Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Trio Enterprise from Enghouse Interactive AB with Avaya Communication Server 1000 and Avaya Aura® Session Manager using a SIP Trunk connection – Issue 1.0

Abstract

These Application Notes describe how to configure an Avaya Communication Server 1000 and an Avaya Aura® Session Manager to interface with Trio Enterprise, which is operating as an attendant answering position. Trio Enterprise is a software application from Enghouse Interactive AB installed on a Windows server that interfaces with Avaya Communication Server 1000 using a SIP connection via Avaya Aura® Session Manager and provides users with the call functions of an attendant console without having to install a hardware attendant position.

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 describe the compliance tested configuration for Avaya Communication Server 1000 and Avaya Aura® Session Manager with Trio Enterprise from Enghouse Interactive AB. Trio Enterprise is a client/server based application running on Microsoft Windows 2012 Server operating systems. Trio Enterprise provides users with an attendant answering position for Avaya Communication Server 1000E that does not require attendant telephony hardware e.g., Avaya 2250 attendant console. Trio Enterprise connects to the Avaya Communication Server 1000 using a SIP connection via Avaya Aura® Session Manager.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using an Avaya Communication Server 1000E (Communication Server 1000). The Trio Enterprise server uses a SIP connection to the Communication Server 1000 call server via Session Manager. See **Figure 1** for a network diagram. A basic Distance Steering Code configuration (DSC) was configured on the Communication Server 1000 to route all calls to the Trio attendant position. If a call is made from the Trio Enterprise attendant console to the PSTN the call will route from the Trio console via a SIP trunk to Session Manager then to the PSTN. During compliance testing simulated PSTN SIP trunks were used. Trio Enterprise can perform the usual range of attendant call functions, i.e., centralized answering position; extend PSTN calls to users, place PSTN calls on behalf of internal users, perform internal telephone directory lookups.

During tests, calls are placed to a number associated with the Trio attendant position. Session Manager routes all calls destined for the Trio Enterprise server over the SIP connection. The Trio Enterprise server then 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 phone, Trio Enterprise server performs a SIP REFER to connect caller and called user directly. It is possible to have multiple Trio attendant positions on a Communication Server 1000 system.

A variety of Avaya telephones were installed and configured on the Communication Server 1000. 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.
Note: The Trio Enterprise server places a call to the attendant's deskphone, for compliance testing an Avaya IP phone was used as the attendant's deskphone. When the attendant is called the Trio Enterprise server calls the Avaya IP phone and bridges the call.

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 this DevConnect Application Note 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 this Application Note, 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 compatibility tests included the following.

- Incoming internal and external calls
- Outgoing internal and external calls
- Supervised and unsupervised transfer with answer
- Directing calls to busy extensions
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions
- Status of the phones

2.2. Test Results

Tests were performed to insure full interoperability between the Trio Enterprise and the Communication Server 1000. The tests were all functional in nature and performance testing was not included. All the 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 shows the network topology during compliance testing. Trio Enterprise is connected to the Communication Server 1000 using a SIP connection via Session Manager. System Manager is used to configure Session Manager. TR87 is accomplished using the SIP CTI Service. Avaya Aura® Communication Manager was used to emulate a 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.

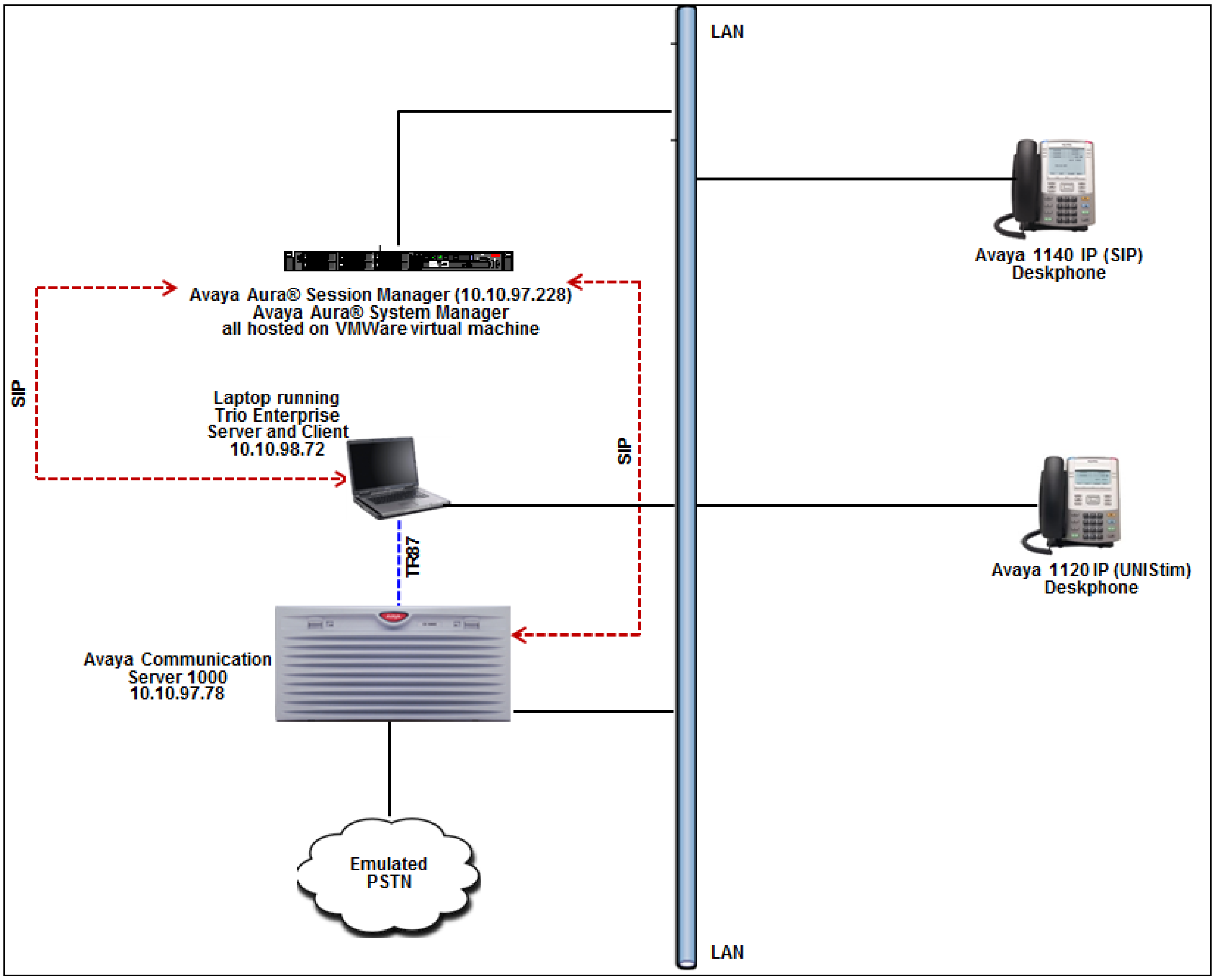


Figure 1: Configuration for Avaya Communication Server 1000, Avaya Aura® Session Manager and Trio Enterprise

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Communication Server 1000	7.65 SP9
Avaya Aura® Session Manager running on virtualized environment	7.1.1.0.711008
Avaya Aura® System Manager running on virtualized environment	7.1.1.0
Avaya 11xx Series IP Telephone • 1120 (UNIStim) • 1140 (SIP)	 C94 4.04.26
Trio Enterprise Server and Client running on Microsoft Windows 2012 R2 Server	7.0

5. Configure Avaya Communication Server 1000E

The configuration operations illustrated in this section were performed using terminal access to the Communication Server 1000 over an "SSH" session using "PuTTY". The information provided in this section describes the configuration of the Communication Server 1000 for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 11**.

Note:

- It is assumed that the SIP connection from Communication Server 1000 to Session Manager is in place and operational and will not be discussed in detail in this application notes. During compliance test, route number (**ROUT**) and route list index (**RLI**) is **6** to Session Manager, this information is needed in Section **02** to configure route to Trio number 71xxx.
- The configuration of the simulated PSTN connections is outside the scope of these Application Notes.
- Not all prompts need a response. The prompts outlined below are mandatory for a basic configuration. Accept the default responses for all other prompts by pressing the return key.

5.1. Verify Licences

Both SIP CTI Licences and AST licenses are required to allow Trio observe TR87 events. To ensure the Communication Server 1000 is licensed for SIP CTI use **LD 22** and type **SLT** at the **REQ** prompt. Check for **SIP CTI TR87** and **AST** (in bold and red below).

```
>ld 22
PT2000

REQ  slt

System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

IPMGs Registered:              1
IPMGs Unregistered:            0
IPMGs Configured/unregistered:  0


TRADITIONAL TELEPHONES 32767    LEFT 32767    USED     0
DECT USERS             32767    LEFT 32767    USED     0
IP USERS               32767    LEFT 32682    USED    85
BASIC IP USERS         32767    LEFT 32764    USED     3
TEMPORARY IP USERS     32767    LEFT 32765    USED     2
DECT VISITOR USER      10000    LEFT 10000    USED     0
ACD AGENTS             32767    LEFT 32739    USED    28
MOBILE EXTENSIONS      32767    LEFT 32761    USED     6
TELEPHONY SERVICES     32767    LEFT 32767    USED     0
CONVERGED MOBILE USERS 32767    LEFT 32767    USED     0
AVAYA SIP LINES        32767    LEFT 32755    USED    12
THIRD PARTY SIP LINES  32767    LEFT 32740    USED    27

PCA                    32767    LEFT 32764    USED     3
ITG ISDN TRUNKS        32767    LEFT 32767    USED     0
H.323 ACCESS PORTS     32767    LEFT 32767    USED     0
AST                    32767    LEFT 32716    USED    51
SIP CONVERGED DESKTOPS 32767    LEFT 32767    USED     0
SIP CTI TR87           32767    LEFT 32734    USED    33
SIP ACCESS PORTS       32767    LEFT 32703    USED    64
```

5.2. Configure Coordinated Dialing Plan

This section show steps on how to create CDP to route the call from CS1000 to Trio Enterprise via Session Manager.

Use the **NEW** command in **LD 87** to create a **CDP** entry for the Trio Enterprise. In the example below, the **DSC** is **71**, **FLEN** is **5** and the **RLI** is **6**.

```
REQ  new
CUST 0
FEAT cdp
TYPE dsc
DSC  71
FLEN 5
DSP  LSC
RRPA NO
RLI  6
CCBA NO
NPA
NXX
```

5.3. Configure TR87 , ELAN and Value Added Server on Communication Server 1000

This section show steps on configure the Class of Service for TR87 events to be allowed from a phone and also to configure the ELAN and Value Added Server (VAS), so that these events can be passed on to Trio Enterprise.

5.3.1. Configure TR87 in Class of Service

To allow Trio observe TR87 events from a specific phoneset TR87, AST and IAPG must be set on a per phoneset basis. Enter overlay 20 to make all of these changes by typing **LD 20** at the **>** prompt. Set the Class of Service (**CLS**) to **T87A** and set the **AST** to **00** (Key 0) and **IAPG** to **1** to allow TR87 events get passed from the phoneset to the Trio application.

```
CLS   CTD FBD WTA LPR PUA MTD FND HTD TDD HFA CRPD
      MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
      POD SLKD CCSD SWD LND CNDA
      CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBD
      ICDD CDMD LLCN MCTD CLBD AUTU
      GPUD DPUD DNDA CFXD ARHD CLTD ASCD
      CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
      UDI RCC HBTD AHD IPND  DDGA NAMA MIND PRSD NRWD NRCD NROD
      DRDD EXR0
      USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
      FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87A SBMD
      KEM3 MSNV FRA  PKCH MUTA MWTD DVLD CROD ELCD VMSA
CPND_LANG ENG
RCO   0
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST  00
IAPG 1
```

5.3.2. Configure ELAN for Trio Enterprise Application

Log in to the command line interface (CLI) of the Communication Server 1000 using the proper credentials (not shown) and issue overlay **LD 17** to create a new ELAN for the Contact Center application. Screen below shows an already configured **ELAN 34**.

```
ADAN     ELAN 34
  CTYP ELAN
  DES  ELAN34
  N1   512
```

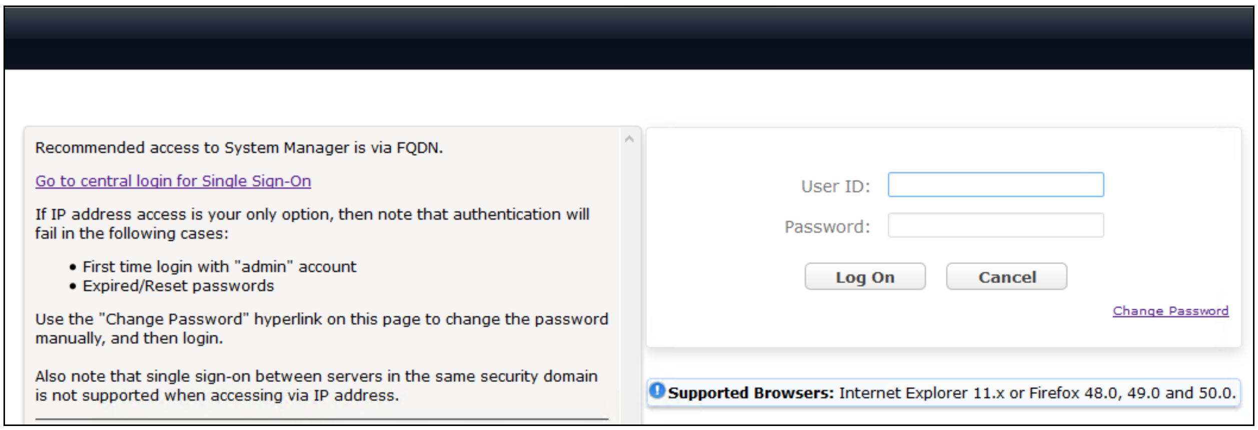
5.3.3. Configure VAS for the ELAN of Trio Enterprise Application

Using the CLI, issue overlay **LD 17** to create a value added server (VAS) for the ELAN 34 that was configured above for the Contact Center application. Screen below shows an already configured **VSID 34**.

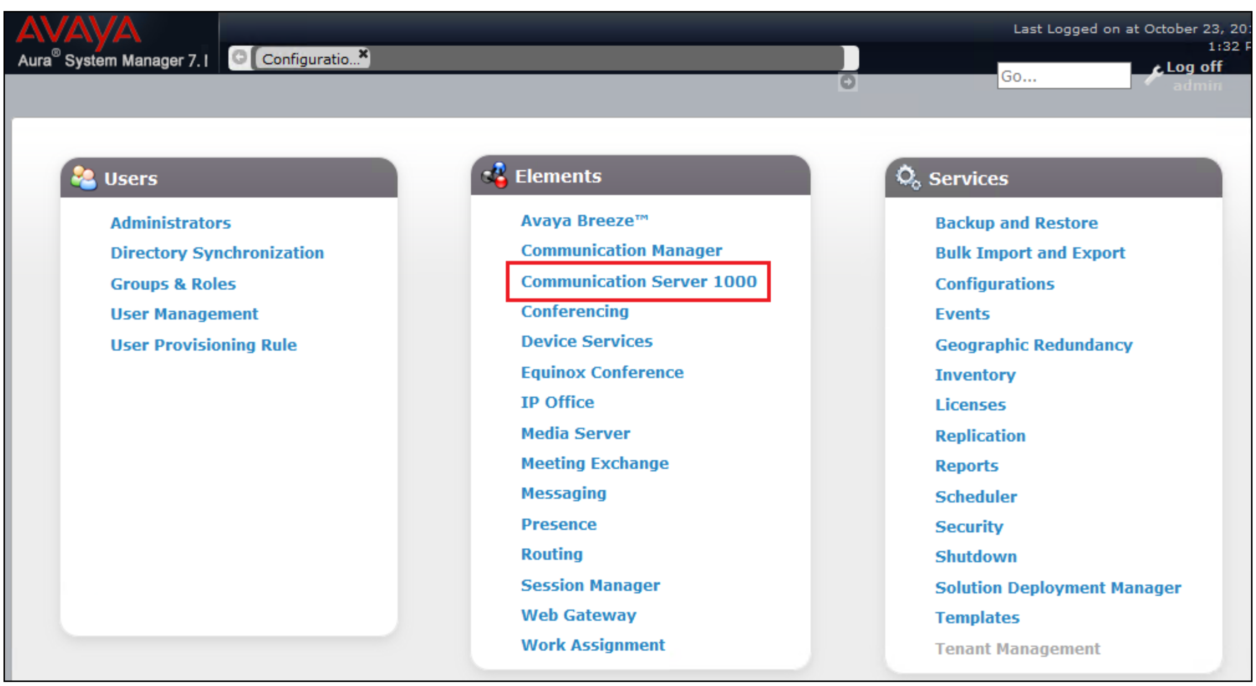
```
VSID   034
    ELAN 034
      SECU YES
      INTL 0001
      MCNT 9999
```

6. Configure Avaya Communication Server 1000 Signalling Server for TR87 events

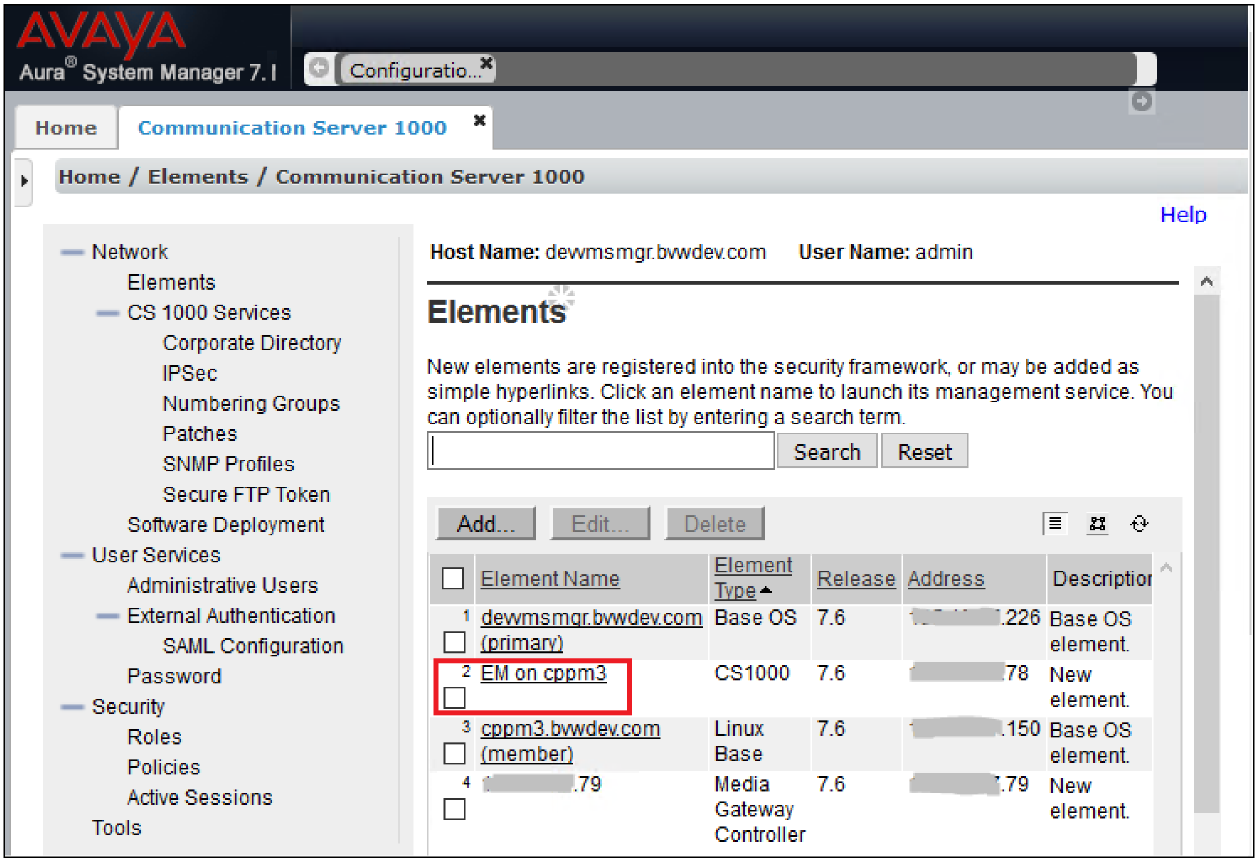
SIP CTI (TR87) services must be enabled and configured on the Communication Server 1000 IP Telephony Node to allow applications obtain presence information or invoke a make-call operation. Changes on the Communication Server 1000 Node are performed using Element Manager which is accessible through the System Manager. To make changes in Element 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.



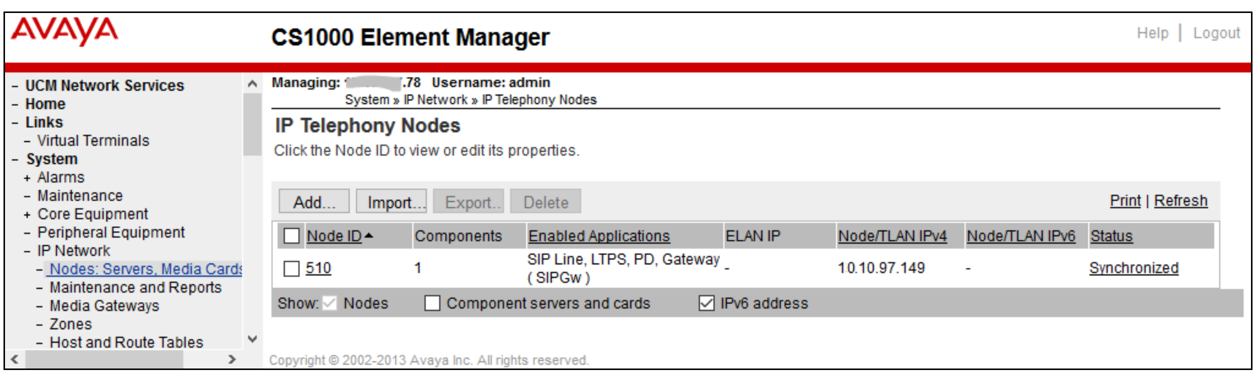
Click on **Communication Server 1000** as shown.



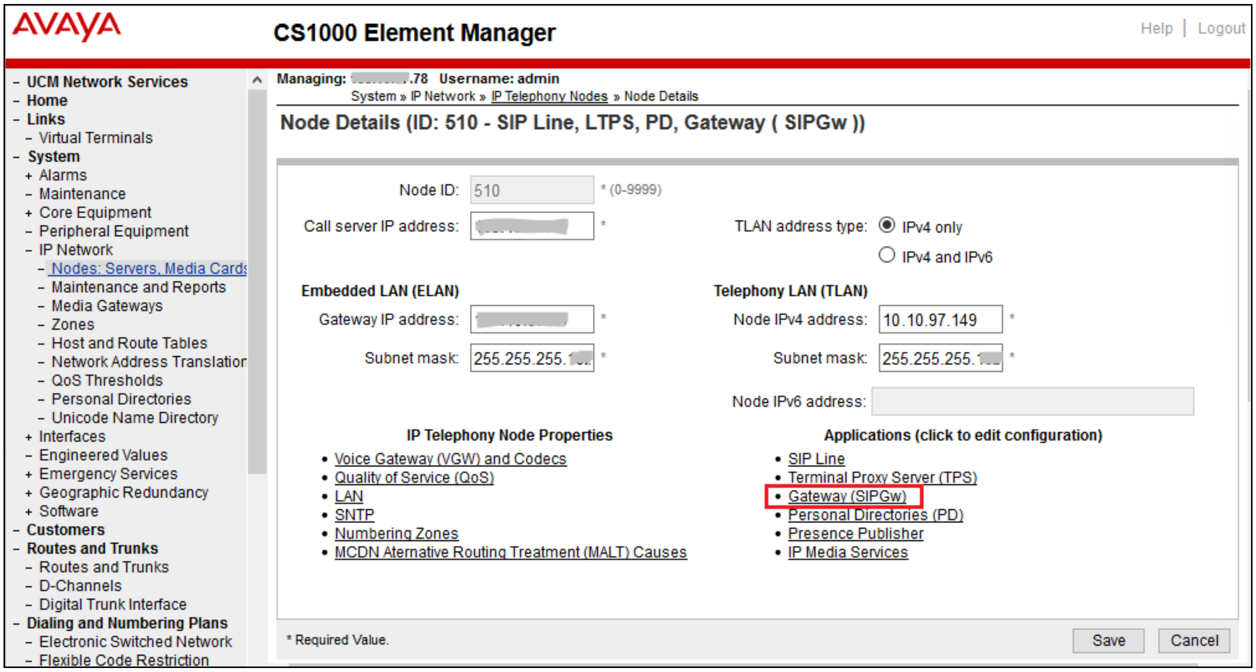
Once **Communication Server 1000** is selected the following screen appears, click on the Element Manager link, in this case click on **EM on cppm3** link.



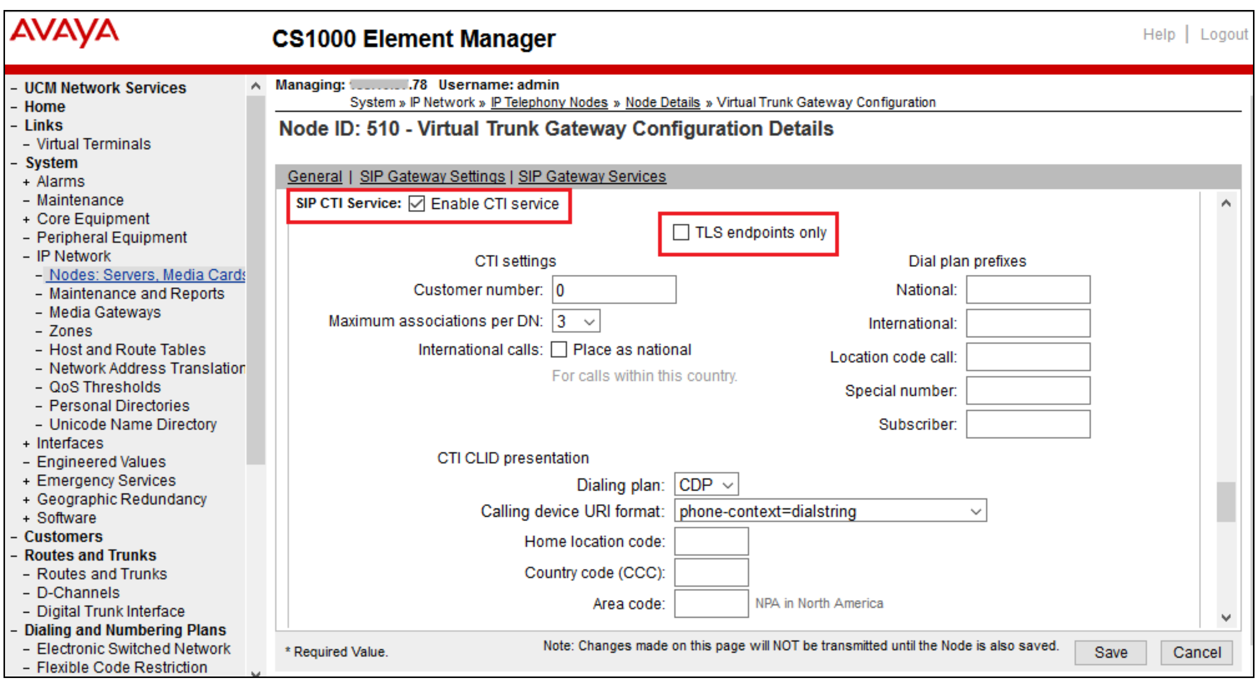
Click on **IP Network → Nodes: Servers, Media Cards** in the left window. Click on the **Node ID** displayed in the right window, during compliance test Node **510** is configured to connect to Session Manager. Note the IP address of this node as it used while configuring Communication Server 1000 as SIP Entity endpoint on Session Manager in **Section 7.5.2.** Trio Enterprise also gets TR87 events via Node **510** and hence this IP address will also be used in **Section 8.4**.



Select **Gateway (SIPGw)** in **Applications (click to edit configuration)** section as shown below.



Ensure that for the field **SIP CTI Service**, the **Enable CTI service** box is selected as shown below and uncheck the **TLS endpoints only** (if this is selected) box; retain default values for all other fields. Click on **Save** once finished.



Save and transmit (not shown) these Node properties to complete the SIPGw configuration. Once the components are synchronized the Signalling Gateway will require a restart.

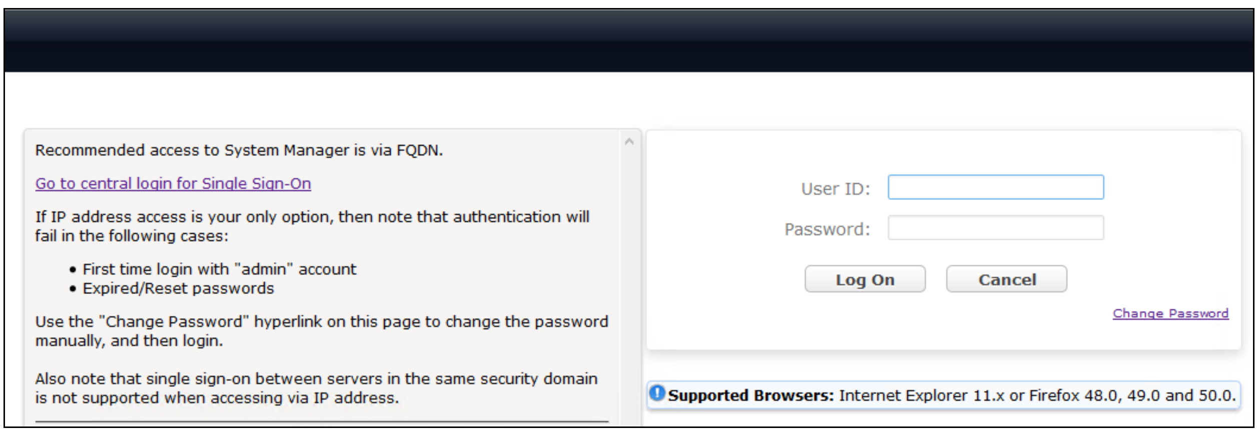
7. Configure Avaya Aura® Session Manager

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

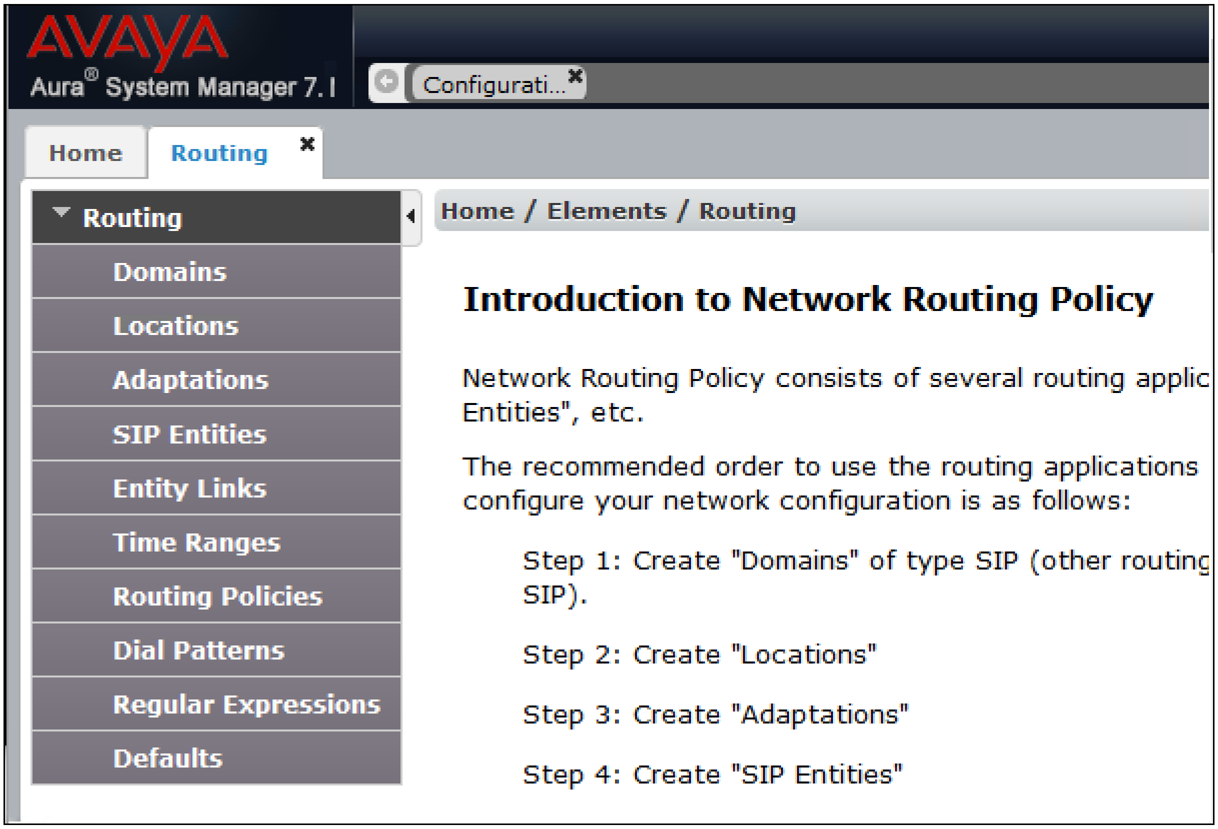
7.1. Launch System Manager

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.

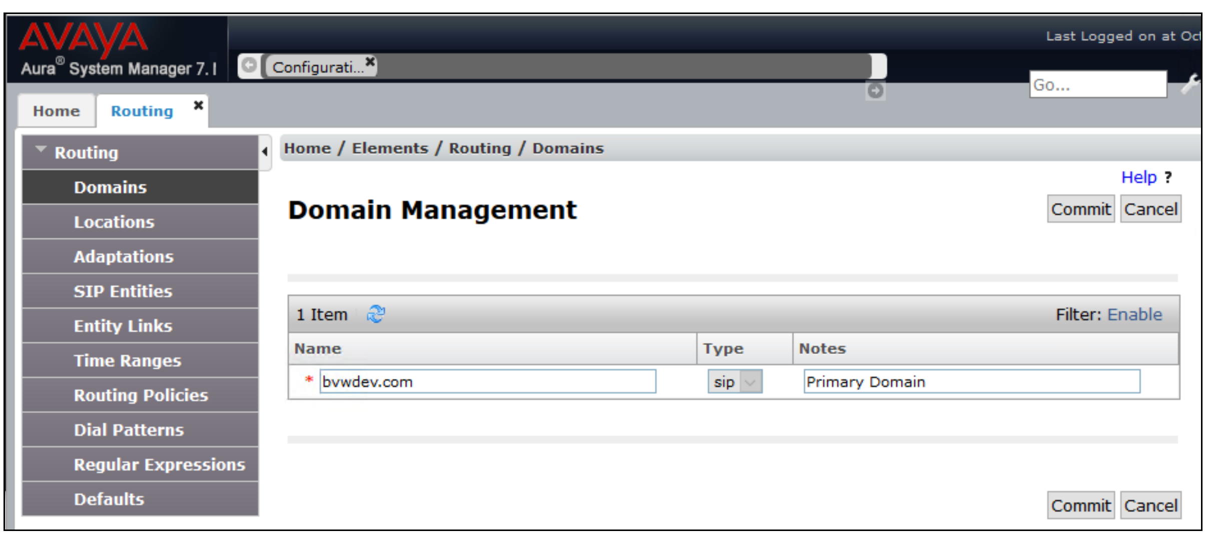


7.2. Administer Domain

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.



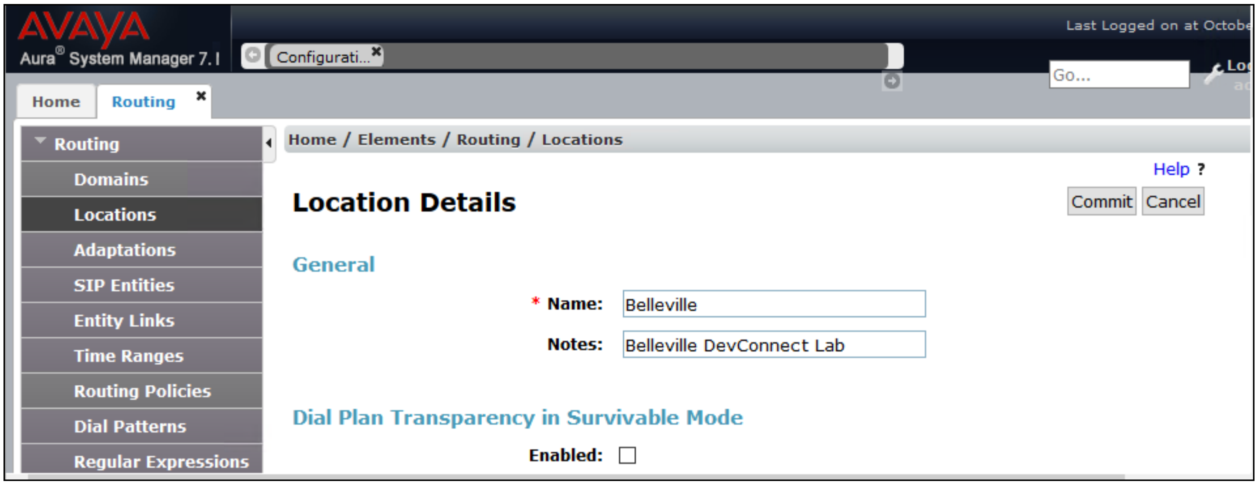
The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select *sip* from the **Type** drop down menu and provide any optional **Notes**.



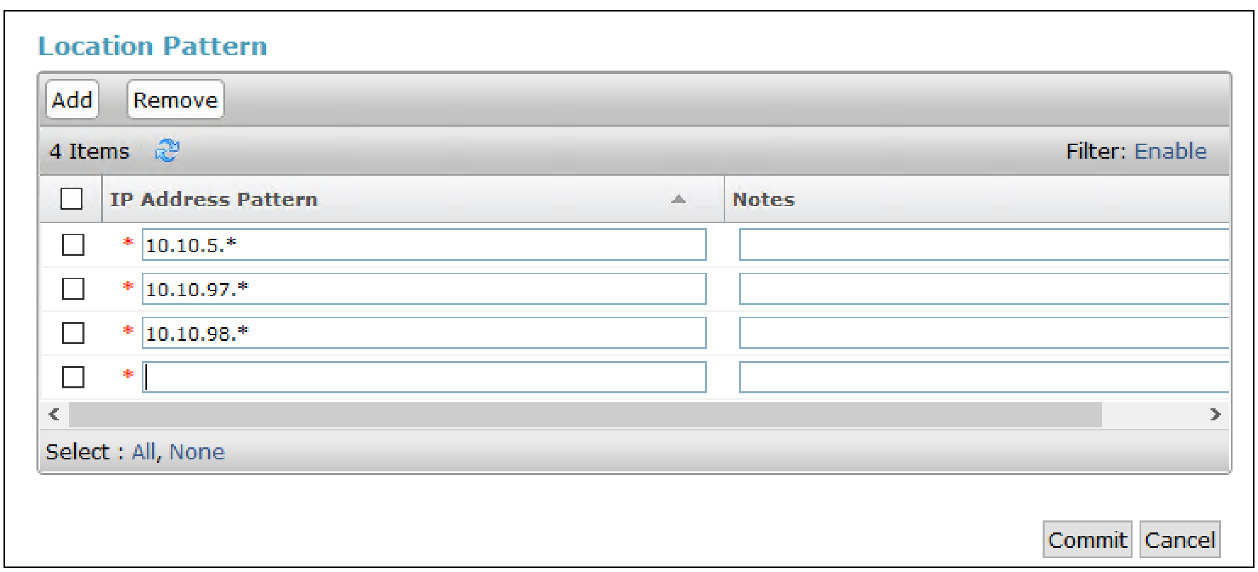
7.3. Administer Locations

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 the simulated PSTN using SIP trunk was between Communication Server 1000 and Avaya Aura® Communication Manager via Session Manager. In order 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. Another adaptation was used for Communication Server 1000 to change the phone context information. Here is step on how to create Adaptation. Select **Adaptations** on the left panel menu and then click on the **New** button in the main window (not shown).

7.4.1. Adaptation for Trio SIP Entity

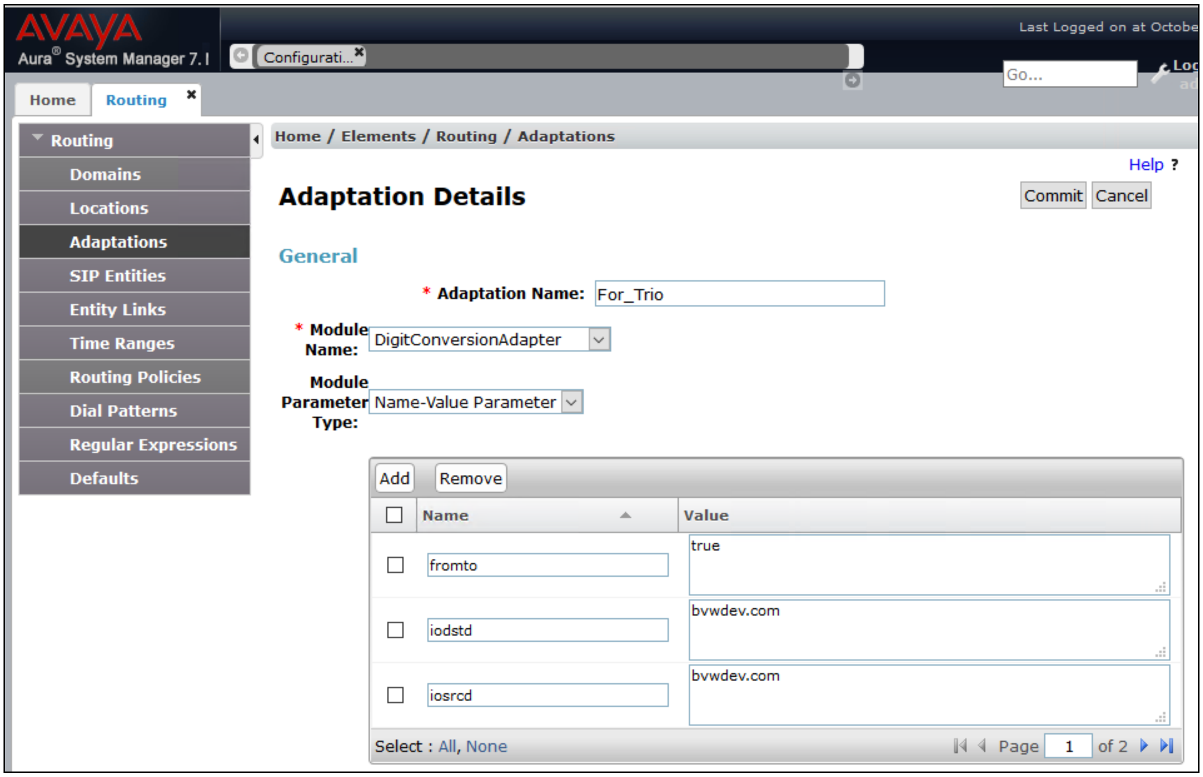
Enter the following for the Trio Adaptation.

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

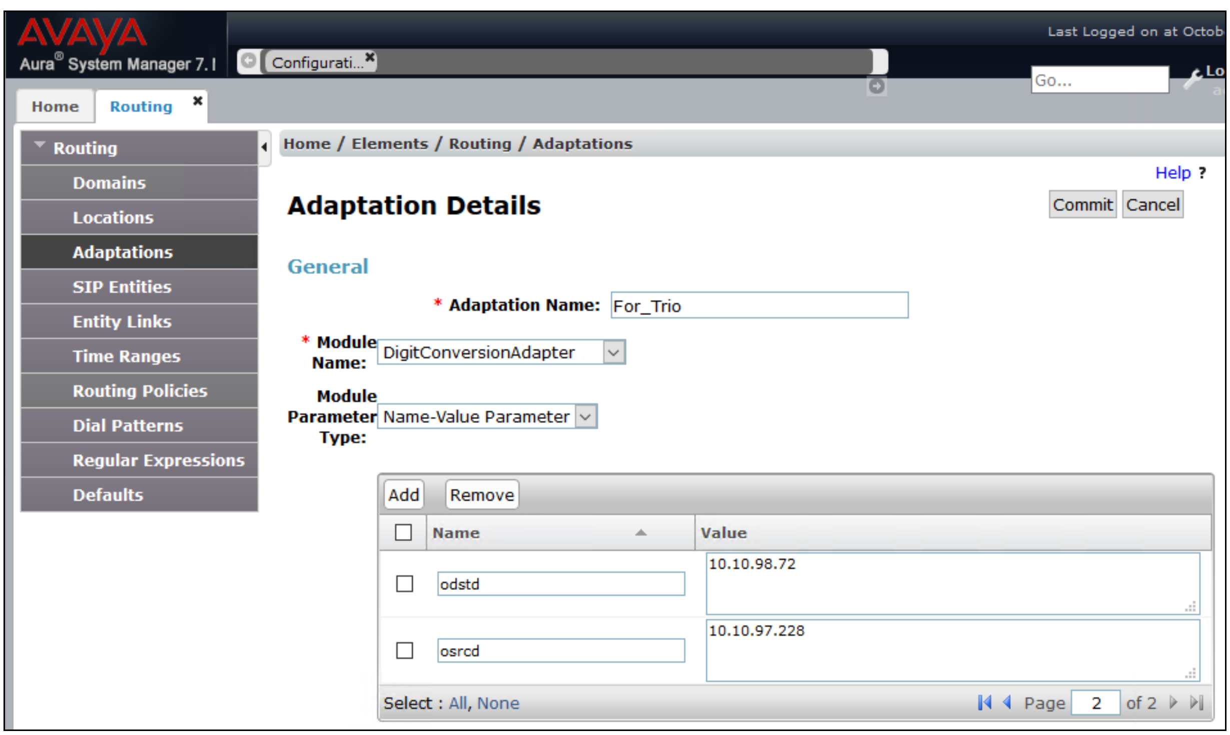
Click **Add** to add a new row for the following values as shown below table:

Name	**Value**
fromto	true
iodstd	Enter the domain name of system, ex: **bvwdev.com**
iosrcd	Enter the domain name of system, ex: **bvwdev.com**
odstd	Enter IP address of Trio, ex: **10.10.98.72**
osrcd	Enter IP Address of Session Manager, ex: **10.10.97.228**

Once the correct information is entered click the **Commit** button. Here the screenshot shows Adaptation created for Trio.



(Continue) the screenshot show Adaptation created for Trio:



7.4.2. Adaptation for Communication Server 1000 SIP Entity

Enter the following for the Communication Server 1000 Adaptation.

- **Adaptation Name:** An informative name (e.g., **CS1000Adapter**)
- **Module Name:** Select "CS1000Adapter"
- **Module Parameter Type:** Select "Name-Value Parameter"

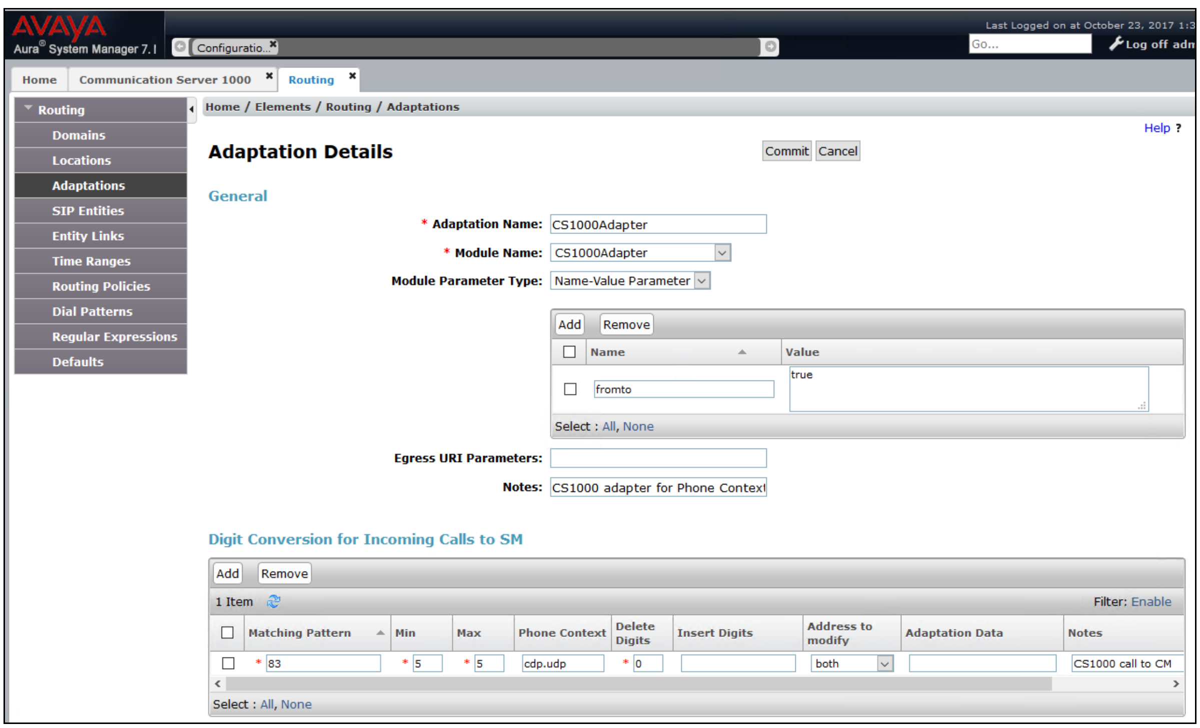
Click **Add** to add a new row for the following values as shown below table:

Name	Value
fromto	true

For **Digit Conversion for Incoming call to SM**, configure the following.

- **Matching Patterns:** Pattern to match for the incoming calls. In this case calls coming from Communication Server 1000 to Session Manager.
- **Min:** Minimum number of digits to be matched.
- **Max:** Maximum number of digits to be matched.
- **Phone Context:** Optional parameter for the ingress adaptation rules. In this case "cdp.udp" based on Communication Server 1000 SIP URI.
- **Address to modify:** "Both". A setting of both will look for adaptations on both origination and destination type headers.
- **Notes:** Any descriptive notes.

Once the correct information is entered click the **Commit** button. Here the screenshot shows Adaptation created for Communication Server 1000.



7.5. Administer SIP Entities

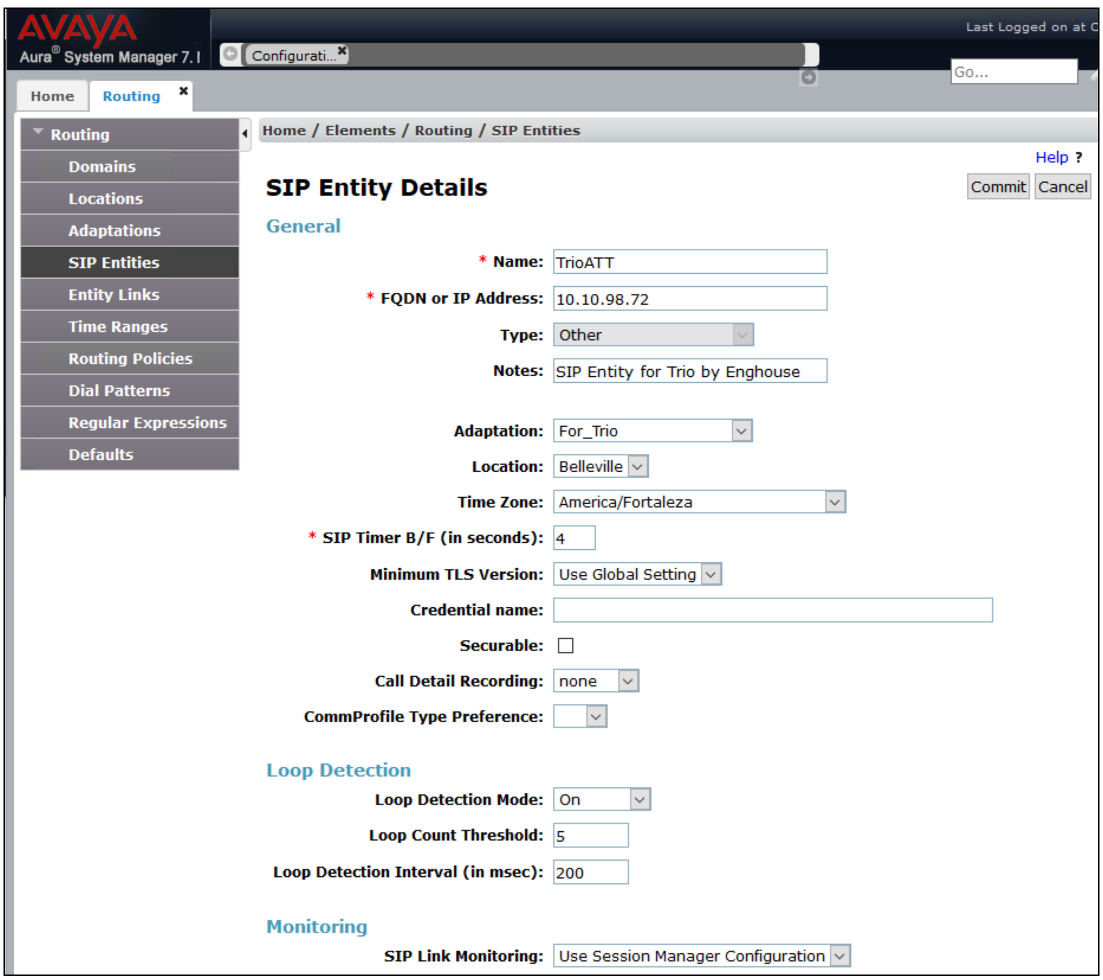
Add two new SIP entities, one for Trio Etnerprise and one for Communication Server 1000.

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.

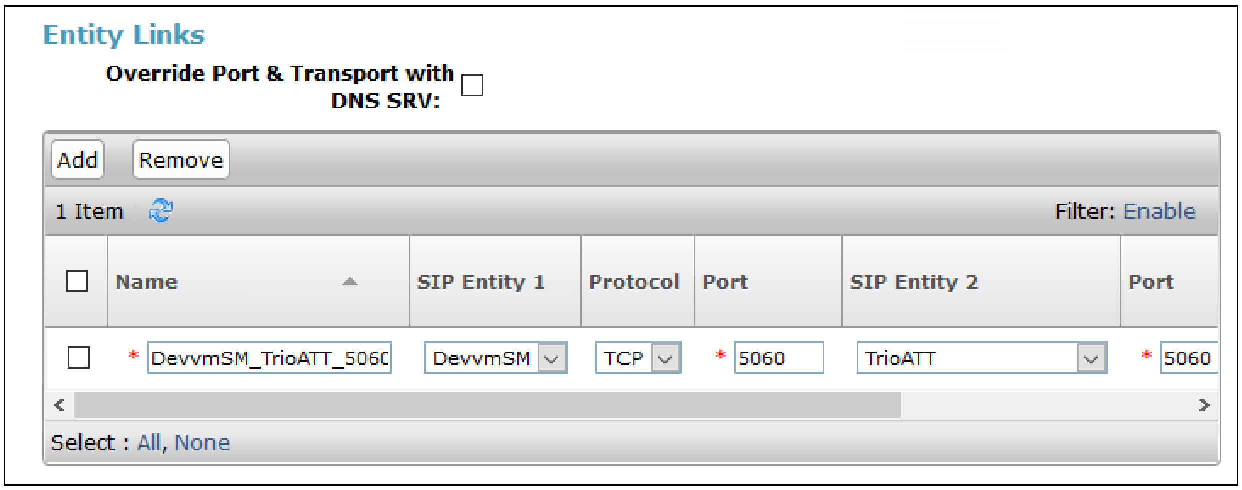
- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the Trio Enterprise server.
- **Type:** "Other"
- **Notes:** Any desired notes.
- **Adaptation:** Select the adaptation configured in **Section 7.4.1**
- **Location:** Select the Trio Enterprise location name from **Section 7.3**.
- **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:** “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 during compliance testing.

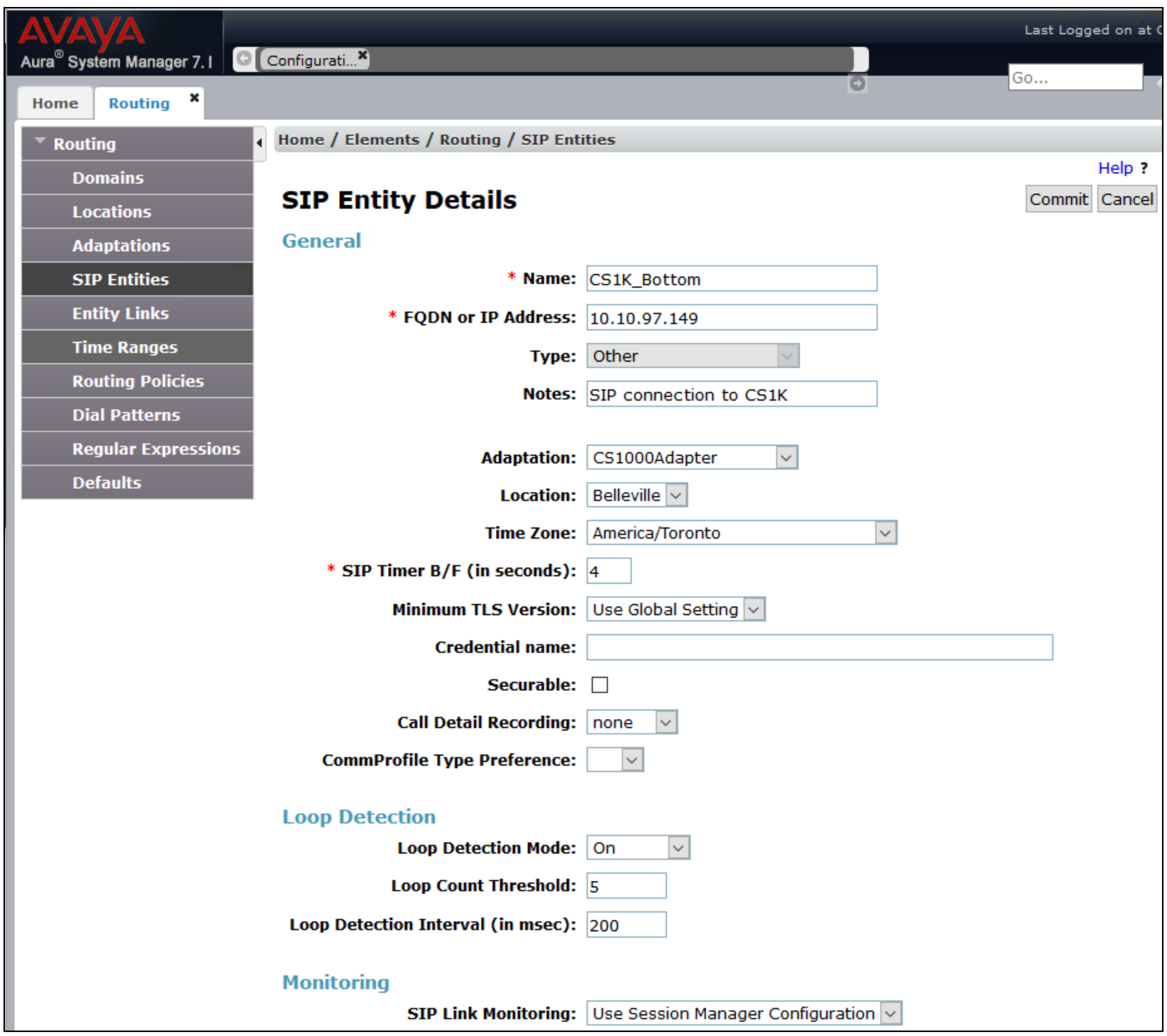


7.5.2. SIP Entity for Communication Server 1000

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 Server 1000. Note that this SIP entity is used for integration with Trio Enterprise.

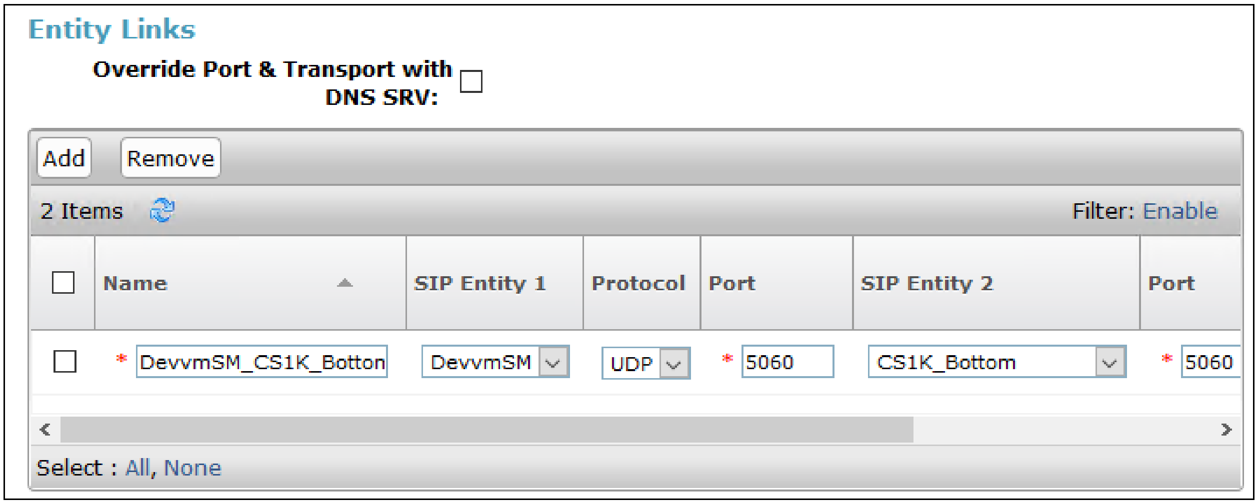
The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing Communication Server 1000 node IP.
- **Type:** "Other"
- **Notes:** Any desired notes.
- **Adaptation:** Select the adaptation configured in **Section 7.4.2**
- **Location:** Select the applicable location for Communication Server 1000.
- **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:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The Communication Server 1000 entity name from this section.
- **Port:** "5060"
- **Connection Policy:** "trusted"



7.6. Administer Routing Policies

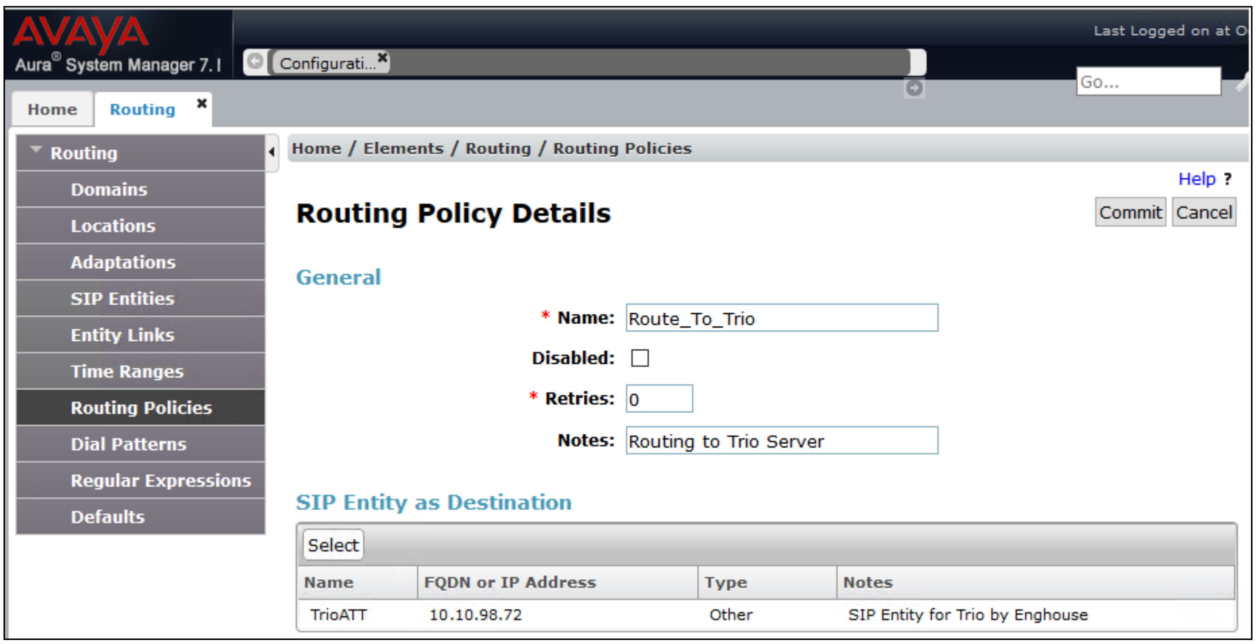
Add two new routing policies, one for Trio Enterprise and one for Communication Server 1000.

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.

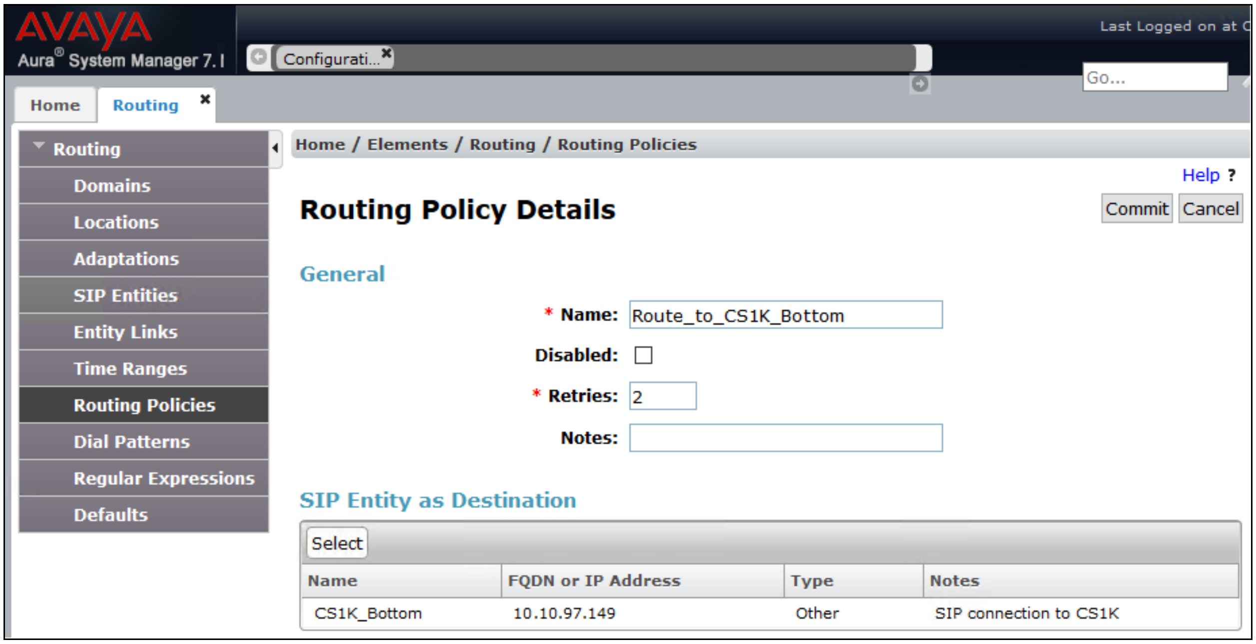


7.6.2. Routing Policy for Communication Server 1000

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 Server 1000.

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 Server 1000 entity name from **Section 7.5.2**. The screen below shows the result of the selection.



7.7. Administer Dial Patterns

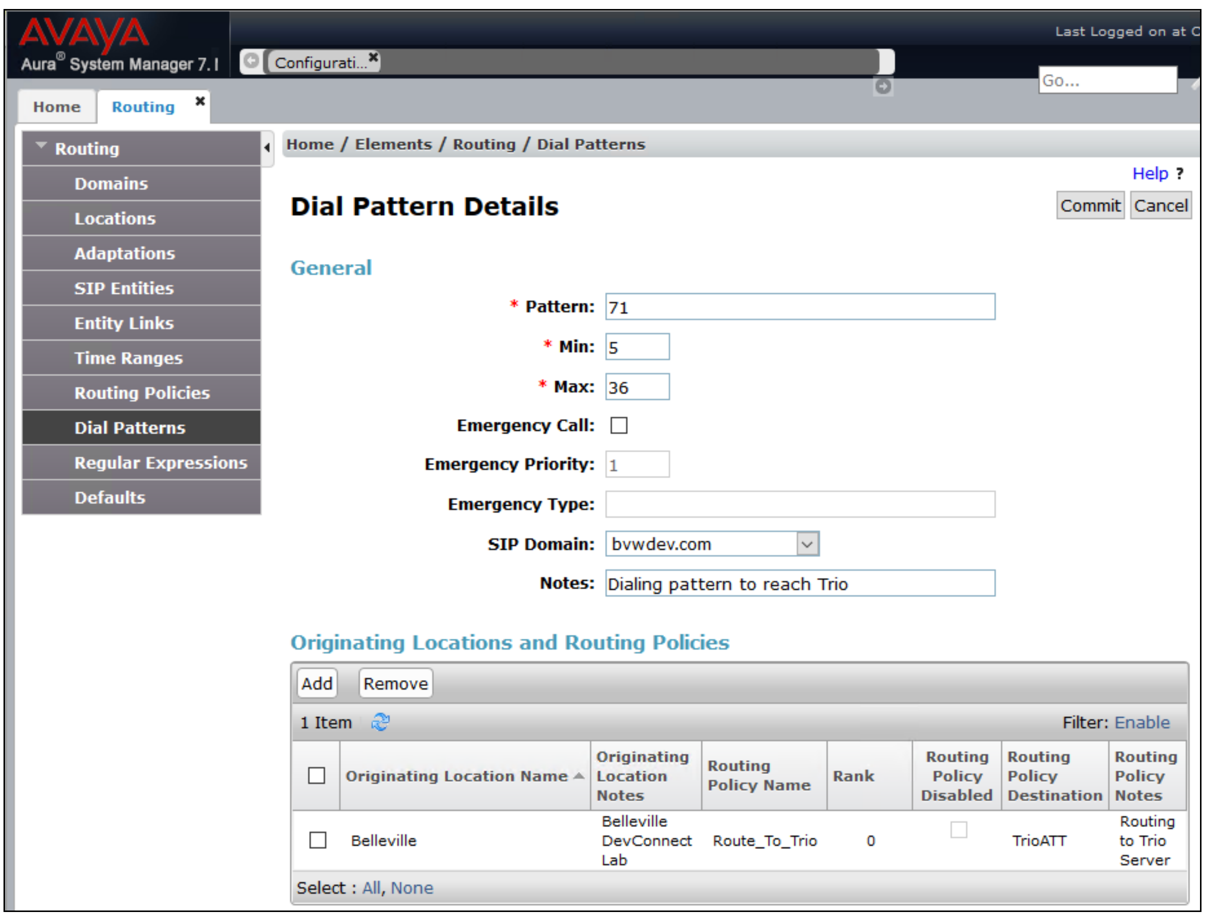
Add a new dial pattern for Trio Enterprise, and update existing dial patterns for Communication Server 1000.

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 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 Server 1000 endpoints in locations "Belleville". The Trio Enterprise routing policy from **Section 7.6.1**was selected as shown below.



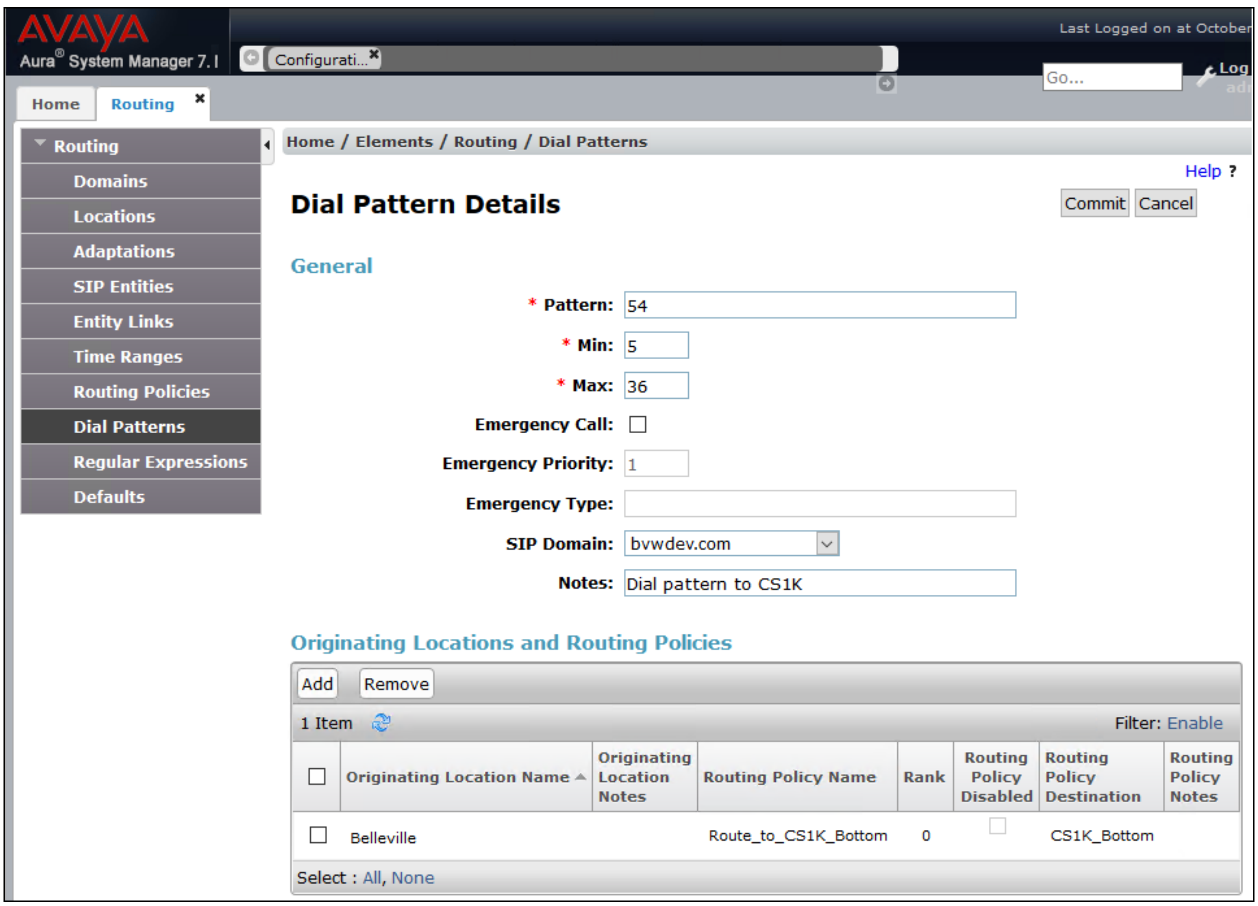
7.7.2. Dial Pattern for Communication Server 1000

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 Server 1000. 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 "54".
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The domain name from **Section 7.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Communication Server 1000. In the compliance testing, the entry allowed for call originations from all Trio Enterprise endpoints in locations "Belleville". The Communication Server routing policy from **Section 7.6.2** was selected as shown below.

Follow the procedures in this section to make similar changes to the applicable Communication Server 1000 dial pattern to reach the PSTN (not shown).



8. Configure TRIO Enterprise

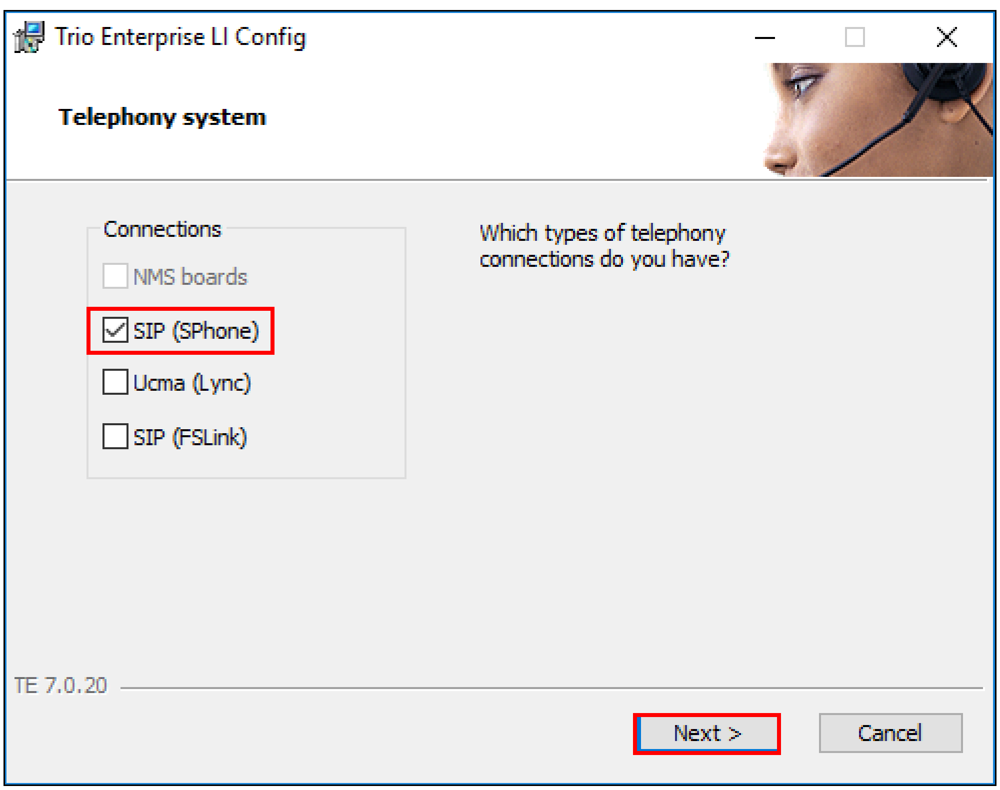
Trio Enterprise connects as a SIP endpoint to the Communication Server 1000 through Session Manager. Trio Enterprise is added to Session Manager as a SIP Entity and calls are routed to the Trio Enterprise server according to the Coordinated Dial Plan setup in **Section 5.2**. This section shows how to configure Trio Enterprise to successfully connect to the Communication Server 1000 using SIP trunks. The installation of the Trio Enterprise software is assumed to be completed and the Trio services are up and running. The steps to configure SIP Trunks are as follows.

8.1. Configure Trio Enterprise to use SIP Trunks

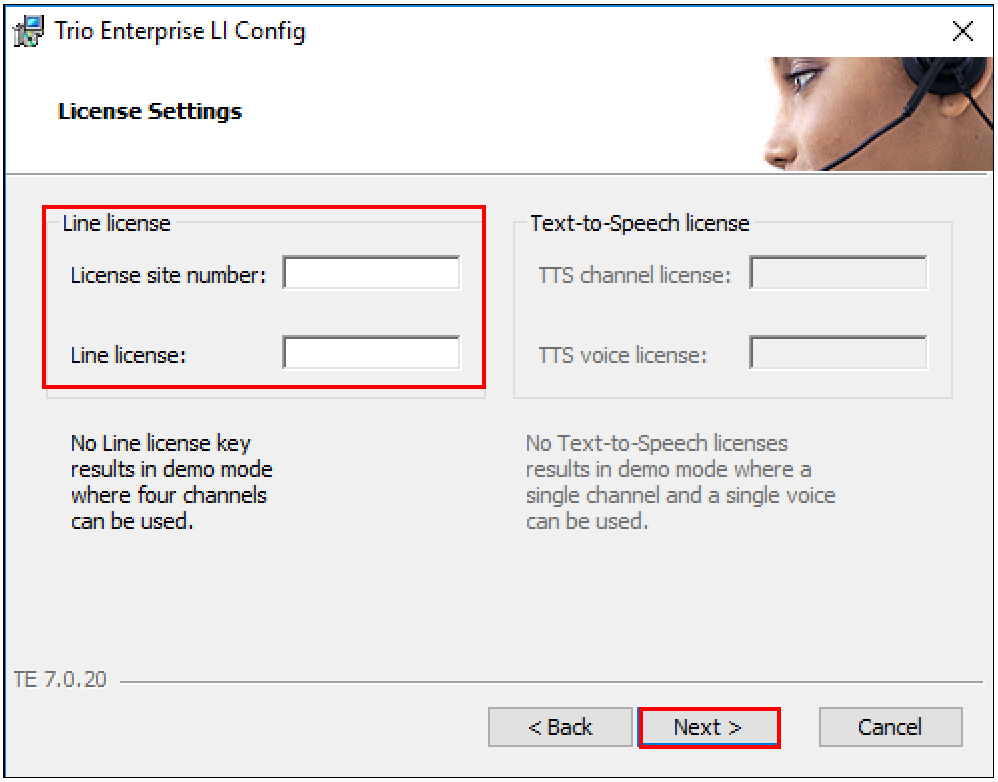
Access Windows services. Select Start → Run, then type **services.msc** into the command line and press return (not shown). When the services window opens, locate the **Trio Televoice service**, right click and select **stop** to stop the service (not shown).

Launch the 'TeleVoice Config' shortcut 

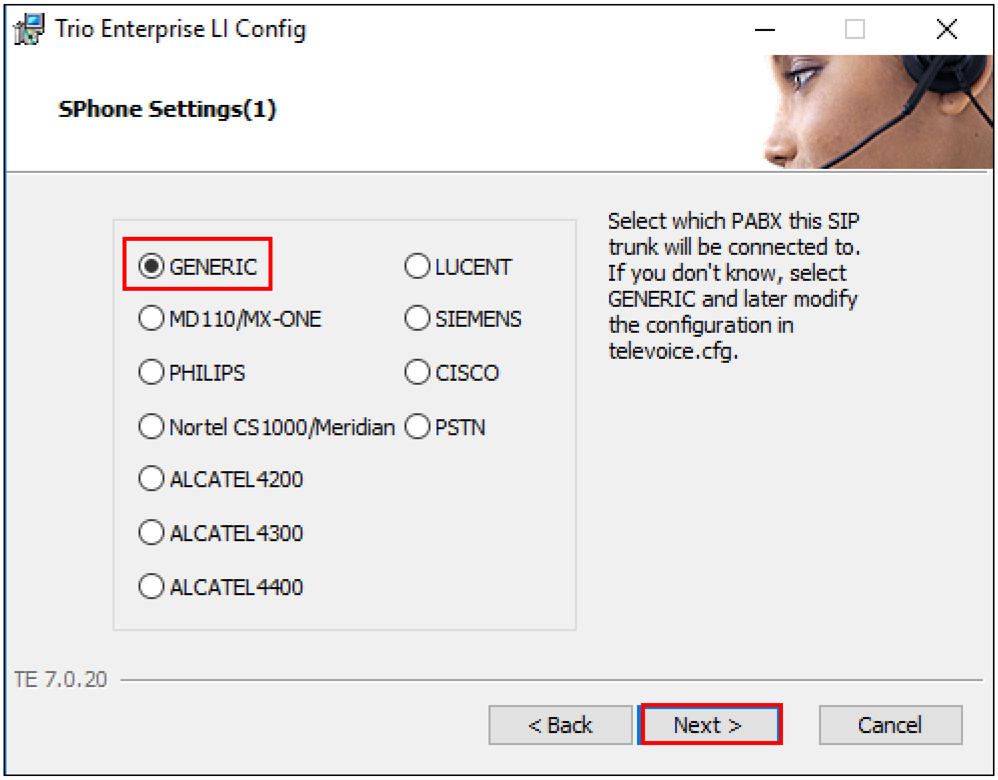
The configuration of the application starts, and when the new window opens, check the **SIP** check box followed by the **Next** button.



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



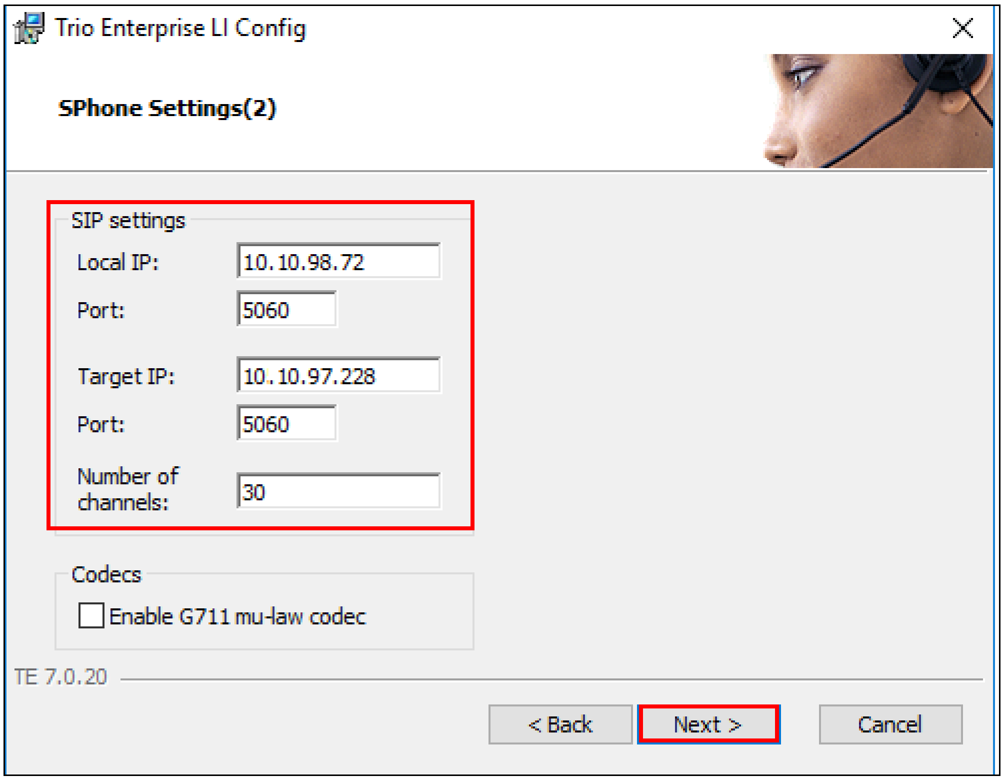
In the subsequent window, click on the **GENERIC** radio button followed by the **Next** button to continue.



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

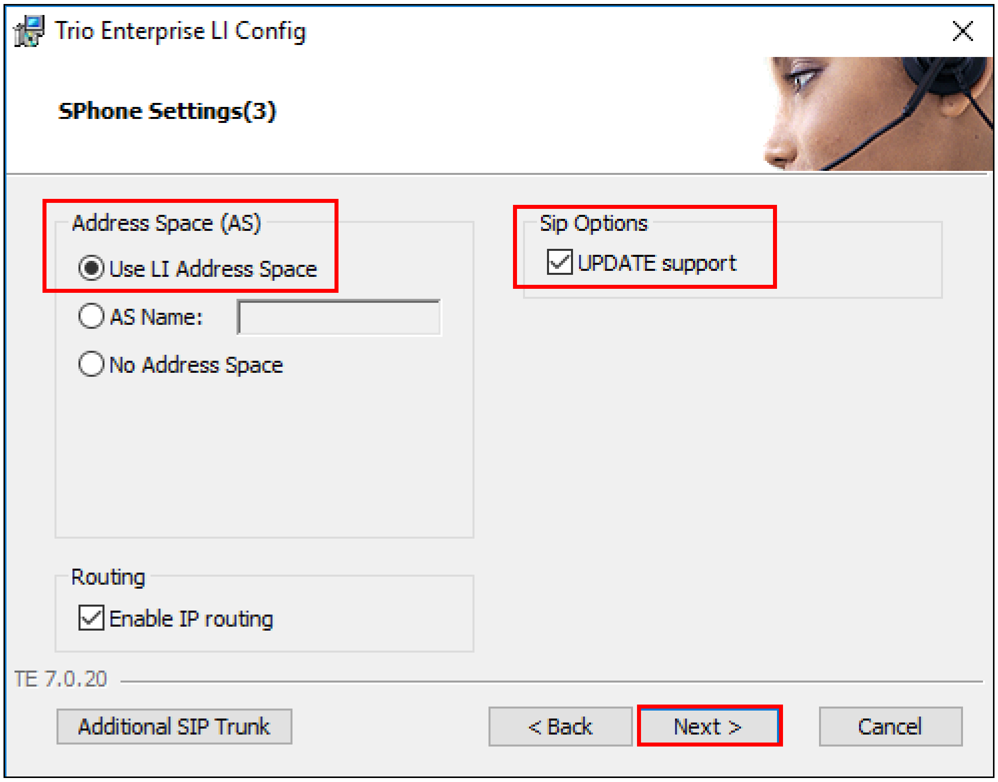
Click on the **Next** button to continue.



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

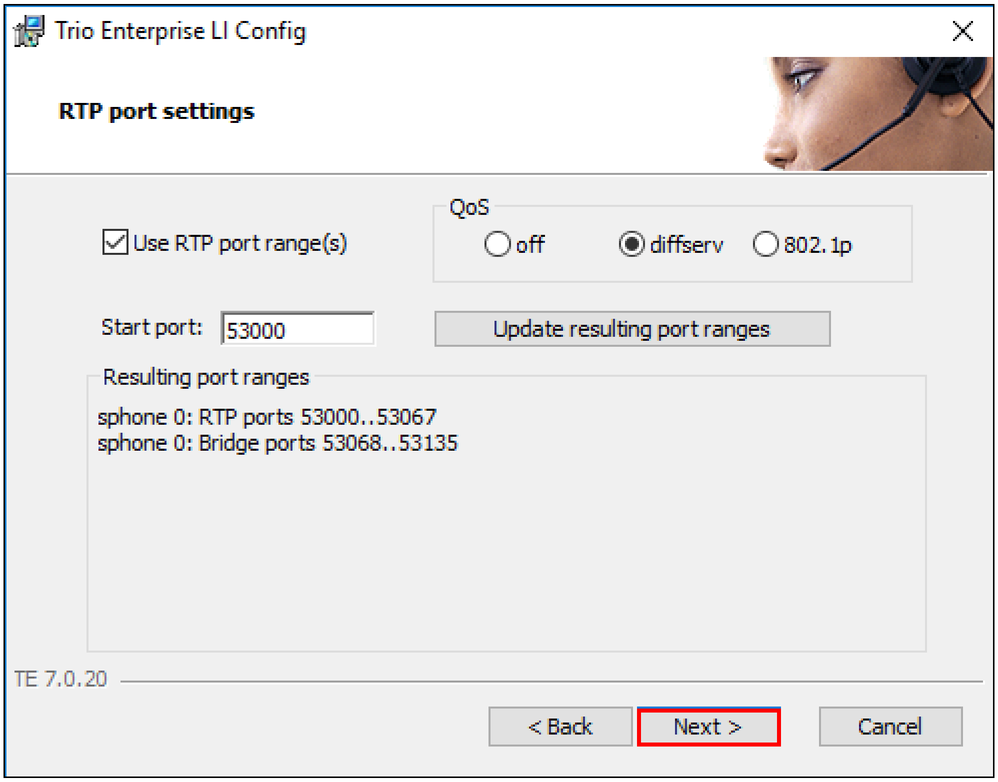
Click on the **Next** button to continue.



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**

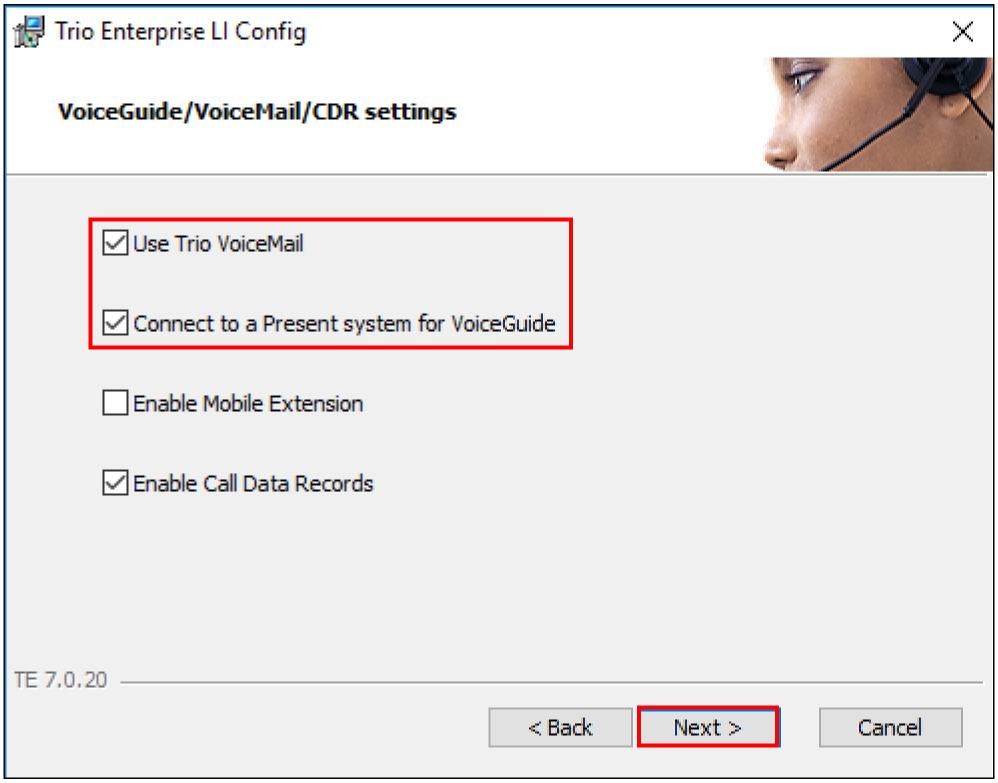
Click on the **Next** button to continue.



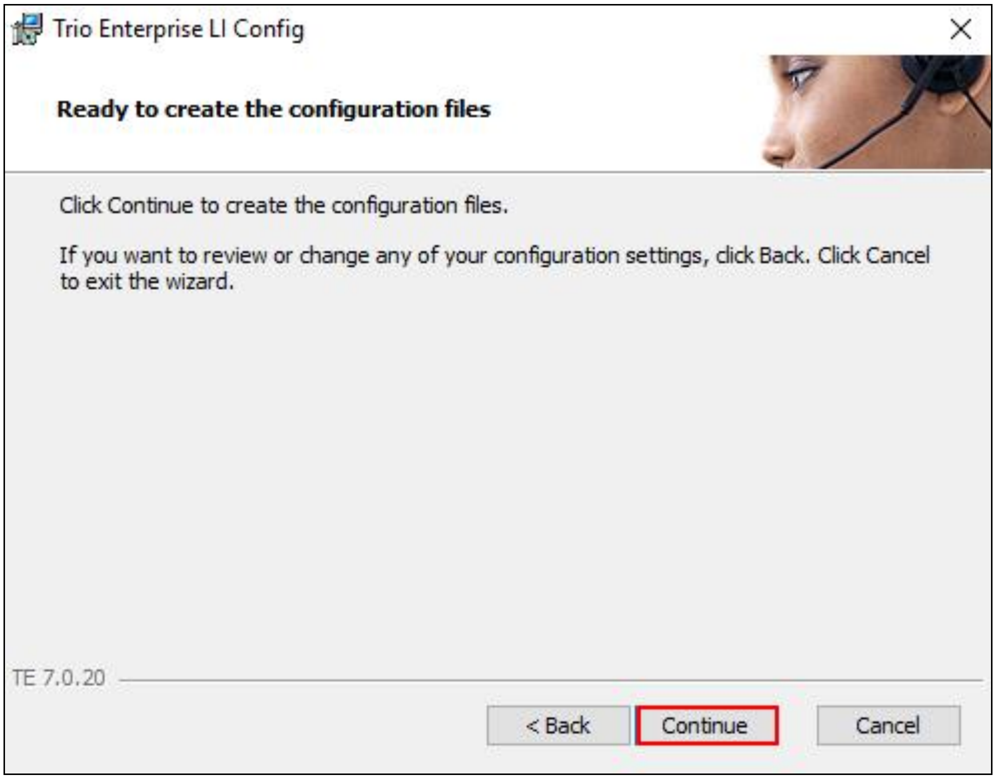
In the subsequent window enter the following settings:

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

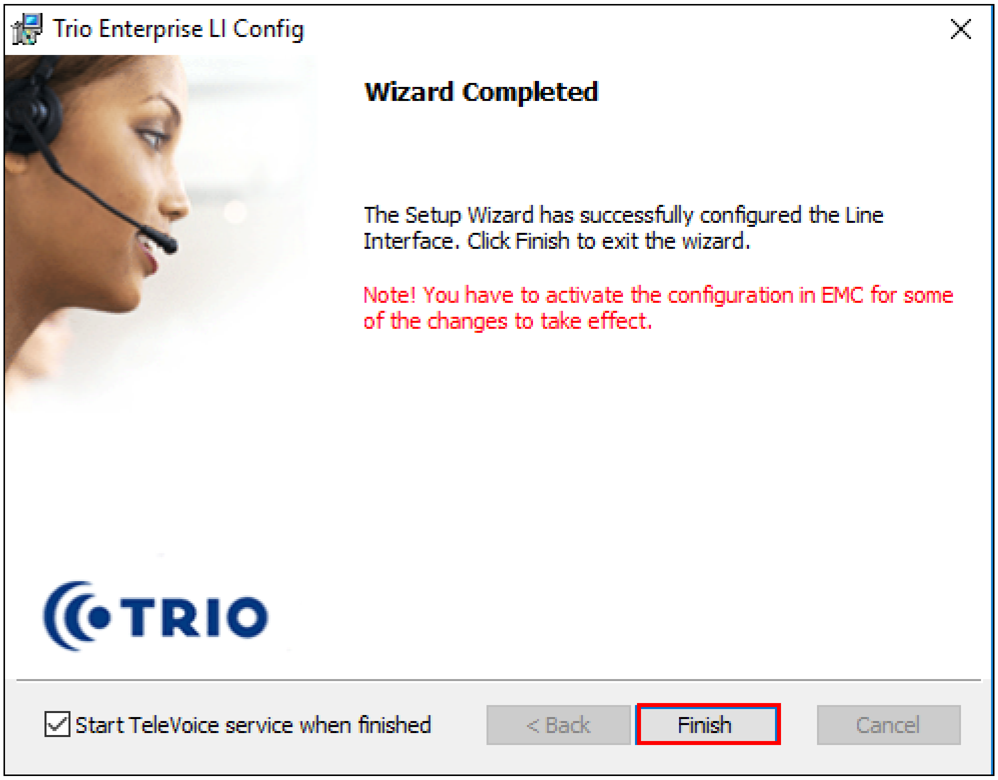
Click on the **Next** button to continue.



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.



8.2. InteractionStudio Configuration

The InteractionStudio is used to configure many features for Trio Enterprise. For compliance testing, the following were configured.

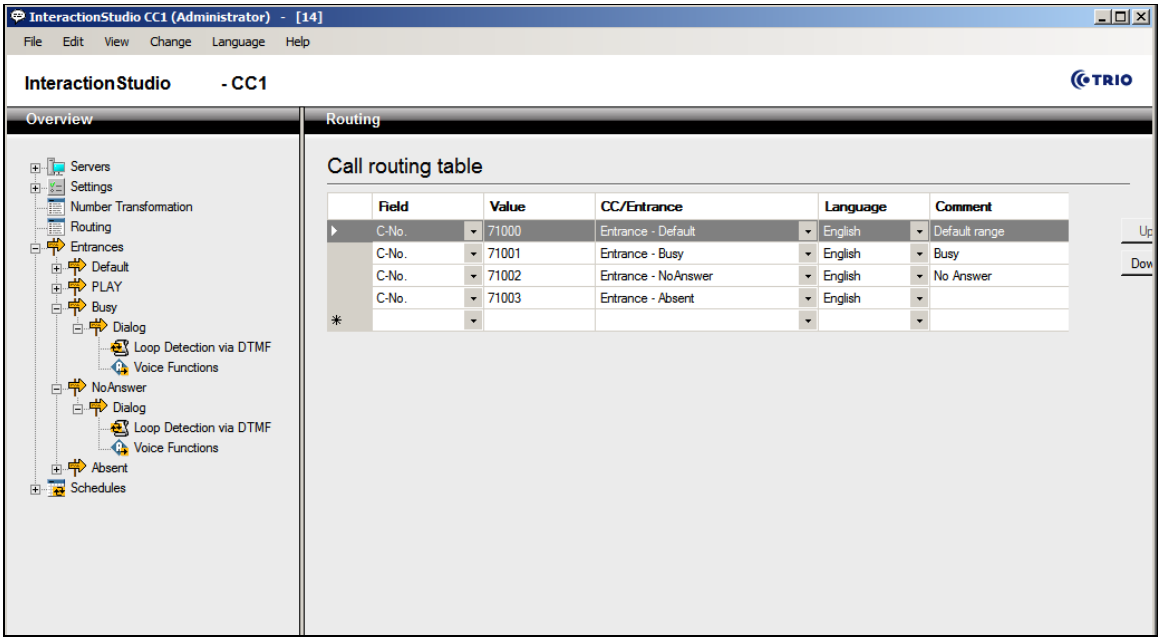
- Configure Call Routing table
- Configure Attendant Service
- Configure Loop Detection via DTMF for Busy signal
- Configure Loop Detection via DTMF for No Answer signal

8.2.1. Configure Call Routing table

On the Trio Enterprise server, launch the 'Interaction Studio' shortcut

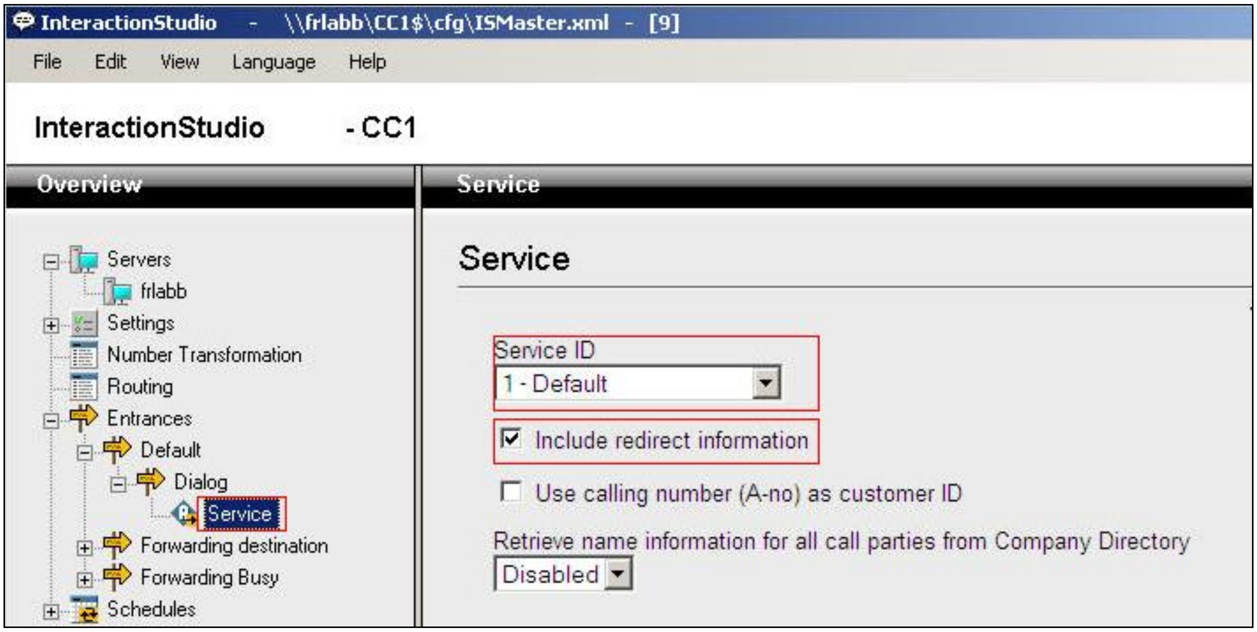
When the Interaction Studio window opens, navigate to **Routing**. A **Call routing table** will open. In the example below:

- Extension **71000** is the main queue number.
- Extension **71001** is the number that calls go to when Call forward Busy is activated.
- Extension **71002** is the number that calls go to when Call forward No Answer is activated.
- Extension **71003** is the number that calls go to when user absent is activated.



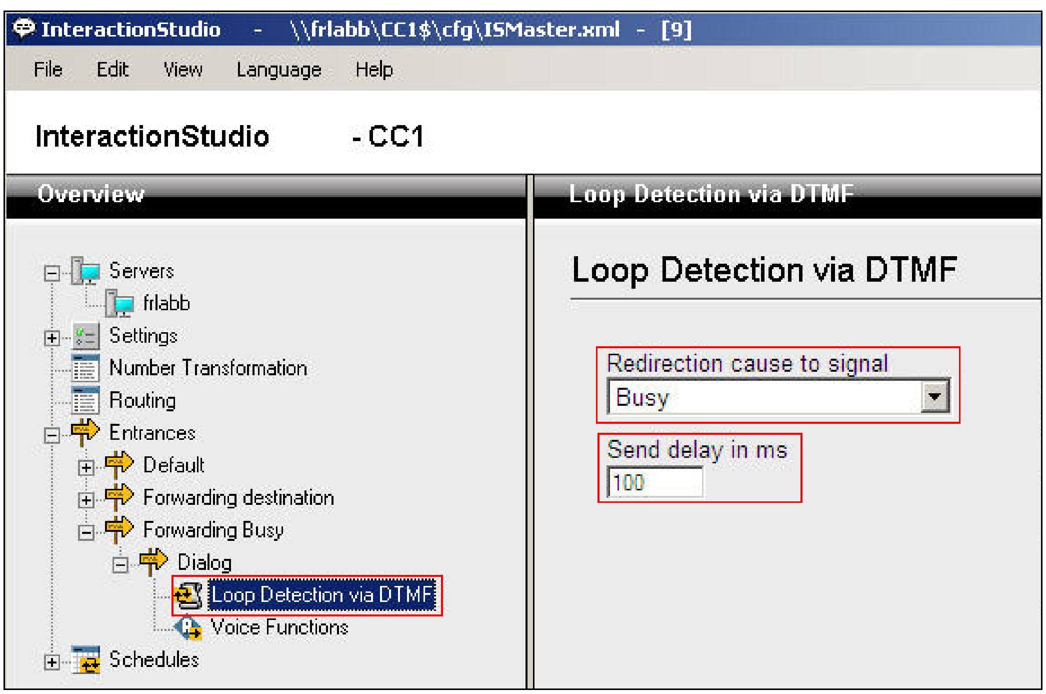
8.2.2. Configure Attendant Service

Navigate to **Entrances → Default → Dialog → Service.** Choose **Default** from the **Service ID** drop down box, and check the **Include redirect information** check box.



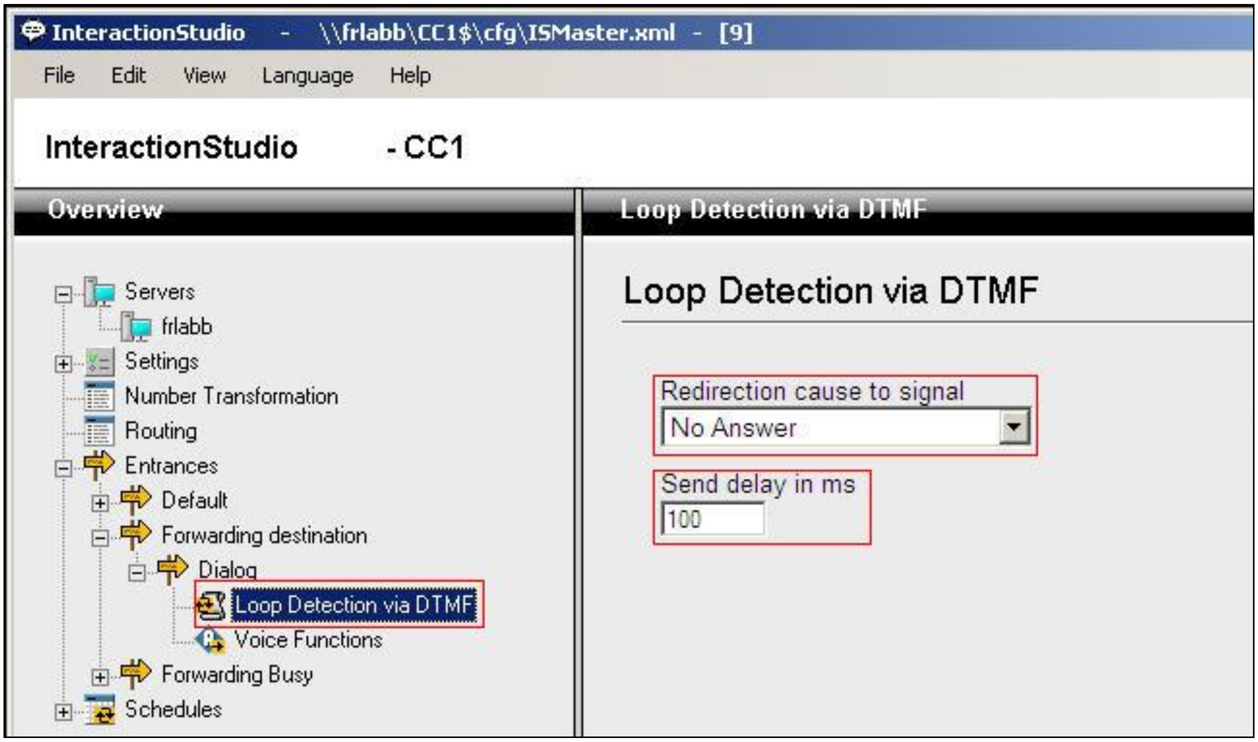
8.2.3. Configure Loop Detection via DTMF for Busy signal

Navigate to **Entrances → Forwarding Busy → Dialog → Loop Detection via DTMF**. Choose **Busy** from the **Redirection cause to signal** drop down box, and enter **100** in the **Send delay in ms** box.



8.2.4. Configure Loop Detection via DTMF for No Answer signal

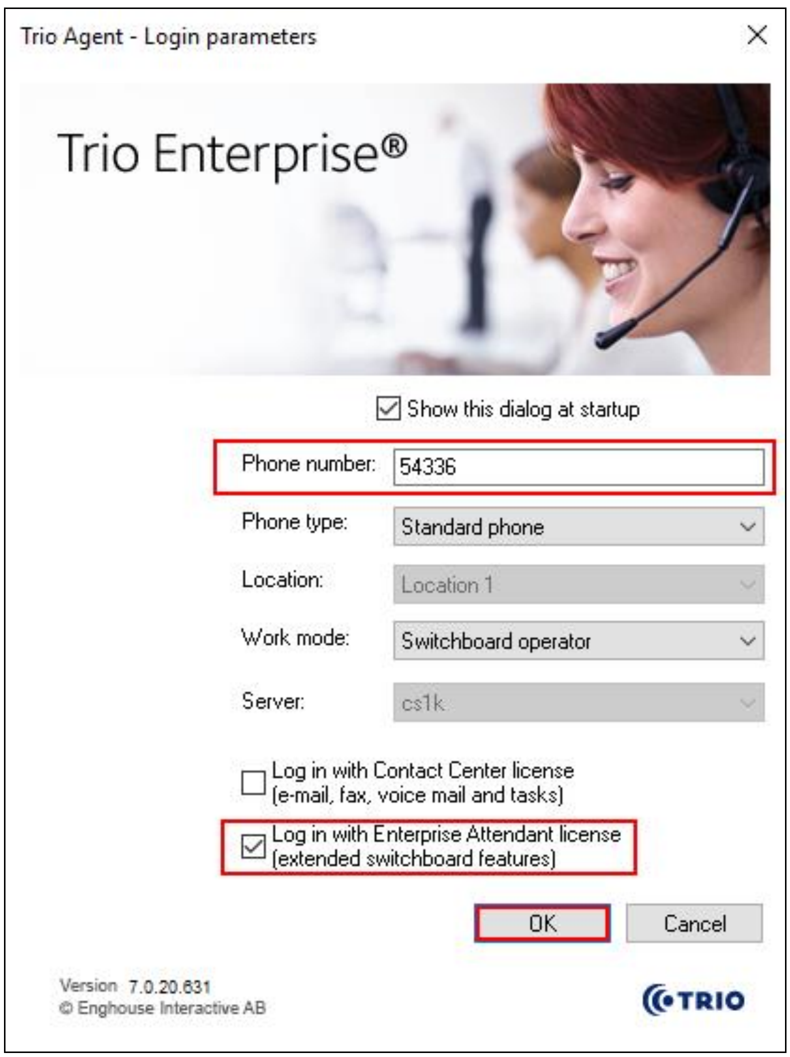
Navigate to **Entrances → Forwarding destination → Dialog → Loop Detection via DTMF**. Choose **No Answer** from the **Redirection cause to signal** drop down box, and enter **100** in the **Send delay in ms** box.



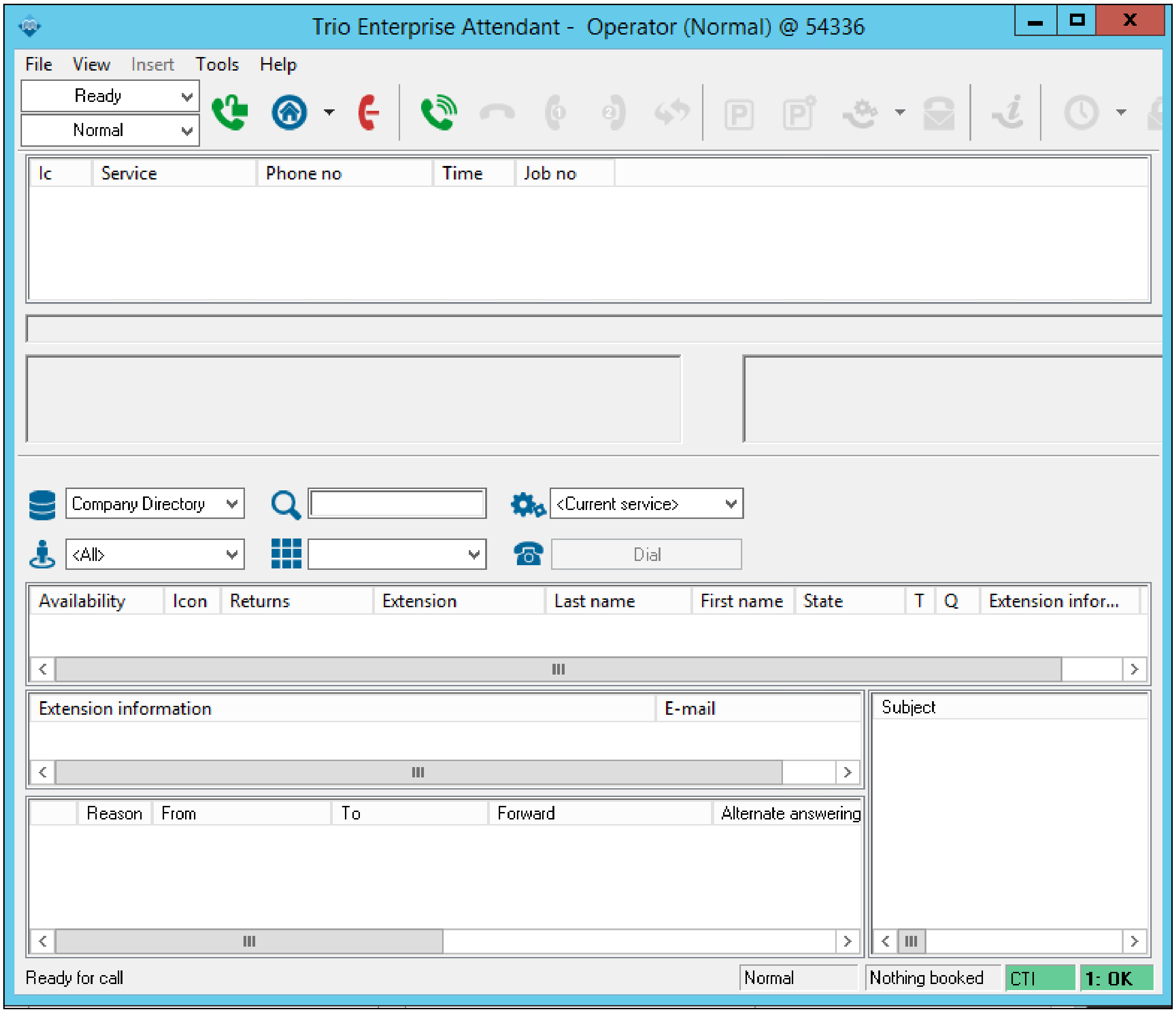
8.3. Configuring Trio Attendant

Trio Attendant is a separate application to Trio Enterprise server and can run concurrently on the same platform. The attendant uses a regular Communication Server 1000 telephone to make and receive calls, which are directed to the phone by Trio Enterprise server. The steps to configure Trio Attendant are to launch the ‘Agent Client’ shortcut.

The window below opens. Enter a valid **User ID** and **Password.** Note this user ID and password is created during the installation of TRIO Enterprise Server. For **Extension**, select the Communication Server 1000 telephone number that will be used as the agent’s audio device (number **54336** in this example). Ensure the correct Trio Enterprise server is selected if there is more than one on the network (default is the current Trio server). Confirm **Phone type** is set to **Standard phone**. Click on the **OK** button when finished.



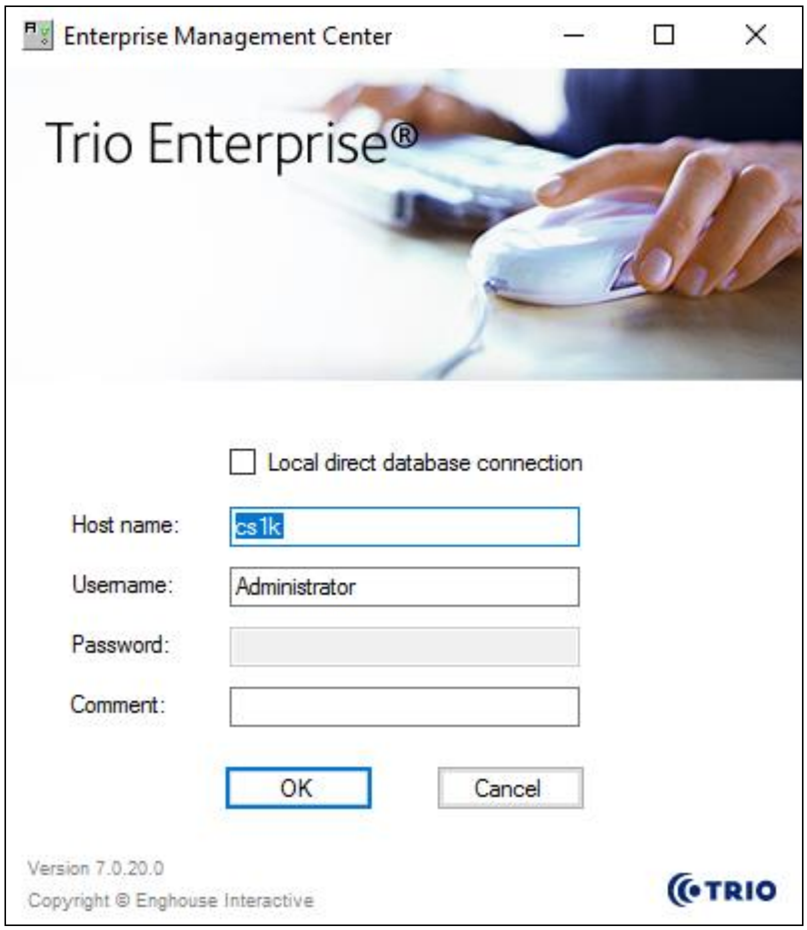
The Trio Agent window appears. Select **Ready** from the drop down box.



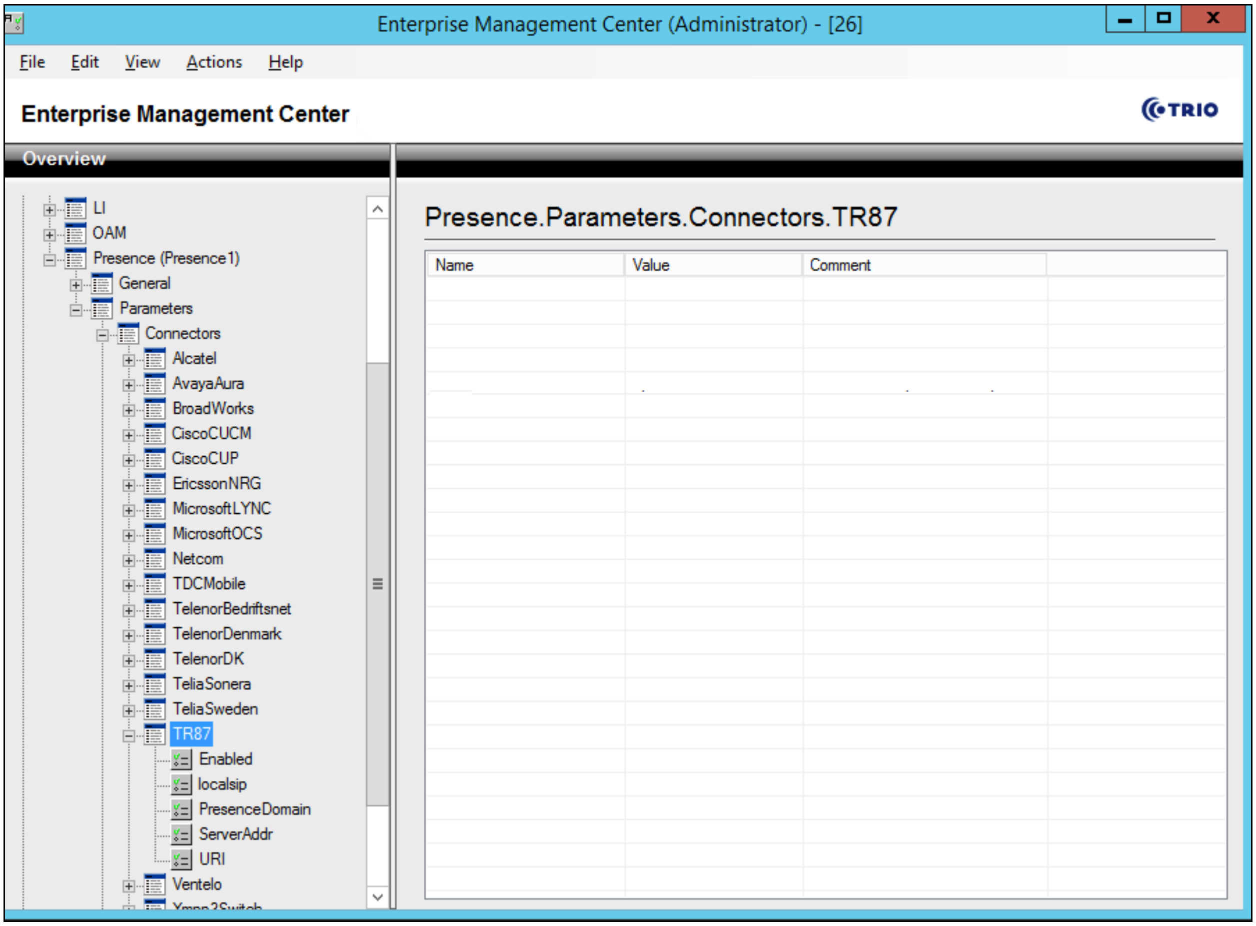
8.4. Configure Presence (TR87) on Trio

Launch the 'Enterprise Management Center' shortcut.

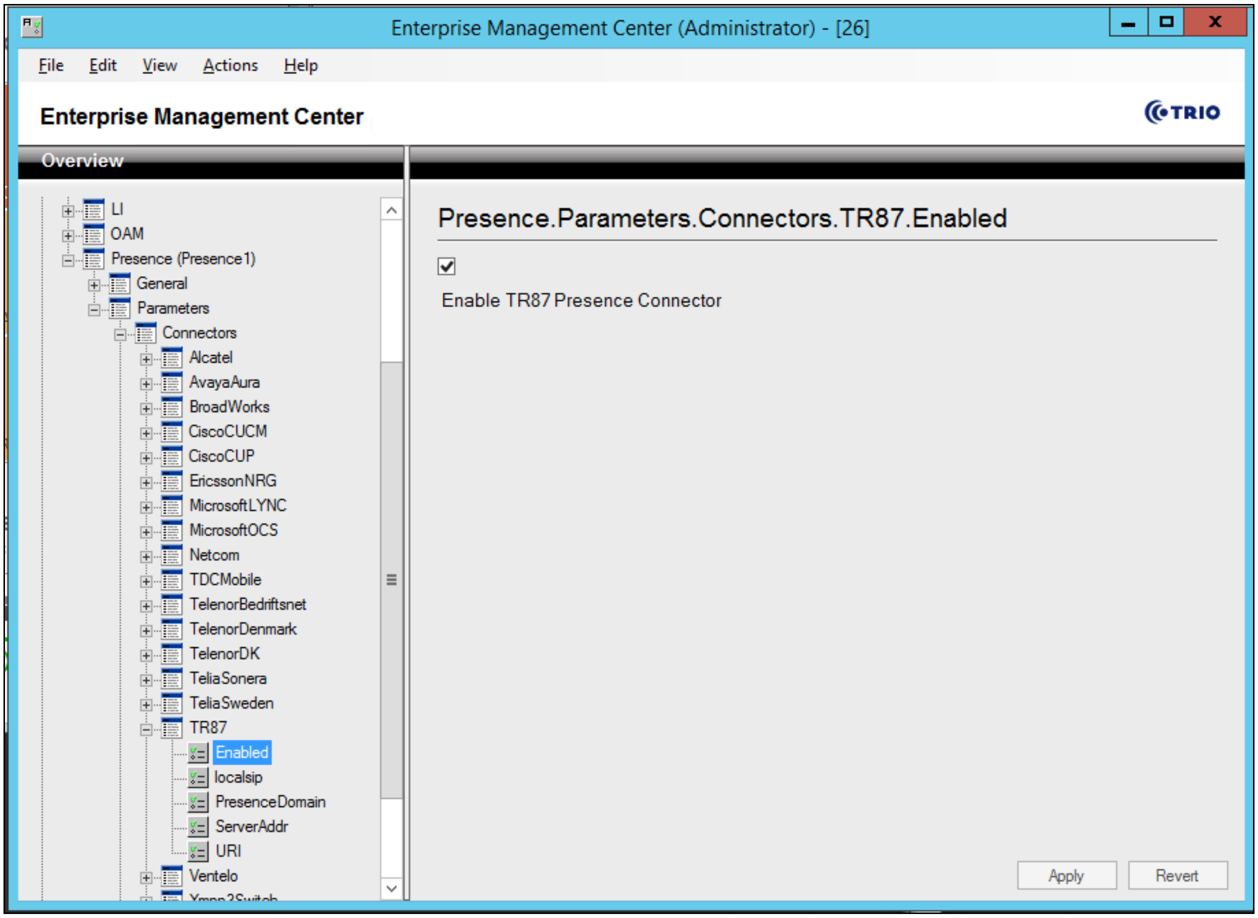
Enter the proper credentials and click on **OK**.



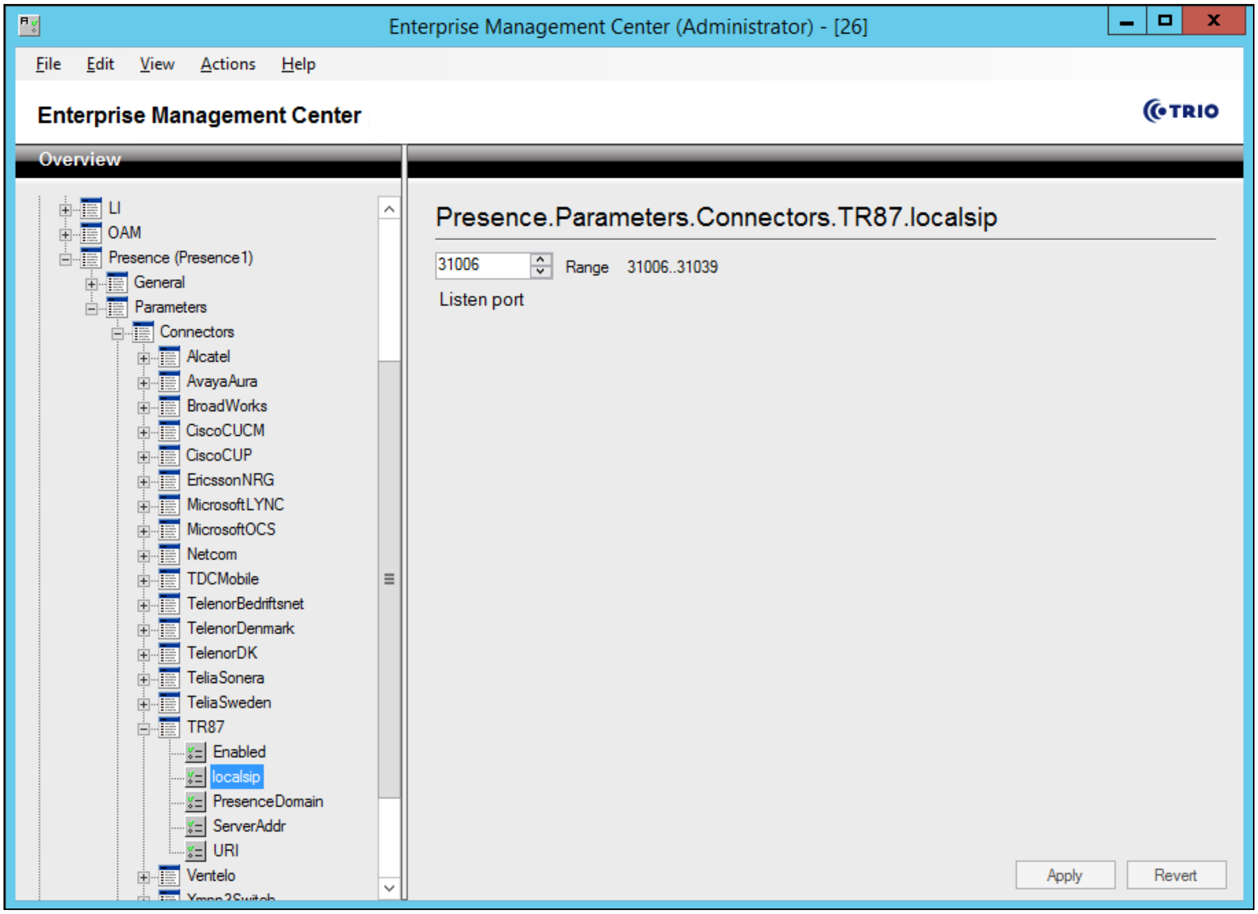
Click on **Parameters → Presence → Parameters → Connectors → TR87** in the left window.



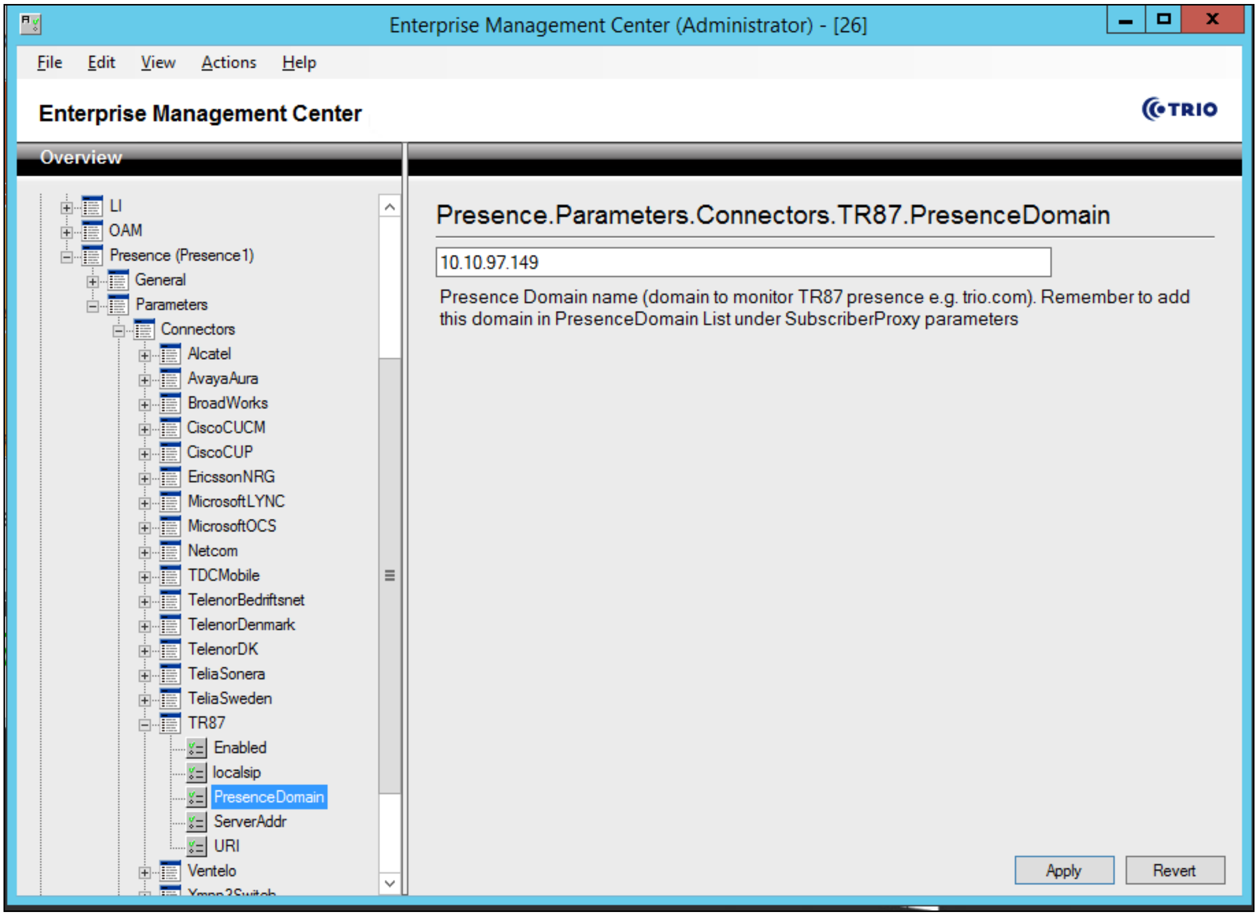
Under **TR87** select **Enabled** in the left window. Ensure that **Enable TR87 Presence Connector** is ticked as shown below. Click **Apply** to continue.



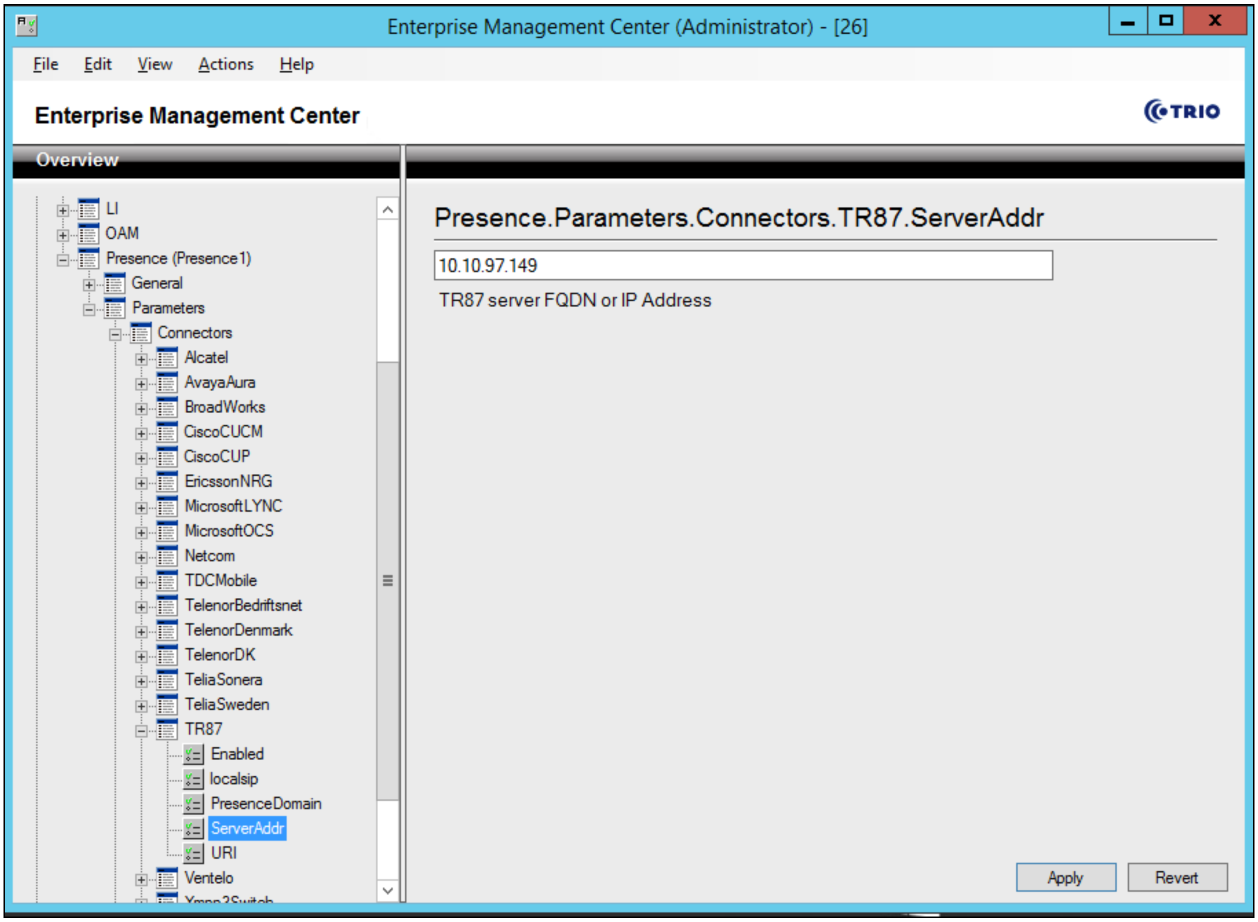
Select **localsip** under **TR87** in the left window and select the **Listen port** for TR87, for compliance testing this was left as default **31006** as shown below. Click **Apply** to continue.



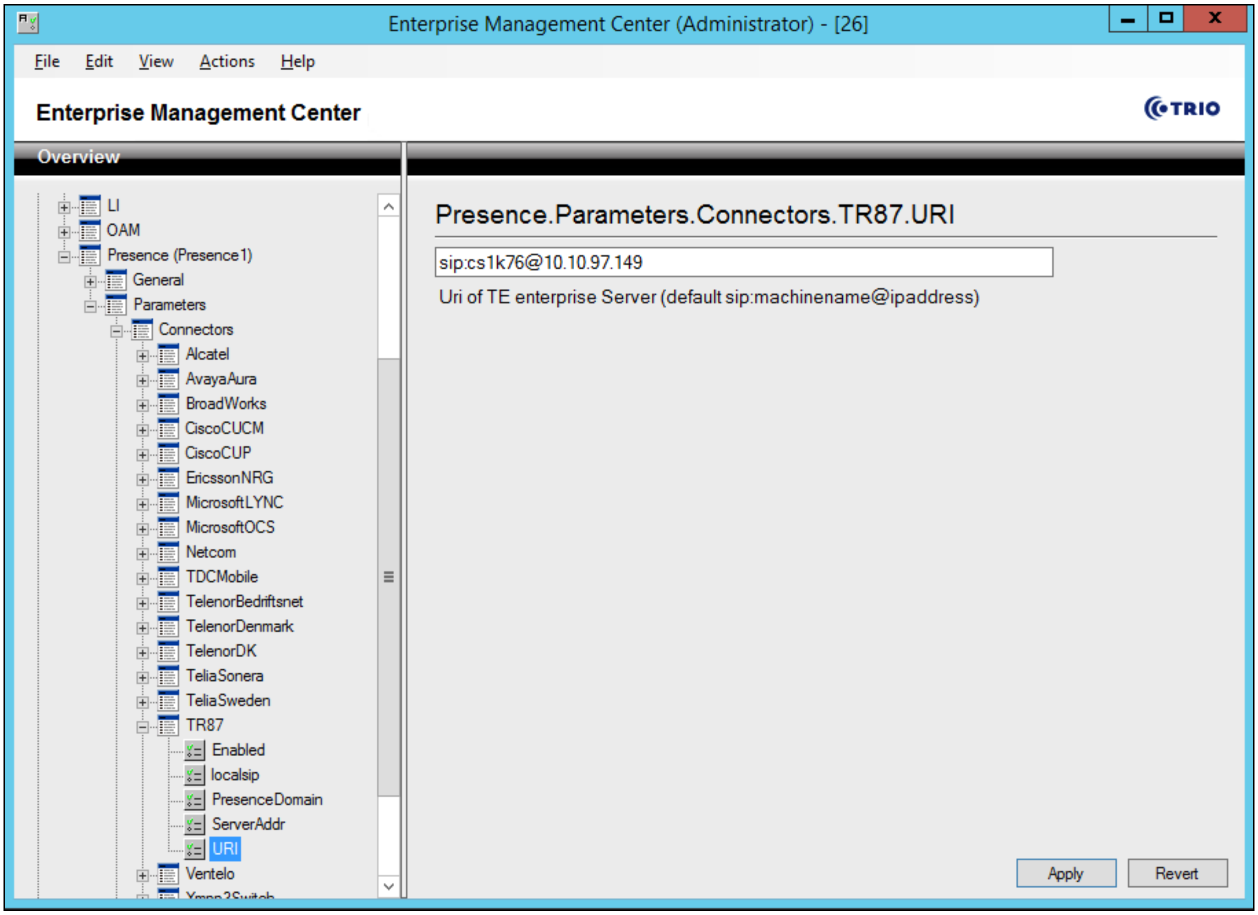
Select **PresenceDomain** under **TR87** in the left window. Enter the Node IP address of the Communication Server 1000 as noted in **Section 6**. Click **Apply** to continue.



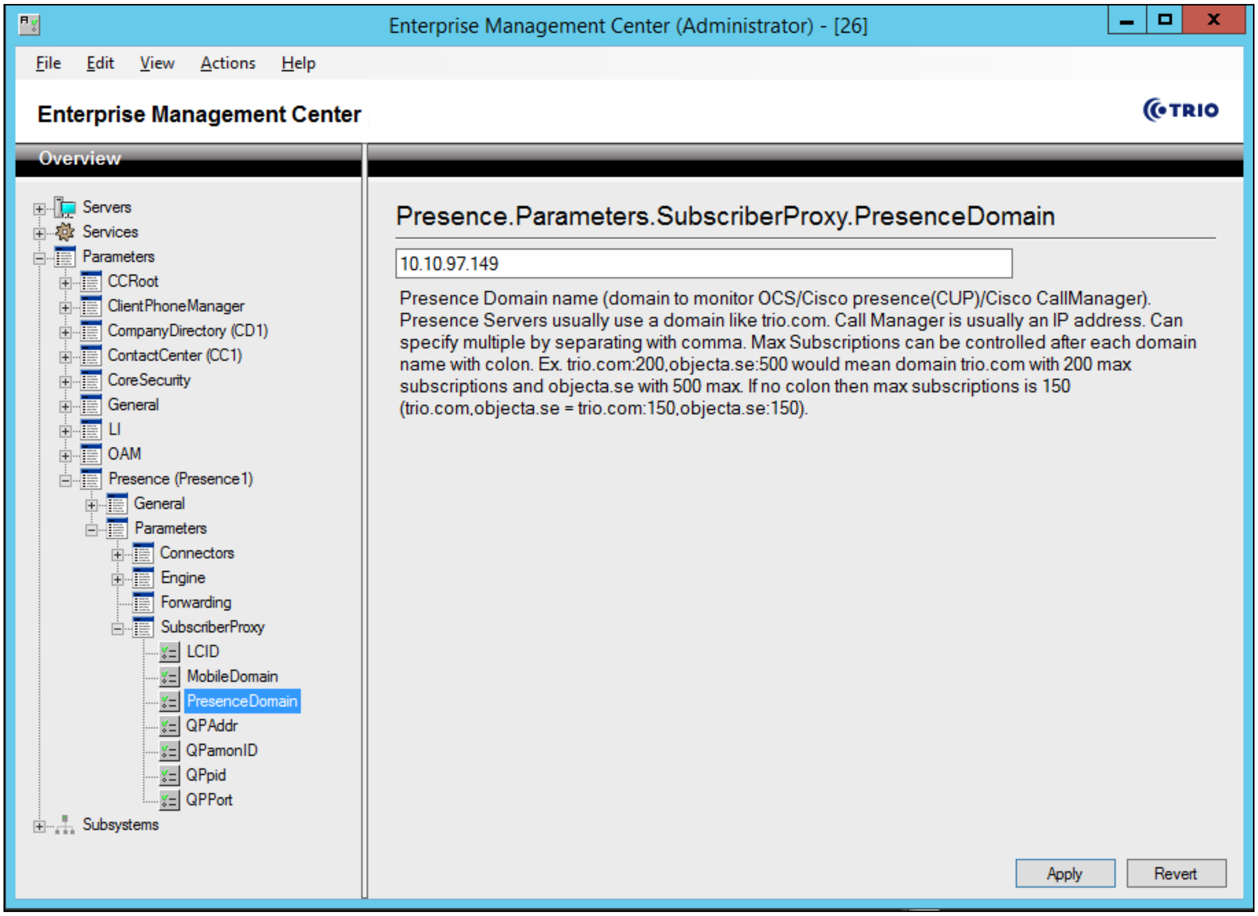
Select **ServerAddr** under **TR87** in the left window and again enter the Node IP address of the Communication Server 1000 as noted in **Section 6**. Click **Apply** to continue.



Select **URI** under **TR87** in the left window and enter the **machinename@ipaddress** preceded with **sip:** as shown below. Click Apply to continue.



Select **PresenceDomain** under **SubscribeProxy** in the left window. Enter the Node IP address of the Communication Server 1000 in the right window as noted in **Section 6**. Click **Apply** to continue.

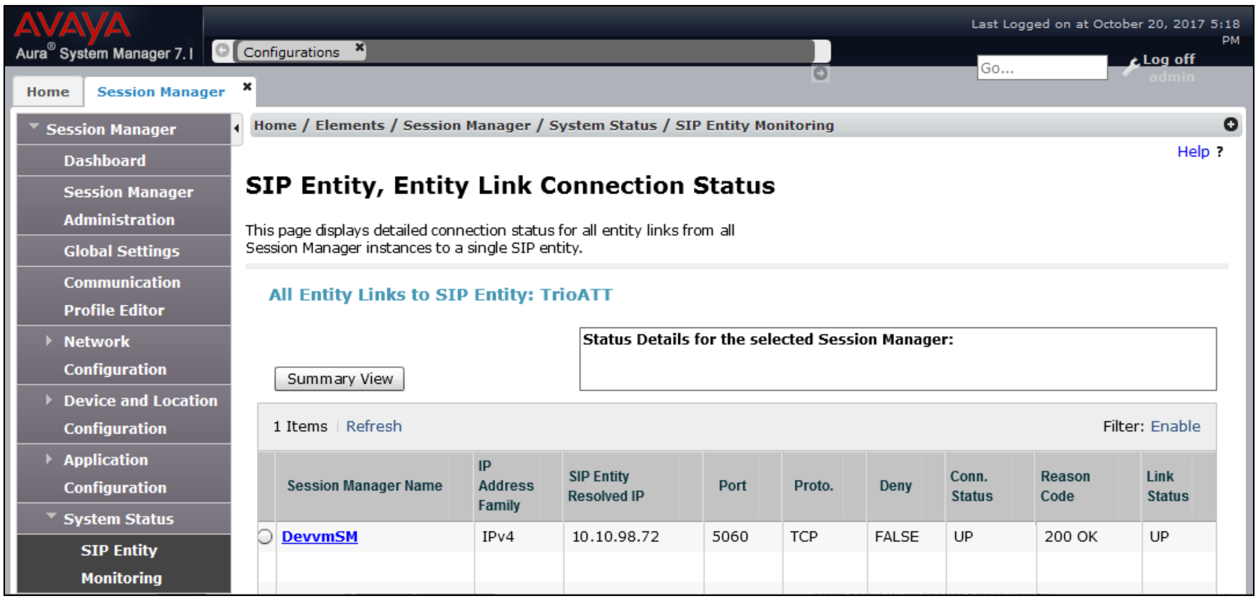


9. Verification Steps

This section provides the tests that can be performed to verify correct configuration of Communication Server 1000 and Session Manager with TRIO Enterprise.

9.1. Verify status of Trio SIP Entity

In System manager web page, to confirm a successful Trio SIP entity connection to Session Manager, click on **Element → Session Manager** and then select **System Status → SIP Entity Monitoring**, click on **TrioATT** entity to verify its status. The detail page shows the link from Trio to Session Manager via **TCP** is **UP**.



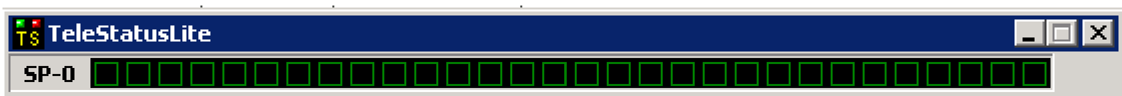
For feature testing the following were verified,

- Incoming internal and external calls
- Outgoing internal and external calls
- Supervised and unsupervised transfer with answer
- Directing calls to busy extensions
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions
- Status of the phones

9.2. SIP Channels on Trio Enterprise

To confirm a successful Trio Enterprise connection with the Session Manager, launch the 'Telestatus' shortcut.

A new window opens, showing the SIP trunk channel status as a series of green squares. Confirm the trunks are all in the idle state (unfilled green squares).



10. Conclusion

These Application Notes describe the configuration steps required for Trio Enterprise from Enghouse Interactive AB to successfully interoperate with Avaya Communication Server 1000 and Avaya Aura® Session Manager using SIP trunks. Trio Enterprise passed all compliance testing successfully; please see **Section 2.2** of these Application Notes for results and observations if any.

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.

Avaya:

1. *Communication Server 1000E Installation and Commissioning*, Release 7.6, NN43041-310
2. *Element Manager System Reference – Administration - Avaya Communication Server 1000*, Release 7.6, NN43001-632.
3. *Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals* Release 7.6, *NN43001*-509.
4. *Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals* -, Release 7.6, NN43001-116.
5. *Avaya Communication Server 1000 - Software Input Output Reference — Administration* Release 7.6, NN43001-611.
6. *Avaya Communication Server 1000 - ISDN Primary Rate Interface Installation and Commissioning*, Release 7.6, NN43001-301.
7. *Deploying Avaya Aura® Session Manager*, Release 7.1 Issue 1 May 2017.
8. *Administering Avaya Aura® Session Manager*, Release 7.1.1 Issue 2 August 2017.
9. *Deploying Avaya Aura® System Manager*, Release 7.1.1 Issue 3 August 2017.
10. *Administering Avaya Aura® System Manager for Release 7.1.1*, Release 7.1.1 Issue 7 October 2017.

All information on the product installation and configuration TRIO Enterprise Server can be found at http://www.trio.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.