Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Trio Enterprise R5.0 from Enghouse Interactive with Avaya Communication Server 1000 R7.6 and Avaya Aura® Session Manager R6.3 using a SIP 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 agent. Trio Enterprise is a software application from Enghouse Interactive 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 hardware.

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.

©2015 Avaya Inc. All Rights Reserved.

1. Introduction

These Application Notes describe the compliance tested configuration for Avaya Communication Server 1000 R7.6 (CS1000) and Avaya Aura® Session Manager R6.3 (Session Manager) with Trio Enterprise R5.0 from Enghouse Interactive. Trio Enterprise is a client/server based application running on Microsoft Windows 2008 Server operating systems.

Trio Enterprise provides users with an attendant client interface for Avaya Communication Server 1000 that does not require attendant telephony hardware e.g., Avaya 2250 attendant console. The Trio attendant client provides a view of contacts, schedules, and communication tasks and was installed on the same server as the Trio Server, but can be installed on a separate platform if required.

Trio Enterprise connects to the Avaya Communication Server 1000 using a SIP connection via Session Manager.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using an CS1000. The Trio Enterprise server uses a SIP connection to the CS1000 via Session Manager. See **Figure 1** for a network diagram. A basic Distance Steering Code (DSC) 71xxx (Trio attendant queue number) was configured on the CS1000 to route call to the Session Manager. Then Dial Pattern 71xxx configured on System Manager to route call to Trio Enterprise Server. If a call is made from the Trio Enterprise attendant client to the PSTN, the call will be routed from the Trio console via a SIP trunk to Session Manager then to the PSTN. During compliance testing a SIP trunk connecting to simulate PSTNwere used. Trio attendant client can perform the usual range of attendant call functions, i.e., centralized answering agent; extend PSTN calls to users, place PSTN calls on behalf of internal users, perform internal telephone directory lookups.

During tests, using simulated PSTN phone place a call to Trio attendant queue number, Session Manager routes the call destined for Trio Enterprise server over a SIP connection. The Trio Enterprise server then automatically places this call to the telephone, which 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.

A variety of Avaya telephones were installed and configured on the CS1000 and assigned as answering phone for internal user on Trio Enterprise server. For compliance testing an Avaya 3904 Digital phone was used as the attendant's deskphone. When a call is made to Trio attendant queue number, Trio Enterprise server calls the 3904 phone; once attendant pickup the phone to answer, Trio server bridges the call between originator and 3904 phone.

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

©2015 Avaya Inc. All Rights Reserved.

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.

©2015 Avaya Inc. All Rights Reserved.

2.1. Interoperability Compliance Testing

The compatibility tests included the following.

- Attendant answers direct calls
- Supervised and unsupervised transfer with answer
- Directing calls to busy extensions
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions

2.2. Test Results

Tests were performed to insure full interoperability between Trio Enterprise and CS1000. The tests were all functional in nature and performance testing was not included. All the test cases passed successfully. With the following observation:

Anonymous calls can result on media path going through CS1000 (instead of directly between the 2 endpoints):

- A call is made from the PSTN and is answered by the Trio Enterprise attendant.
- The attendant transfers the call to an internal station on the CS1000 by doing a SIP Refer.
- When the internal station answers the call, the direct path request is declined with reason 603, and the CS1000 stays in the media path.
- Both endpoints can hear each other.

For non-anonymous call, the same scenario applied, when the internal station answers the call, the direct path request is accepted and media goes directly between the endpoints.

2.3. Support

For technical support on Trio products, please use the following web link.
http://www.trio.com/web/Support.aspx

Enghouse Interactive can also be contacted as follows.

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

©2015 Avaya Inc. All Rights Reserved.

3. Reference Configuration

Figure 1 shows the network topology during compliance testing. Trio Enterprise is connected to CS1000 using a SIP connection via Session Manager. The Trio Enterprise Server is configured as a SIP Endpoint. System Manager is used to configure Session Manager.

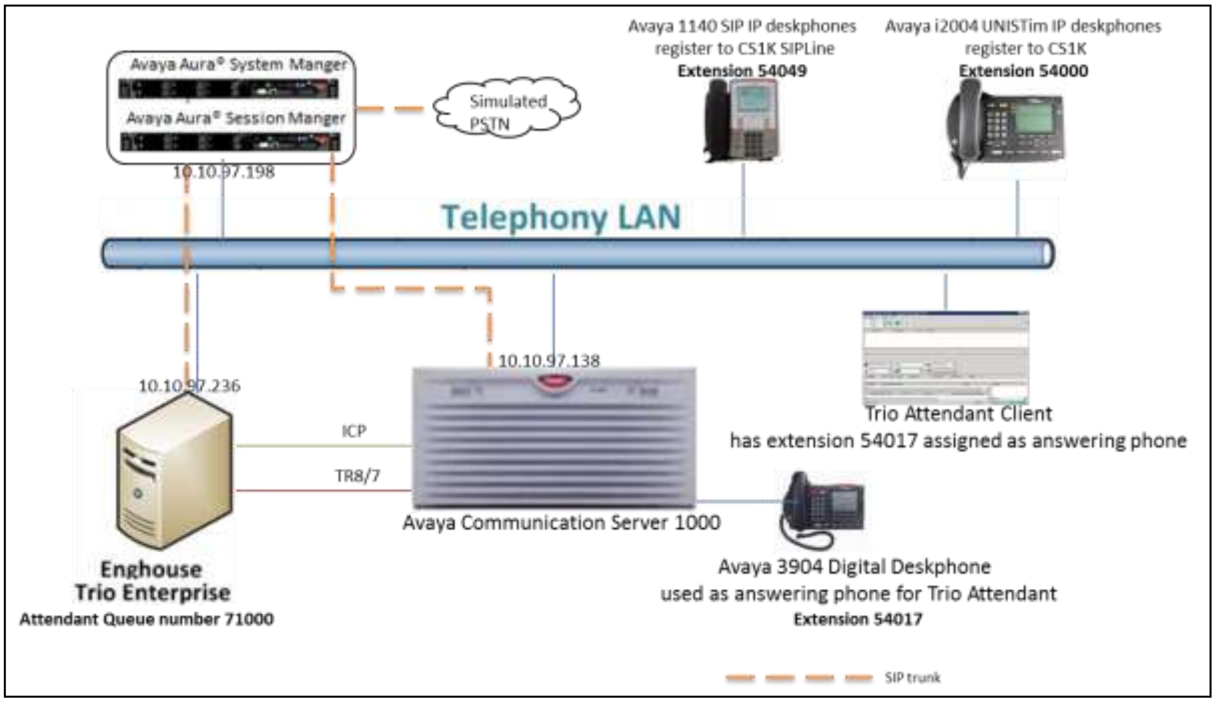


Figure 1: Configuration for Avaya Communication Server 1000 R7.6, Avaya Aura® Session Manager R6.3 and Trio Enterprise R5.0

©2015 Avaya Inc. All Rights Reserved.

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 on CPPM	R7.6 SP5
Avaya Aura® System Manager	System Manager 6.3.0 – SP 10
Avaya Aura® Session Manager	Session Manager R6.3 – SP 10
Avaya 3904 Digital Deskphone	Core V2.4 Flash V9.4
Trio Enterprise Running on a Windows 2008 R2 64-bit server	Version 5.0

©2015 Avaya Inc. All Rights Reserved.

5. Configure Avaya Communication Server 1000

The configuration operations illustrated in this section were performed using terminal access to the CS1000 over an "SSH" session using "PUTTY". The information provided in this section describes the configuration of the CS1000 for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 11**.

Note: SIP connection from CS1000 to Session Manager is in place and operational. During compliance test, route number (**ROUT**) and route list index (**RLI**) is **1** to Session Manager, this information is needed in Section **5.2** to configure route to Trio number 71xxx.

Note: The configuration of the simulated PSTN connections are outside the scope of these Application Notes.

Note: Not all prompts need an answer. The prompts outlined below are mandatory for a basic configuration. Accept the default responses for all other prompts by pressing the return key.

©2015 Avaya Inc. All Rights Reserved.

5.1. Verify Licences

Both SIP CTI Licences and AST licenses are required to allow Trio observe TR87 events. To ensure the CS1000 is licensed for SIP CTI use **LD 22** and type **SLT** at the **REQ** prompt. Check for **SIP CTI TR87** and **AST** (in bold below).

```
>ld 22

REQ  slt

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

IPMGs Registered:               2
IPMGs Unregistered:             0
IPMGs Configured/unregistered:  0
TRADITIONAL TELEPHONES 32767    LEFT 32762    USED     5
DECT USERS             32767    LEFT 32767    USED     0
IP USERS               32767    LEFT 32617    USED   150
BASIC IP USERS         32767    LEFT 32766    USED     1
TEMPORARY IP USERS     32767    LEFT 32767    USED     0
DECT VISITOR USER      10000    LEFT 10000    USED     0
ACD AGENTS             32767    LEFT 32743    USED    24
MOBILE EXTENSIONS      32767    LEFT 32767    USED     0
TELEPHONY SERVICES     32767    LEFT 32767    USED     0
CONVERGED MOBILE USERS 32767    LEFT 32767    USED     0
AVAYA SIP LINES        32767    LEFT 32730    USED    37
THIRD PARTY SIP LINES  32767    LEFT 32717    USED    50

PCA                    32767    LEFT 32757    USED    10
ITG ISDN TRUNKS        32767    LEFT 32767    USED     0
H.323 ACCESS PORTS     32767    LEFT 32671    USED    96
AST                    32767    LEFT 32737    USED    30
SIP CONVERGED DESKTOPS 32767    LEFT 32761    USED     6
SIP CTI TR87           32767    LEFT 32757    USED    10
SIP ACCESS PORTS       32767    LEFT 32725    USED    42
RAN CON                32767    LEFT 32767    USED     0
MUS CON                32767    LEFT 32766    USED     1    ACTIVE     0

IP RAN CON             16384    LEFT 16384    USED     0
IP MUS CON             16896    LEFT 16896    USED     0    ACTIVE     0
IP MEDIA SESSIONS      35842    LEFT 35842    USED     0    ACTIVE     0
TNS                    32767    LEFT 32097    USED   670
ACDN                   24000    LEFT 23984    USED    16
AML                       16    LEFT     7    USED     9
IDLE_SET_DISPLAY CS1K_Middle 7.65
LTID                   32760    LEFT 32760    USED     0
RAN RTE                  512    LEFT   512    USED     0
ATTENDANT CONSOLES     32767    LEFT 32767    USED     0
IP ATTENDANT CONSOLES  32767    LEFT 32767    USED     0
BRI DSL                10000    LEFT 10000    USED     0
MPH DSL                  100    LEFT   100    USED     0
DATA PORTS             32767    LEFT 32767    USED     0
PHANTOM PORTS          32767    LEFT 32767    USED     0
TRADITIONAL TRUNKS     32767    LEFT 32638    USED   129
ELC ACCESS PORTS       32767    LEFT 32767    USED     0
DCH                      255    LEFT   245    USED    10
```


©2015 Avaya Inc. All Rights Reserved.

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. For each extension, a CDP entry needs to be created. In the example below, the **DSC** is **71000**, **FLEN** is **5** and the **RLI** is **1**.

Prompt	Response	Description
>	**LD 87**	Enter Overlay 87
REQ	**NEW**	Create new
CUST	0	Customer Number as defined in LD15
FEAT	**CDP**	Coordinated dialing plan
TYPE	DSC	Distance Steering code
DSC	**71000**	Distant Steering code
FLEN	**5**	**Flexible Length number of digits**
RLI	**1**	**Route list index Number**

5.3. Configure TR87 on CS1000

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 **TR87A** 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.

Prompt	Response	Description
>	**LD 20**	Enter Overlay 20
REQ	CHG	Change
TYPE	1140	Change phoneset type 1140
TN	L S C U	Terminal Number Loop Shelf Card Unit
CLS	**TR87A**	Change TR87 to “Allowed”
AST	**00**	Set AST for key 00
IAPG	**1**	Set CTI messaging to “Yes”

©2015 Avaya Inc. All Rights Reserved.

5.4. Configure Intercept Computer Update on CS1000

Trio uses Intercept Computer Update (ICP) on the CS1000 to change the presence state of the phoneset. A physical port on the CS1000 must be configured for ICP along with the ICP configuration in the Customer Data Block.

5.4.1. Configuration of ICP Port

Enter overlay 17 to add a new terminal to connect to the Trio for ICP use. Follow the instructions below to configure a physical connection on port 2 connected to MGC card 4 0. Type **LD 17** at the > prompt to enter overlay 17.

Prompt	**Response**	**Description**
>	**LD 17**	Enter Overlay 17
REQ	**CHG**	Change
TYPE	**ADAN**	Change the Action Device and Number
ADAN	New TTY x	New tty port x
CTYP	MGC	Nedia Gateway Controller
IPMG	4 0	Loop and Shelf
DNUM	15	Data number
PORT	0	Port number
DES	ICP2	Description
BPS	1200	Bits per Second
BITL	7	Bit Length
STOP	1	Stop bit
PARY	EVEN	Parity
FLOW	NO	Flow
USER	ICP	User type is set to ICP

5.4.2. Configuration of ICP in the Customer Data Block

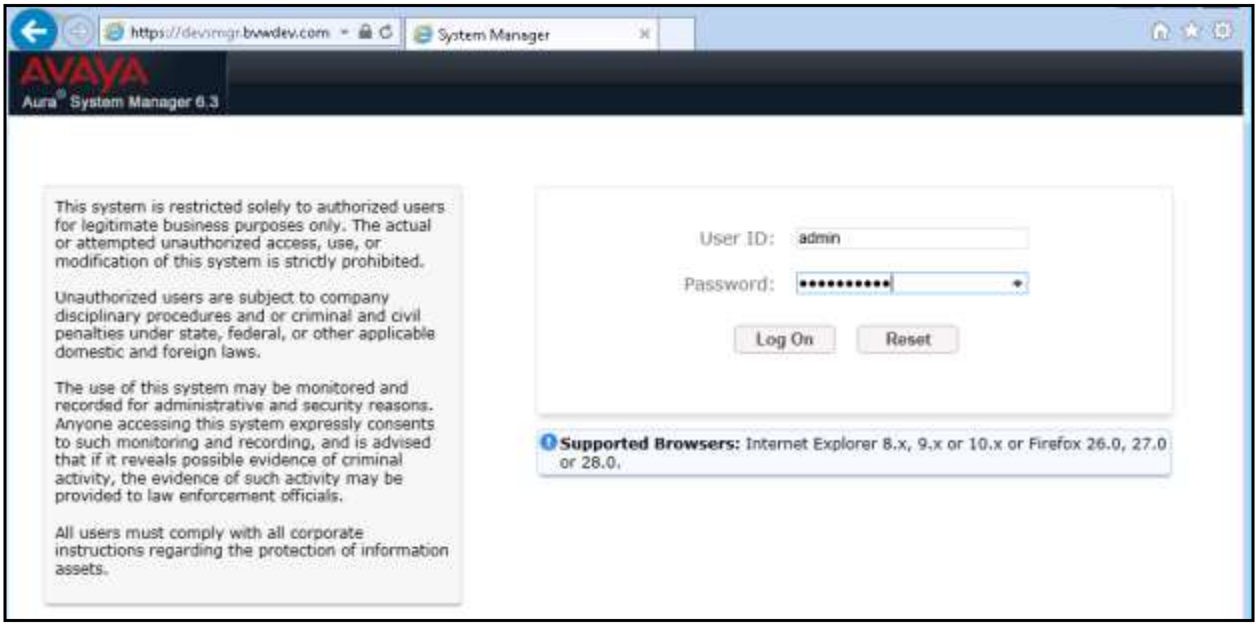
Enter Overlay 15 to change the Intercept Computer Update (ICP) data block by typing **LD 15** at the > prompt and follow the instructions as shown below to configure ICP for Trio.

Prompt	**Response**	**Description**
>	**LD 15**	Enter Overlay 15
REQ	**CHG**	Change
TYPE	**icp**	Change the Intercept Computer Update
CUST	0	Customer Number
APL	15	Auxiliary Processor Link used
NIPN	9	Number of Intercept positions
ICCR	NO	Intercept Position Cancelling Reply
ICDN	71003	Trio DN diversion call for absence
ECDN	71003	Trio DN diversion call for absence
CPS	CIR	
ICDL	5	CP DN Length
ICPD	0	ICP Padding Digit
ICTD	YES	Intercept Terminal Dail from Directory

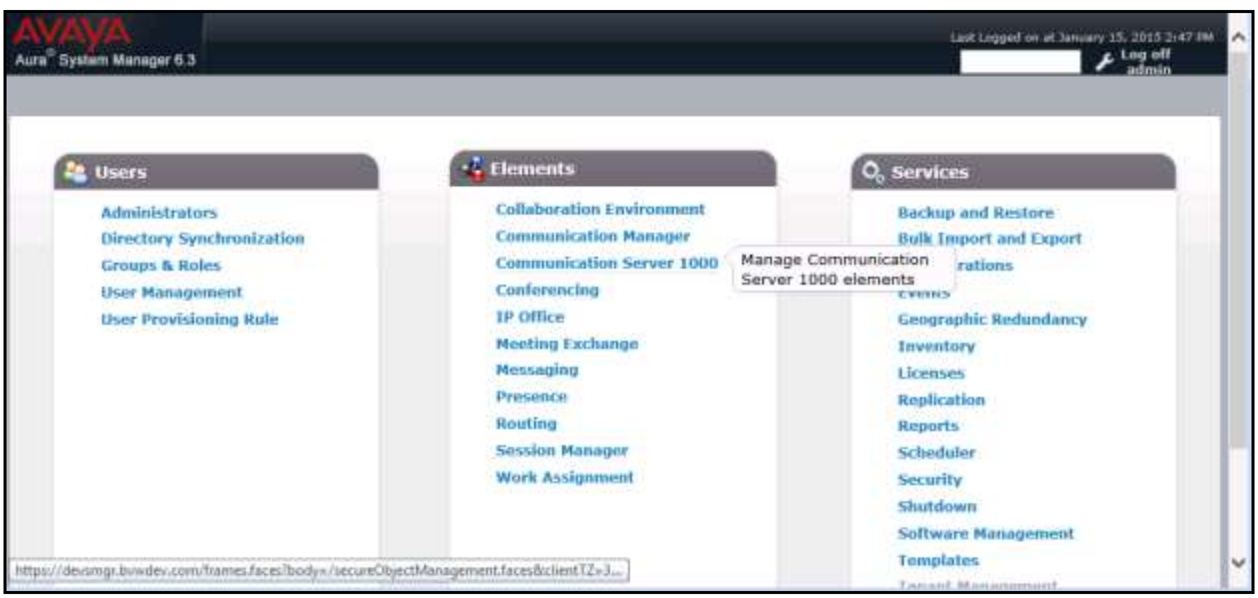
©2015 Avaya Inc. All Rights Reserved.

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

SIP CTI (TR/87) services must be enabled and configured on the CS1000 IP Telephony Node to allow applications obtain presence information or invoke a make-call operation. Changes on the CS1000 Node are performed using Element Manager which is only accessible through the System Manager. To make changes in Element Manager, log in to System Manager using the URL **https://<fqdn>/SMGR** or **https://<ip-address>/SMGR**. Log in with the appropriate credentials and click **Log On** highlighted below.

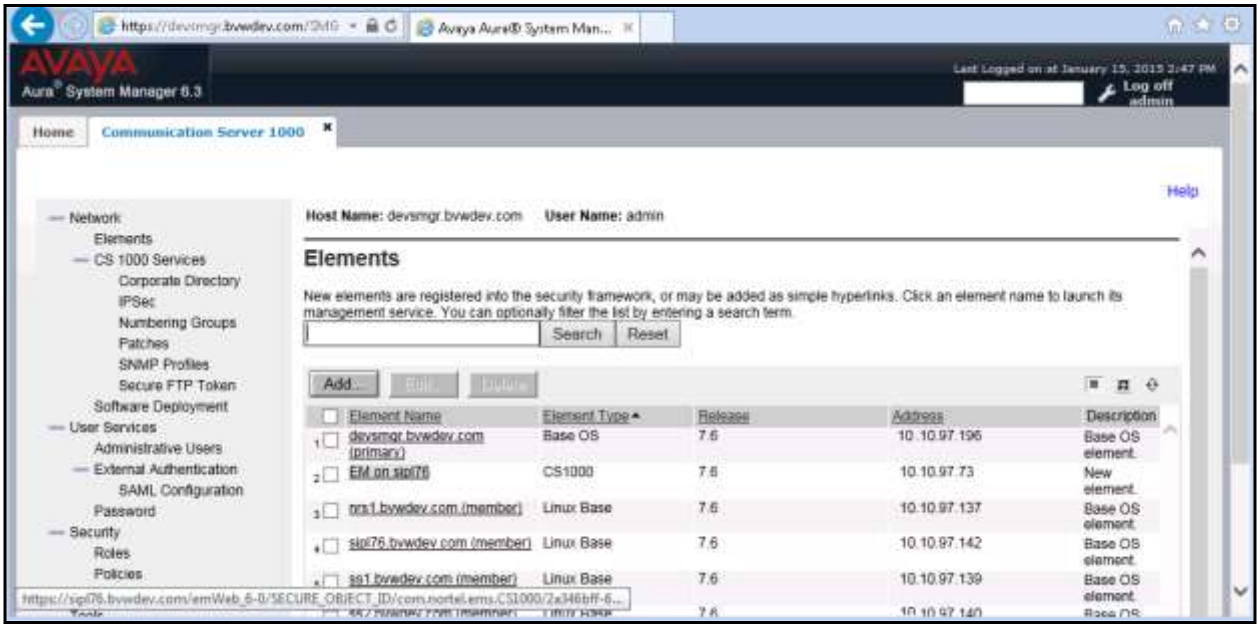


Click on **Communication Server 1000** as shown.

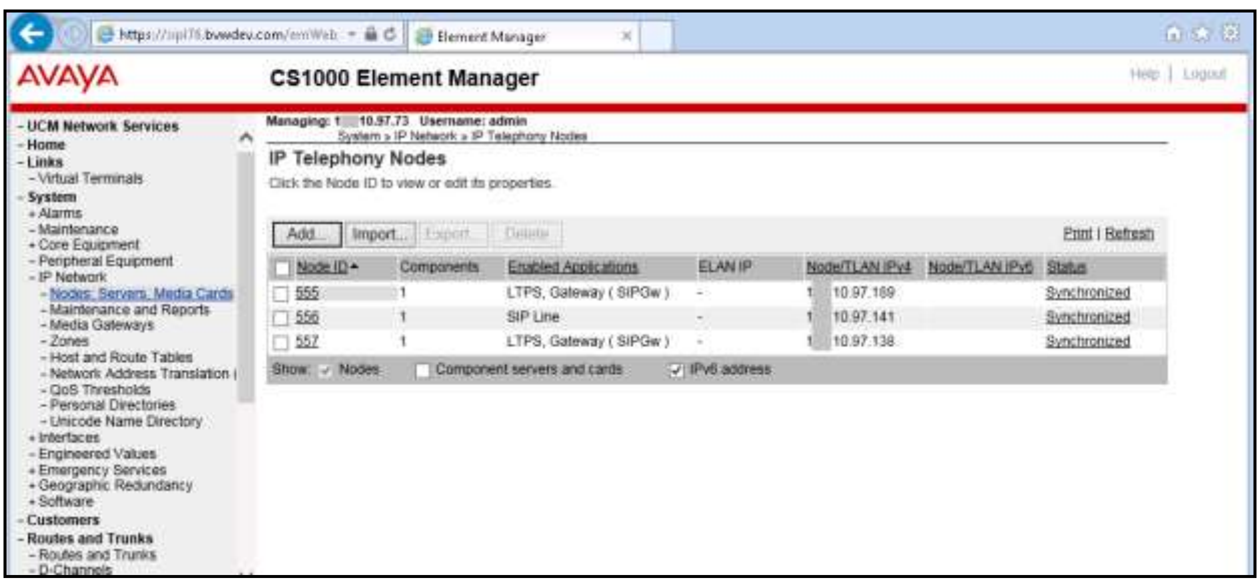


©2015 Avaya Inc. All Rights Reserved.

Once **Communication Server 1000** is selected the following screen appears, click on the Element Manager link, in this case click on **EM on sipl76** link.

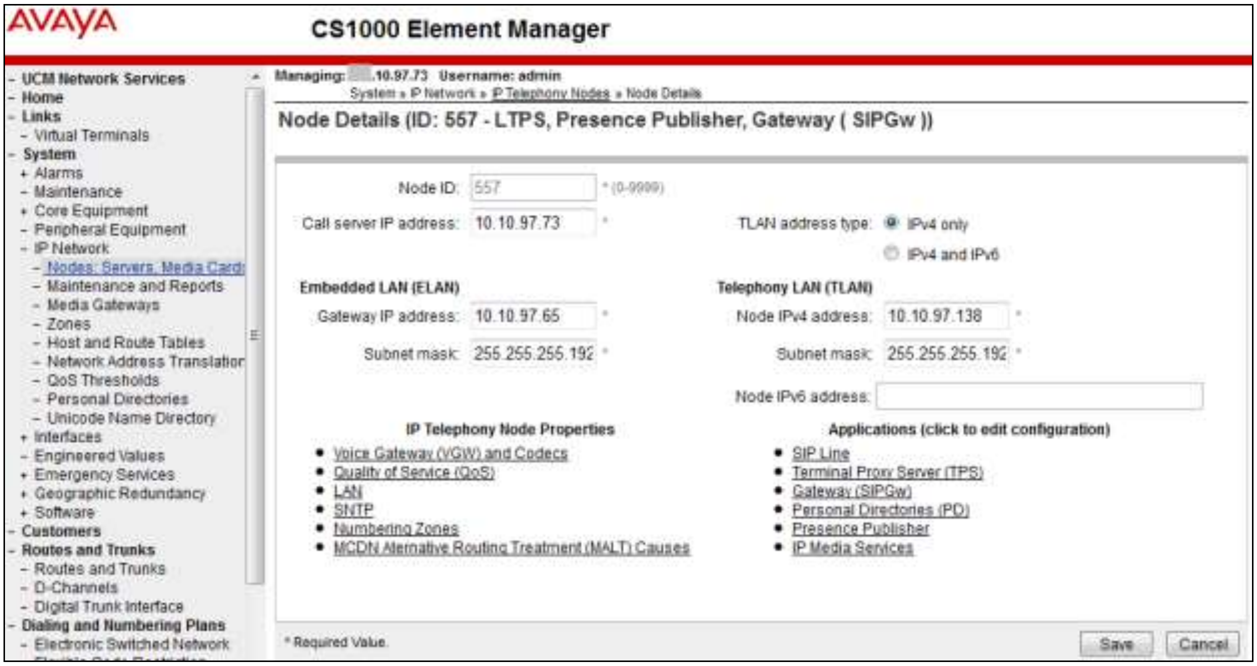


Click on **IP Network → Nodes: Servers, Media Cards** in the left window. Click on the **Node ID** displayed in the right window, during compliance test Node **557** is configured to connect to Session Manager. Note the IP address of this node as it used CS1000 as SIP Entity endpoint on Session Manager. Trio Enterprise also gets TR87 events via Node **557** note this address as it will be used in Section Error! Reference source not found..

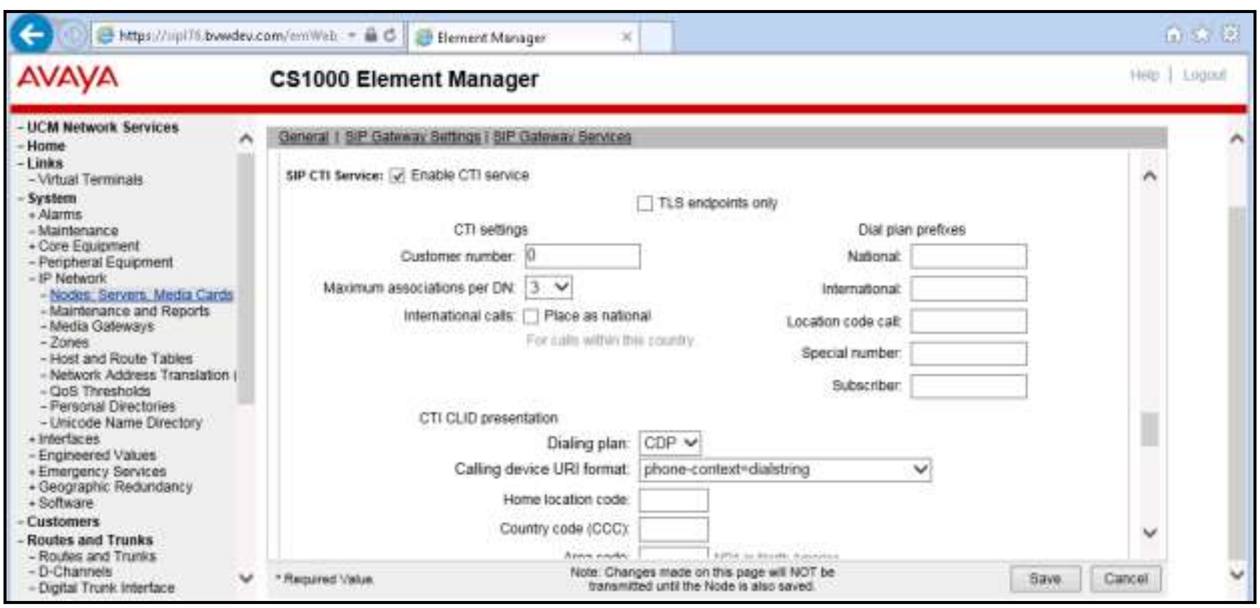


©2015 Avaya Inc. All Rights Reserved.

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

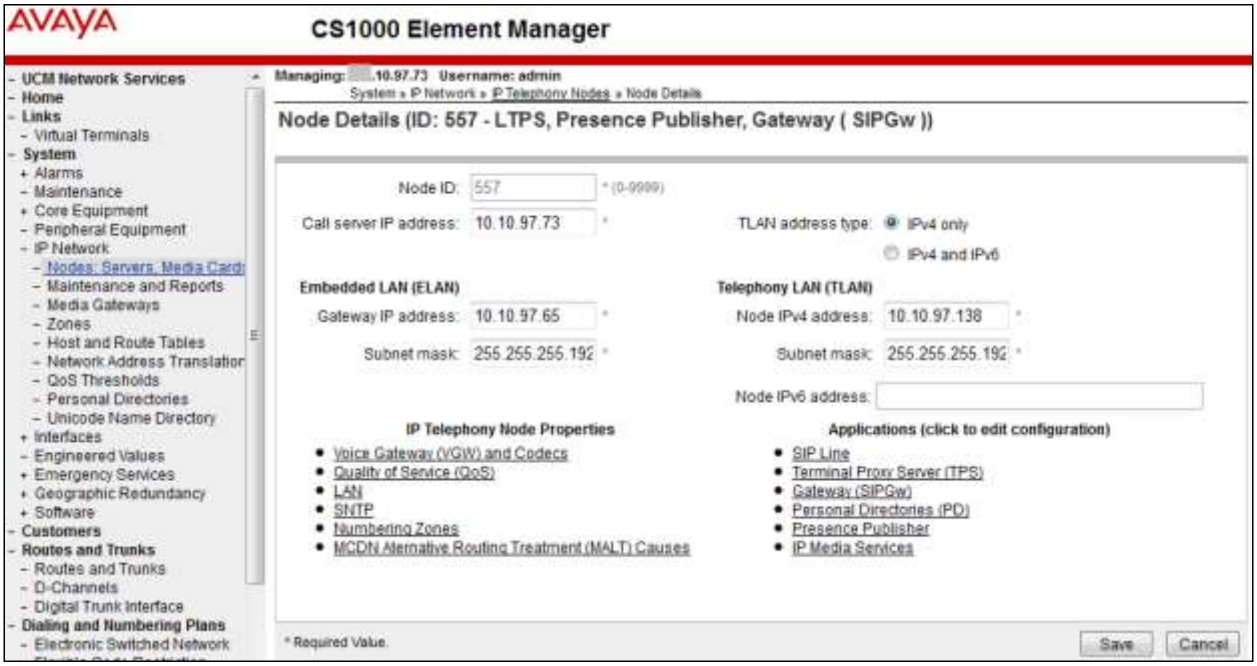


Ensure that **SIP CTI Service** is ticked as shown below and un-tick the **TLS endpoints only** (if this is ticked); everything else can be left as default. Click on **Save** once finished.

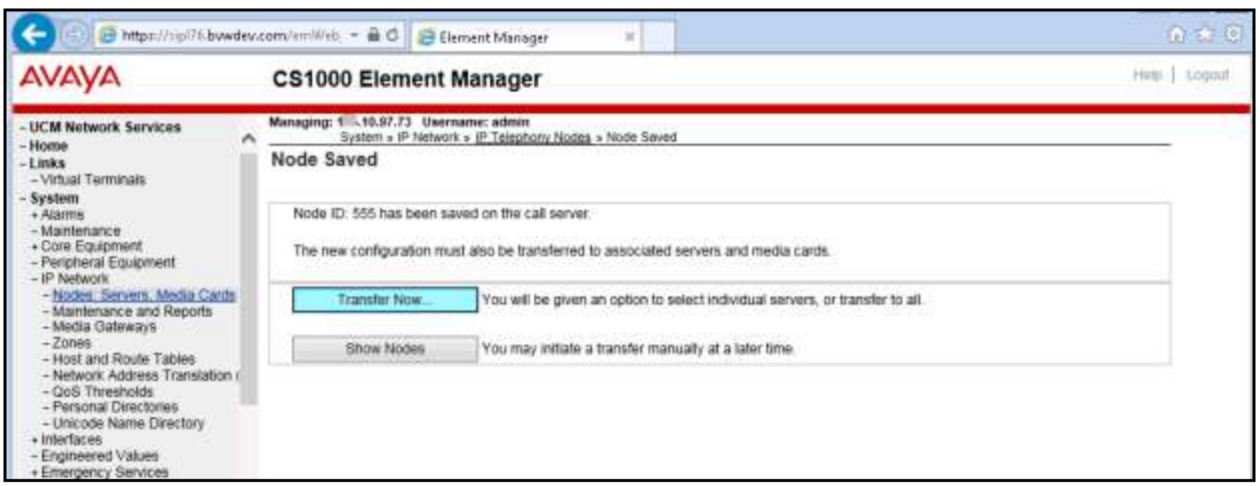


©2015 Avaya Inc. All Rights Reserved.

Once **Save** is clicked the following screen appears. Click on **Save** as shown below.

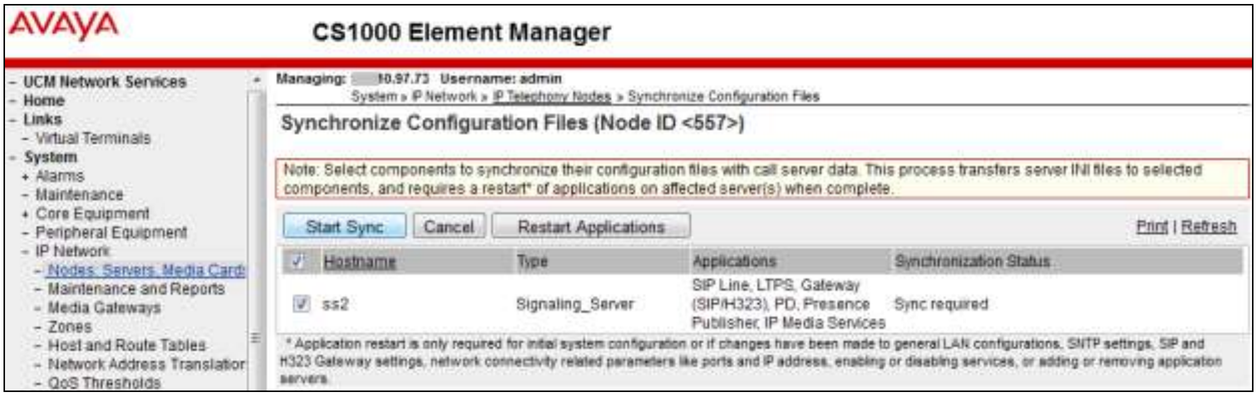


Once Save is clicked the following screen appears. Select **Transfer Now** as highlighted below.

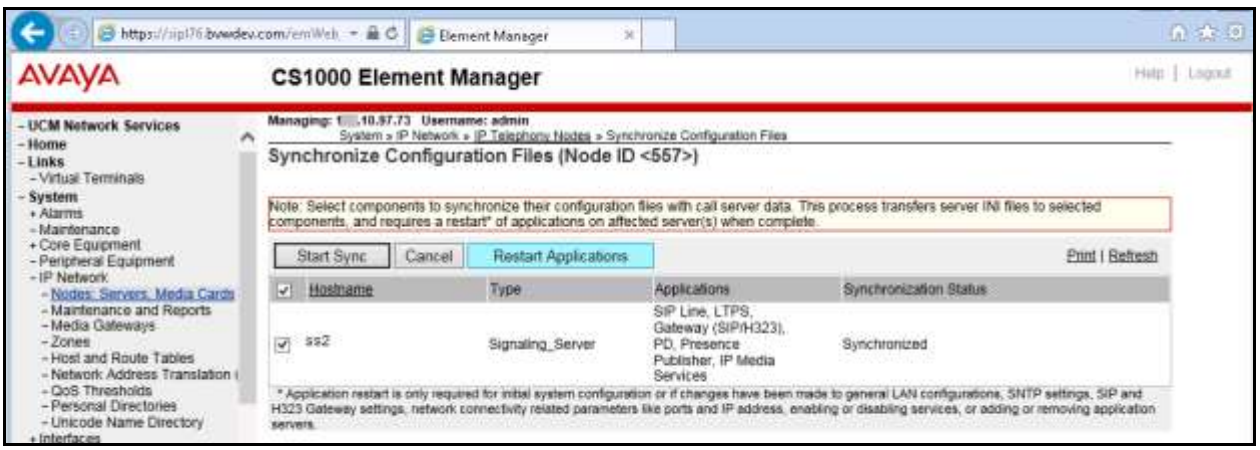


©2015 Avaya Inc. All Rights Reserved.

Once the information is transferred over then the components need to synchronize their configuration files with the CS1000 call server. Select the **Hostname** as shown below and click on **Start Sync**.



Once the components are synchronized the application will require a restart, select the **Hostname** and click on **Restart Applications** as highlighted below.



©2015 Avaya Inc. All Rights Reserved.

7. Creating SIP Entities on the Avaya Aura® Session Manager

It is assumed that Session Manager and CS1000 were configured and operational. During compliance test the system location is **Belleville** and domain name is **bvwdev.com**. To add the Trio Enterprise Server as a SIP Entity on the Session Manager existing system, the following must be configured.

- Create Adaptation
- Create a SIP Entity
- Create an Entity Link
- Create a Routing Policy
- Create a Dial Pattern

©2015 Avaya Inc. All Rights Reserved.

Configuration of Session Manager is achieved by accessing the browser-based GUI of System Manager, using the URL **https://<fqdn>/SMGR** or **https://<ip-address>/SMGR**. Log in with the appropriate credentials and click **Log On** as shown in Section **6**. Once logged in select the **Routing** link under the **Elements** column (not shown).

7.1. Create Adaptation

During compliance test, in order to make the call from and to Communication Manager used as simulated PSTN via Session Manager, Adaptation to translate IP address into domain name is used for Trio SIP entity. 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).

Enter the following for the Trio Adaptation.

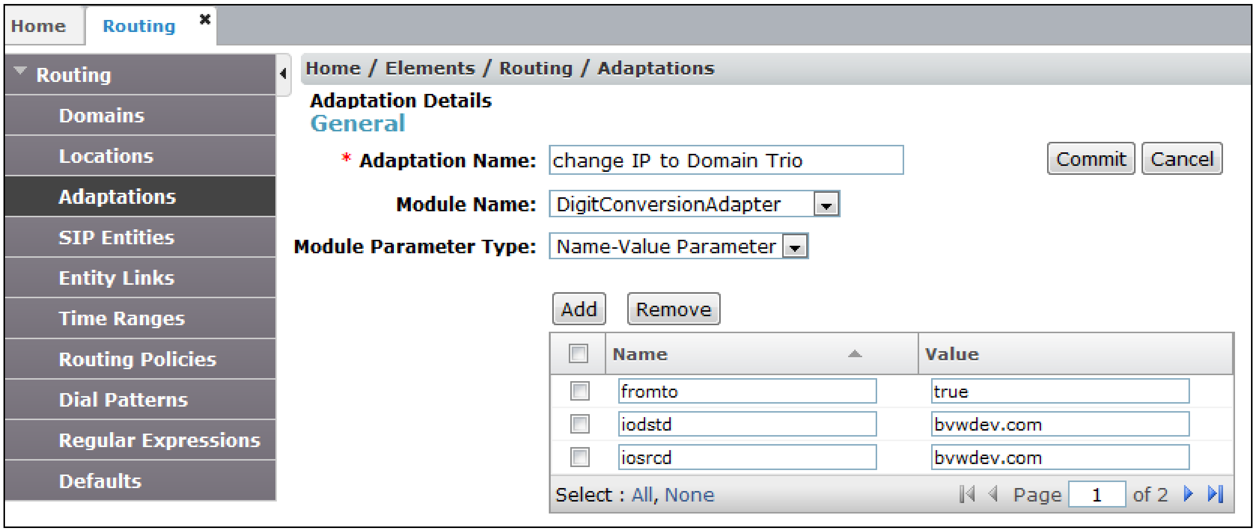
- **Adaptation Name** An informative name (e.g., **change IP to Domain Trio**)
- **Module Name** Select **DigitConversionAdapter**
- **Module Parameter Type** Select Name-Value Parameter

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

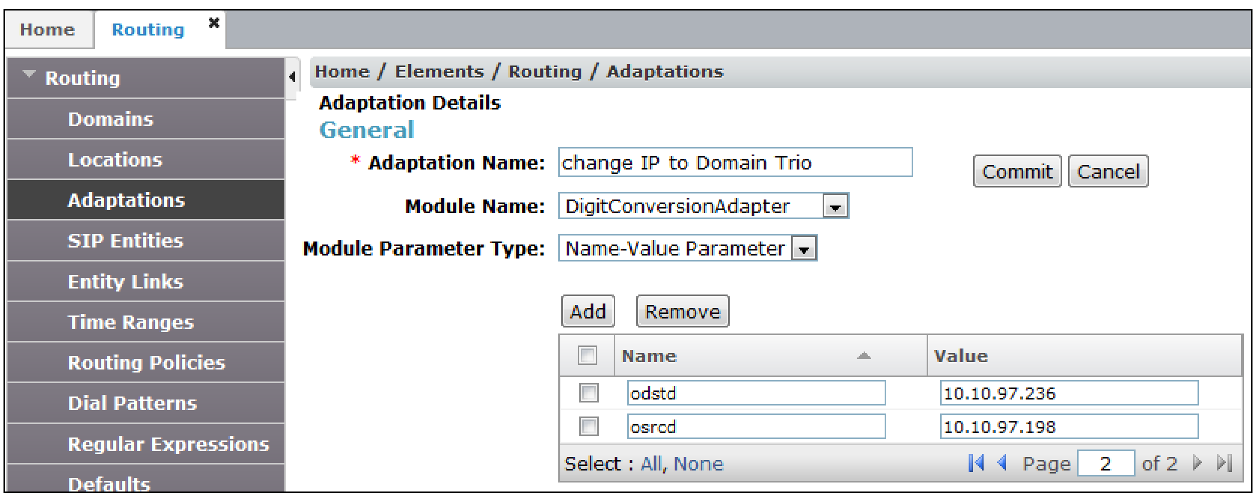
Name	**Value**
fromto	true
iodstd	Enter the domain name of system, ex: **bvwdev.com**
iosrcd	Enter the domain name of system, ex: **bvwdev.com**
odstd	Enter IP address of Trio, ex: **10.10.97.236**
osrcd	Enter IP Address of Session Manager, ex: **10.10.97.198**

©2015 Avaya Inc. All Rights Reserved.

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



(Continue) the screenshot show Adaptation created for Trio:



©2015 Avaya Inc. All Rights Reserved.

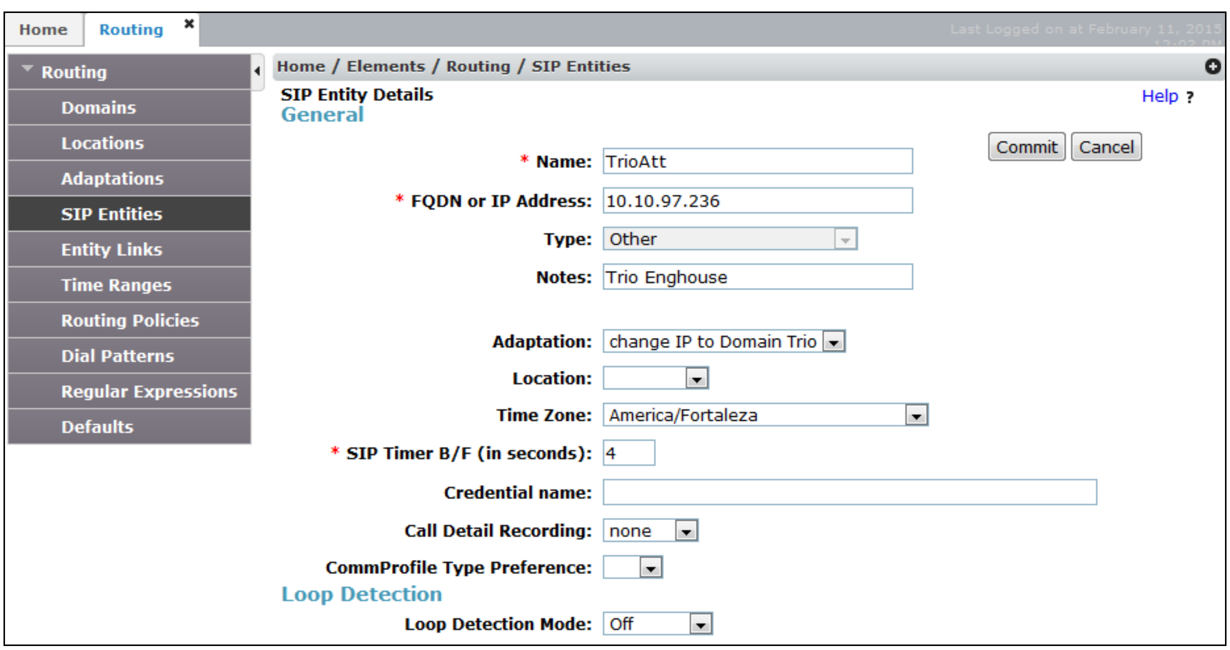
7.2. Create Trio Enterprise SIP Entity

A SIP Entity must be added for Trio Enterprise Endpoint. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button in the main window.

Enter the following for the Trio SIP Entity.

- **Name** An informative name (e.g., **TrioAtt**)
- **FQDN or IP Address** IP address of the signalling interface on the Trio Enterprise
- **Type** **Other** for Trio.
- **Adaptation** Select adaptation created above if needed. Ex: **change IP to Domain Trio**
- **Time Zone** Select appropriated time zone for this location.

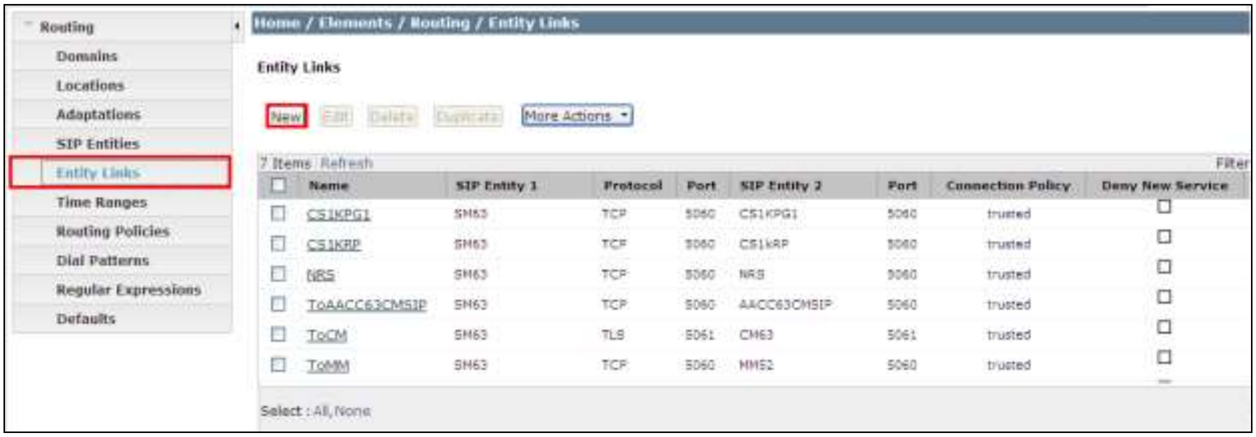
Once the correct information is entered click the **Commit** button.



©2015 Avaya Inc. All Rights Reserved.

7.3. Create Entity Link to Trio Enterprise

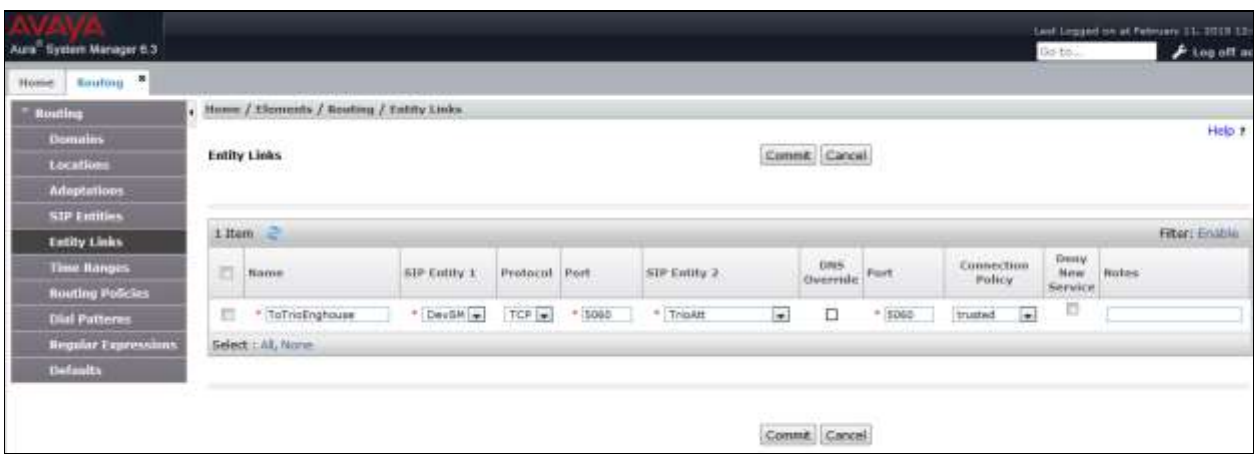
To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button in the main window.



Fill in the following fields in the new row that is displayed.

- **Name** An informative name, e.g. **ToTrioEnghouse**
- **SIP Entity 1** Select **Session_Manager** from the **SIP Entity 1** dropdown box
- **Protocol** Select **TCP** from the Protocol
- **Port** Enter **5060** as the Port
- **SIP Entity 2** Select **TrioAtt** from the **SIP Entity 2** dropdown box
- **Port** Enter **5060** as the Port
- **Trusted** Check the **Trusted** check box

Click **Commit** to save changes. The following screen shows the Entity Links used.

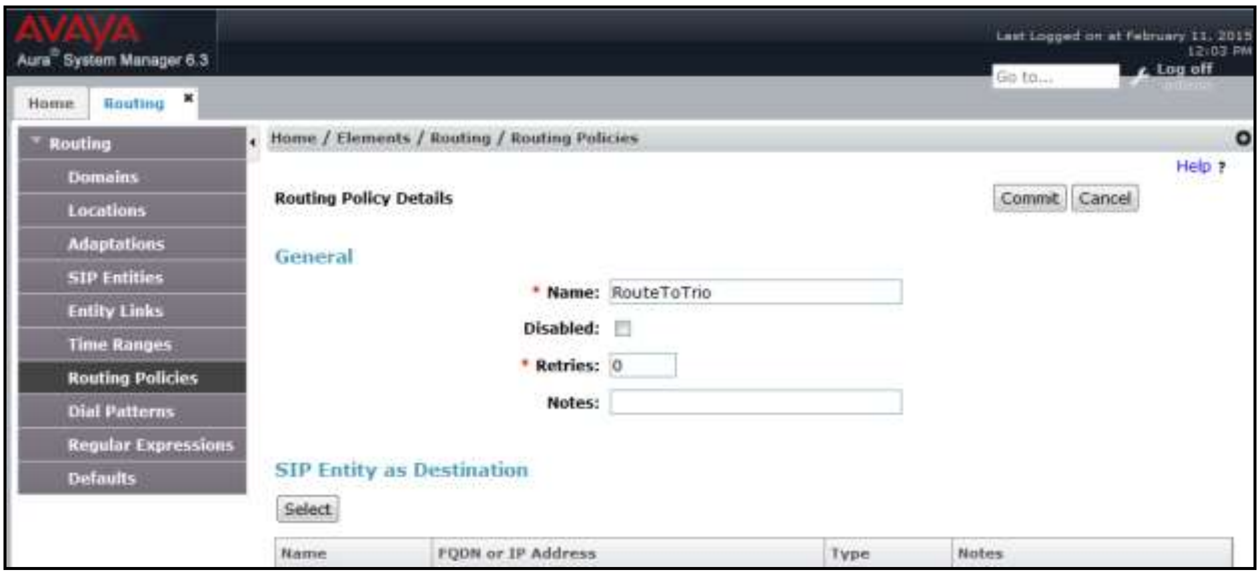


©2015 Avaya Inc. All Rights Reserved.

7.4. Create Routing Policy for Trio Enterprise

Create routing policies to direct calls to Trio Enterprise. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button in the main window.

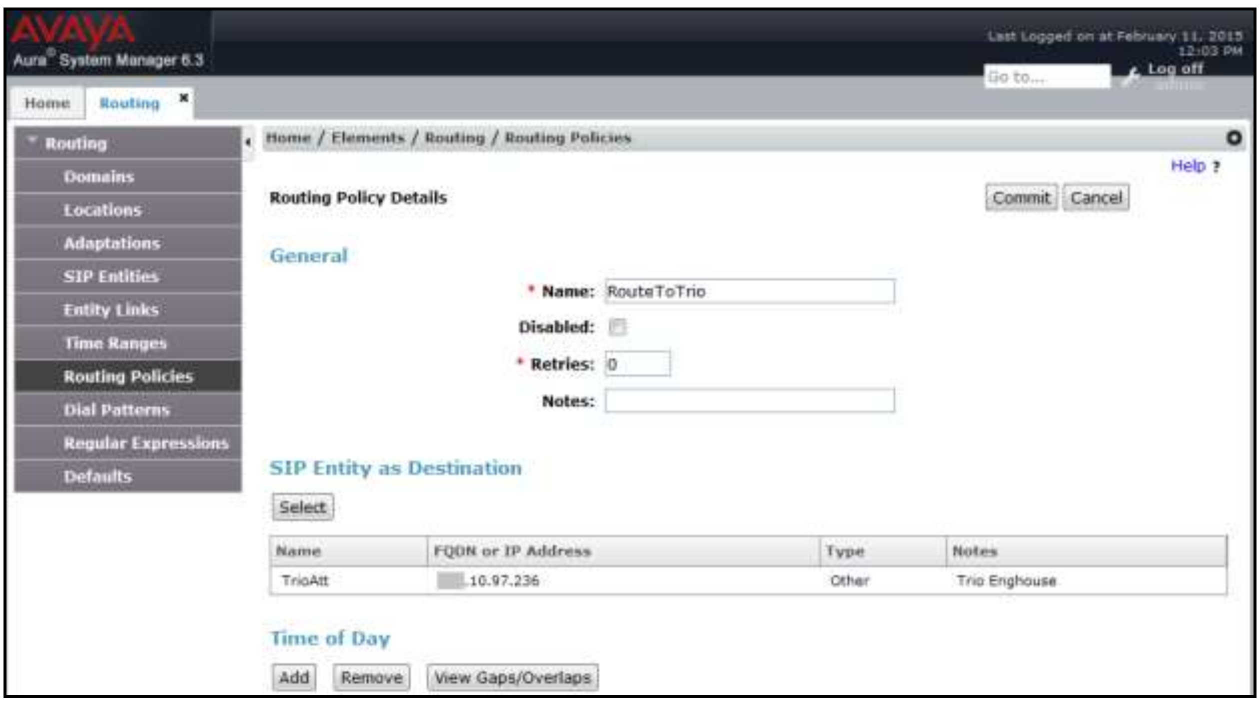
Enter an informative name, (e.g., **RouteToTrio**), under **SIP Entity as Destination,** click **Select**.



©2015 Avaya Inc. All Rights Reserved.

Check the **Trio** radio button and click on the **Select** button to confirm the chosen options and then be returned to the **Routing Policies Details** screen.

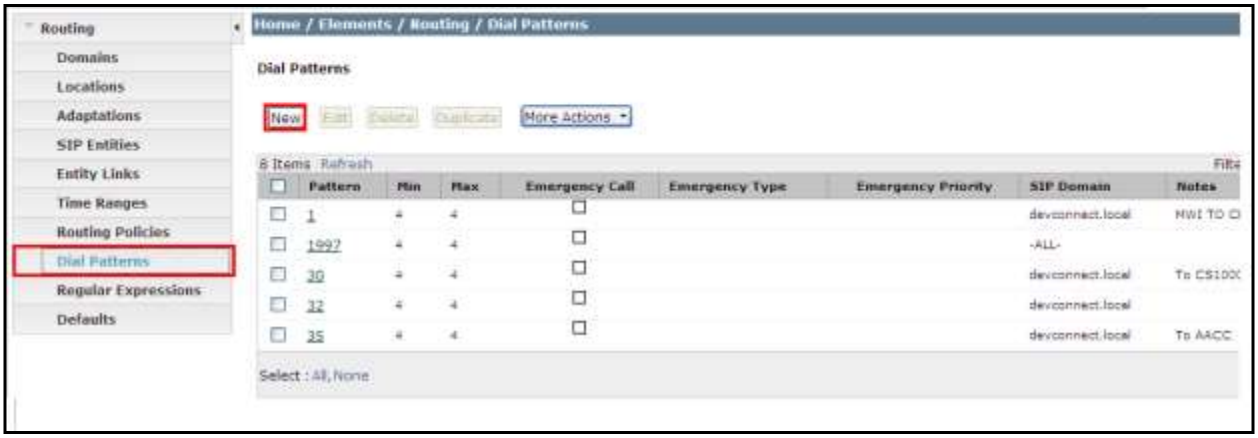
Click the **Commit** button to save. The following screen shows the **Routing Policy Details** for calls to Trio.



©2015 Avaya Inc. All Rights Reserved.

7.5. Create Trio Enterprise Dial Pattern

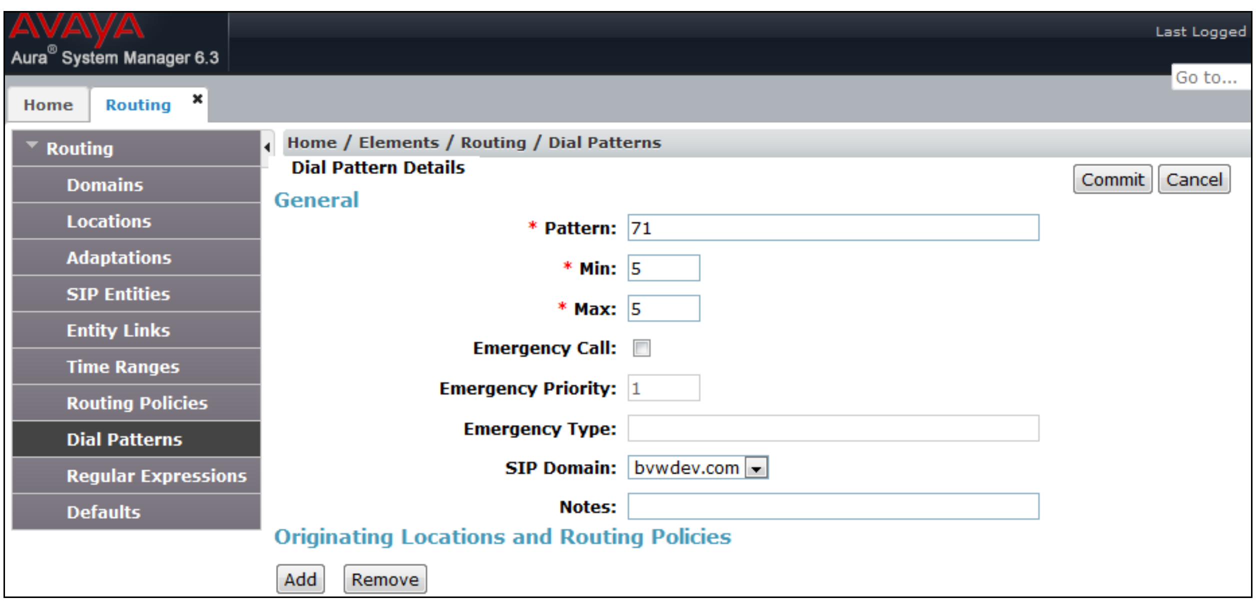
A dial pattern must be defined that will direct calls to Trio Enterprise. During testing there was several numbers used beginning with **71xxx** in conjunction with the domain name called **bvwdev.com**. To configure the Trio Enterprise dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button in the main window.



Enter the following information.

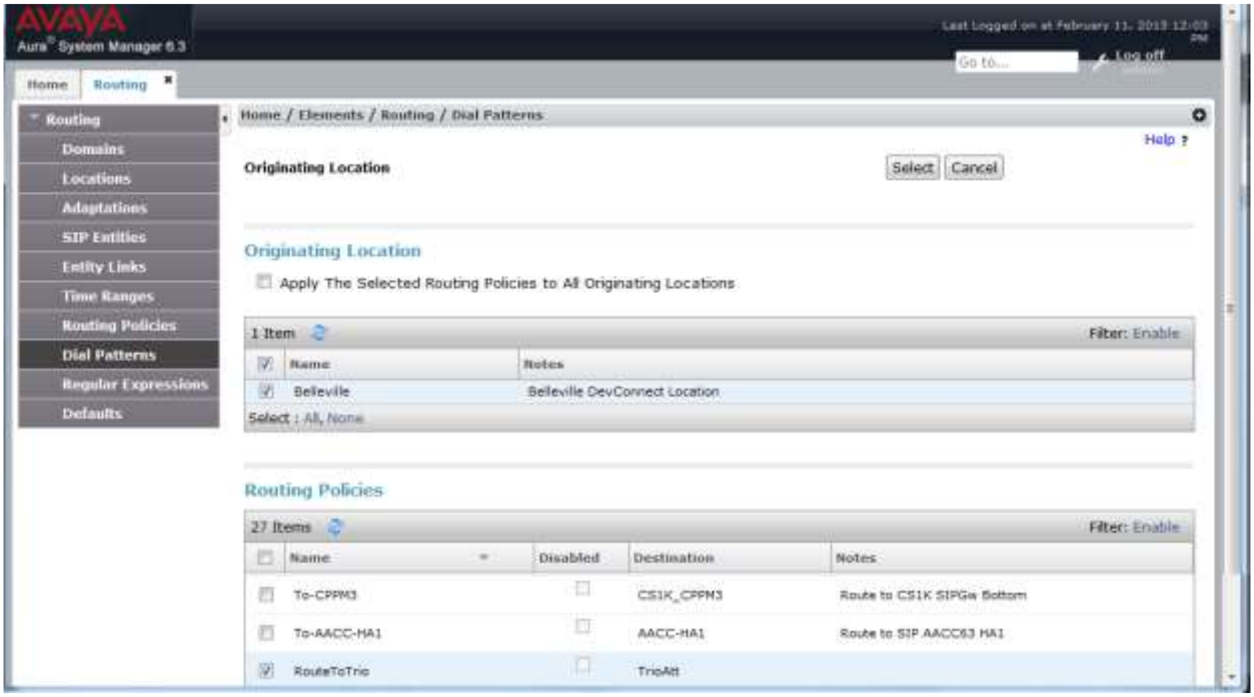
- **Pattern** Dialed number or prefix **71**
- **Min** Minimum length of dialled number **5**
- **Max** Maximum length of dialled number **5**
- **SIP Domain** Domain name of **bvwdev.com** was used in the compliance testing

Under **Originating Locations and Routing Policy List,** click on **Add.**

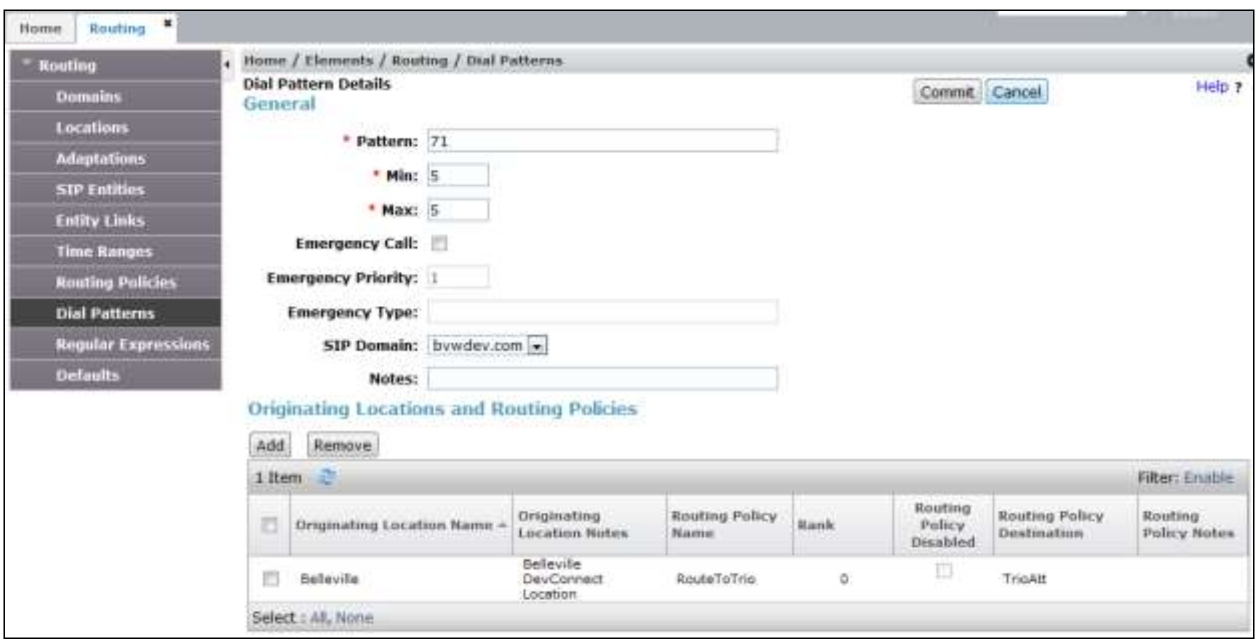


©2015 Avaya Inc. All Rights Reserved.

Check the **Belleville** check box, and under **Routing Policies** check **RouteToTrio** check box. Click on the **Select** button to confirm the chosen options and then be returned to the Dial Pattern screen (shown previously).



Click the **Commit** button to save.



©2015 Avaya Inc. All Rights Reserved.

8. Configure TRIO Enterprise

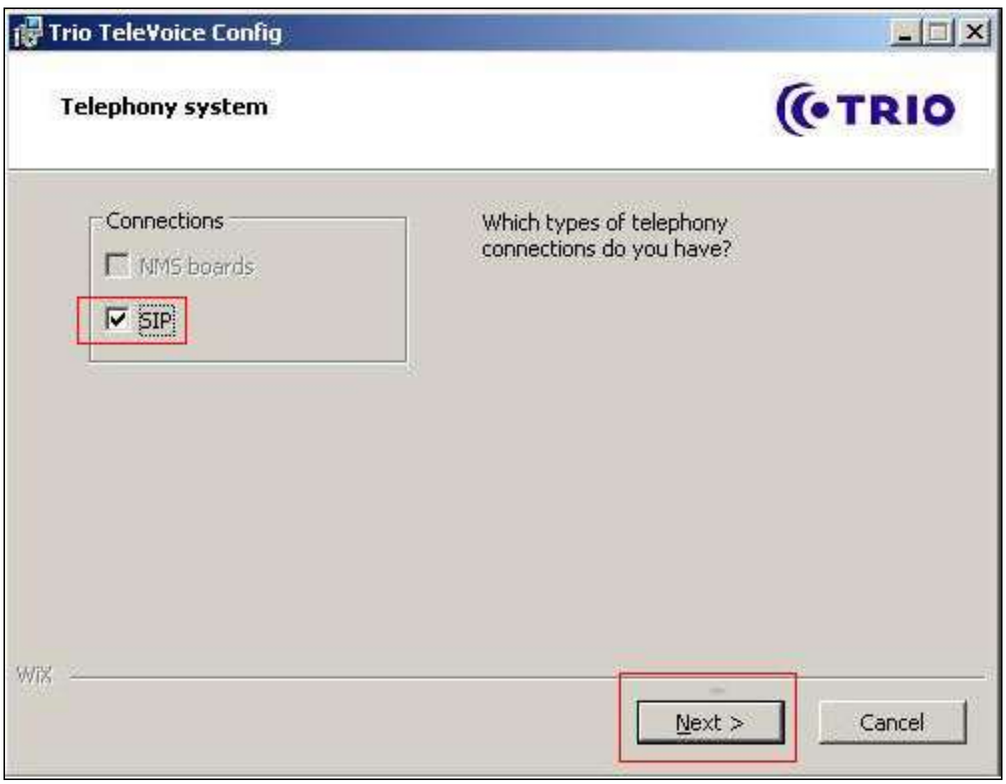
Trio Enterprise connects as a SIP endpoint to the CS1000 through Session Manager. Trio Enterprise is added to Session Manager as a SIP Entity and calls are routed to the Trio Enterprise server according to the Coordinated Dial Plan setup in **Section 5.3**. This section shows how to configure Trio Enterprise to successfully connect to the CS1000 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 s**ervice, right click and select **stop** to stop the service (not shown).
Launch the Trio configuration application. Select **Start → Programs → Trio Enterprise → Line Interface** and click on the **Config** entry (not shown). The configuration application starts up and presents the screenshot below.

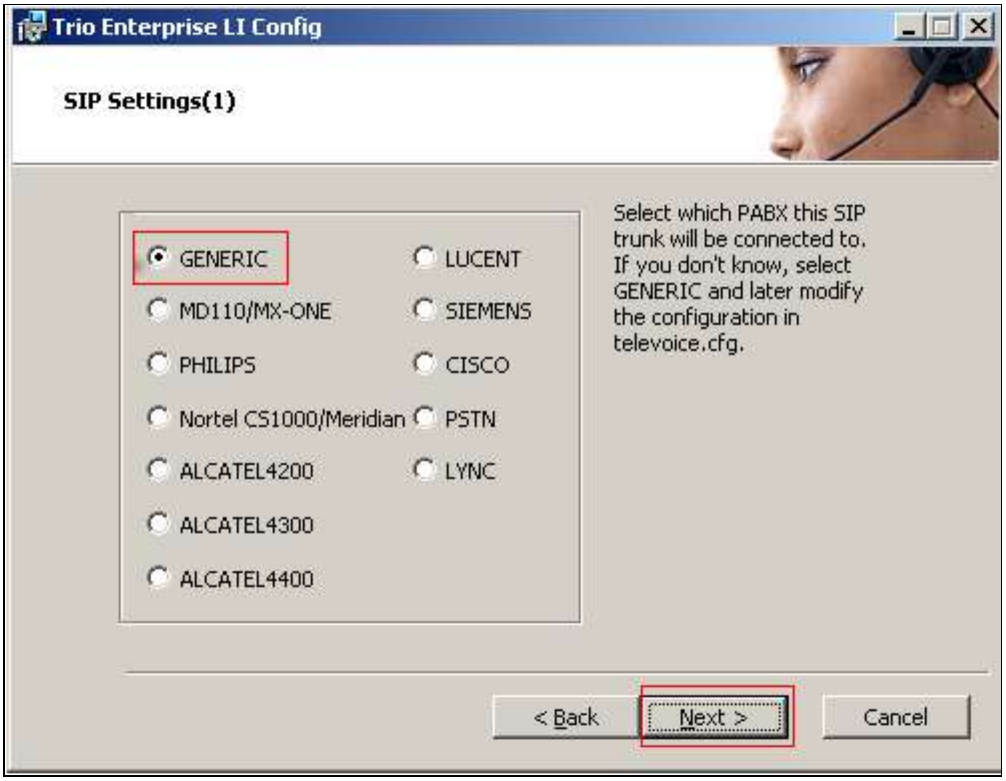
- Ensure the **SIP** entry in the **Connections** area is checked.

Click **Next** >to continue.



©2015 Avaya Inc. All Rights Reserved.

Select **GENERIC** under **SIP Settings**. Click **Next** to continue.

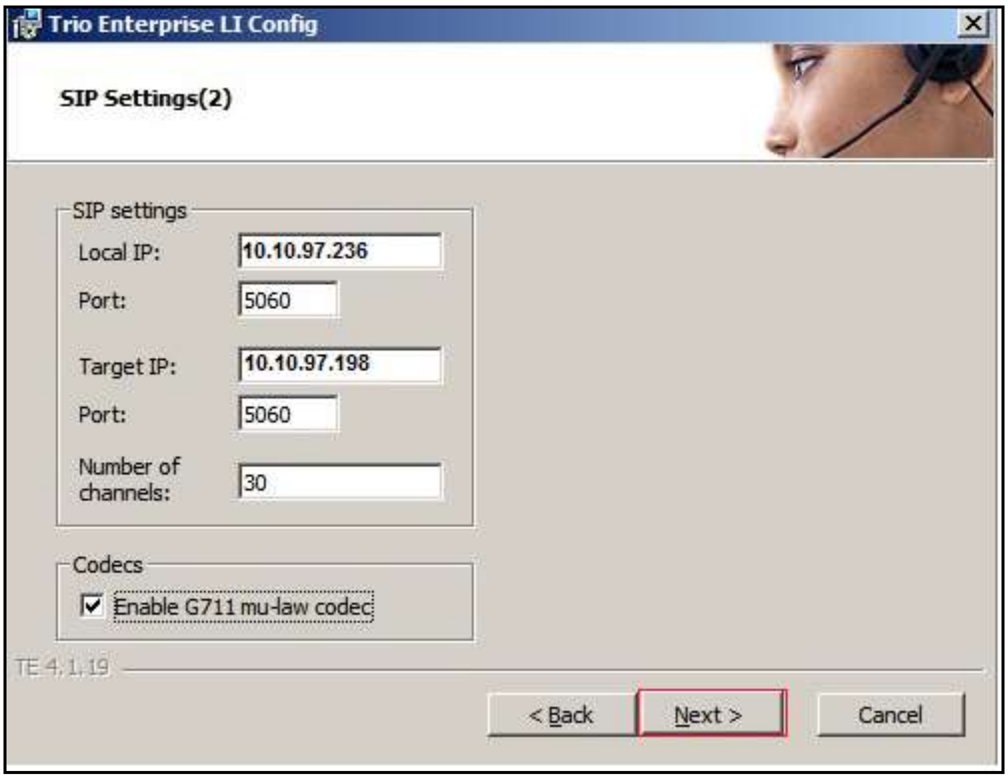


©2015 Avaya Inc. All Rights Reserved.

On the next **SIP settings** page, enter the following SIP settings.

- **Local IP** Enter the local IP address of the Trio Enterprise server, ex: **10.10.97.236**
- **Port** Enter the SIP **Port 5060**
- **Target IP** Enter the IP address of the Session Manager, ex: **10.10.97.198**
- **Port** Enter the SIP **Port 5060**
- **Number of channels** Enter **30** as the number of channels

Click **Next** to continue.

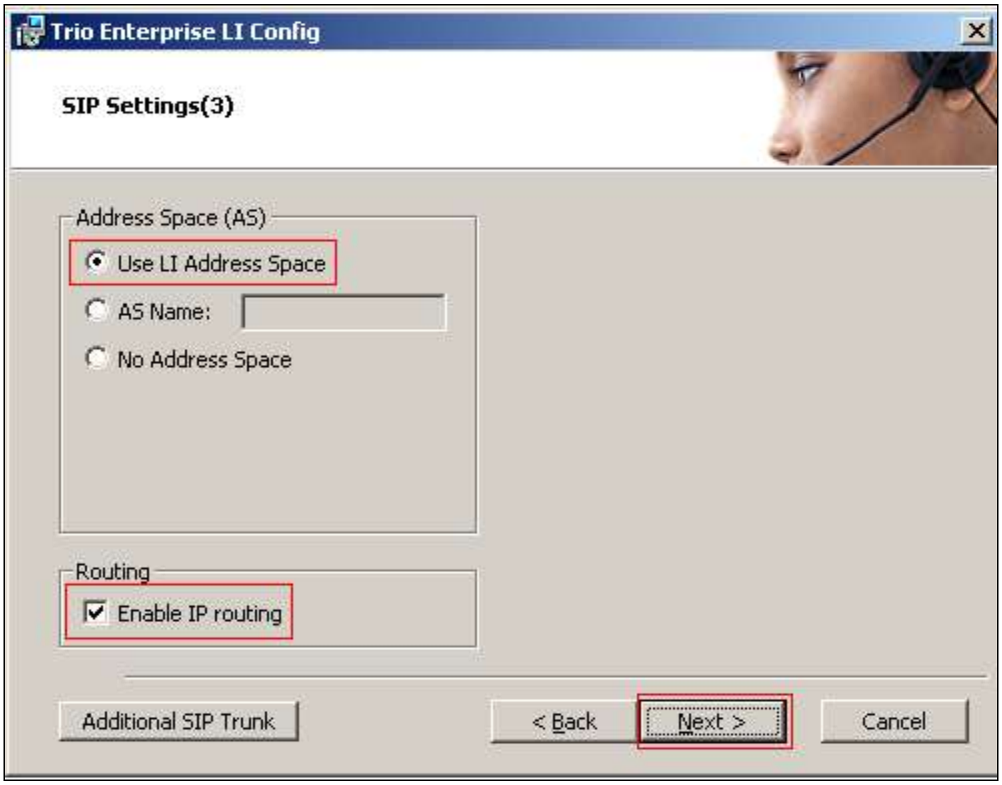


©2015 Avaya Inc. All Rights Reserved.

On the next **SIP settings** page, enter the following SIP settings.

- Select **Use LI Address Space**
- Check **Enable IP routing**

Click **Next** to continue.

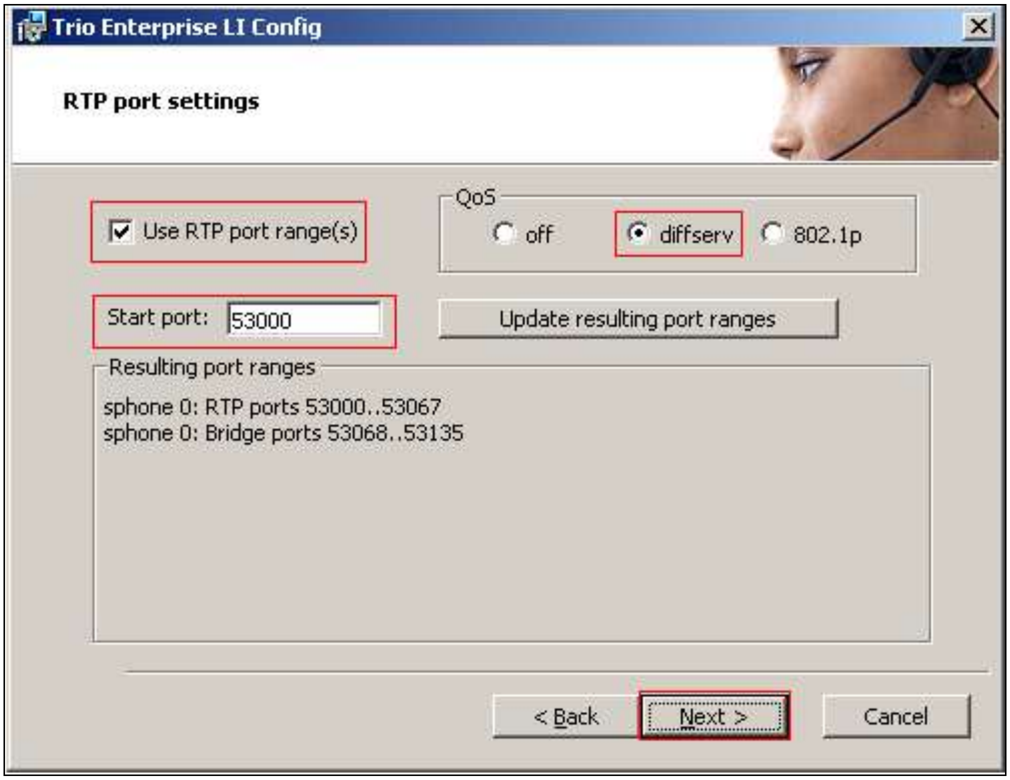


©2015 Avaya Inc. All Rights Reserved.

On the **RPT port settings** page, enter the following settings.

- Check the **Use RPT port range(s)**
- Select **diffserv**
- **Start port,** enter **53000**

Click **Next** to continue.

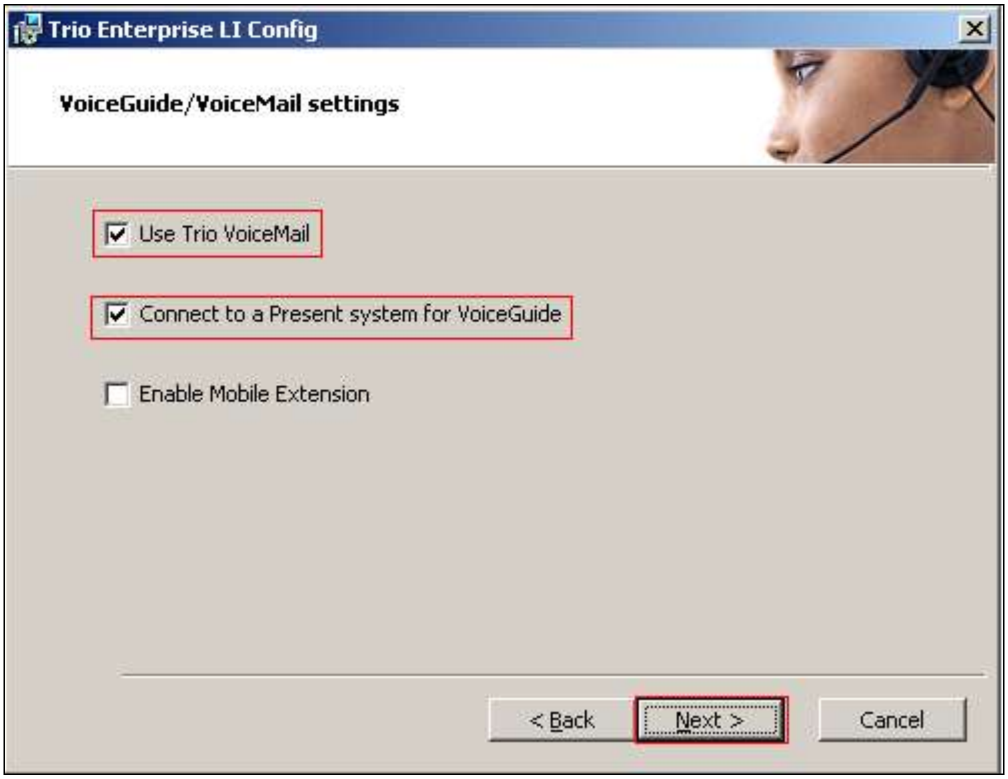


©2015 Avaya Inc. All Rights Reserved.

On the **VoiceGuide/VoiceMail settings** page, enter the following settings.

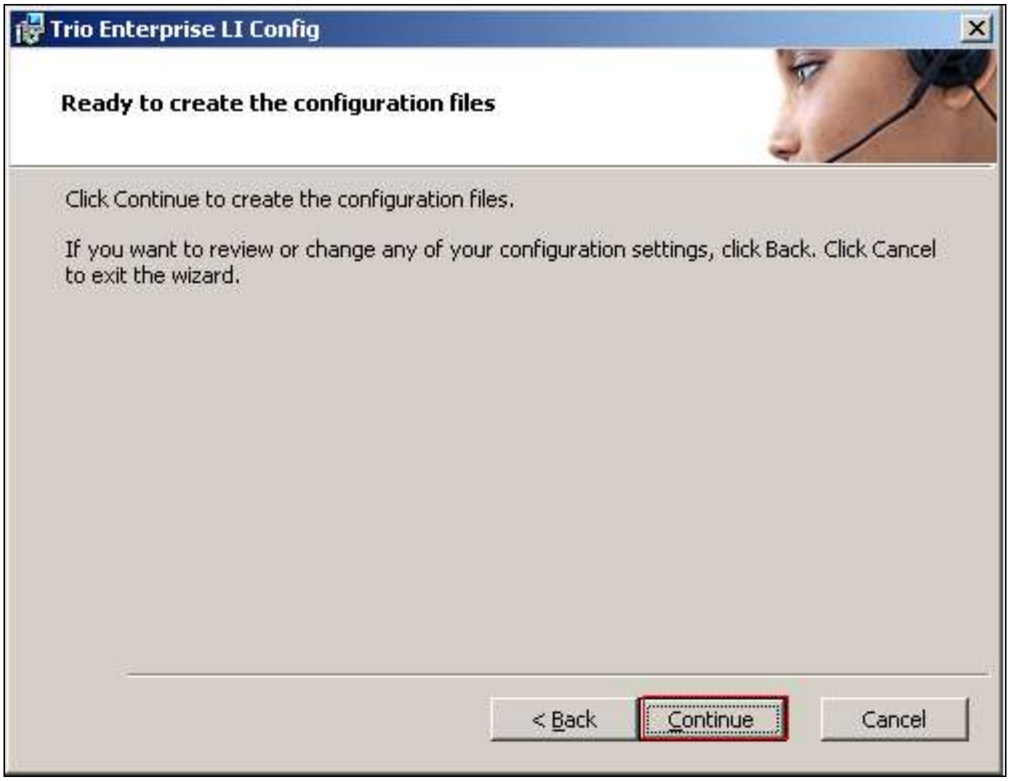
- Check **Use Trio VoiceMail**
- Check **Connect to a Present system for VoiceGuide**

Click **Next** to continue.



©2015 Avaya Inc. All Rights Reserved.

On the **Ready to create the configuration files** page click on **Continue** button.



©2015 Avaya Inc. All Rights Reserved.

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

©2015 Avaya Inc. All Rights Reserved.

8.2. InteractionStudio Configuration

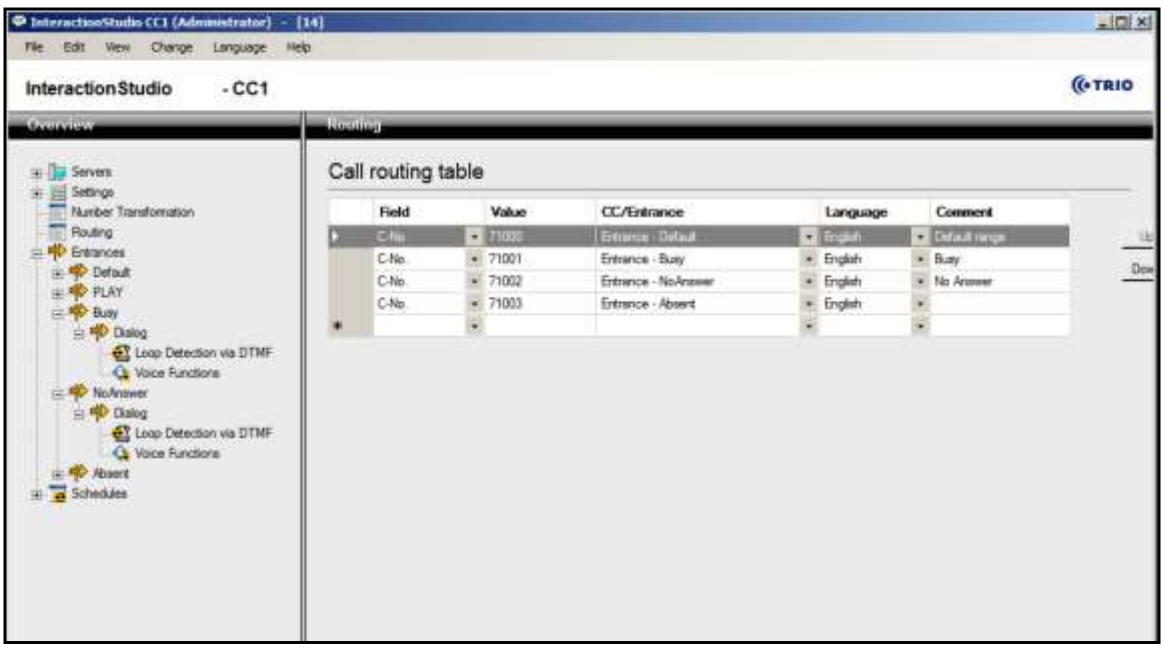
The InteractionStudio is used to configure many features for Trio Enterprise. For compliance testing, the following were configured.

- 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, click the **Start** button → **Programs** → **Trio Enterprise** → **Contact Center** → **CC1** → **Interaction Studio** (not shown). When the InteractionStudio window opens, navigate to **Routing**. A **Call routing table** will open. In the example below:

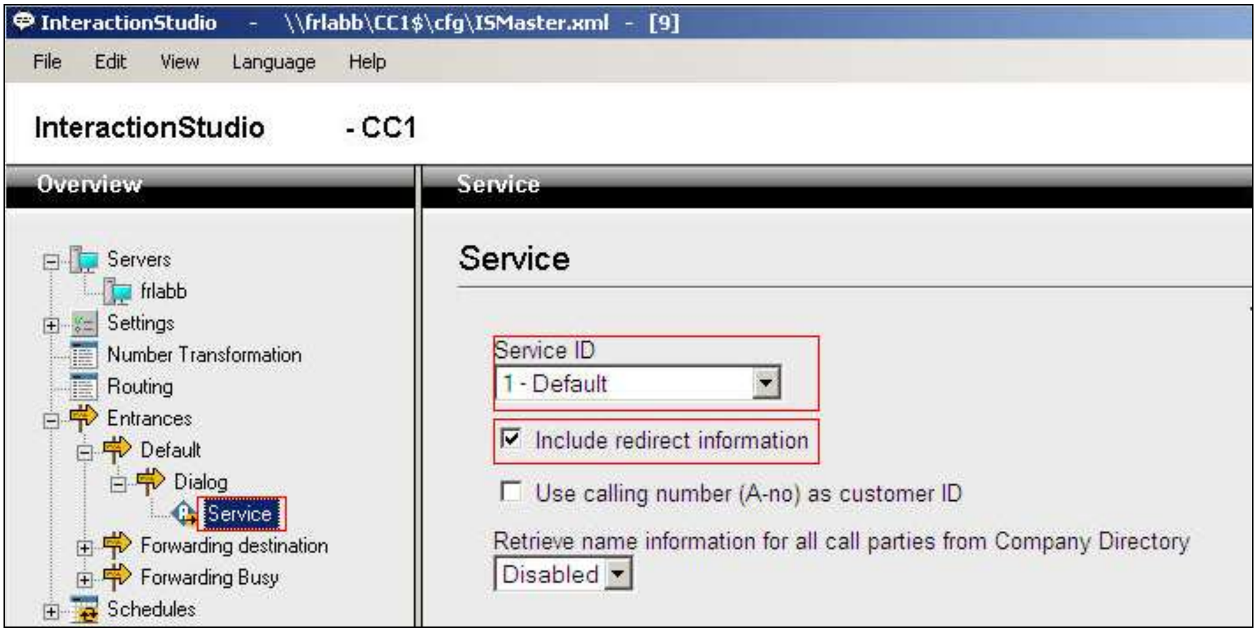
- Extension **71000** is the main queue number.
- Extension **71001** is the number that calls go to when Call forward Busy is activated.
- Extension **71002** is the number that calls go to when Call forward No Answer is activated.
- Extension **71003** is the number that calls go to when user absent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



©2015 Avaya Inc. All Rights Reserved.

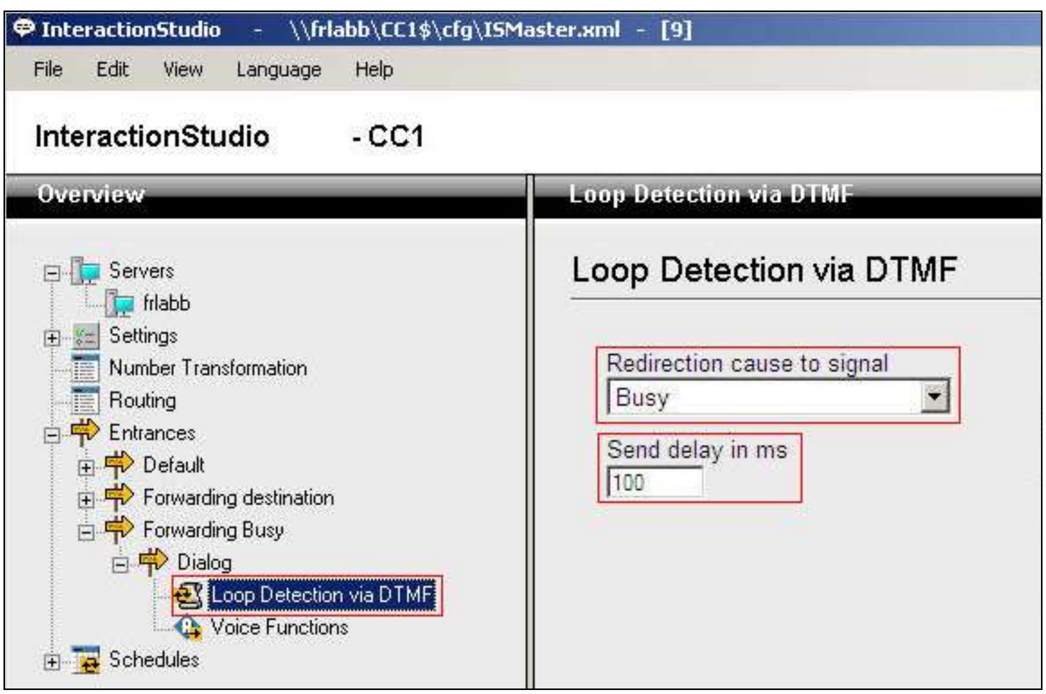
8.2.2. Configure Attendant Service

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

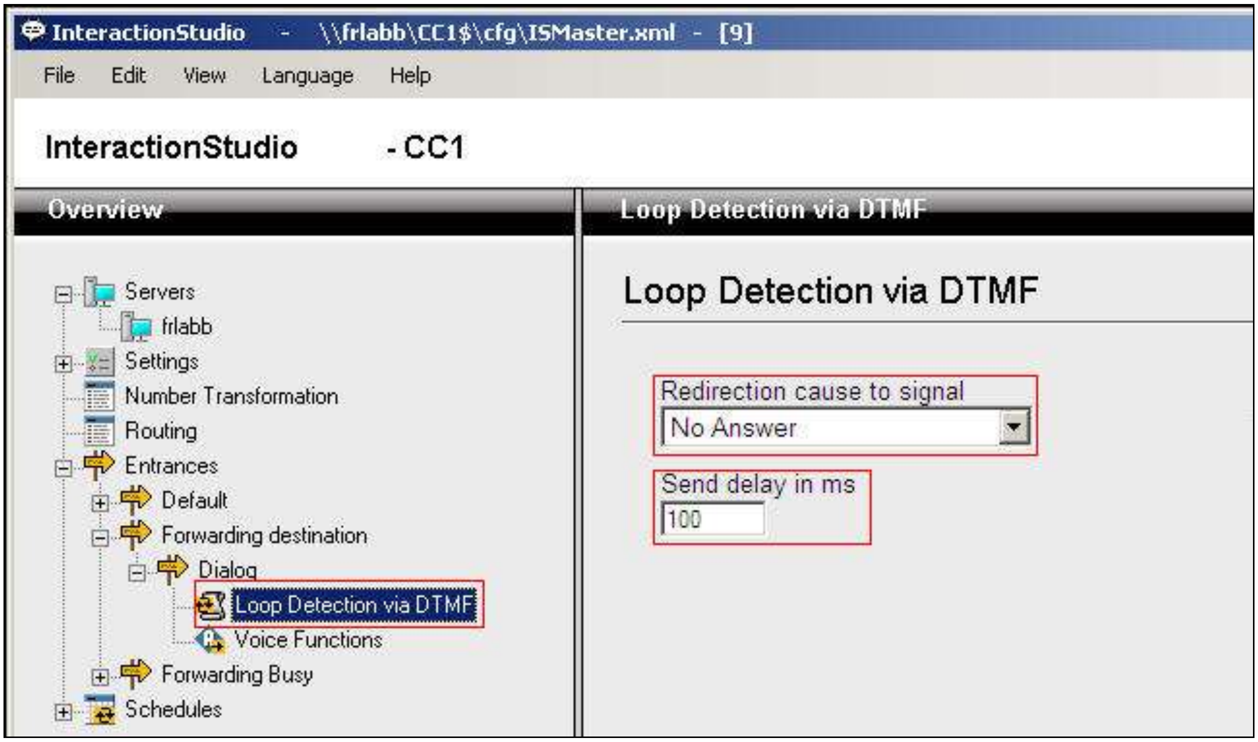
Navigate to **Entrances → Forwarding Busy → Dialog → Loop Detection via DTMF.** Choose **Busy** from the **Redirection cause to signal** drop down box, and enter **100** in the **Send delay in ms** box.



©2015 Avaya Inc. All Rights Reserved.

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.



©2015 Avaya Inc. All Rights Reserved.

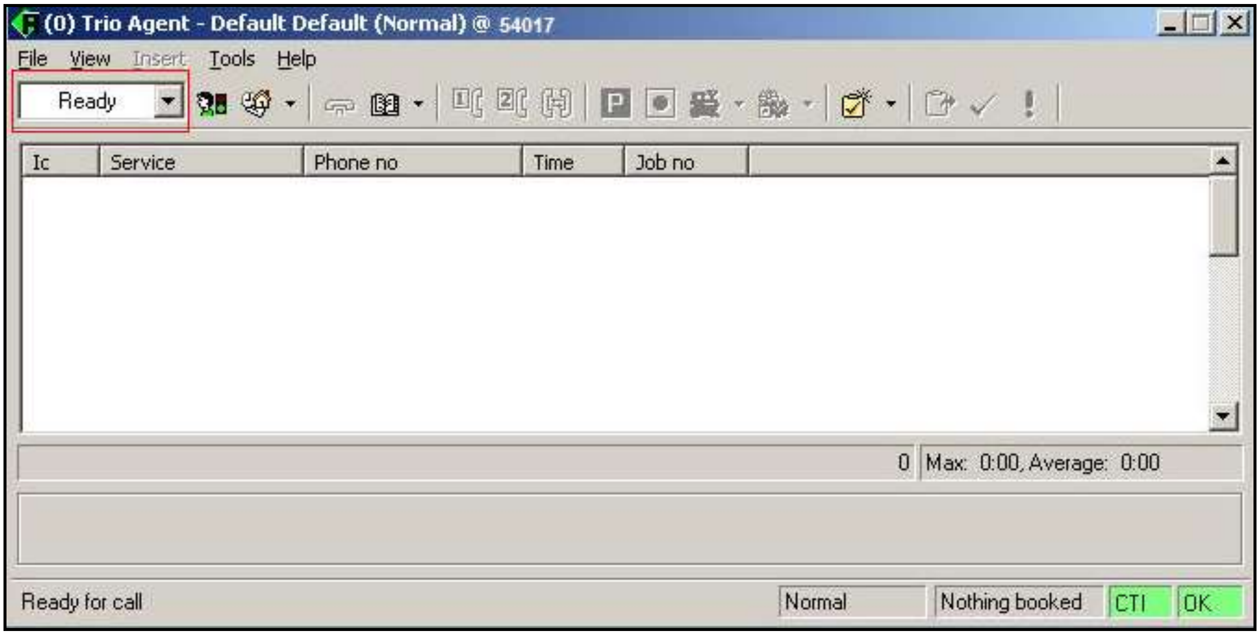
8.3. Configuring Trio Attendant

Trio Attendant is a separate application to Trio Enterprise server and can run concurrently on the same platform. The attendant uses a regular CS1000 telephone to make and receive calls, which are directed to the phone by Trio Enterprise server. The steps to configure Trio Attendant are to click on **Start → Programs → Trio Enterprise → Contact Centre → Agent Client** (not shown).

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 CS1000 telephone number that will be used as the agent's audio device (number **54017** 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.

©2015 Avaya Inc. All Rights Reserved.

The Trio Agent window appears. Select **Ready** from the drop down box (confirm the traffic light goes green in the small icon to the right of the drop down box).



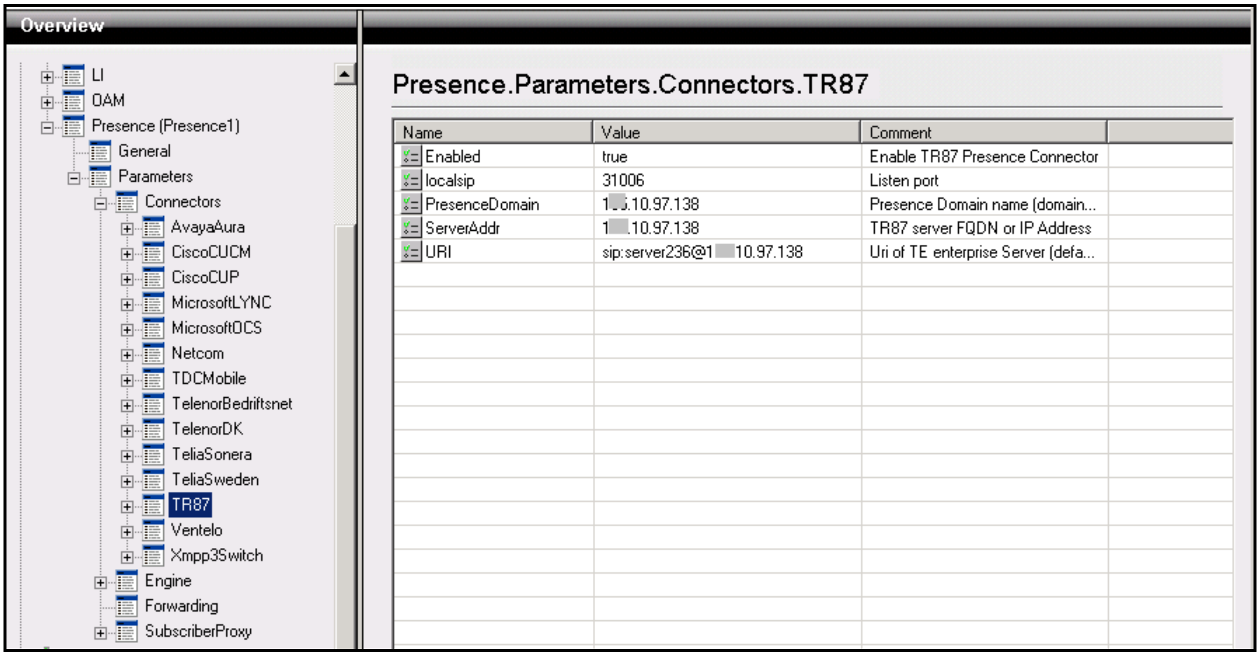
©2015 Avaya Inc. All Rights Reserved.

Click on **Start → Programs → Enterprise Management Center**.

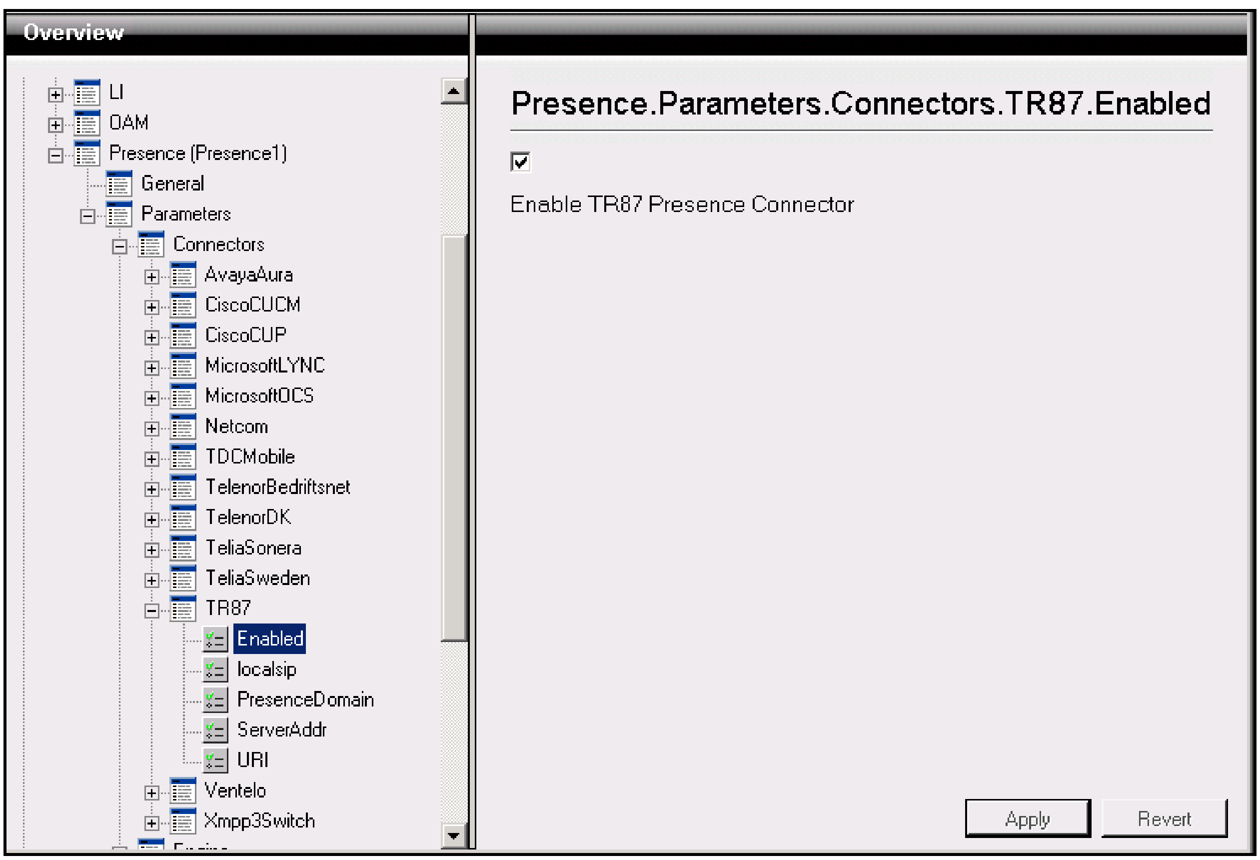
Enter the proper credentials and click on **OK**.

©2015 Avaya Inc. All Rights Reserved.

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

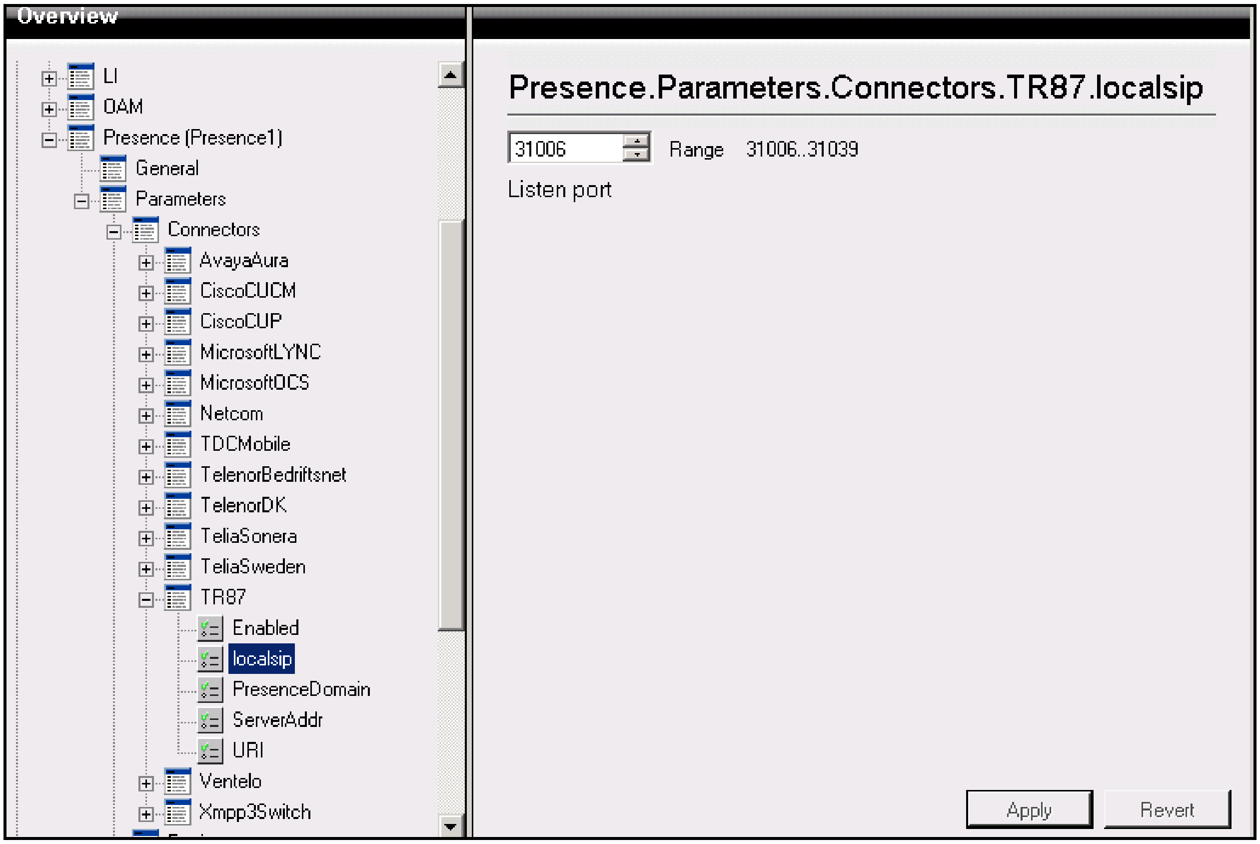


Under **TR87** select **Enabled** in the left window. Ensure that **Enable TR87 Presence Connector** is ticked as shown below. Click **Apply** to continue.



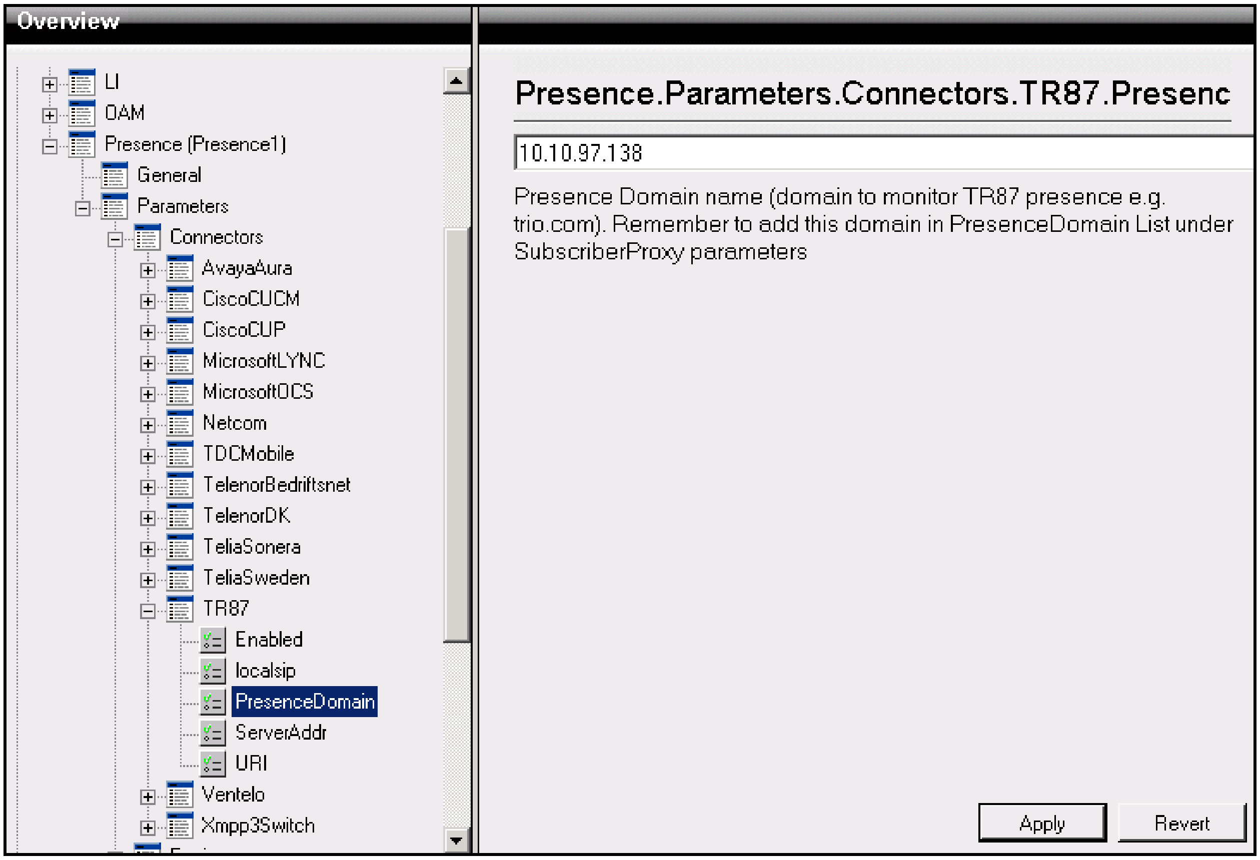
©2015 Avaya Inc. All Rights Reserved.

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.



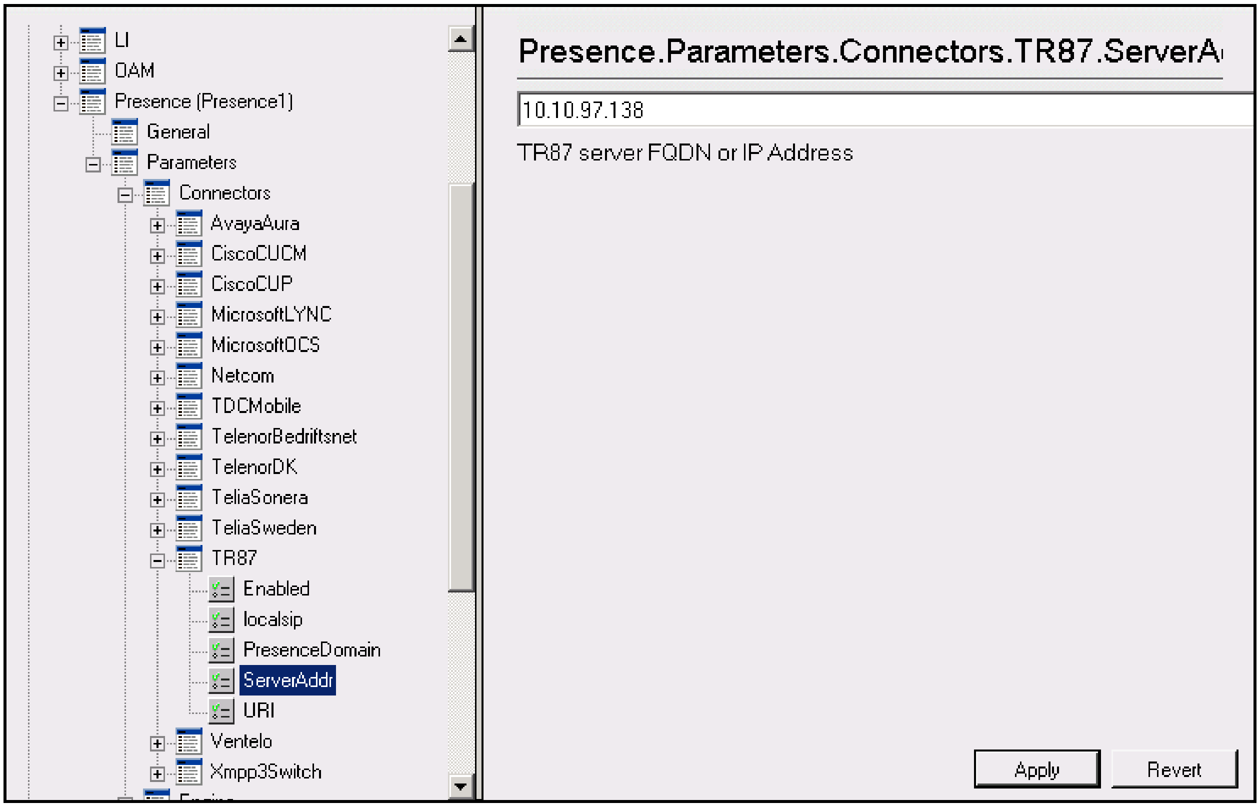
©2015 Avaya Inc. All Rights Reserved.

Select **PresenceDomain** under **TR87** in the left window. Enter the Node IP address of the CS1000 as per **Section 6**. Click **Apply** to continue.



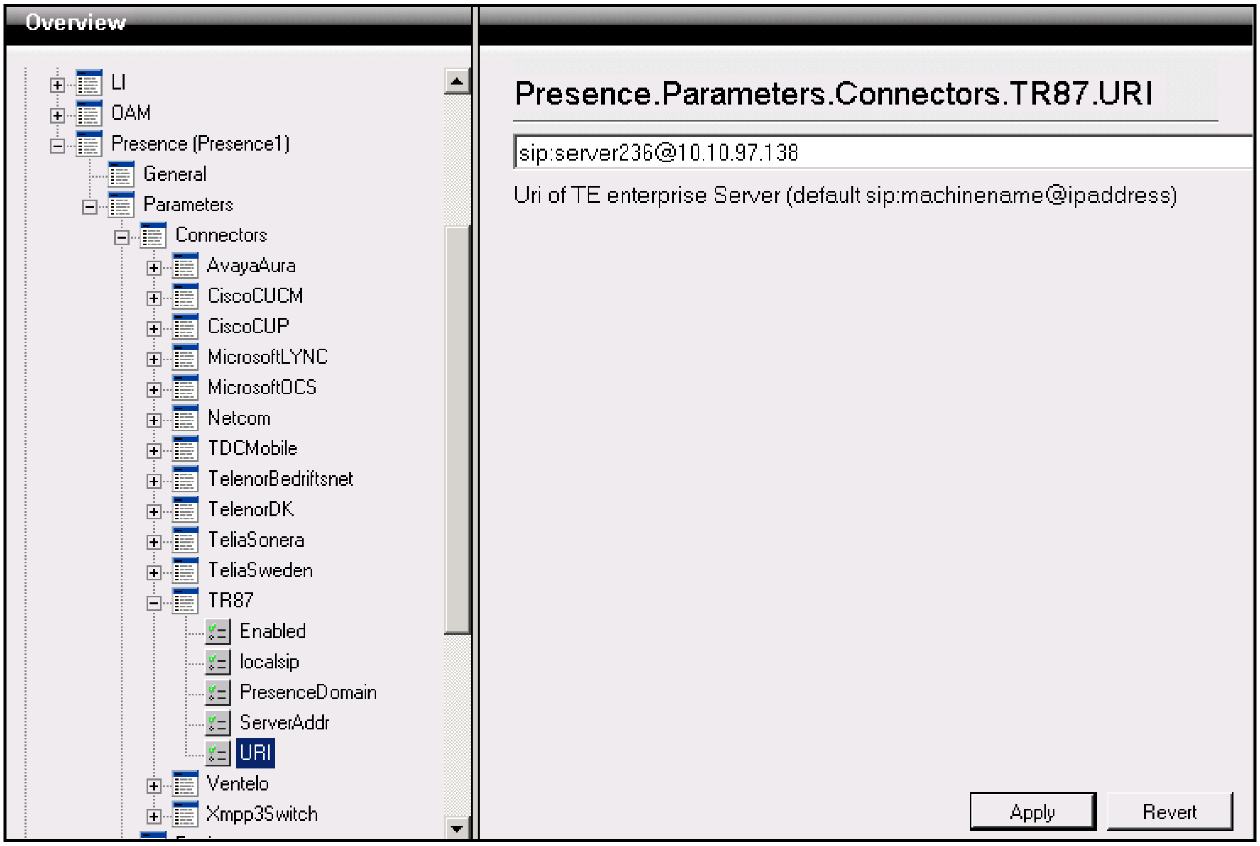
©2015 Avaya Inc. All Rights Reserved.

Select **ServerAddr** under **TR87** in the left window and again enter the Node IP address of the CS1000. Click **Apply** to continue.



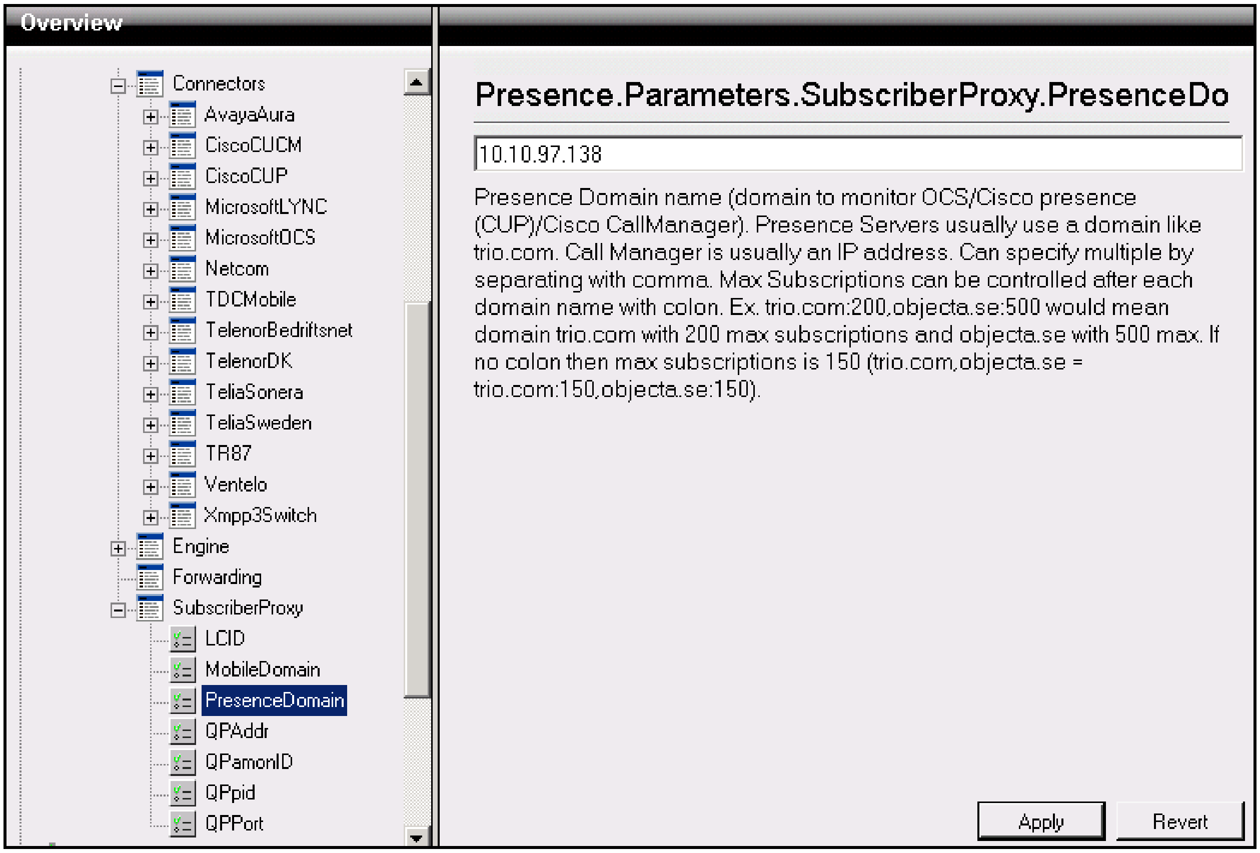
©2015 Avaya Inc. All Rights Reserved.

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



©2015 Avaya Inc. All Rights Reserved.

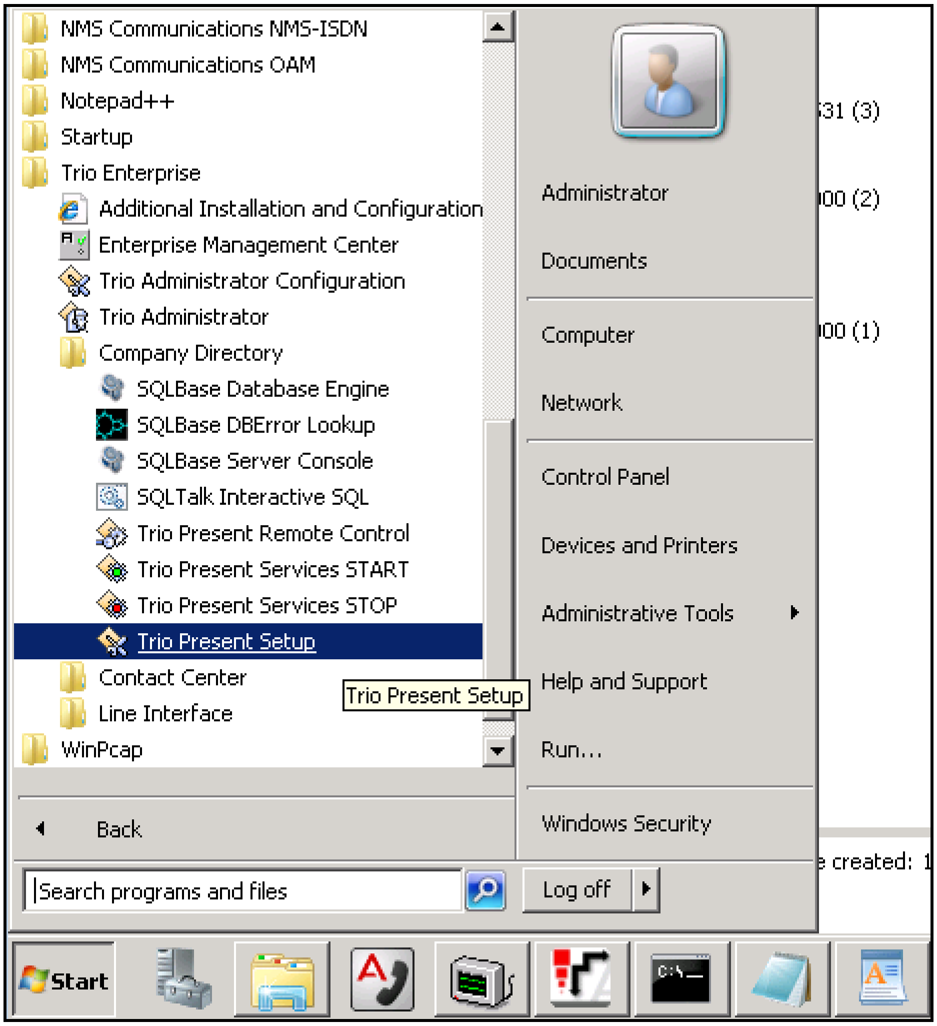
Select **PresenceDomain** under **SubscribeProxy** in the left window. Enter the Node IP address of the CS1000 in the right window. Click **Apply** to continue.



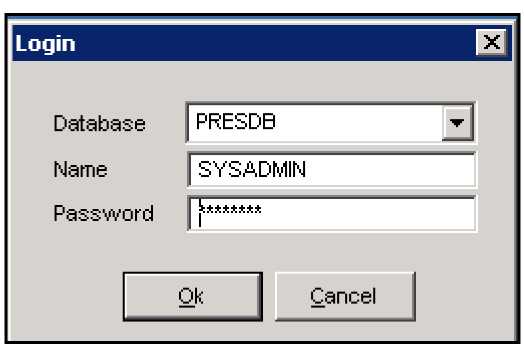
©2015 Avaya Inc. All Rights Reserved.

8.4. Configure ICP on Trio

Select **Start → Programs → Trio Enterprise → Company Directory → Trio Present Setup** as shown below.



Enter the proper credentials and click **Ok**.

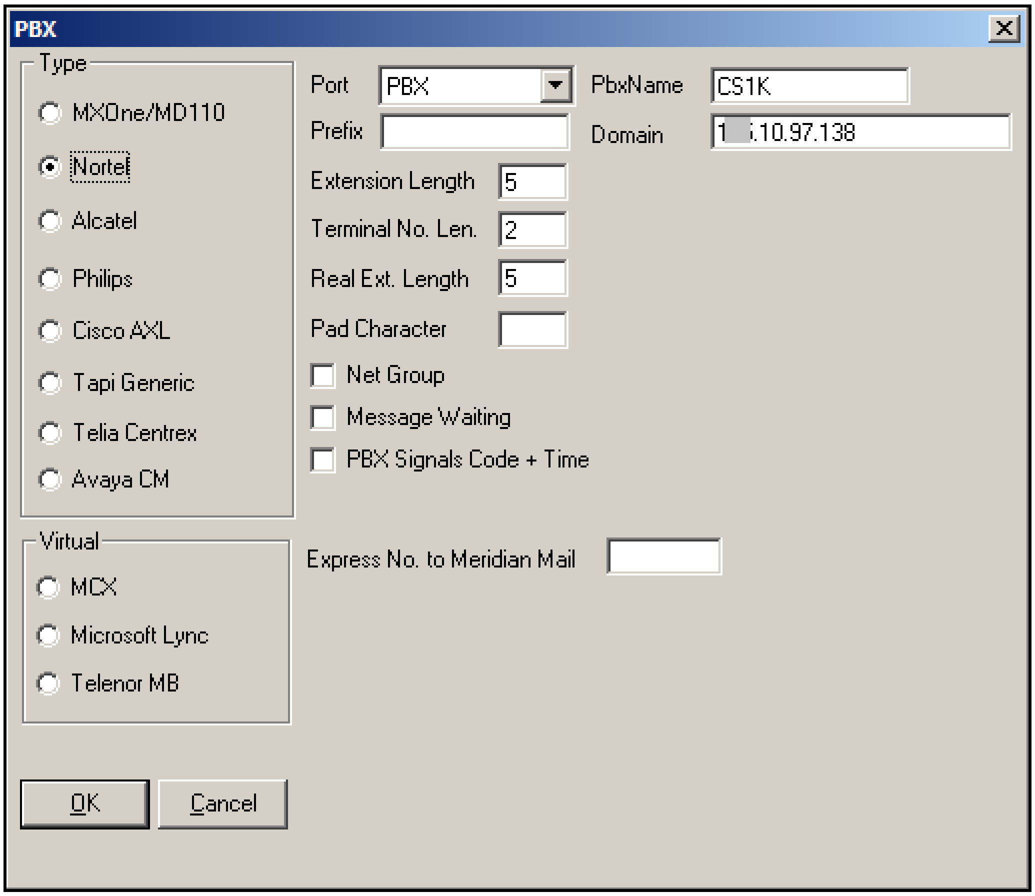


©2015 Avaya Inc. All Rights Reserved.

Highlight the selected PBX under the PBX tab and click on **Change** (not shown). This opens the window displayed on the right. Ensure the following are selected.

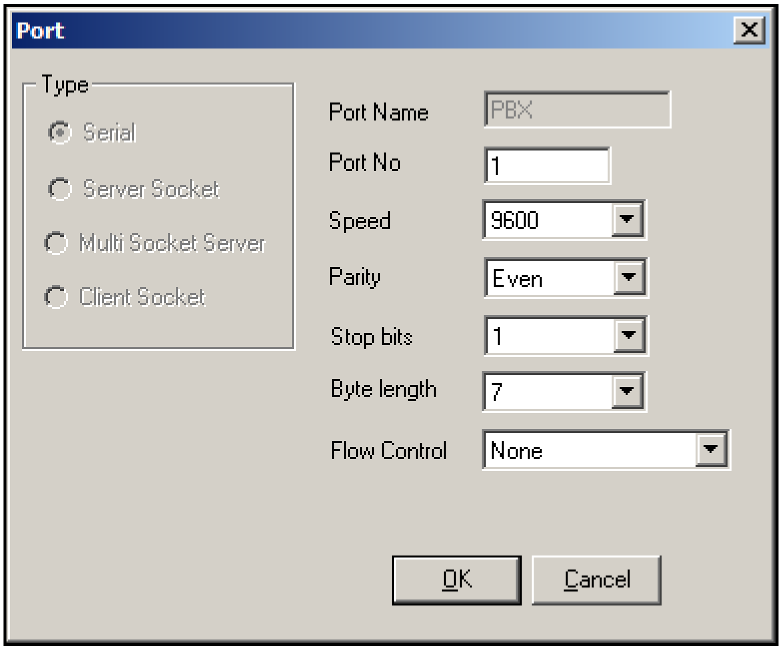
- **Type** **Nortel**
- **Port** **PBX**
- **Domain** Node IP Address of CS1000, in this case it is IP address of Node **557**

Select **OK** once the correct information is entered.



©2015 Avaya Inc. All Rights Reserved.

Under the **Communications** tab select **ICP** and click **Change** (not shown). Enter the information that was entered in **Section 5.5.1**. Click **OK** once all correct information is added.



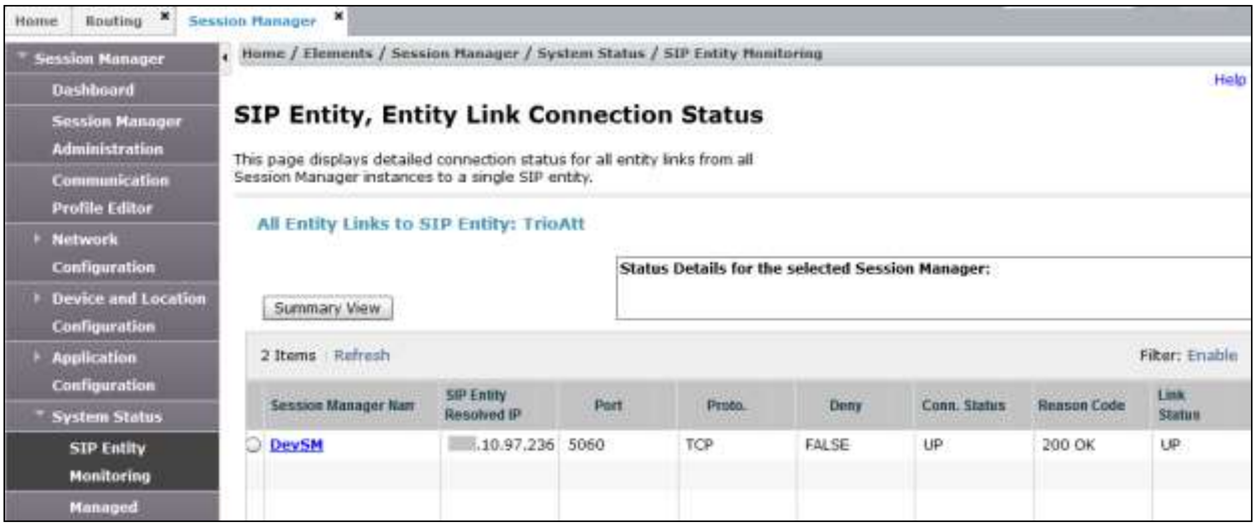
©2015 Avaya Inc. All Rights Reserved.

9. Verification Steps

This section provides the tests that can be performed to verify correct configuration of CS1000 and Session Manager with TRIO Enterprise.

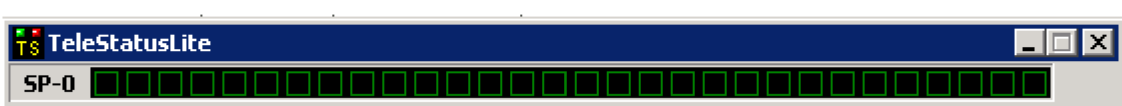
9.1. Verify status of Trio SIP Entity

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. In the detail page it show the link from **Trio** to **Session Manager** via **TCP** is **UP**.



9.2. SIP Channels on Trio Enterprise

To confirm a successful Trio Enterprise connection with the Session Manager, click on **Start → Programs → Trio Enterprise → Line Interface** and then select the **Telestatus** entry. 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).



©2015 Avaya Inc. All Rights Reserved.

10. Conclusion

These Application Notes describe the configuration steps required for Trio Enterprise R5.0 from Enghouse Interactive to successfully interoperate with Avaya Communication Server 1000 R7.6 and Avaya Aura® Session Manager R6.3 using SIP trunks. Trio Enterprise passed all compliance testing successfully; please see **Section 2.2** of these Application Notes for results and observations.

11. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com where the following documents can be obtained.

[1] *Software Input Reference Administration Avaya Communication Server 1000, Release 7.6*; Document No. NN43001-611_05.02

[2] *Administering Avaya Aura® Session Manager*; Doc # 03-603324, Release 6.3

[3] *Unified Communications Management Common Services Fundamentals Avaya Communication Server 1000*; Doc # NN43001-116, 05.08

[4] *Element Manager System Reference –Administration Avaya Communication Server 1000*; Doc # NN43001-632, 05.04

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

©2015 Avaya Inc. All Rights Reserved.

©2015 Avaya Inc. All Rights Reserved.
Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

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.