: Diversion
                           Destination: Vector Number        8
                   Attendant Vectoring? n
                  Meet-me Conferencing? n
                    Allow VDN Override? n
                                   COR: 1
                                   TN*: 1
                              Measured: none     Report Adjunct Calls as ACD*? n


        VDN of Origin Annc. Extension*:
```

5.16.4. Configure Vector 8

Configure the Vector that was used as the "Vector Number" in **Section 5.16.3**. Use the "add vector 8" command, and configure the following:

- **Name:** Enter an informative name (i.e. Diversion).
- **Line 01** Enter "wait-time 100 secs hearing ringback".
- **Line 02** Enter "stop".

```
display vector 8                                              Page   1 of   6
                                  CALL VECTOR

    Number: 8                Name: Diversion
Multimedia? n      Attendant Vectoring? n    Meet-me Conf? n           Lock? n
     Basic? y   EAS? y   G3V4 Enhanced? y   ANI/II-Digits? y   ASAI Routing? y
 Prompting? y   LAI? y  G3V4 Adv Route? y   CINFO? y   BSR? y   Holidays? y
 Variables? y   3.0 Enhanced? y
01 wait-time    100 secs hearing ringback
02 stop
03
```

6. Configuration of Avaya Aura® Application Enablement Services

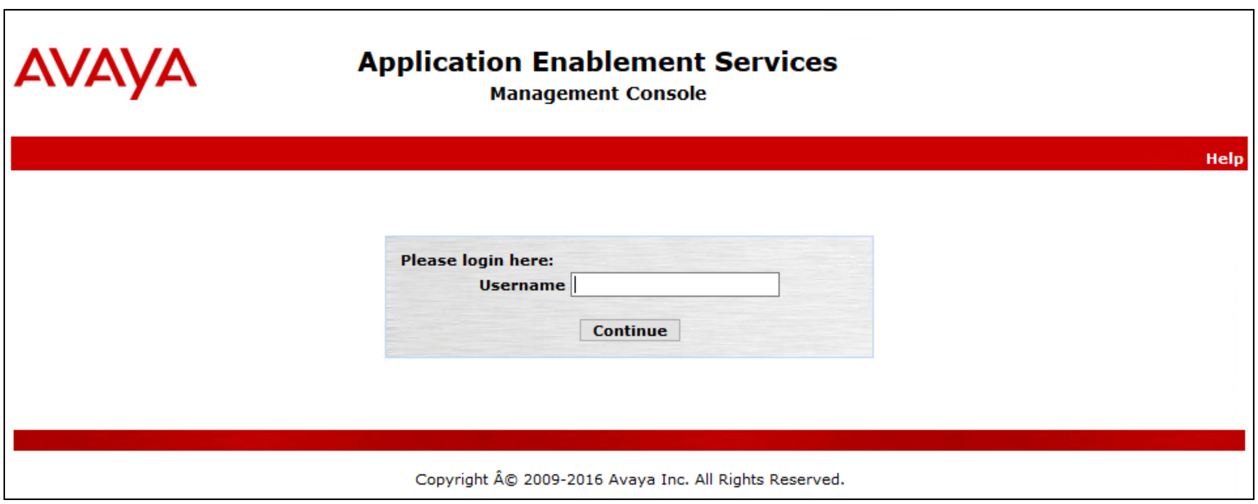
This section provides the procedures for configuring Application Enablement Services. It is implied a working Application Enablement Services is already in place and the Security Database (SDB) is configured. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 11**. The configuration operations described in this section can be summarized as follows:

- Logging into Avaya Aura® Application Enablement Services
- Verify Avaya Aura® Application Enablement Services License
- Create an Avaya Aura® Communication Manager Switch Connection
- Create a TSAPI Link
- Create CTI User
- Configure Security Database
- Obtain Tlink Name
- Disable Security Database
- Enable Ports
- Restart TSAPI Service

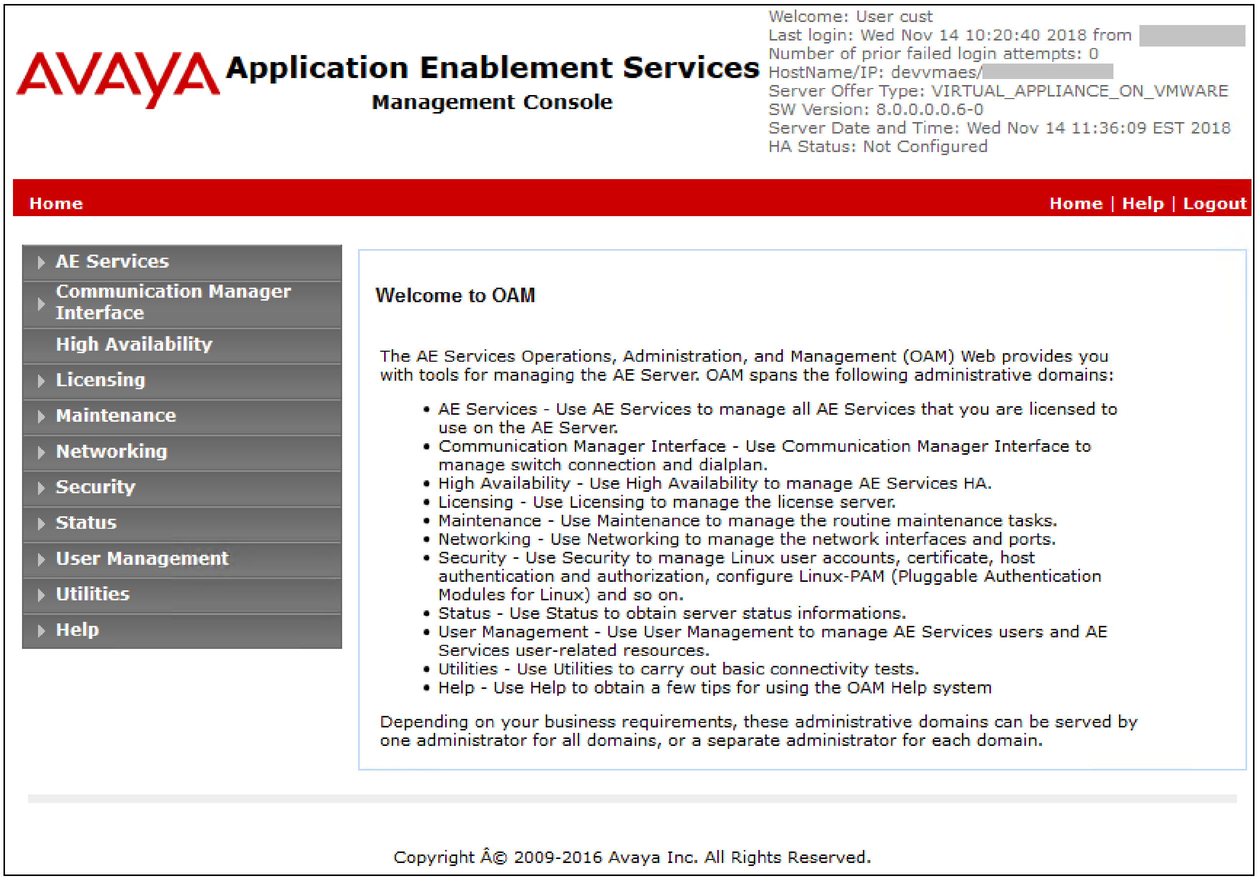
6.1. Logging into the Avaya Aura® Application Enablement Services

Access the OAM web-based interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the Application Enablement Services server.

The **Please login here** screen is displayed. Log in using the appropriate credentials.

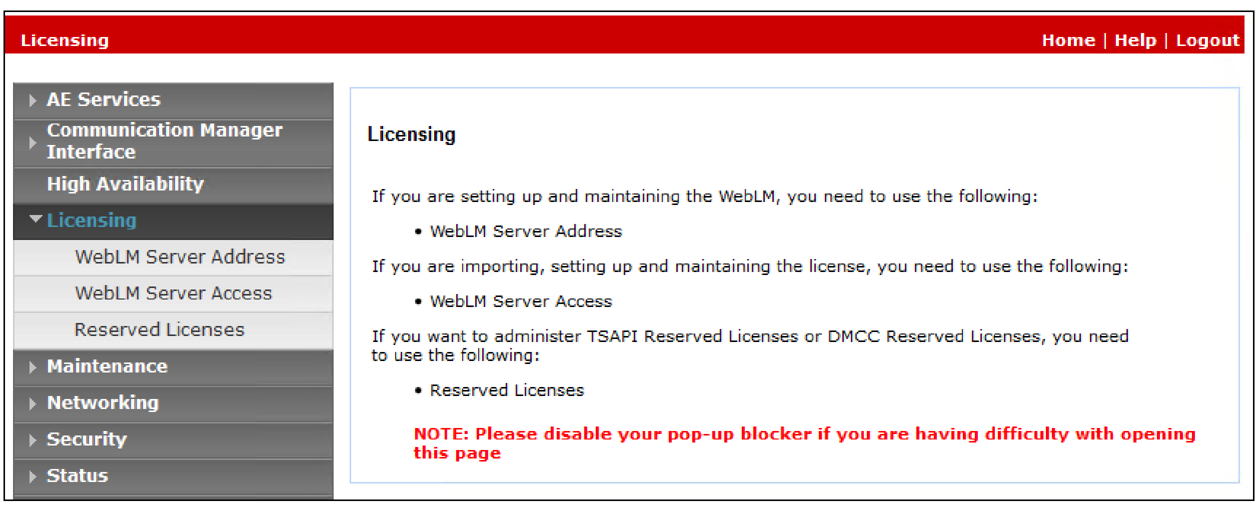


The **Welcome to OAM** screen is displayed next.



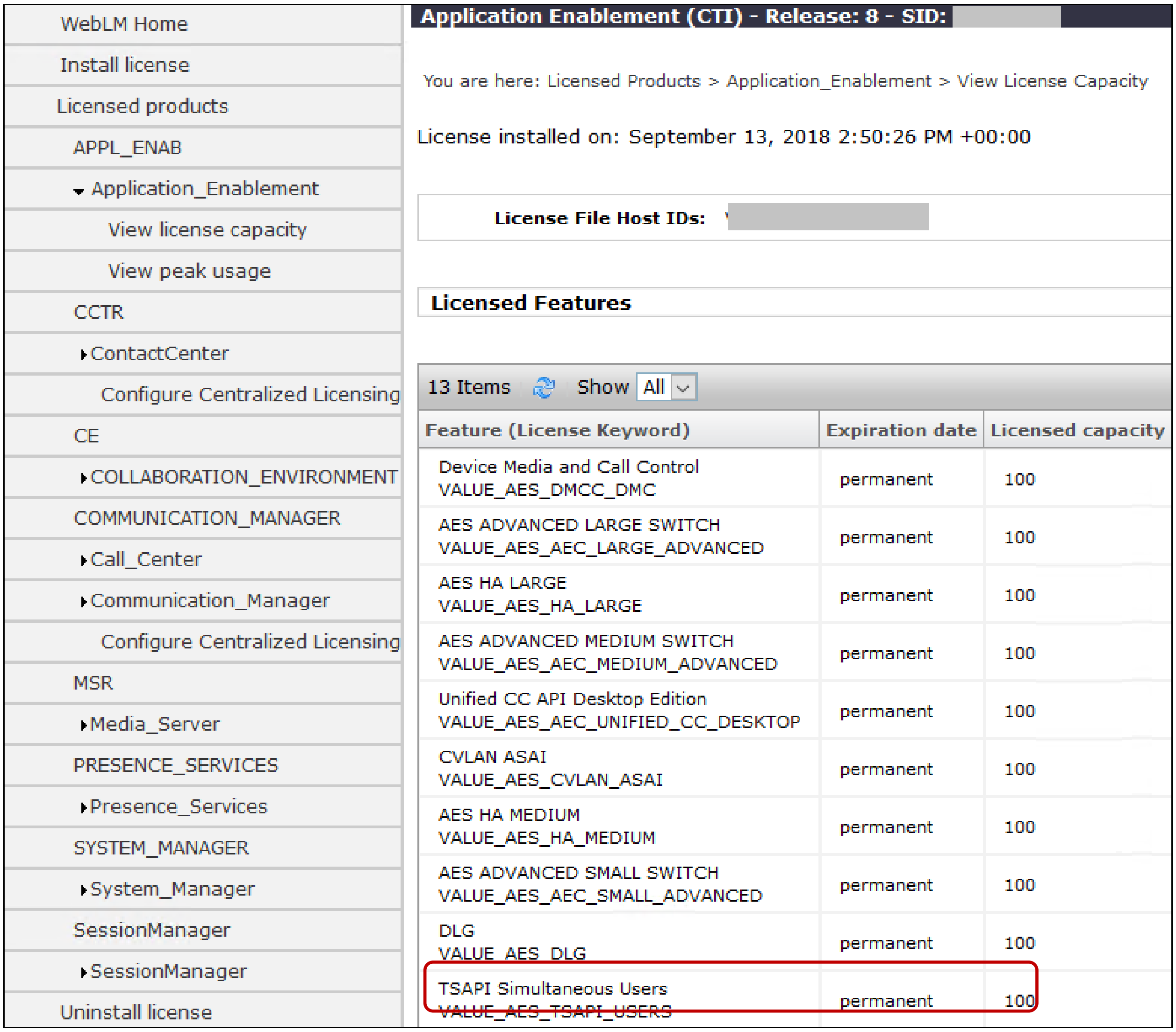
6.2. Verify Avaya Aura® Application Enablement Services License

Select **Licensing → WebLM Server Access** in the left pane, to display the **Web License Manager** pop-up screen (not shown), and log in using the appropriate credentials.



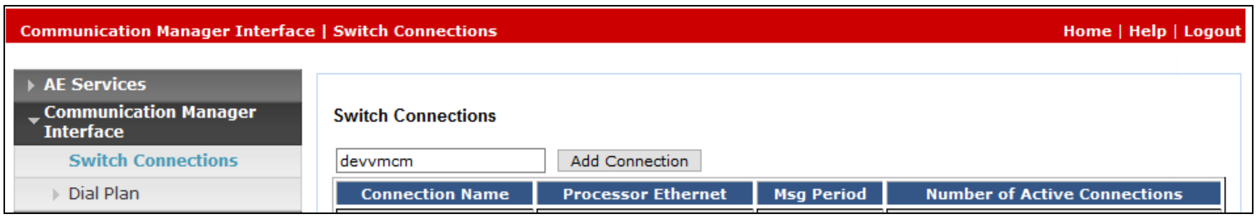
The **Web License Manager** screen below is displayed. Select **Licensed products → APPL_ENAB → Application_Enablement** in the left pane, to display the **Application Enablement (CTI)** screen in the right pane.

Verify that there are sufficient licenses for **TSAPI Simultaneous Users** as shown below. Note that the TSAPI license is required for Telephony Web Service.

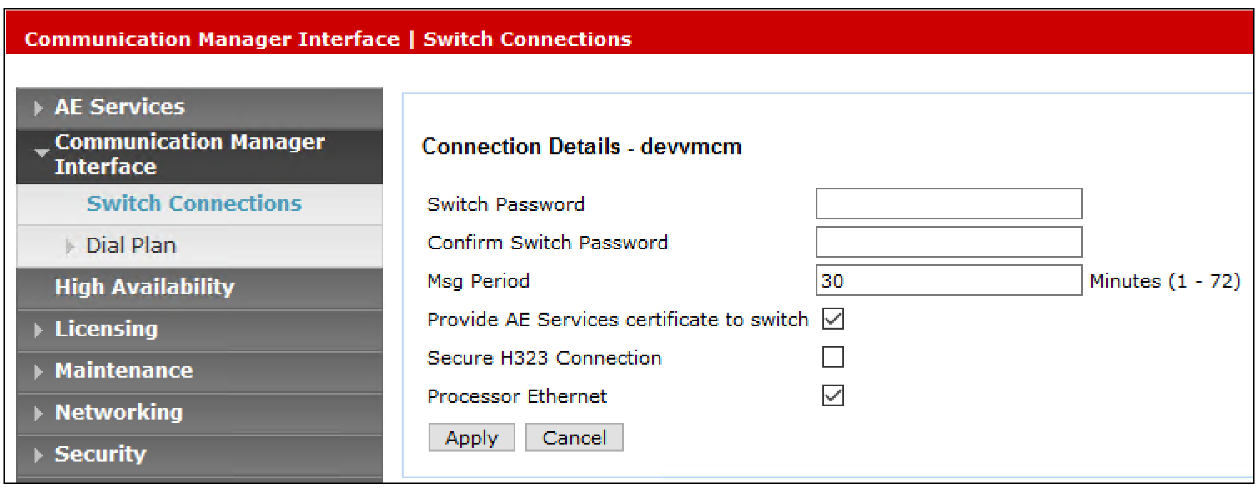


6.3. Create an Avaya Aura® Communication Manager Switch Connection

A Communication Manager Switch Connection needs to be created to enable the Application Enablement Services to communicate with the Communication Manager. Navigate to **Communication Manager Interface → Switch Connections**. In the **Switch Connections** page, enter an informative name for the Communication Manager (i.e. devvmcm). Click on the **Add Connection** button.

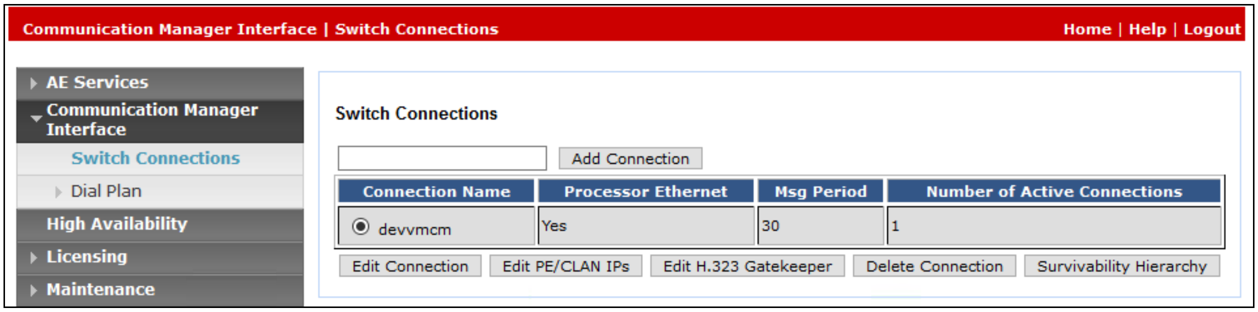


In the **Connection Details** window, enter the **Switch Password** as was configured in **Section 5.14** and **Confirm Switch Password**. Click on the **Apply** button.

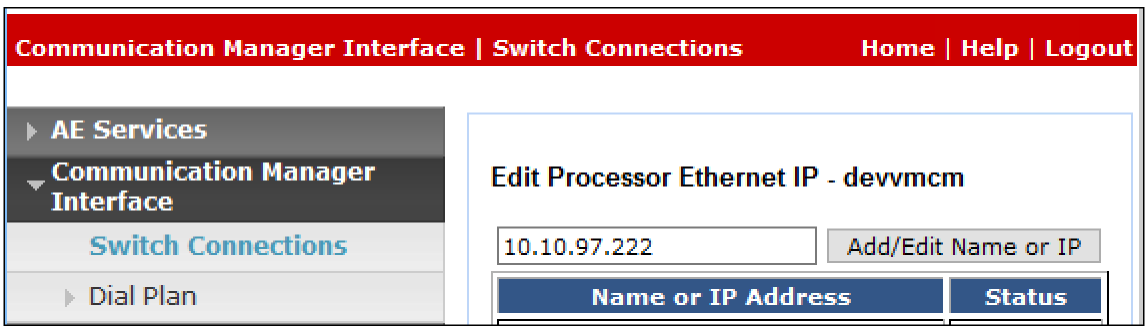


Select **Communication Manager Interface → Switch Connections** from the left pane. The **Switch Connections** screen shows a listing of the existing switch connections.

Locate the connection name associated with the relevant Communication Manager, in this case "devvmcm", and select the corresponding radio button. Click **Edit PE/CLAN IPs**.

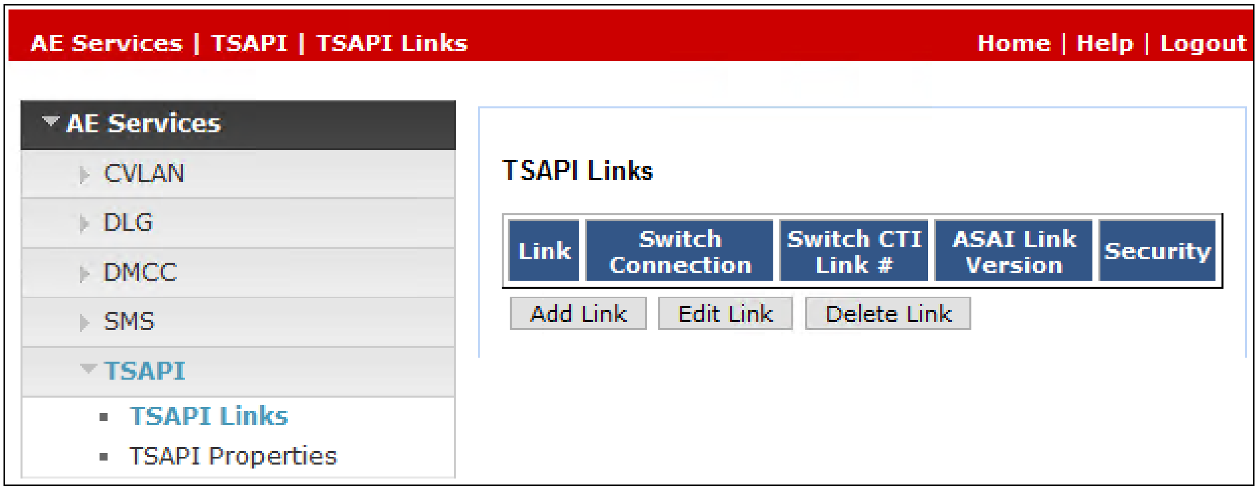


The **Edit Processor Ethernet IP** screen is displayed. Enter the IP address of a C-LAN circuit pack or the Processor C-LAN on Communication Manager to be used, in this case "10.10.97.222" as shown below, which is the Processor C-LAN on Communication Manager. Click **Add/Edit Name or IP**.



6.4. Create a TSAPI Link

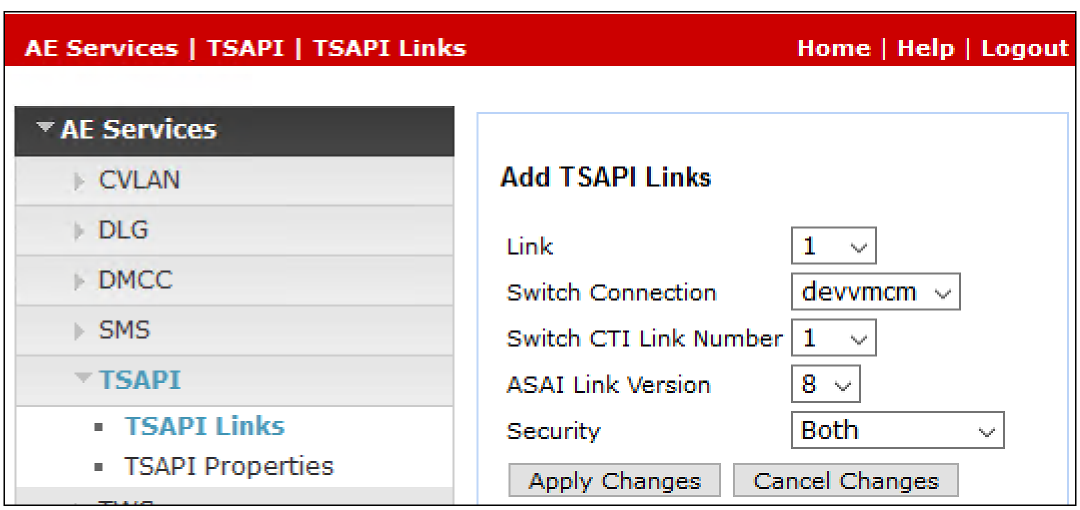
A TSAPI Link needs to be created to interoperate with Trio Enterprise. Navigate to **AE Services → TSAPI → TSAPI Links** and click on the **Add Link** button.



Once the **Add TSAPI Links** window opens enter the following:

- **Link:** Select the next available Link from the drop-down box
- **Switch Connection:** Select "devvmcm" from the drop-down box. (The Switch connection as created in **Section 6.3**)
- **Switch CTI Link Number:** Select "1" from the drop-down box. (The CTI link as created in **Section 5.15**)
- **ASAI Link Version:** Select "8" from the drop-down box.
- **Security:** Select "Both" from the drop-down box

Click on the **Apply Changes** button.

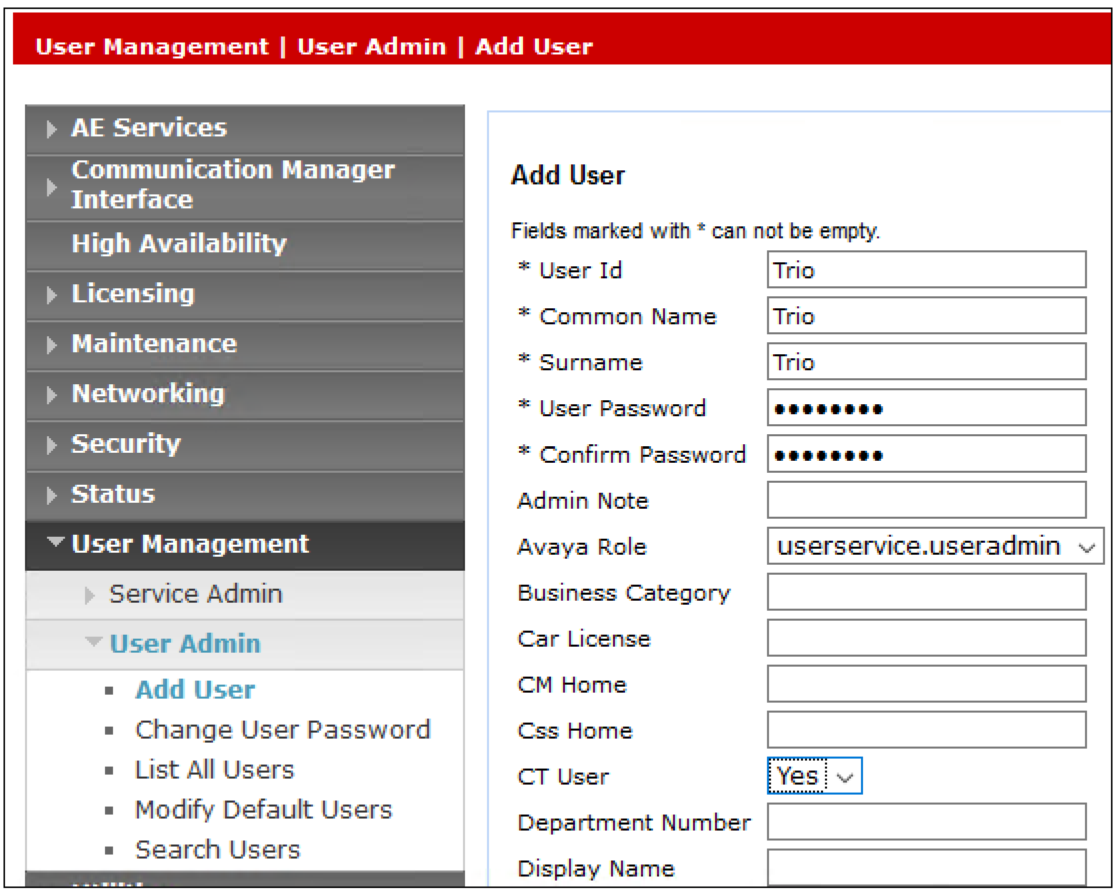


6.5. Create CTI User

Navigate to **User Manager → User Admin** and select **Add User**. On the **Add User** screen enter the following:

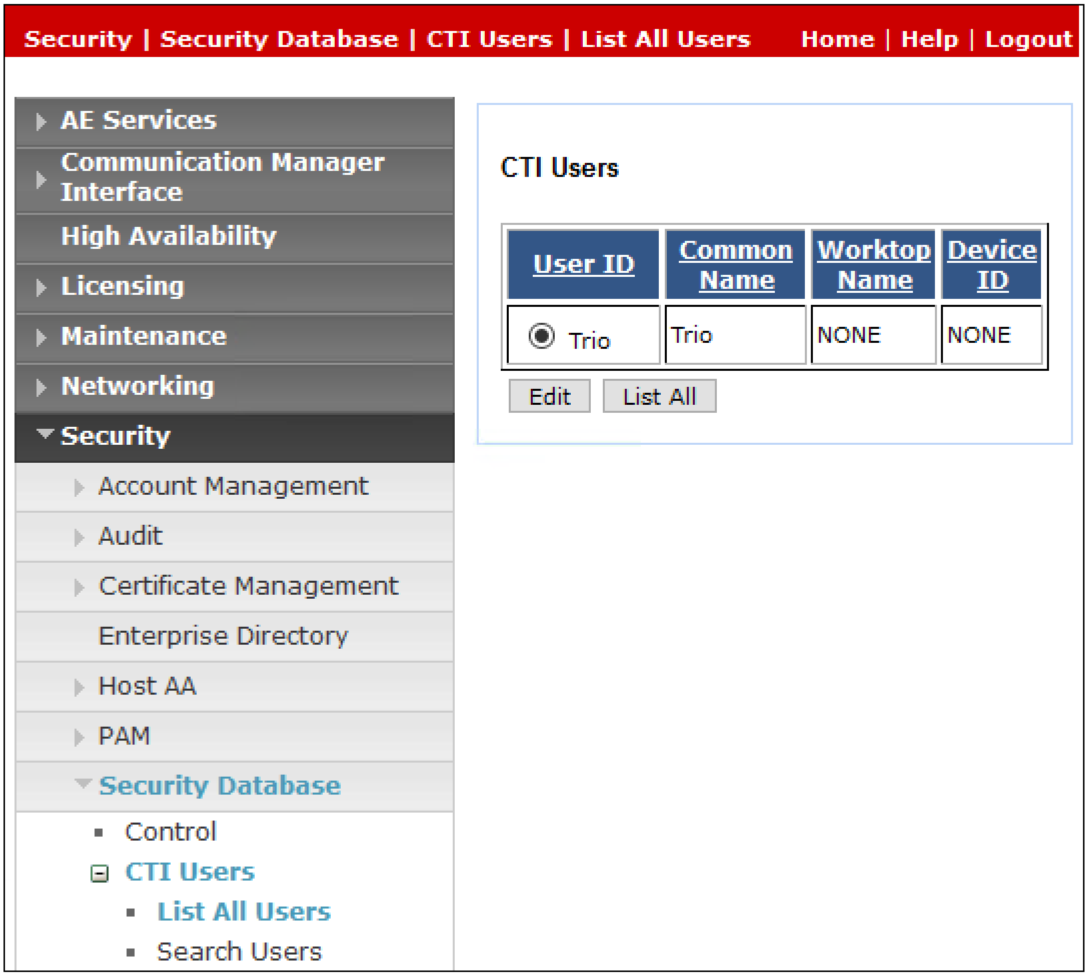
- **User Id**: Enter an informative name (i.e. T**rio**. This ID is required for the Trio Enterprise installation
- **Common Name**: Enter a Common Name (i.e. T**rio**)
- **Surname**: Enter a Surname (i.e. T**rio**)
- **User Password:** Enter a password. This password is being required for the Trio Enterprise Installation
- **Confirm Password:** Confirm the password
- **Avaya Role** Select "userservice.useradmin" from the drop-down box
- **CT User:** Select "Yes" from the drop-down box

Click the **Apply** button at the bottom of the screen (not shown).

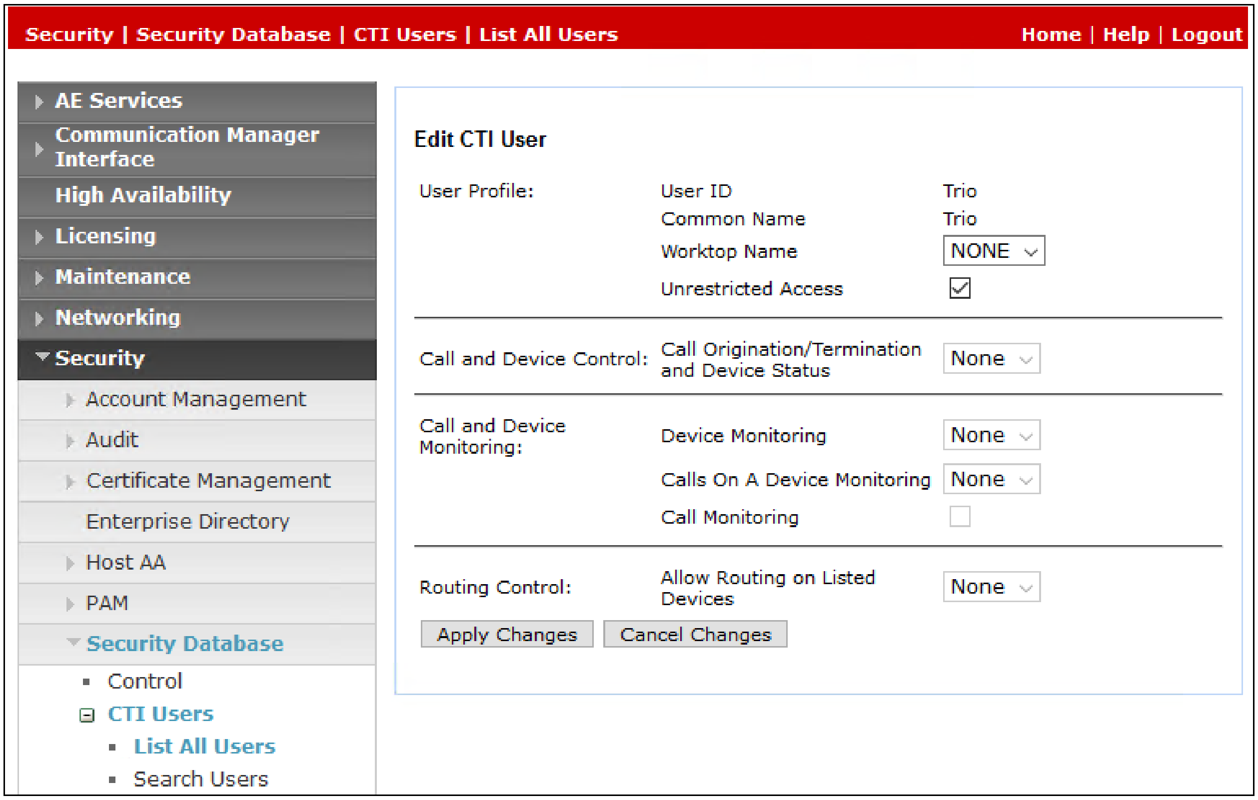


6.6. Configure Security Database

Navigate to the All Users screen by selecting **Security → Security Database → CTI Users → List All Users.** In the **CTI Users** window, select the radio button relating to the CTI user created in **Section 6.5** (**Trio**) and click on the **Edit** button.



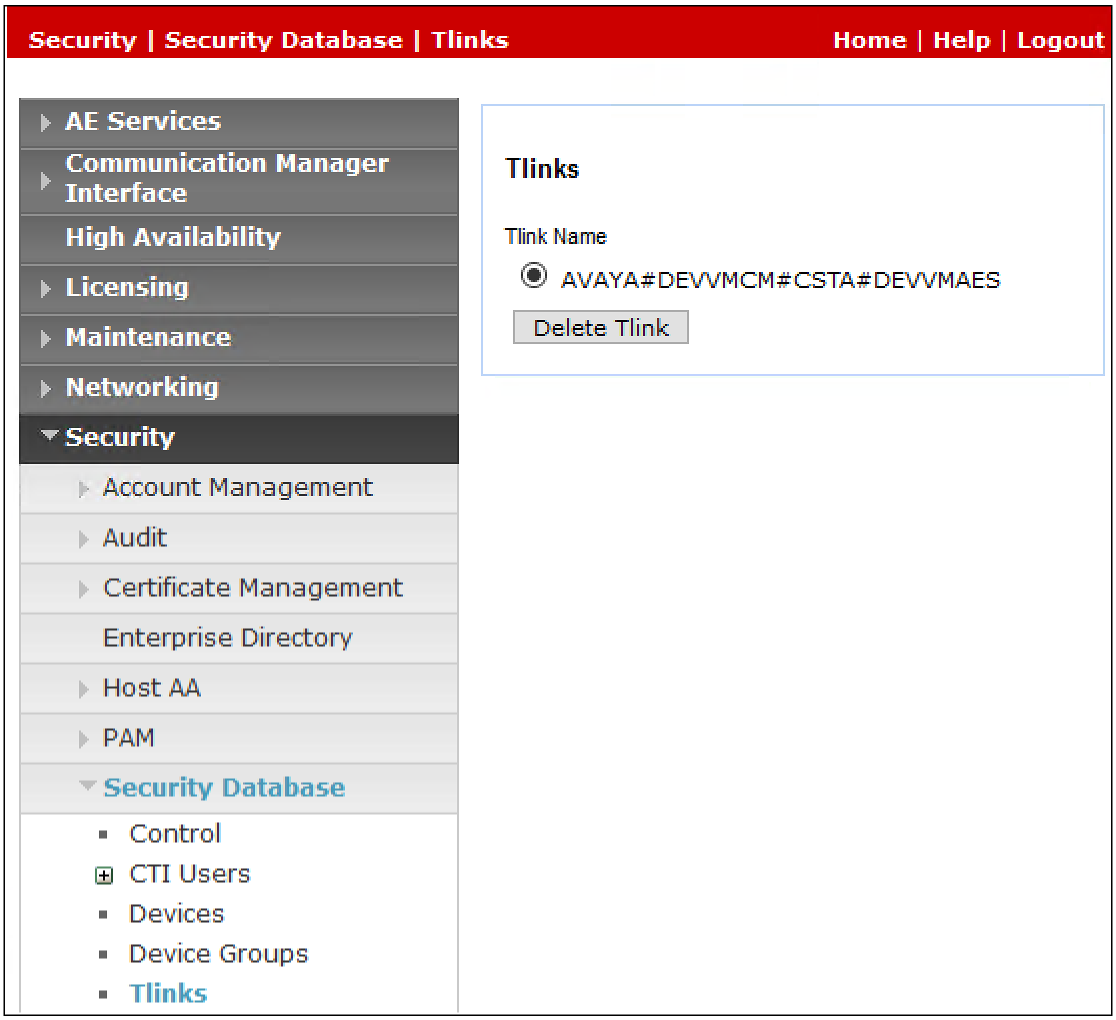
Once the **Edit CTI User** page appears, select the **Unrestricted Access** check box and click on the **Apply Changes** button.



6.7. Obtain Tlink Name

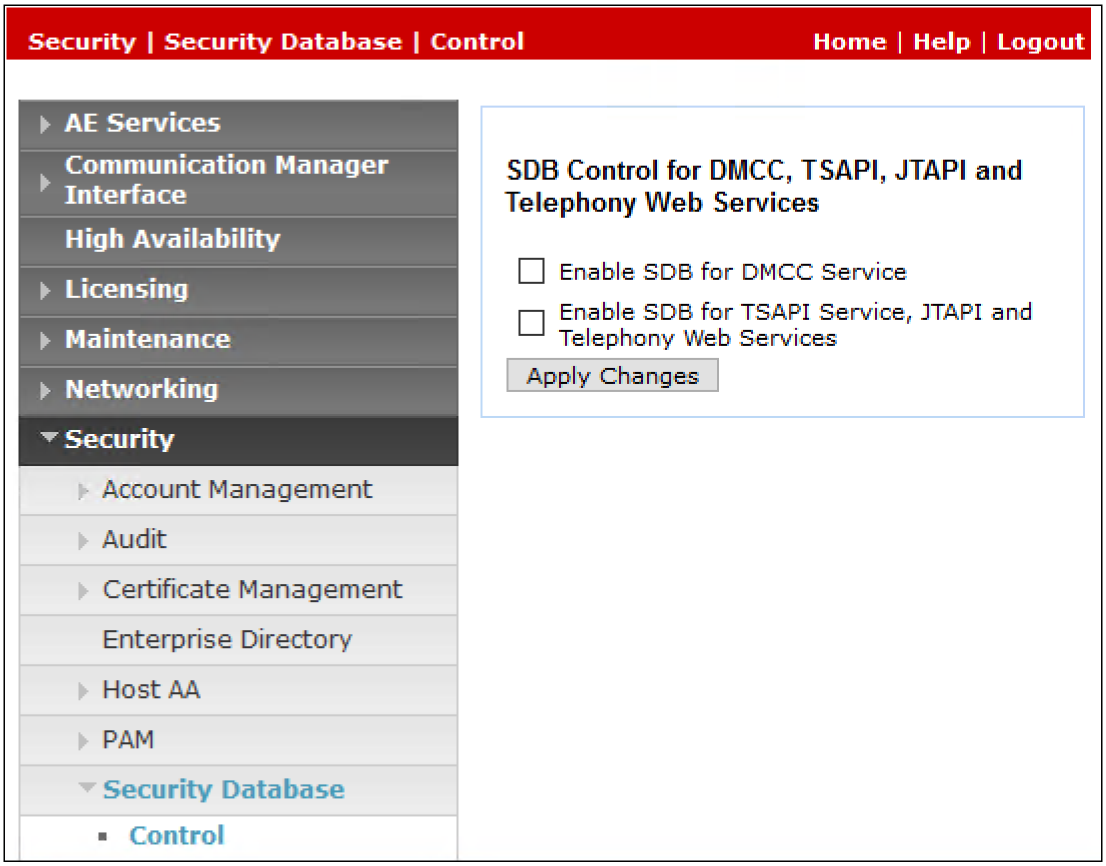
Select **Security → Security Database → Tlinks** from the left pane. The **Tlinks** screen shows a listing of the Tlink names. A new Tlink name is automatically generated for the TSAPI service. Locate the Tlink name associated with the relevant switch connection, which would use the name of the switch connection as part of the Tlink name. Make a note of the associated Tlink name, to be used later for configuring Trio Enterprise.

In this case, the associated Tlink name is "AVAYA#DEVVMCM#CSTA#DEVVMAES". Note the use of the switch connection "devvmcm" from **Section 6.3** as part of the Tlink name.



6.8. Disable Security Database

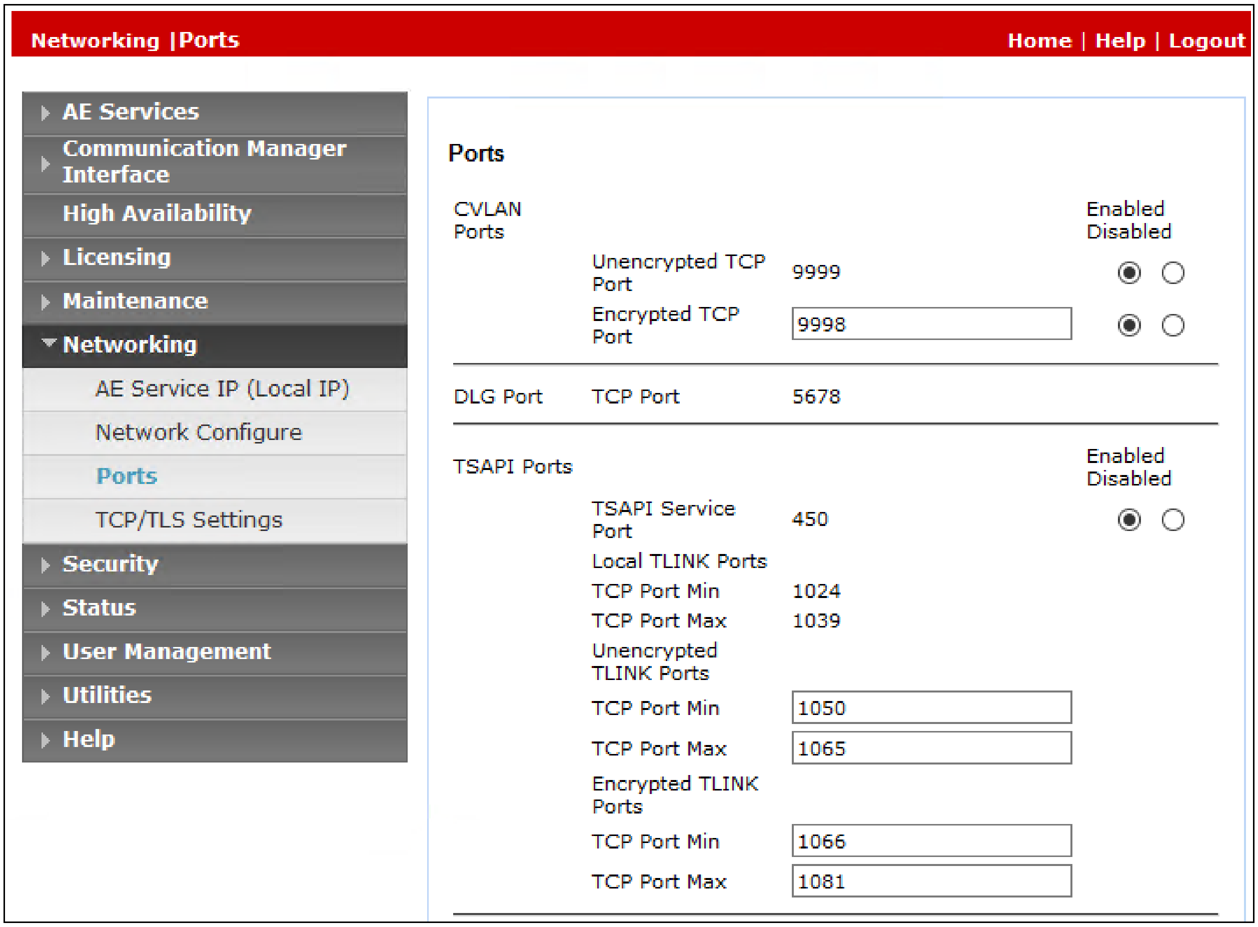
Select **Security → Security Database → Control** from the left pane, to display the **SDB Control for DMCC, TSAPI, JTAPI and Telephony Web Services** screen in the right pane. Uncheck both fields below.



6.9. Enable Ports

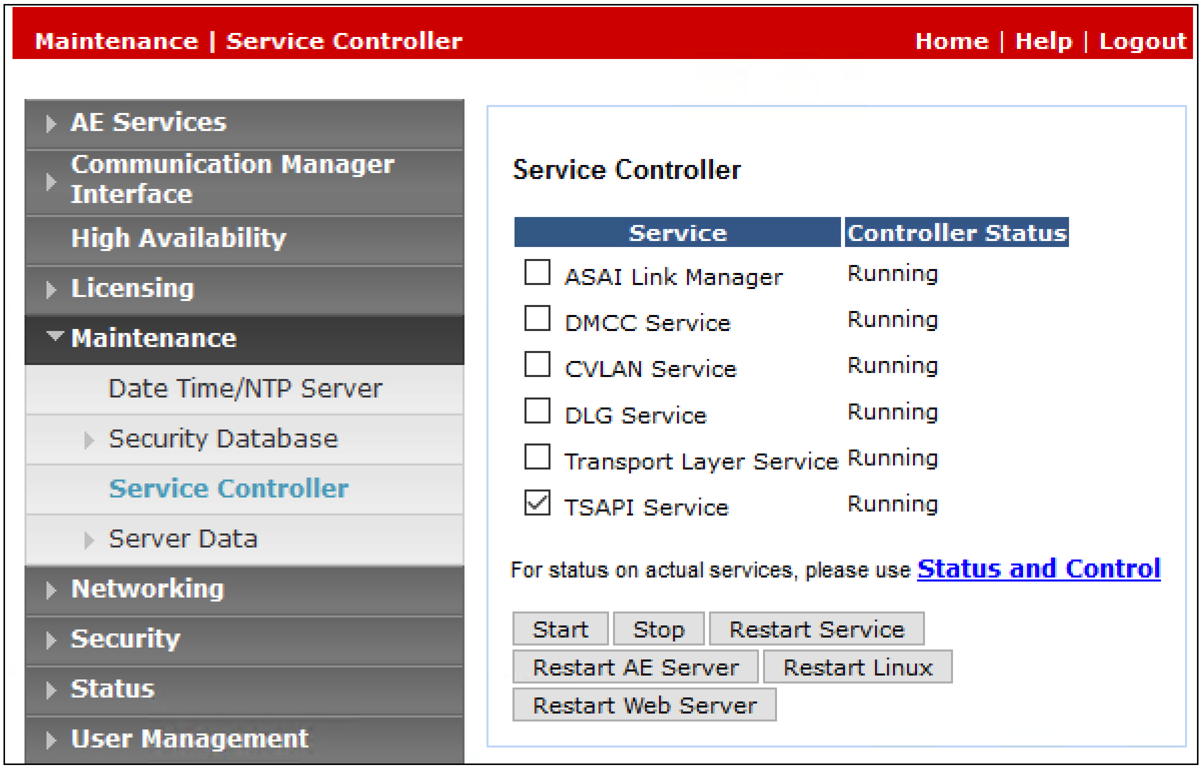
Select **Networking → Ports** from the left pane, to display the **Ports** screen in the right pane.

In the **TSAPI Ports** section, select the radio button for **TSAPI Service Port** under the **Enabled** column, as shown below. Retain the default values in the remaining fields.



6.10. Restart TSAPI Service

After the Application Enablement Services configuration is completed the TSAPI service needs to be restarted. To restart, navigate to **Maintenance → Service Controller**. Check the **TSAPI Service** check box and click on the **Restart Service** button.



When the Restart page opens click on the **Restart button** (not shown).

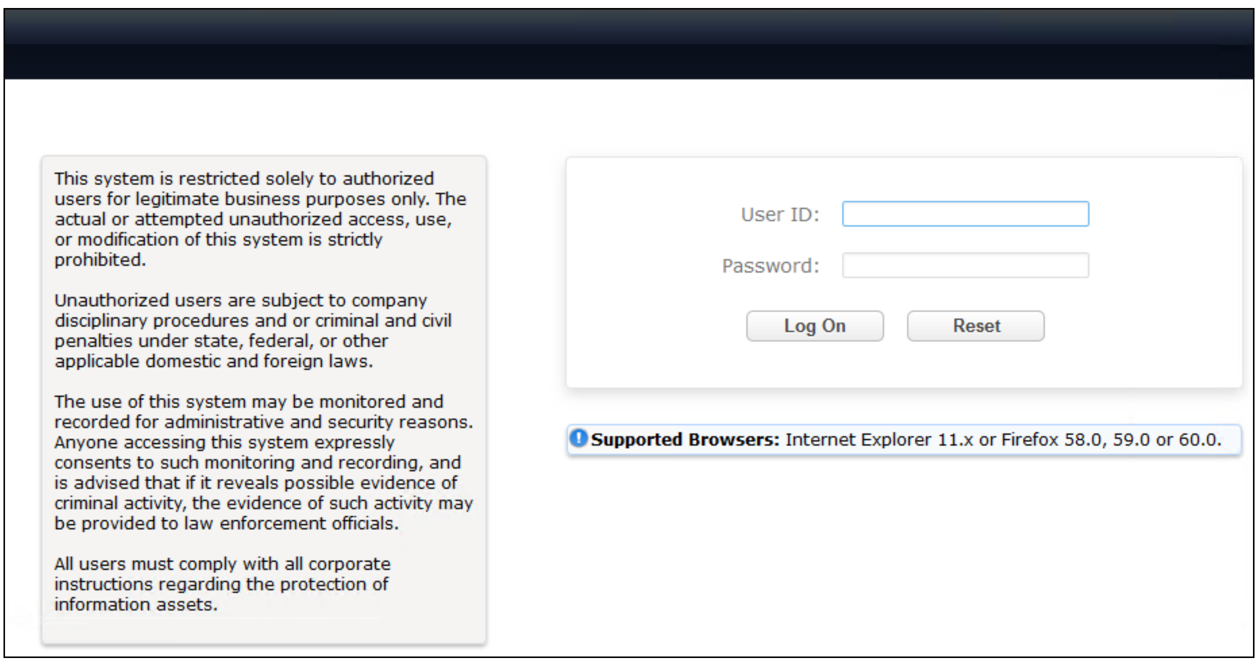
7. Configure Avaya Aura® Session Manager

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

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

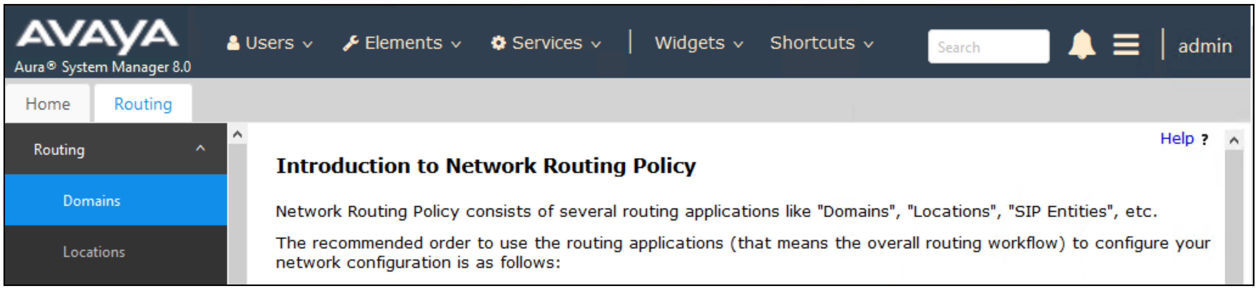
7.1. Launch System Manager

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.

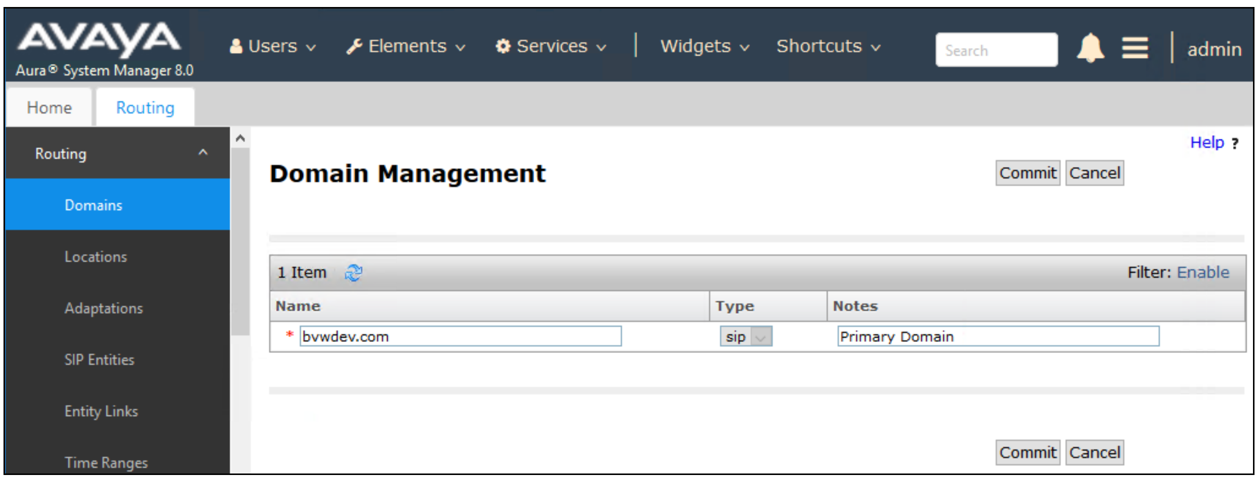


7.2. Administer Domain

In the subsequent screen (not shown), select **Elements → Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing → Domains** from the left pane, and click **New** in the subsequent screen (not shown) to add a new domain



The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select “sip” from the **Type** drop down menu and provide any optional **Notes**.

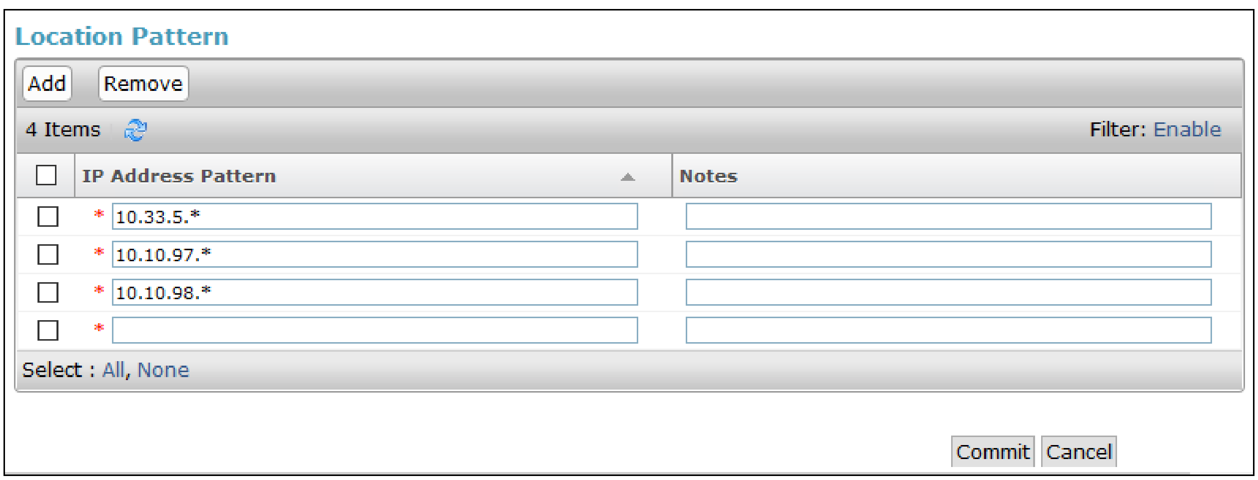


7.3. Administer Locations

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

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 all devices involved in the compliance testing in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.



7.4. Administer Adaptation

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 Enterprise 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).

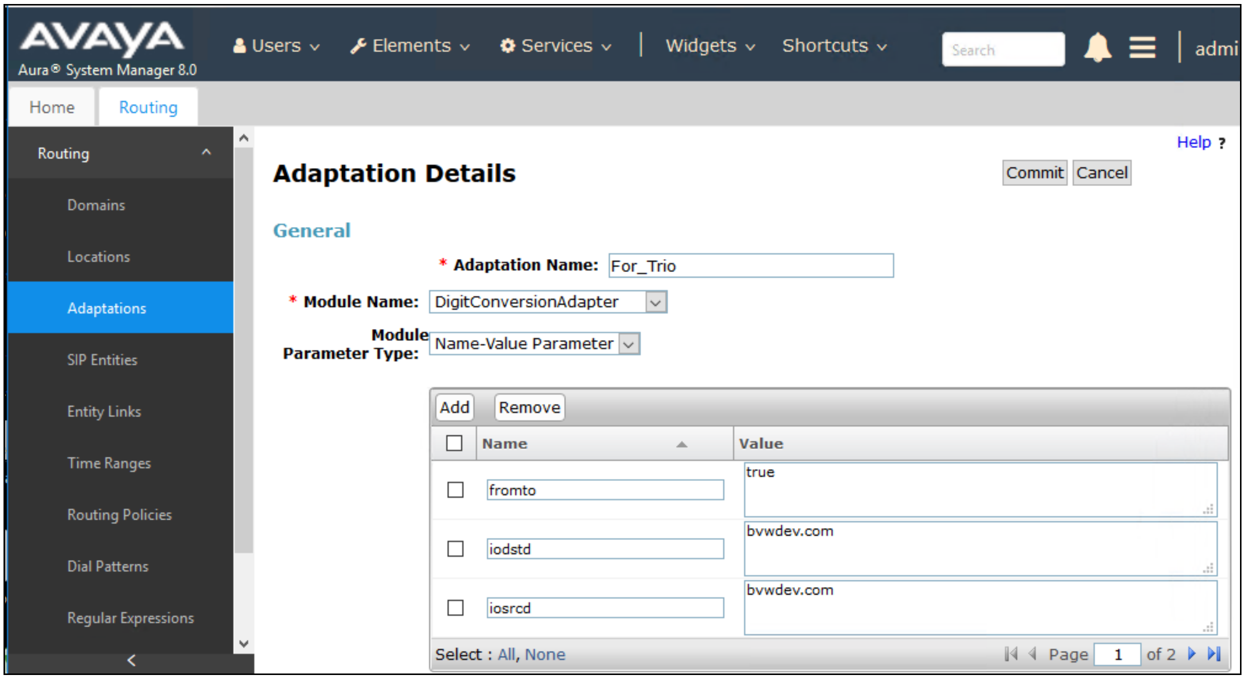
Enter the following for the Trio Enterprise Adaptation.

- **Adaptation Name:** An informative name (e.g., change IP to Domain of Trio Enterprise).
- **Module Name:** Select "DigitConversionAdapter".
- **Module Parameter Type:** Select "Name-Value Parameter".

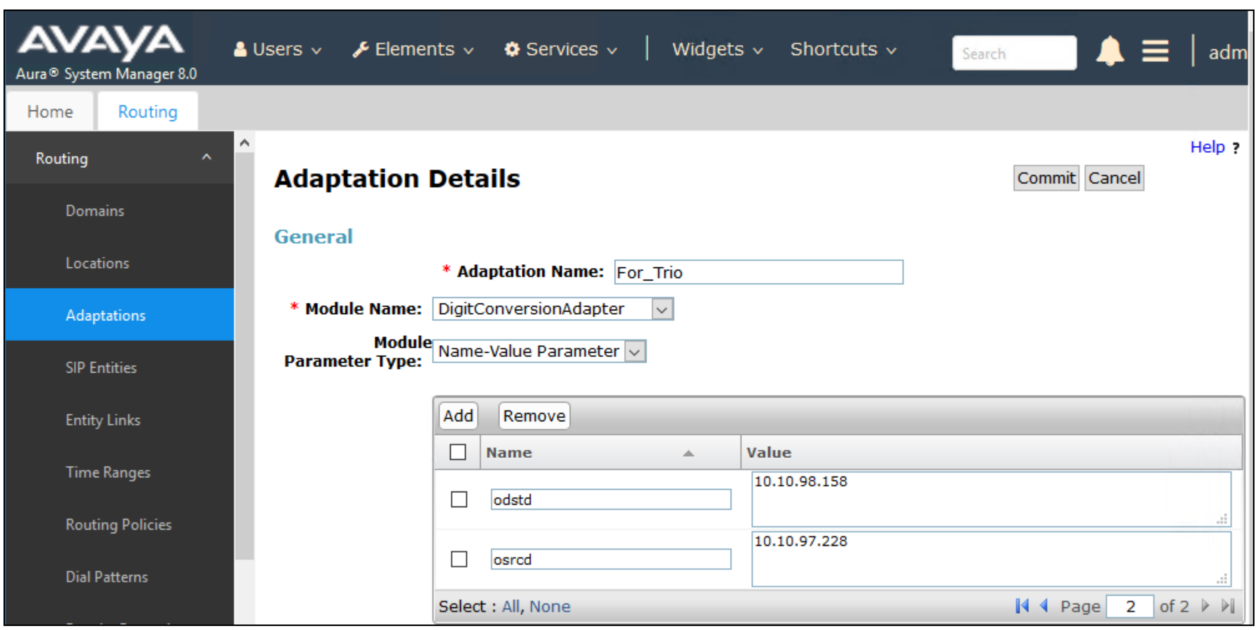
Click **Add** to add a new row for the following values as shown below table:

Name	**Value**
fromto	true
iodstd	Enter the domain name of system, e.g.: **bvwdev.com**
iosrcd	Enter the domain name of system, e.g.: **bvwdev.com**
odstd	Enter IP address of Trio Enterprise SIP Server, e.g.: **10.10.98.158**
osrcd	Enter IP address of Session Manager Server, e.g.: **10.10.97.228**

Once the correct information is entered click the **Commit** button. Below is the screenshot showing the Adaptation created for Trio Enterprise.



The screenshot showing the continuation of the Adaptation values configured for Trio Enterprise.



7.5. Administer SIP Entities

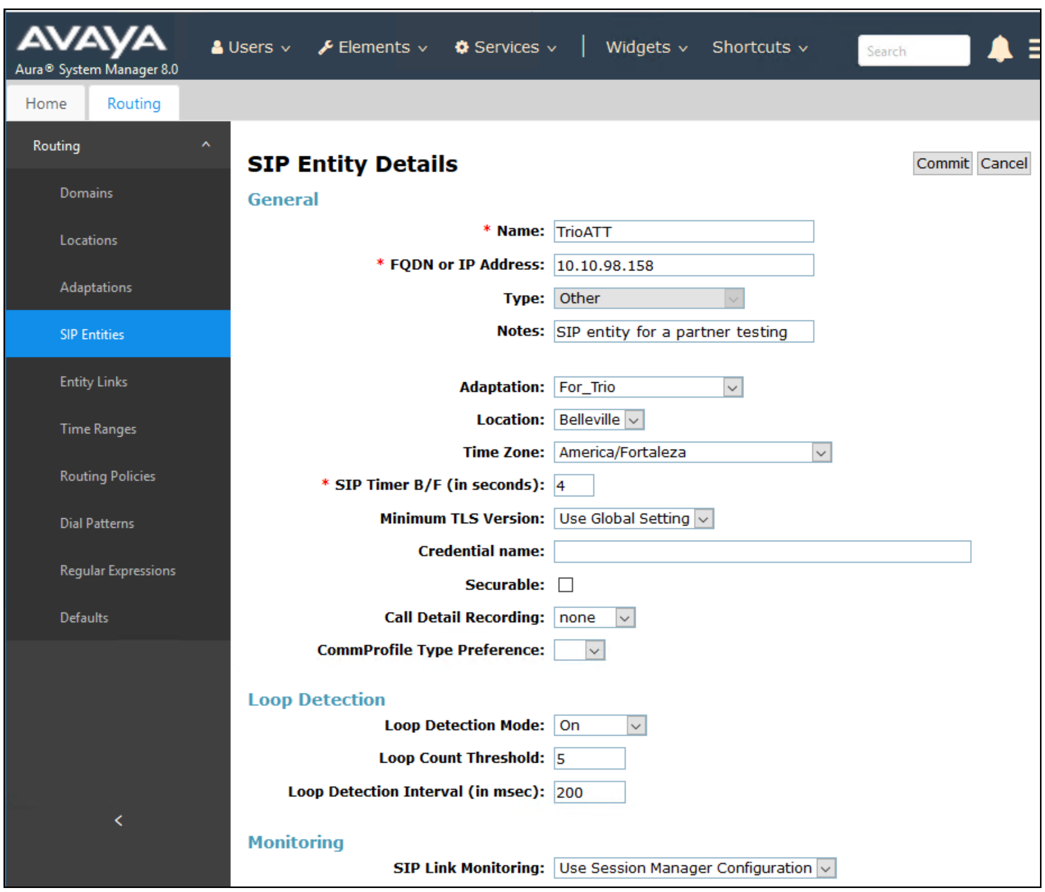
Add two new SIP entities, one for Trio Enterprise and one for the new SIP trunks with Communication Manager.

7.5.1. SIP Entity for Trio Enterprise

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 Enterprise.

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

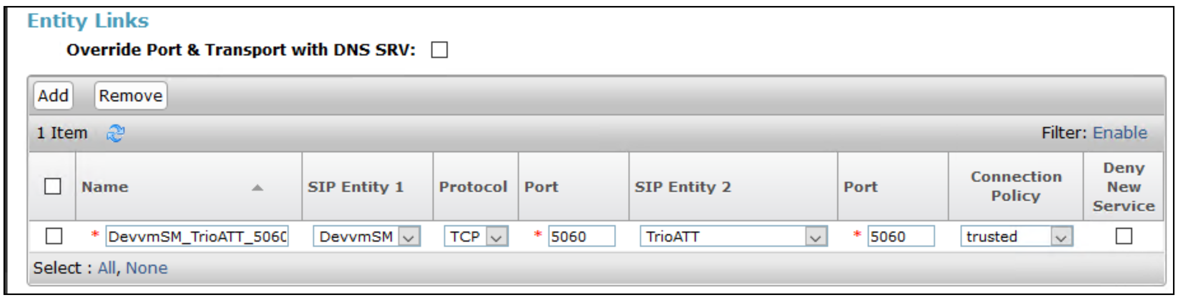
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of Trio Enterprise SIP Server.
- **Type:** "Other"
- **Notes:** Any desired notes.
- **Adaptation:** Select the adaptation configured in **Section 7.4**
- **Location:** Select the Trio Enterprise location name from **Section 7.3**.
- **Time Zone:** Select the applicable time zone.
- **SIP Link Monitoring:** Select "Use Session Manager Configuration".



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

Note that only TCP protocol was tested.

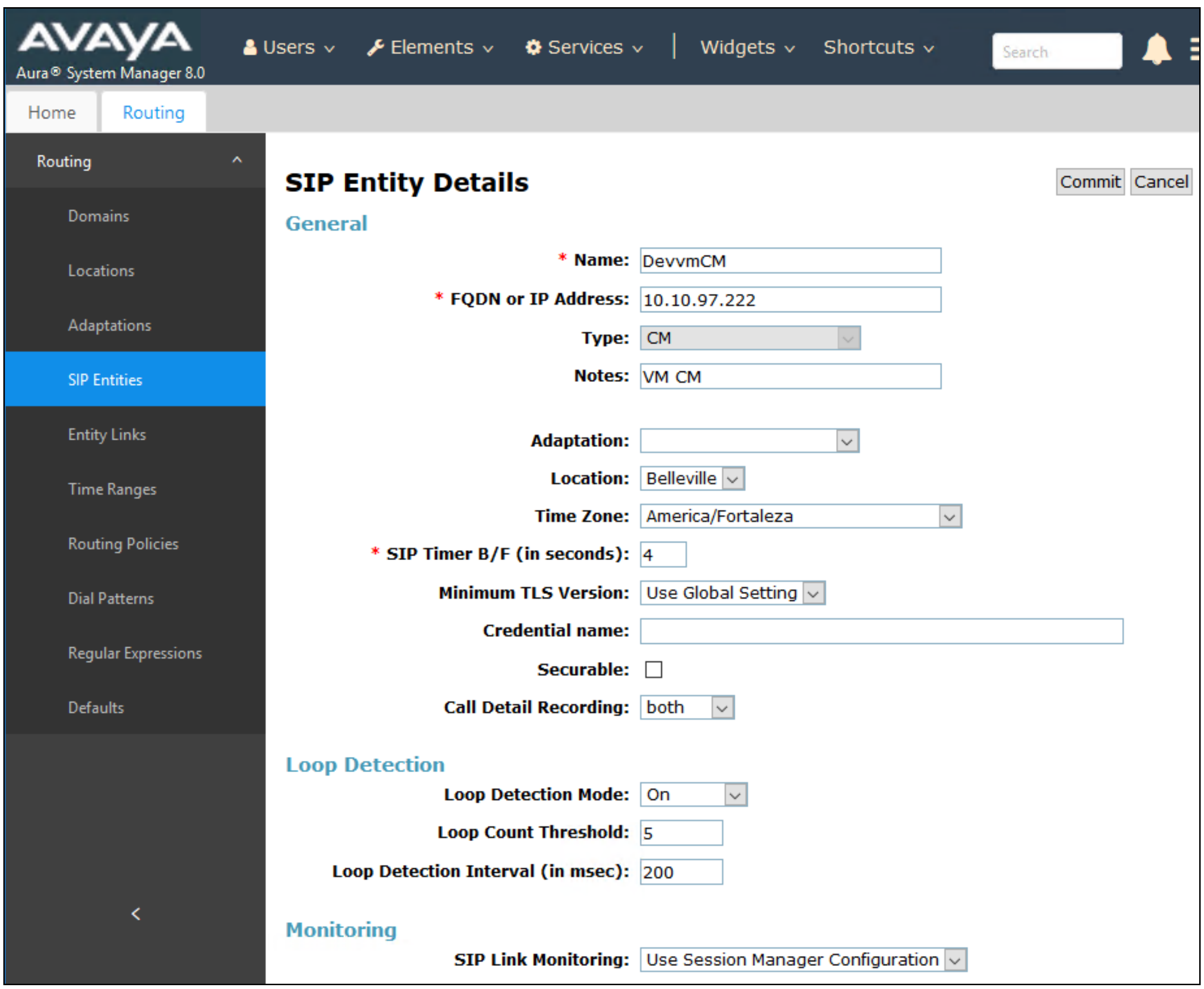


7.5.2. SIP Entity for Communication Manager

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 Enterprise.

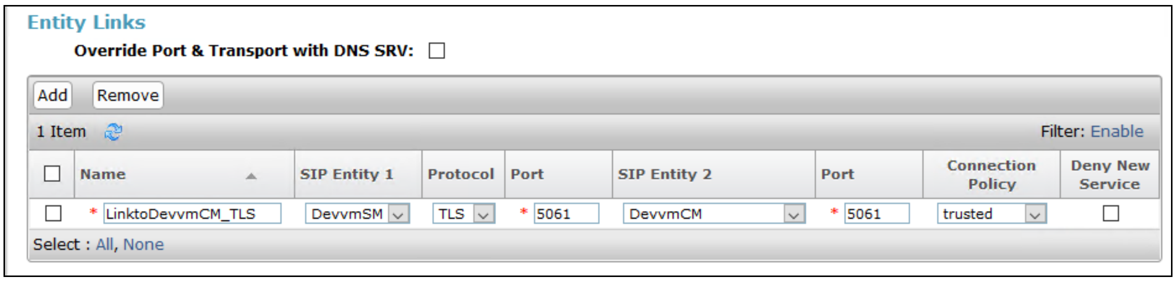
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields.

- **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 "DevvmSM".
- **Protocol:** The signalling group transport (TLS) method from **Section 5.5**.
- **Port:** The signalling group listen port (5061) number from **Section 5.5**.
- **SIP Entity 2:** The Communication Manager entity name from this section.
- **Port:** The signalling group listen port (5061) number from **Section 5.5**.
- **Connection Policy:** "trusted"



7.6. Administer Routing Policies

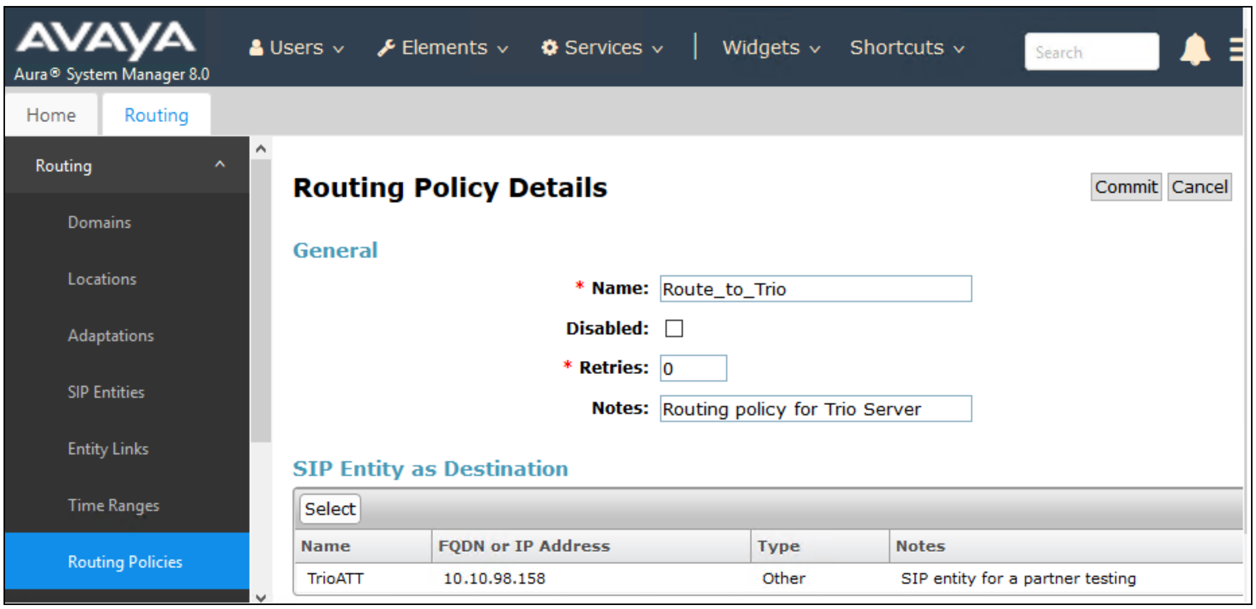
Add two new routing policies, one for Trio Enterprise and one for the new SIP trunks with Communication Manager.

7.6.1. Routing Policy for Trio Enterprise

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 Enterprise.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Trio Enterprise entity name from **Section 7.5.1**. The screen below shows the result of the selection.

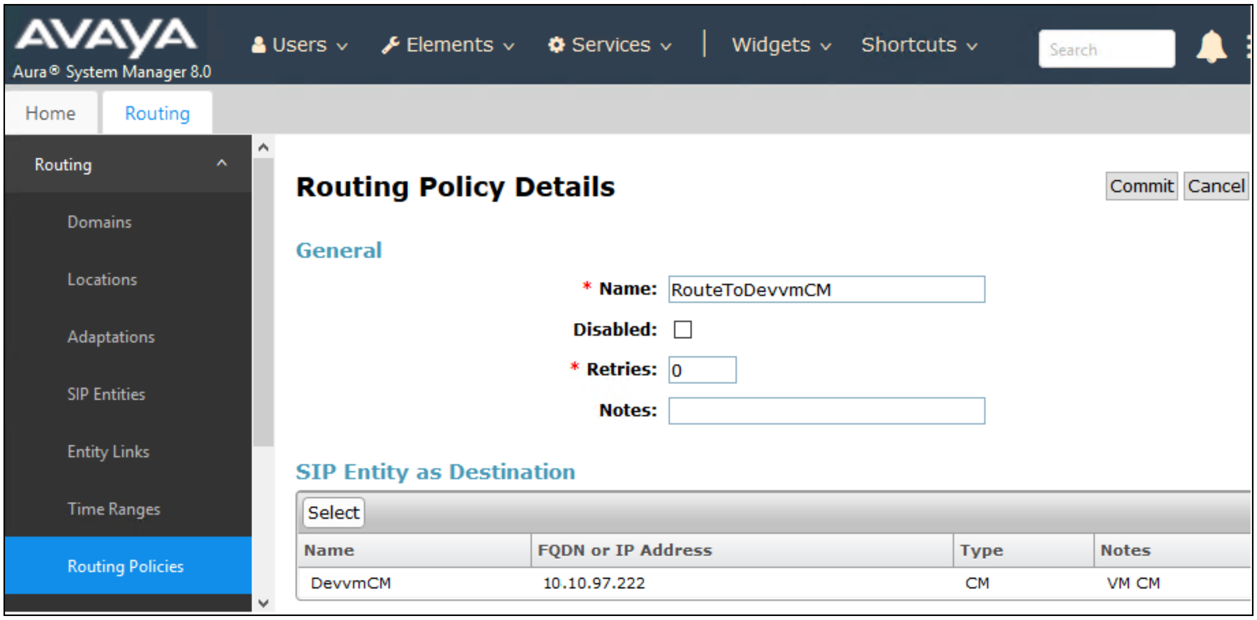


7.6.2. Routing Policy for Communication Manager

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**, 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 7.5.2**. The screen below shows the result of the selection.



7.7. Administer Dial Patterns

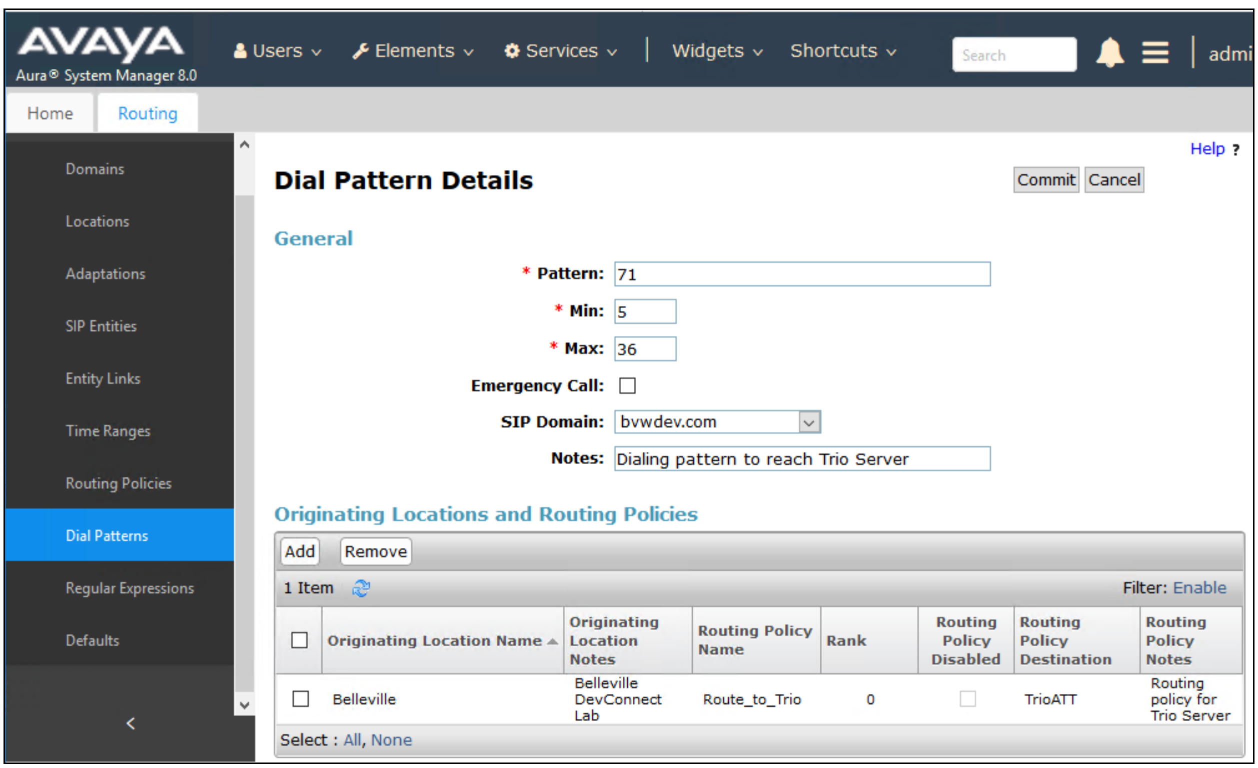
Add a new dial pattern for Trio Enterprise and Communication Manager.

7.7.1. Dial Pattern for Trio Enterprise

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 Trio Enterprise. 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 "71".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signalling group domain name from **Section 7.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Trio Enterprise. In the compliance testing, the entry allowed for call originations from all Communication Manager endpoints in locations "Belleville". The Trio Enterprise routing policy from **Section 7.6.1** was selected as shown below.

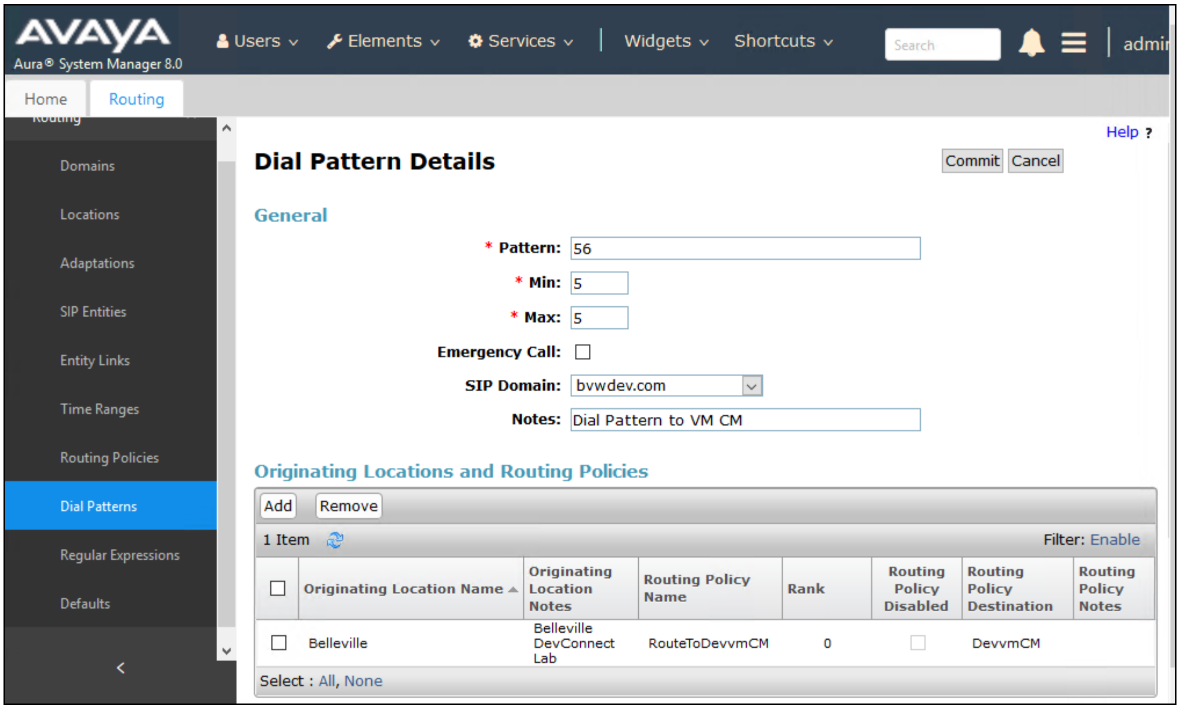


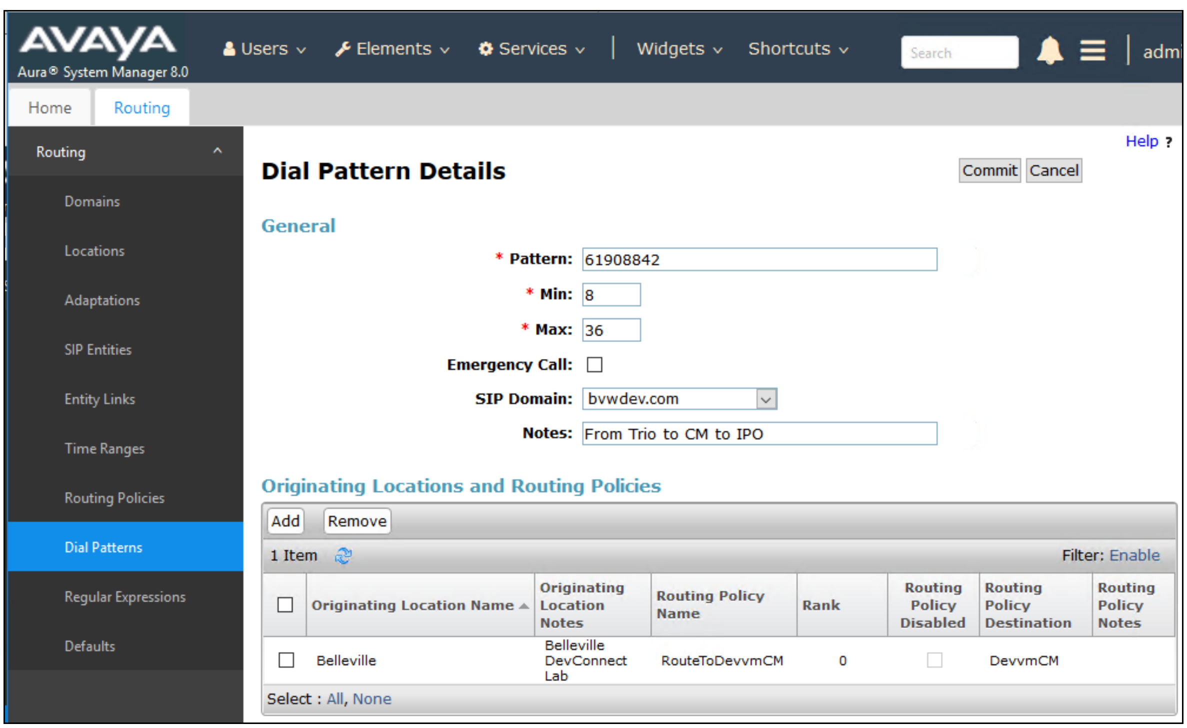
7.7.2. Dial Pattern for Communication Manager

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 "56" and "6190842".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The signalling group domain name from **Section 7.2**.

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 all Trio Enterprise endpoints in locations "Belleville". The Communication Manager routing policy from **Section 7.6.2** was selected as shown below.





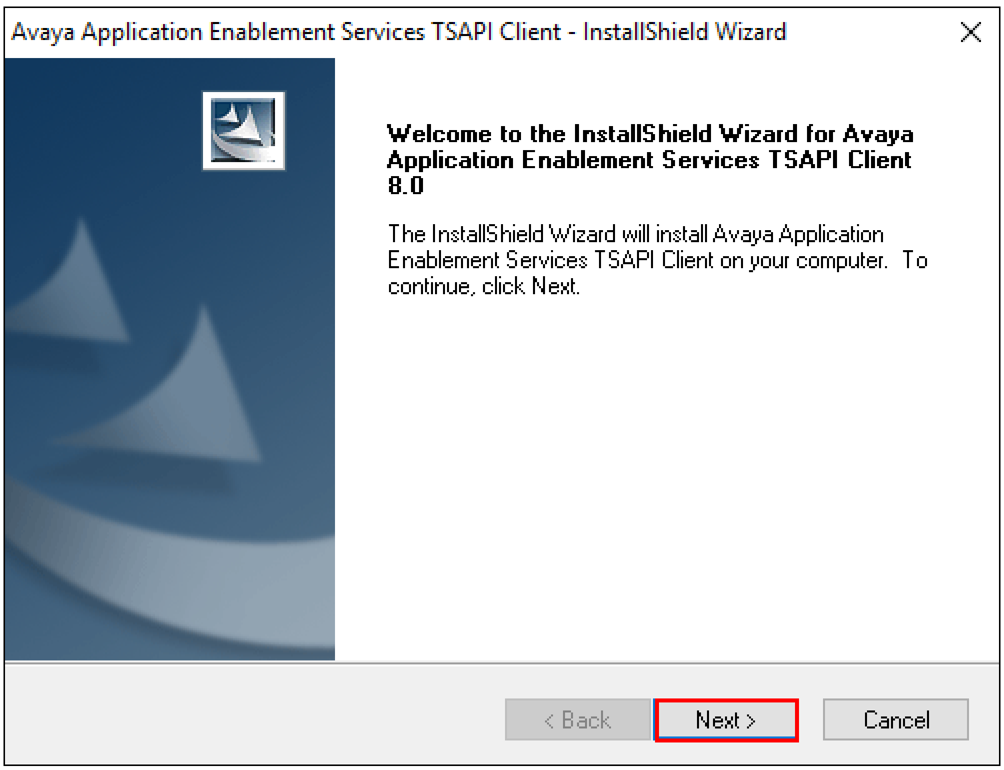
8. Configure Trio Enterprise

This section shows how to configure Trio Enterprise to successfully connect to Communication/Application Enablement Services. The installation of the Trio Enterprise software is assumed to be completed and the Trio Enterprise services are up and. The steps to configure SIP Trunks are as follows:

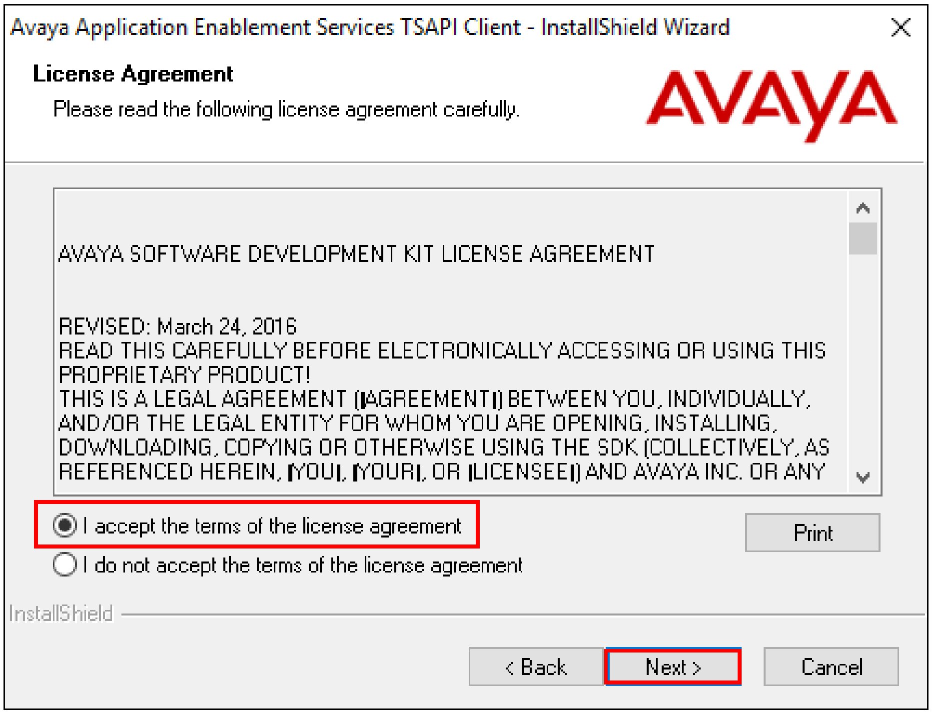
- Install Avaya Application Enablement Services TSAPI Client
- Configure Trio Enterprise to use SIP Trunks
- Configure Absence
- Configure Trio Enterprise Attendant

8.1. Install Avaya Application Enablement Services TSAPI Client

An InstallShield Wizard is used to install the Avaya Application Enablement Services TSAPI Client. Locate the InstallShield Wizard and once opened click on **Next**.



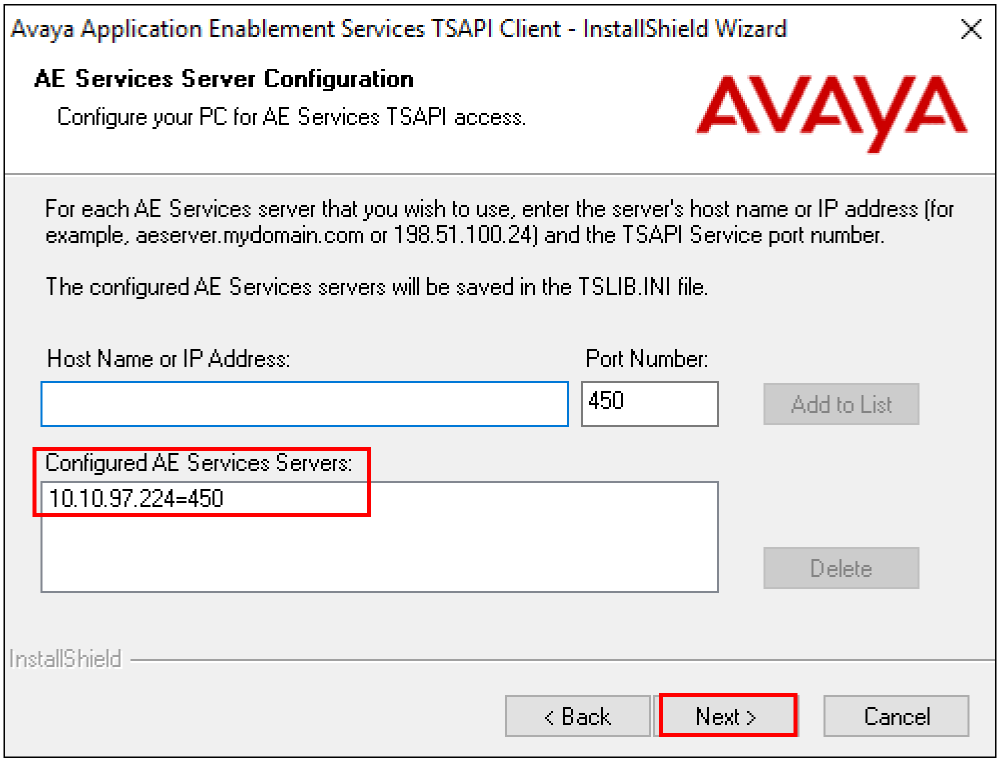
Accept the license agreement as shown below and click on **Next**.



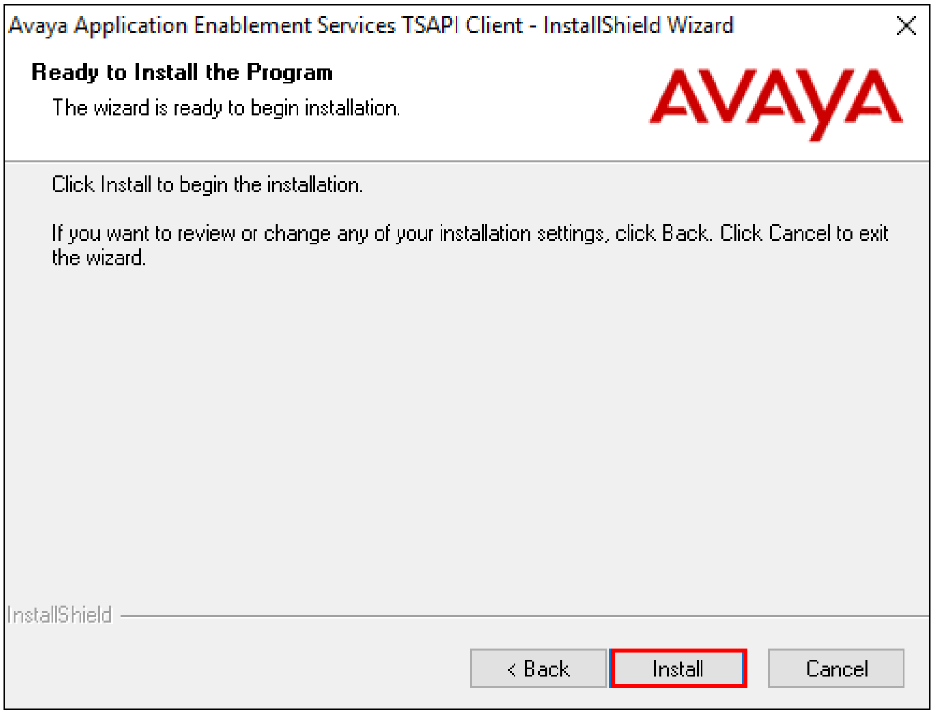
In the subsequent window, enter the following and select **Add to List**:

- **Host Name or IP Address:** Enter the IP address of the Application Enablement Services
- **Port Number:** Enter **450**

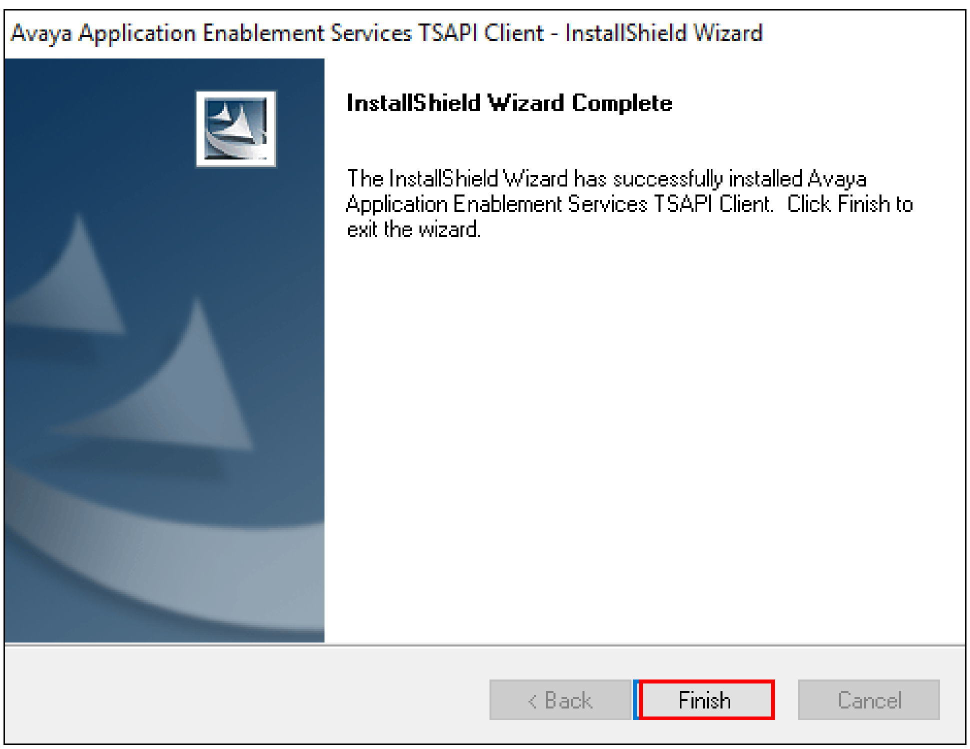
Click on the **Next** button to continue.



In the subsequent window shown below, click on the **Install** button.



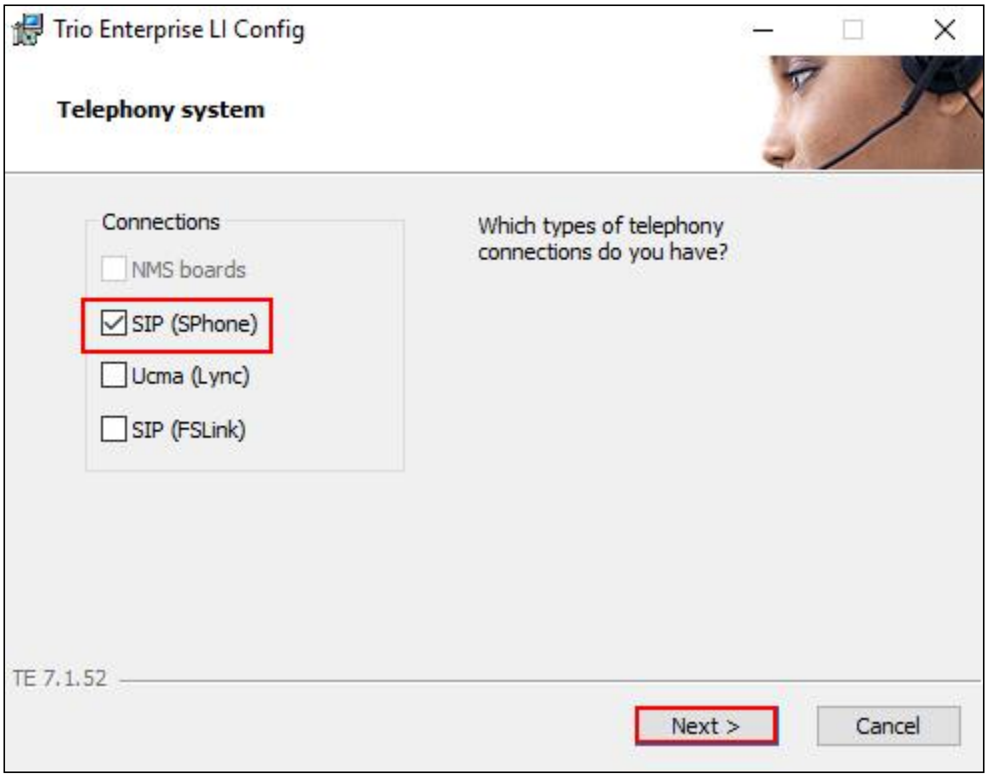
When the **InstallShield Wizard Complete** window appears click on the **Finish** button.



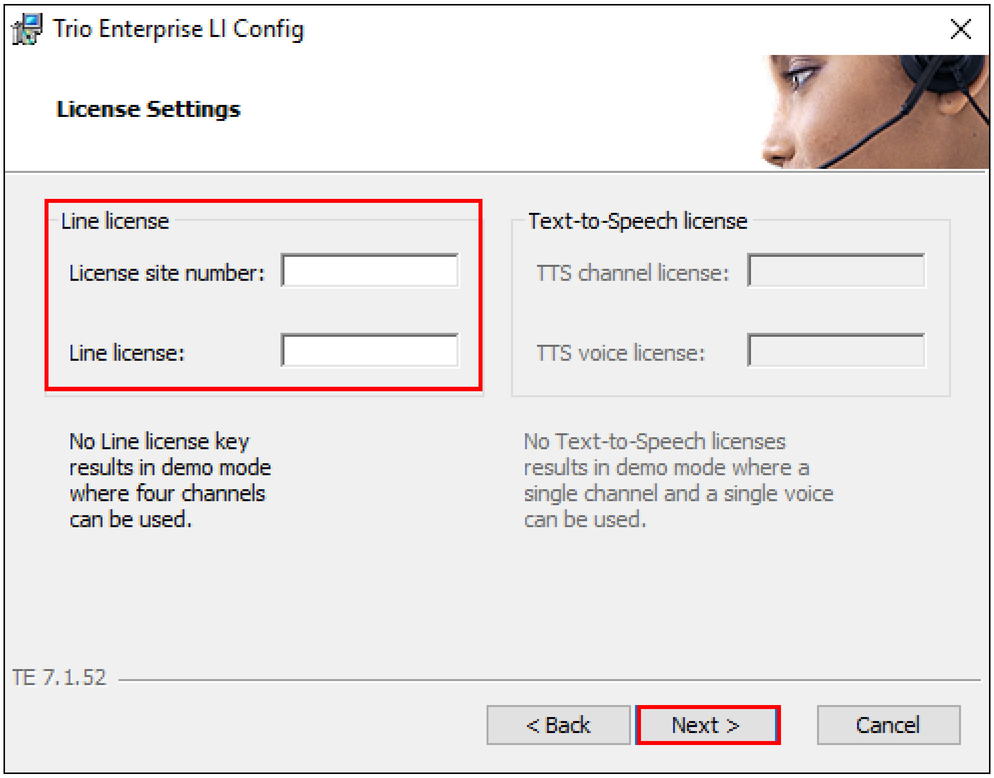
8.2. Configure Trio Enterprise to use SIP Trunks

Access Windows services. Select **Start → Run**, then type **services.msc** into the command line and press return (not shown). When the services window opens, locate the **Trio Televoice service**, right click and select **stop** to stop the service (not shown).

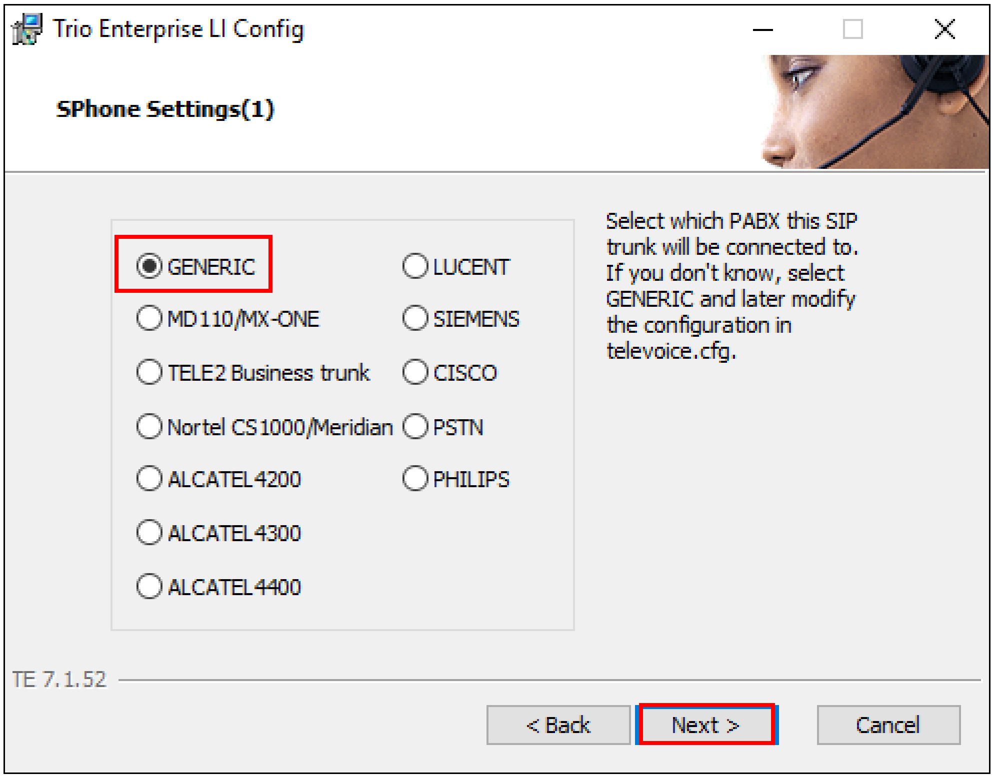
Launch the Trio configuration application. Select **Start → Programs → Trio Enterprise → TeleVoice Config** (not shown). The configuration of the application starts, and when the new window opens, check the **SIP** check box followed by the **Next** button.



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.



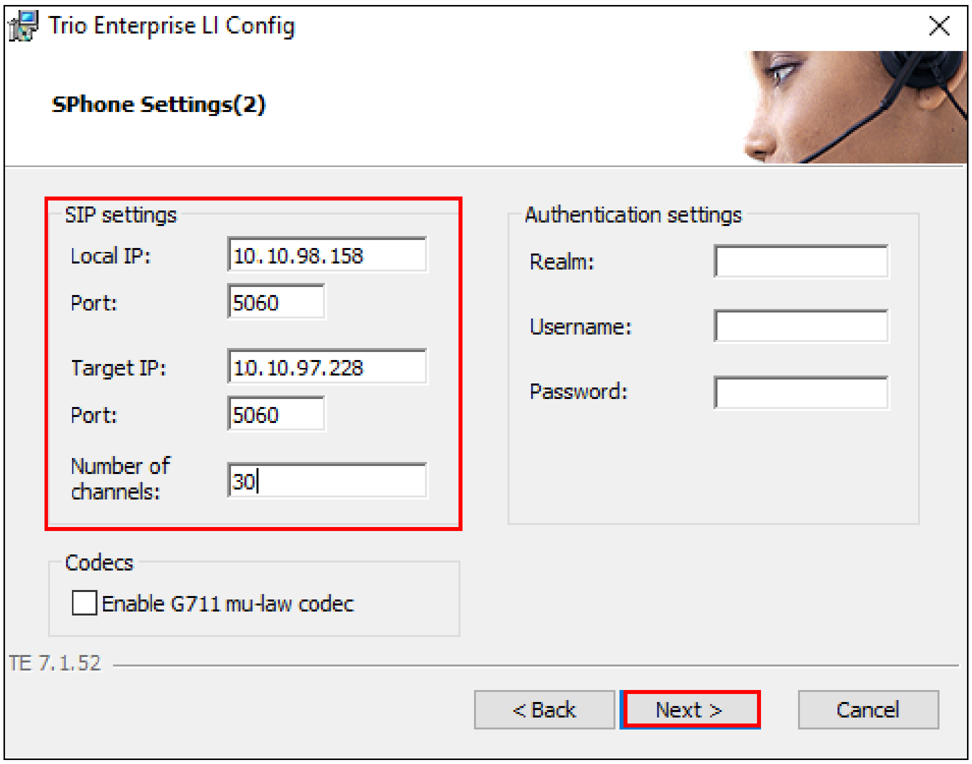
In the subsequent window, select on the **GENERIC** radio button followed by the **Next** button to continue.



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 Communication Manager (procr IP address)
- **Port:** Enter the SIP Port "5060"
- **Number of channels:** Enter "30" as the number of channels

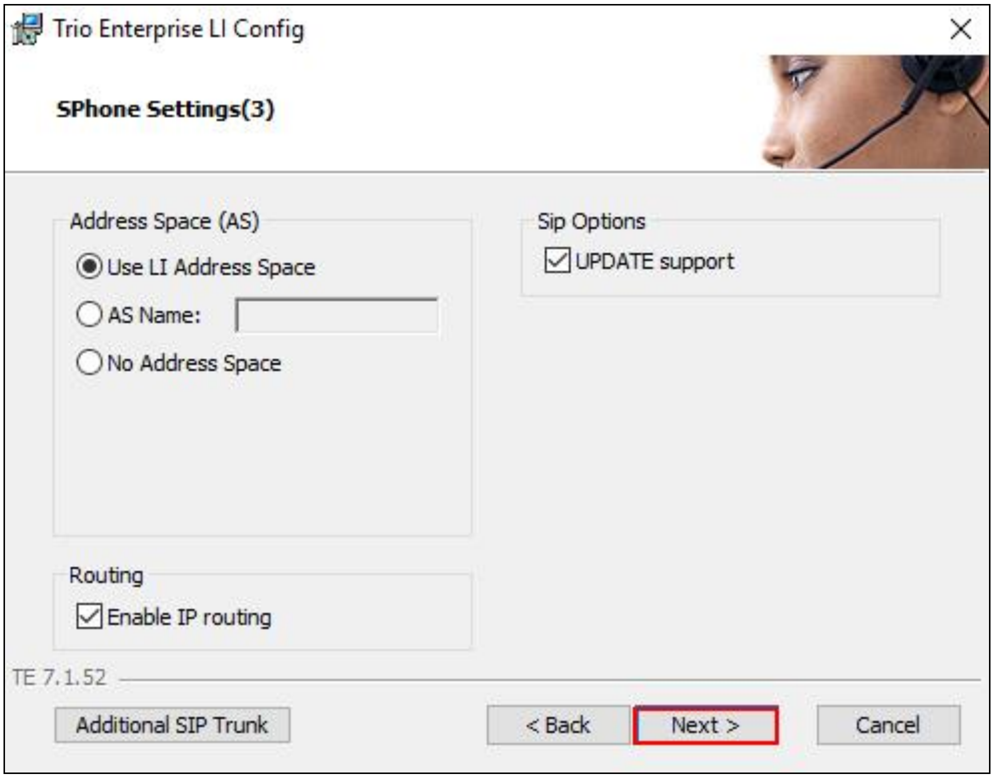
Click on the **Next** button to continue.



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

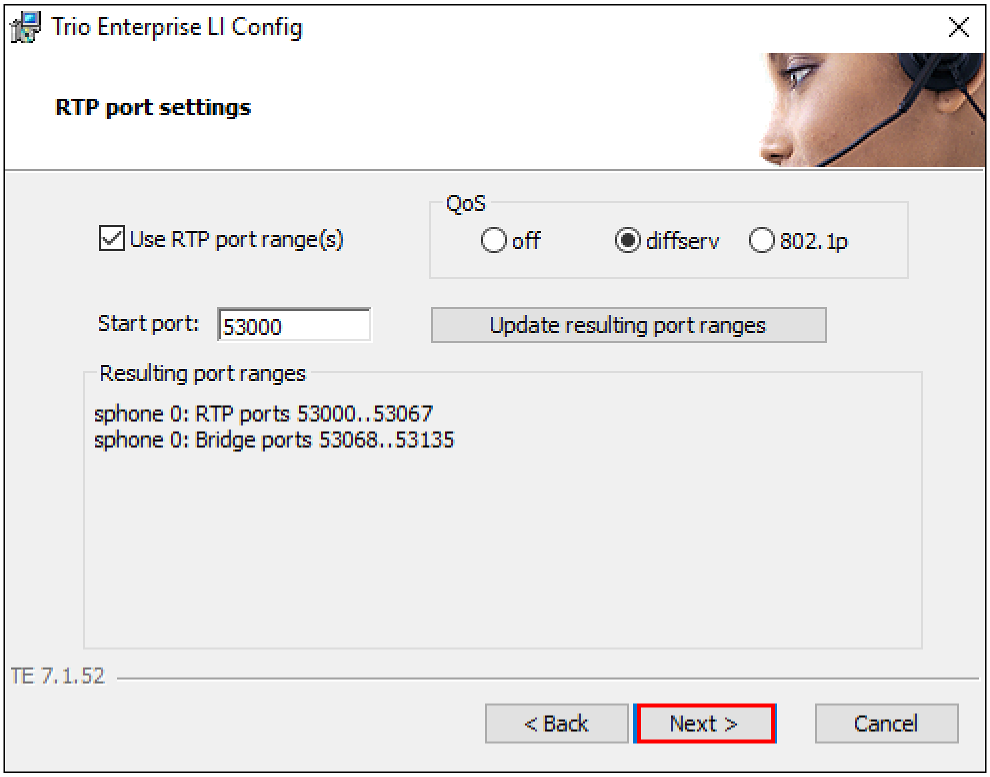
Click on the **Next** button to continue.



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"

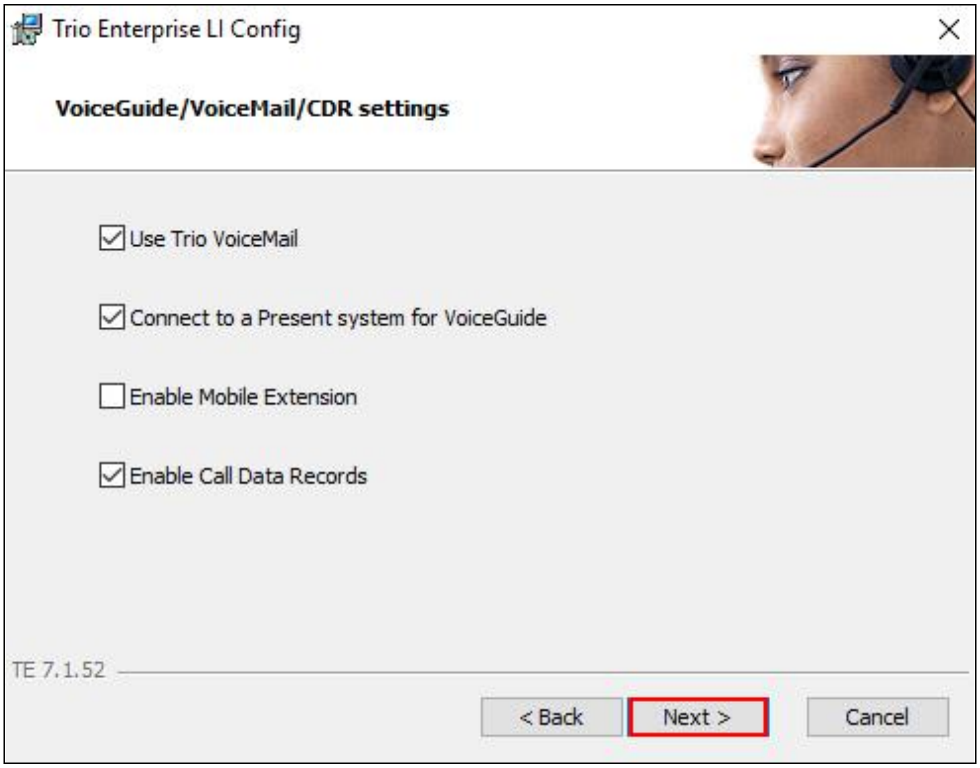
Click on the **Next** button to continue.



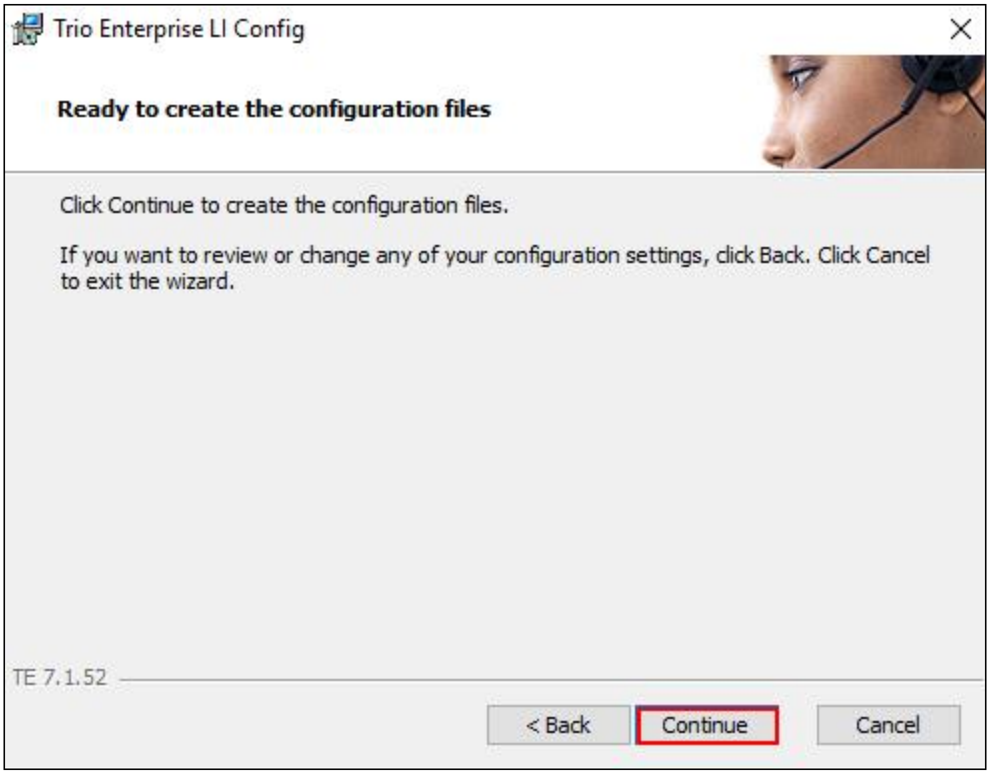
In the subsequent window enter the following settings:

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

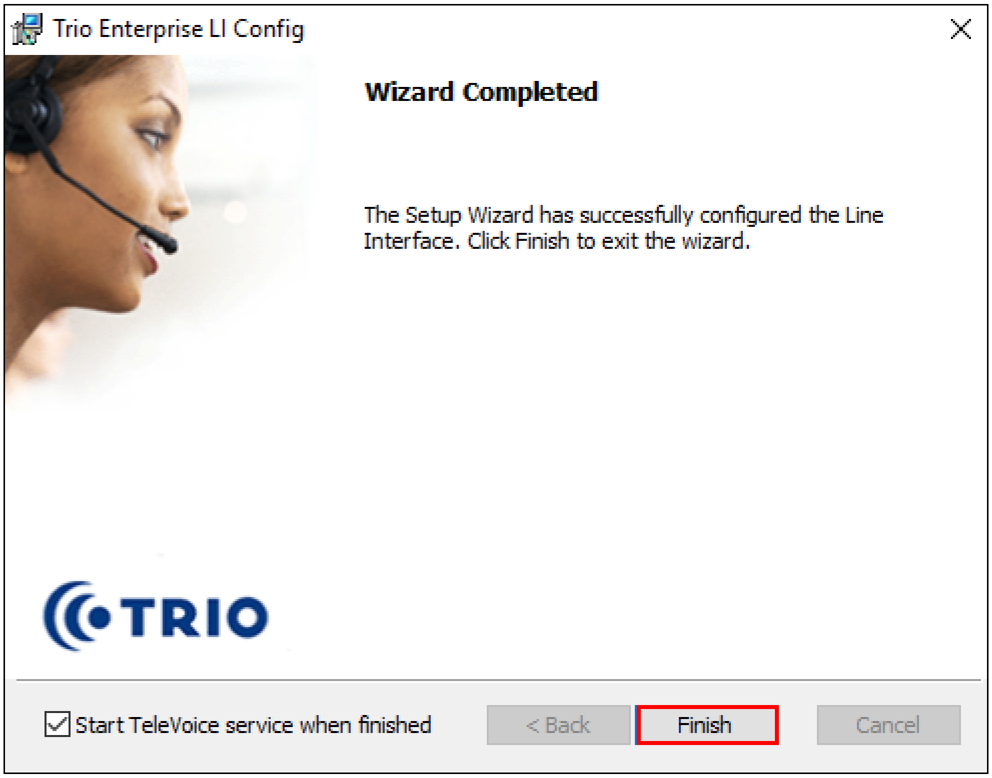
Retain default values for other fields and click on the **Next** button to continue.



In the subsequent window shown below, click on **Continue** button.



On the **Wizard Completed** page check the **Start TeleVoice service when finished** check box, followed by the **Finish** button.



8.3. Special Configuration for Avaya Aura® Session Manager

Access the template for televoice.cfg. This is typically found in \TE\ProgramData\LI\templates folder.

Find the [sip_x] section and add the row “usetcp =1” as shown below,

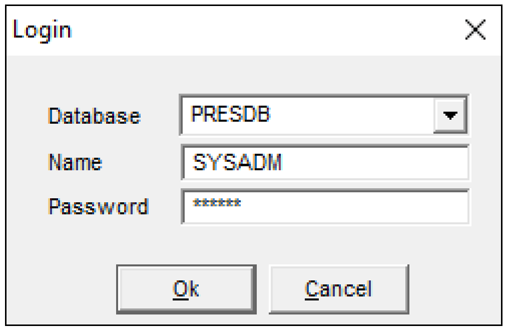
```
[sip_1]
        signallingprotocol=sip
        localHost=10.10.98.158
        targetHost=10.10.97.228
        uriScheme=1
        transferPoint=afterAnswer
        update=1
        mwiMethod=unsolicited
        rel100=false
        allowTransferMedia=false
        usetcp=1
```

Find the [device_0] section and set the “autype” records as shown in the example below, this will prioritize G.711MU-Law or G.711A-Law as required.

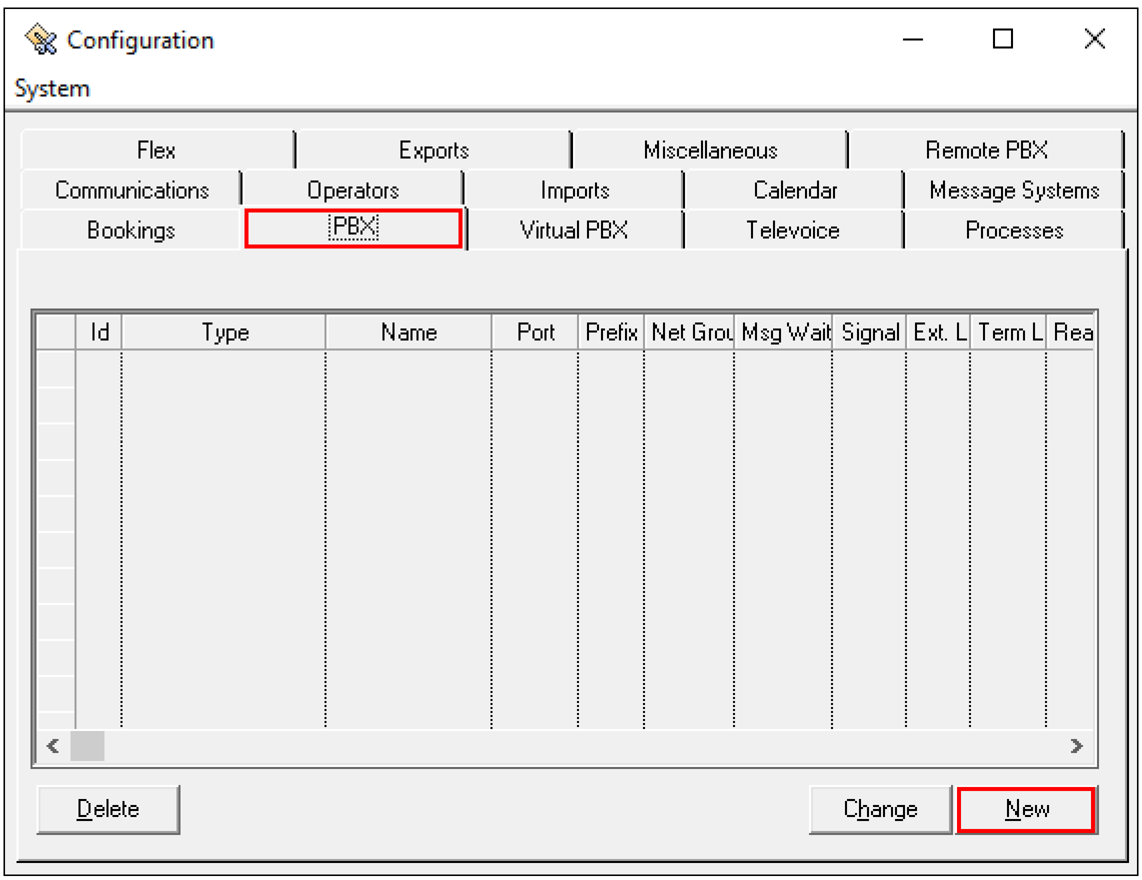
```
[device_0]
        type=sphone
        access=127.0.0.1:33109
        voiceserver_1=localhost:33813
        sphone=0
        localip=10 .10.98.158
        mf=SipGw_QSIG=0x3ff
        rtpsendlog=f=127.0.0.1:33109
        autype_1=sdp=pcma
        autype_2=sdp=telephone-event,payload=101
        autype_3=sdp=pcmu
        rtpportrange=53000..53067,dscp
        rtpbridgeportrange=53068..53135,dscp
```

8.4. Configure Absence connection

To configure the Absence connect; navigate to **Start → Programs → Trio Enterprise → Trio Present Setup** (not shown). Use the correct credentials to login as shown below.



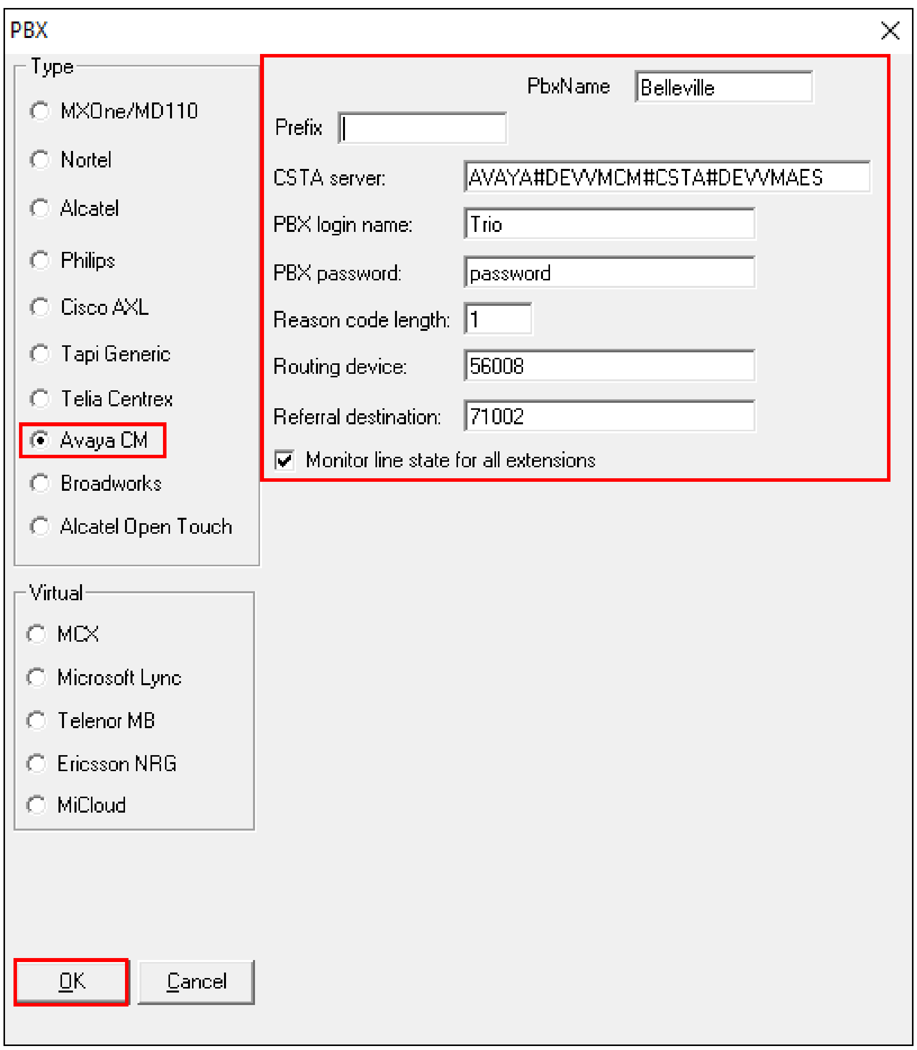
From the screen shown below, select **PBX** and then click on **New**.



Configure the **PBX** window as shown below.

- **Type:** Click on the "Avaya CM" radio button.
- **PbxName:** Enter an informative name.
- **CSTA server:** Enter the appropriate Tlink name as seen in **Section 6.7**.
- **PBX login name:** Enter the CTI Username as configured in **Section 6.5**.
- **PBX password:** Enter the CTI password as configured in **Section 6.5**.
- **Reason code length:** Enter "1"
- **Routing device:** Enter the extension assigned to the diversion VDN used for activating referrals from the phone set as configured in **Section 5.16.3**.
- **Referral destination:** Enter the number "71002" that the extensions should be forwarded to when a referral is activated. This number is configured on the Trio Enterprise server for absence treatment.

Click on the **OK** button.



8.5. Configure Trio Enterprise Attendant

Trio Enterprise Attendant is a separate application to Trio Enterprise server and can run concurrently on the same platform. The attendant uses a regular Communication Manager telephone to make and receive calls, which are directed to the telephone by Trio Enterprise server. The steps to configure Trio Attendant are to click on **Start → Programs → Trio Enterprise → Agent Client** (not shown).

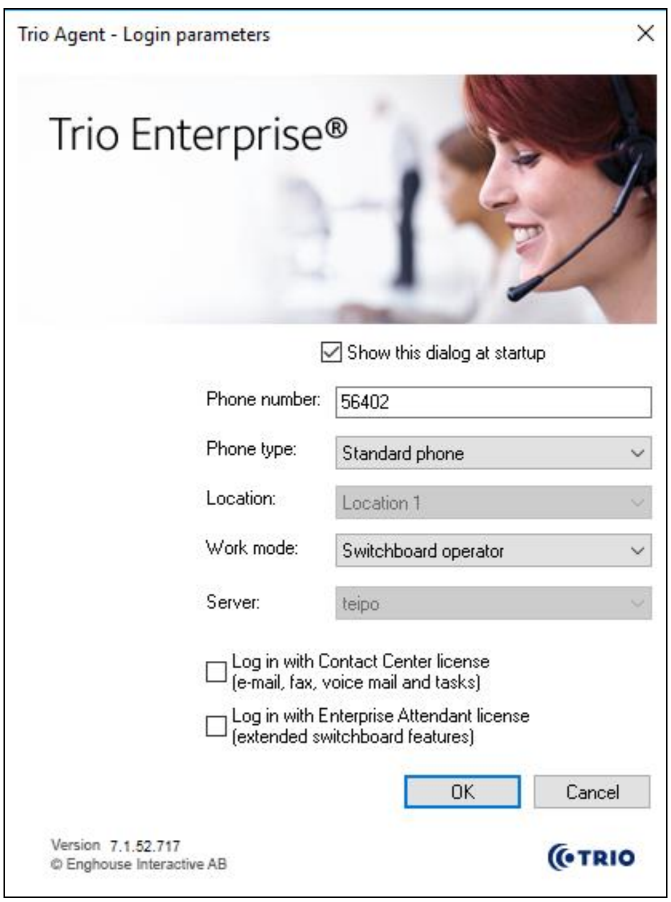
When the Trio Agent window opens enter the following:

- **User ID:** Enter a valid user ID
- **Password:** Enter a valid Password

Note this user ID and password is created during the installation of Trio Enterprise Server.

- **Extension:** Select the Communication Manager telephone number that will be used as the agent's audio device (number 56402 in this example).
- **Phone type:** Select "Standard phone" from the dropdown menu
- **Server:** Select the correct Trio Enterprise server (default is the current Trio server).

Click on the **OK** button to continue with log in.



9. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the Avaya and Trio Enterprise solution.

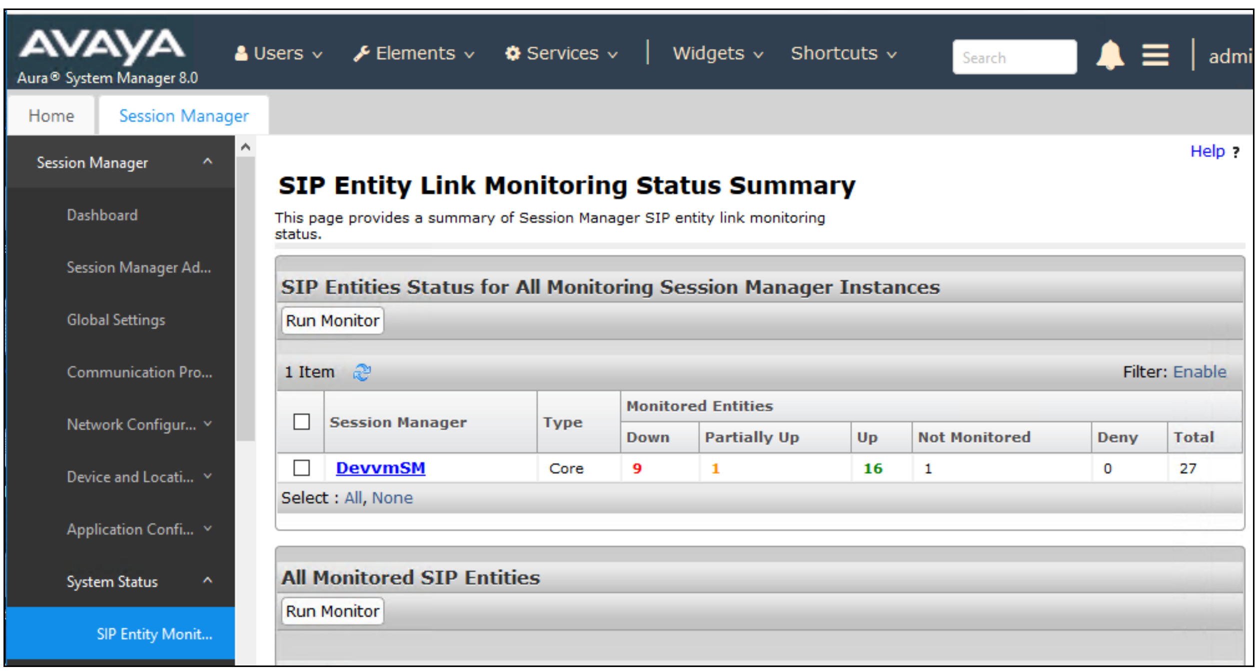
9.1. Verify Avaya Aura® Communication Manager CTI Service State

The following steps can ensure that the communication between Communication Manager and the Application Enablement Services server is functioning correctly. Using SAT, connect to Communication Manager and check the AESVCS link status with Application Enablement Services by using the command “status aesvcs cti-link”. The CTI Link is 1. Verify that the **Service State** of the CTI link is **established**.

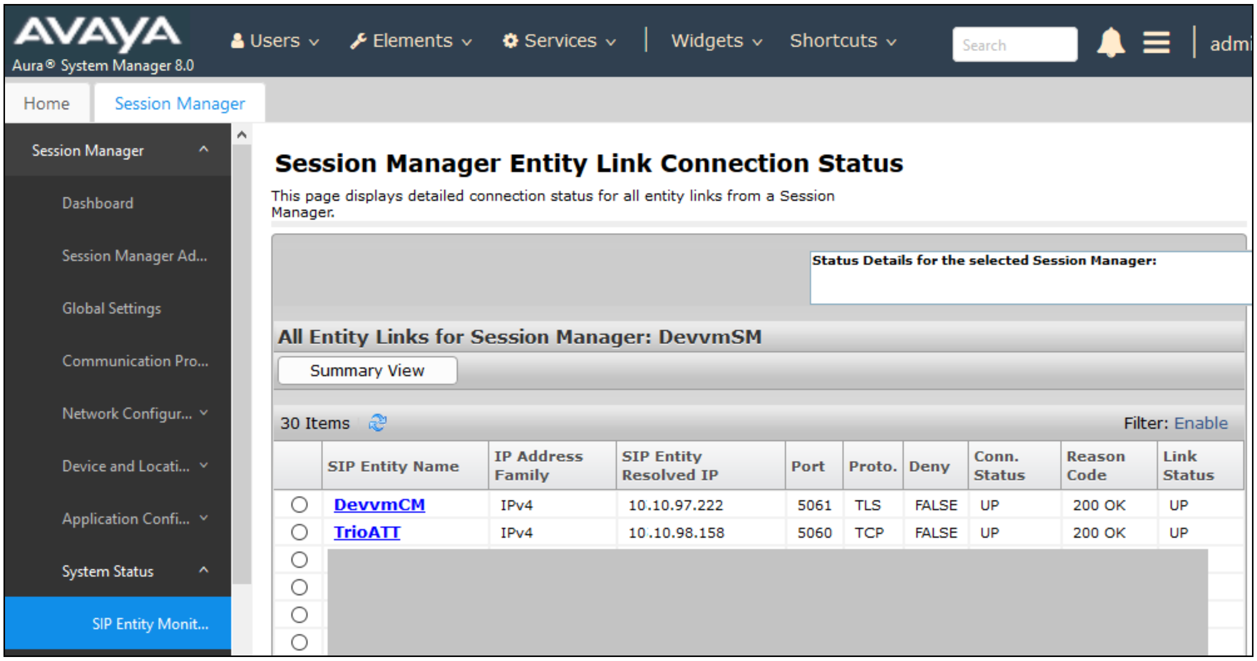
```
status aesvcs cti-link

                          AE SERVICES CTI LINK STATUS

CTI   Version  Mnt   AE Services      Service        Msgs      Msgs
Link           Busy  Server           State          Sent      Rcvd

1     7        no    devvmaes         established    15        15
```

9.2. Verify Session Manager

Log in to System Manager. Under the **Elements** section, navigate to **Session Manager → System Status → SIP Entity Monitoring**. Click the Session Manager instance (**DevvmSM** in the example below).

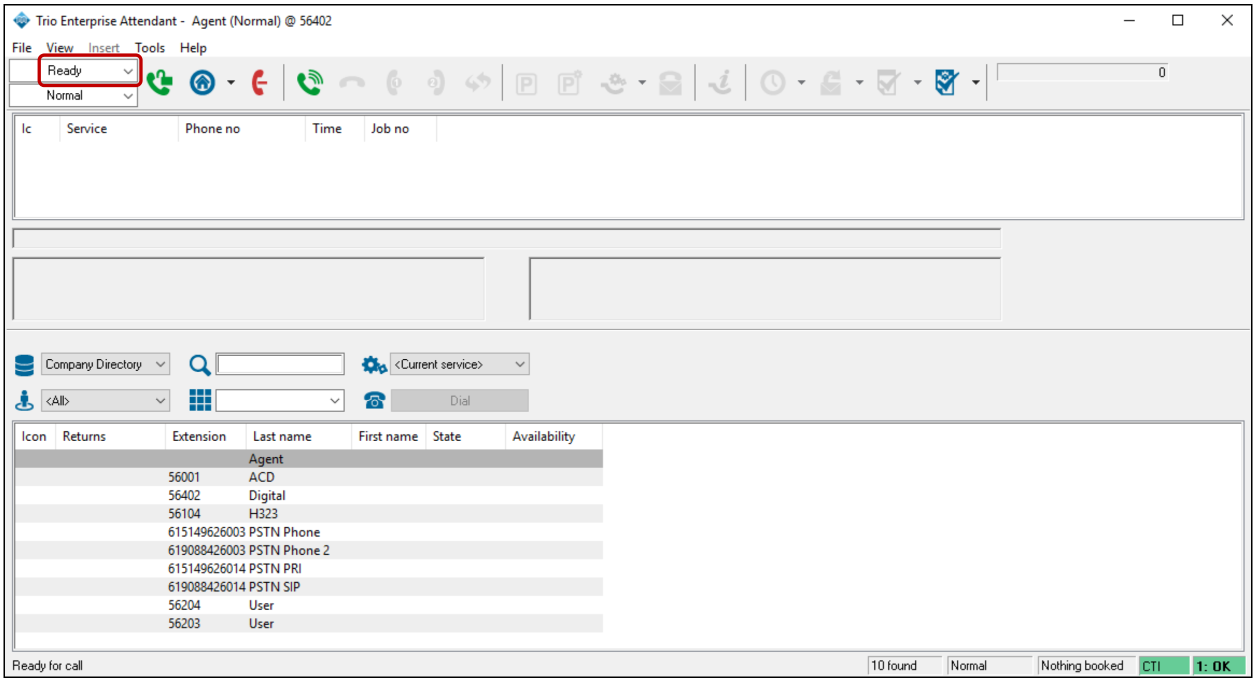


Verify that the state of the Session Manager links to Communication Manager and Trio Enterprise under the **Conn. Status** and **Link Status** columns is **UP**, as shown in the screen below.



9.3. Verify Trio Enterprise Attendant

To verify that Trio Enterprise is connected to Communication Manager via Session Manager, log in to the Trio Enterprise Attendant at **Start → Programs → Trio Enterprise → Contact Centre → Agent Client** (not shown) or launch the shortcut mentioned in **Section 8.5**. Complete log in with the appropriate credentials as shown in **Section 8.5**. The Trio Enterprise Attendant window appears as shown below. Select **Ready** from the drop-down box.



The following scenarios were also tested:

- 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
- Absence detection
- Message Waiting Indicator

10. Conclusion

These Application Notes describe the procedures required to configure Trio Enterprise from Enghouse Interactive AB to interoperate with Avaya Aura® Communication Manager, Avaya Aura® Application Enablement Services and Avaya Aura® Session Manager using SIP Trunks and TSAPI.

All feature functionality test cases described in **Section 2.1** were passed with the observations pointed in **Section 2.2**.

11. Additional References

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

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

1. *Deploying Avaya Aura® Communication Manager in Virtual Appliance*, Release 8.0, Issue 3 September 2018.
2. *Avaya Aura® Communication Manager Feature Description and Implementation*, Release 8.0, Issue 1 July 2018.
3. *Administering Avaya Aura® Communication Manager*, Release 8.0, Issue 1 July 2018.
4. *Avaya Aura® Communication Manager Screen Reference*, Release 8.0, Issue 2 August 2018.
5. *Deploying Avaya Aura® Session Manager in Virtual Appliance*, Release 8.0, Issue 2 September 2018.
6. *Administering Avaya Aura® Session Manager*, Release 8.0, Issue 2 August 2018.
7. *Deploying Avaya Aura® System Manager in Virtualized Environment*, Release 8.0, Issue 2 September 2018.
8. *Administering Avaya Aura® System Manager for Release 8.0*, Release 8.0, Issue 4 September 2018.
9. *Deploying and Updating Avaya Aura® Media Server Appliance*, Release 8.0, Issue 2 July 2018.
10. *Implementing and Administering Avaya Aura® Media Server*, Release 8.0, Issue 2 July 2018.

Product Documentation for Enghouse Interactive AB can be obtained in the installed software or at: http://enghouseinteractive.com

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.