Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Trio Enterprise R4.1 from Enghouse Interactive AB with Avaya Communication Server 1000E 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 1000E 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 1000E 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.

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 1000E R7.6 and Avaya Aura® Session Manager R6.3 with Trio Enterprise R4.1 from Enghouse Interactive AB. Trio Enterprise is a client/server based application running on Microsoft Windows 2008 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 1000E 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 (CS1000E). The Trio Enterprise server uses a SIP connection to the CS1000E call server via Session Manager. See **Figure 1** for a network diagram. A basic Distance Steering Code configuration (DSC) was configured on the CS1000E 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 the CS1000E and then to the PSTN using the CS1000E PSTN connection. During compliance testing three different simulated PSTN trunks were used which included a QSIG ISDN trunk, a H.323 trunk and a SIP trunk. 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. The CS1000E 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 path replacement and the caller and the called user are now directly connected. It is possible to have multiple Trio attendant positions on a CS1000E system.

A variety of Avaya telephones were installed and configured on the CS1000E. 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 1140E was used as the attendant's deskphone. When the attendant is called the Trio Enterprise server calls the 1140E 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.

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 the Trio Enterprise and the CS1000E. 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 on Trio 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: infosweden@enghouse.com

3. Reference Configuration

Figure 1 shows the network topology during compliance testing. Trio Enterprise is connected to the CS1000E 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.

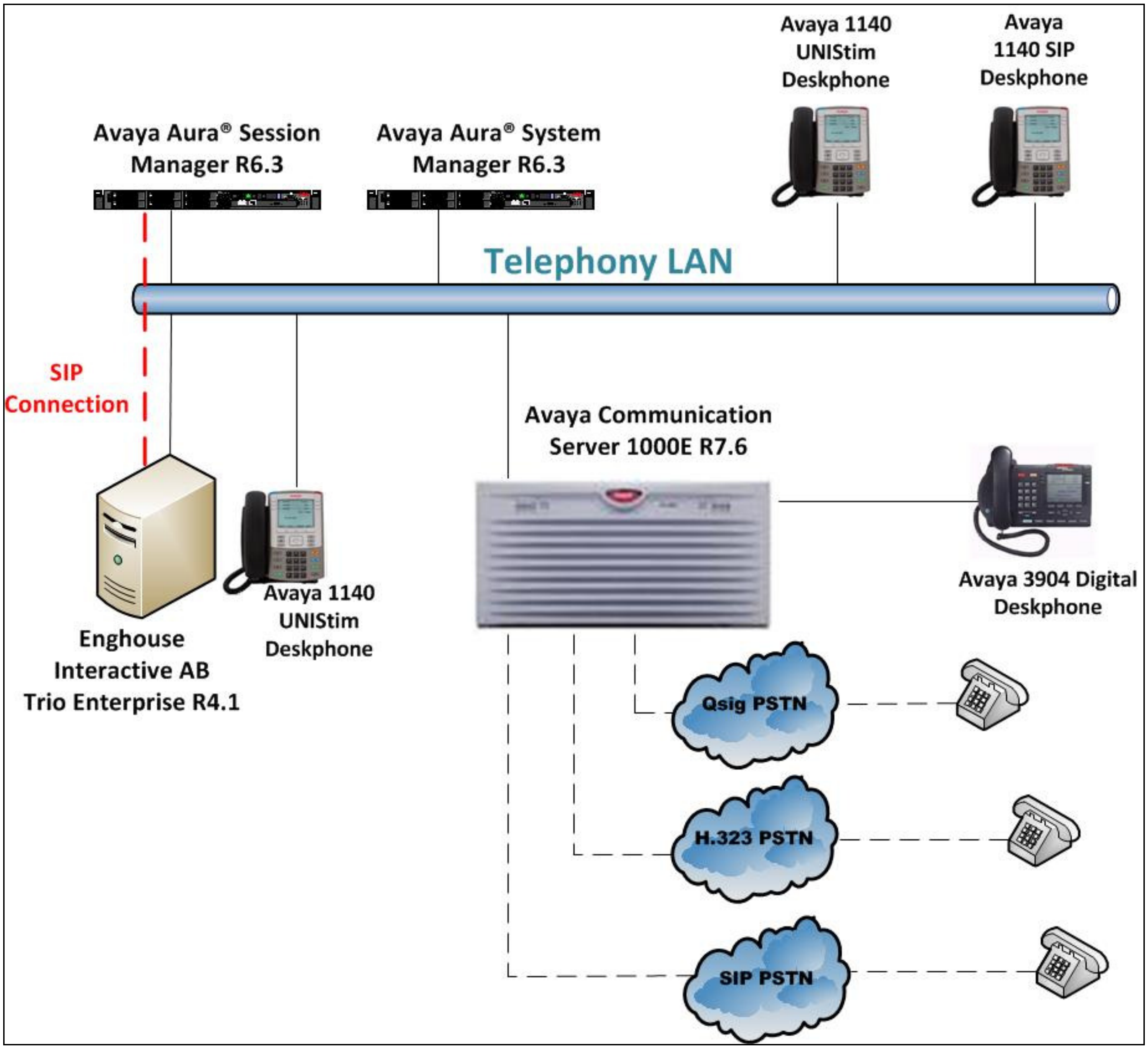


Figure 1: Configuration for Avaya Communication Server 1000E R7.6, Avaya Aura® Session Manager R6.3 and Trio Enterprise R4.1

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Communication Server 1000E on CPPM	R7.6 SP2 (See Appendix A for list of patches)
Avaya Aura® System Manager	System Manager 6.3.0 - FP2 Build No. - 6.3.0.8.5682-6.3.8.1814 Software Update Revision No: 6.3.3.5.1719
Avaya Aura® Session Manager	Session Manager R6.3 SM 6.3.3.0.633004
Avaya 1140 UNIStim Deskphone	UNIStim V0625C8D
Avaya 1140 SIP Deskphone	SIP 04.03.12
Avaya 3904 Digital Deskphone	Core V2.4 Flash V9.4
Trio Enterprise Running on a Windows 2008 R2 64-bit server	Version 4.1

Note: The Avaya Aura® System Manager and Session Manager are virtual instances running on VMware ESXi 5.0

5. Configure Avaya Communication Server 1000E

The configuration operations illustrated in this section were performed using terminal access to the CS1000E over an "SSH" session using "PUTTY". The information provided in this section describes the configuration of the CS1000E for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 11**.

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.

5.1. Verify Licences

Both SIP CTI Licences and AST licenses are required to allow Trio observe TR87 events. To ensure the CS1000E 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).

```
Prompt              Response            Description
>                   LD 22               Enter Overlay 22
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  2000    LEFT  1992    USED     8
DECT USERS              2000    LEFT  2000    USED     0
IP USERS                4000    LEFT  3978    USED    22
BASIC IP USERS          2000    LEFT  1998    USED     2
TEMPORARY IP USERS      2000    LEFT  2000    USED     0
DECT VISITOR USER       2000    LEFT  2000    USED     0
ACD AGENTS              2000    LEFT  1995    USED     5
MOBILE EXTENSIONS       2000    LEFT  2000    USED     0
TELEPHONY SERVICES      2000    LEFT  2000    USED     0
CONVERGED MOBILE USERS  2000    LEFT  2000    USED     0
AVAYA SIP LINES         2000    LEFT  1997    USED     3
THIRD PARTY SIP LINES   2000    LEFT  1998    USED     2

PCA                     2000    LEFT  2000    USED     0
ITG ISDN TRUNKS         2000    LEFT  2000    USED     0
H.323 ACCESS PORTS      2000    LEFT  1990    USED    10
AST                     2000    LEFT  1981    USED    19
SIP CONVERGED DESKTOPS  2000    LEFT  2000    USED     0
SIP CTI TR87            2000    LEFT  1992    USED     8
SIP ACCESS PORTS        2000    LEFT  1970    USED    30
RAN CON                 2000    LEFT  2000    USED     0
MUS CON                 2000    LEFT  2000    USED     0

IP RAN CON              2000    LEFT  2000    USED     0
```

```
IP MUS CON              2000    LEFT  2000    USED     0
IP MEDIA SESSIONS       2000    LEFT  1997    USED     3
TNS                    10000    LEFT  9805    USED   195
ACDN                   24000    LEFT 23979    USED    21
AML                       16    LEFT    12    USED     4
IDLE_SET_DISPLAY Cores3 Rls 7.5
LTID                    2000    LEFT  2000    USED     0
RAN RTE                  512    LEFT   510    USED     2
ATTENDANT CONSOLES       100    LEFT    99    USED     1
IP ATTENDANT CONSOLES   2000    LEFT  1999    USED     1
BRI DSL                10000    LEFT 10000    USED     0
MPH DSL                  100    LEFT   100    USED     0
DATA PORTS              2000    LEFT  2000    USED     0
PHANTOM PORTS           2000    LEFT  1995    USED     5
TRADITIONAL TRUNKS      2000    LEFT  1962    USED    38
ELC ACCESS PORTS        2000    LEFT  2000    USED     0
DCH                      255    LEFT   252    USED     3
```

5.2. Configuring a SIP Connection on CS1000E

To configure the SIP connection there are a number of steps.

- Create a D-channel for the SIP trunk
- Create Route Data Block
- Add TIE Trunks

5.2.1. Create a D-Channel

Use the **CHG** command in **LD 17** to create a D-channel for the SIP connection. In the example below, D-Channel 66 (i.e. **DCH 66**) was created. At the **CTYP** prompt, enter **DCIP.** This signifies the SIP D-Channel.

LD 17

Prompt	Response	Description
>	**LD 17**	Enter Overlay 17
REQ	**CHG**	Change
TYPE	ADAN	Change the Action Device and Number
ADAN	NEW	Create New Action Device and Number
TYPE	**DCH 66**	Create new D-Channel 66
CTYP	**DCIP**	Card type is IP D-Channel
USR	ISDL	Integrated Services Digital Line
IFC	SL1	D-Channel interface type

5.2.2. Create Route Data Block

Use the **NEW** command in **LD 16** to create a Route Data Block. The route created is a **TIE** route in order to connect to the Trio system. Ensure **VTRK** is set to **YES** and **PCID** is **SIP.**

LD 16

Prompt	**Response**	**Description**
>	**LD 16**	Enter Overlay 16
REQ	**NEW**	Create new
TYPE	RDB	Route Data block
CUST	0	Customer Number as defined in LD15
ROUT	**20**	Route Number
TKTP	**TIE**	Route Type
VTRK	**YES**	Virtual Route
PCID	**SIP**	Protocol ID for route
DTRK	NO	Digital Trunk Route
ISDN	YES	Integrated Services Digital Network
MODE	ISDL	mode of operation
IFC	SL1	Interface type
ACOD	8020	Access Code for trunk route

5.2.3. Adding TIE Trunks

Use the **NEW** command in **LD 14** to add (**IPTI**) **TIE** trunks to the new route created in **Section 5.2.2.** If adding multiple trunks for each route, use **NEW XX**, where XX is the number of trunks. In the example below **10** trunks were added.

LD 14

Prompt	**Response**	**Description**
>	**LD 14**	Enter Overlay 14
REQ	**NEW 10**	Create 10 New Trunks
TYPE	**IPTI**	IP TIE trunk
TN	96 0 3 0	Loop Shelf Card Unit
CUST	0	Customer Number as defined in LD15
RTMB	20 1	Route number and Member number

5.3. Configure a Coordinated Dialing Plan

In order to setup a Coordinated Dialing Plan (CDP) both a route list index and a CDP are added.

5.3.1. Create a Route List Index

Use the **NEW** command in **LD 86** to create a **RLI**. Enter the route (**ROUT)** that was created in **Section 5.2.2.**

LD 86

Prompt	Response	Description
>**LD 86**	Enter Overlay 86	
REQ	**NEW**	Create New
CUST	0	Customer Number as defined in LD15
FEAT	RLB	Route list Block
TYPE	**RLI**	Route list Index
RLI	36	Route list Index number
ENTR	0	First entry for the RLI
ROUT	20	Enter the route number

5.3.2. Create CDP

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 **4000**, **FLEN** is **4** and the **RLI** is **36**.

Note: The RLI number used is the one created in **Section 5.3.1**.

LD 87

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	**4000**	Distant Steering code
FLEN	4	Flexible Length number of digits
RLI	36	Route list index Number

5.4. Configure TR87 on CS1000E

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”

5.5. Configure Intercept Computer Update on CS1000E

Trio uses Intercept Computer Update (ICP) on the CS1000E to change the presence state of the phoneset. A physical port on the CS1000E must be configured for ICP along with the ICP configuration in the Customer Data Block.

5.5.1. 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	14	Data number
PORT	2	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

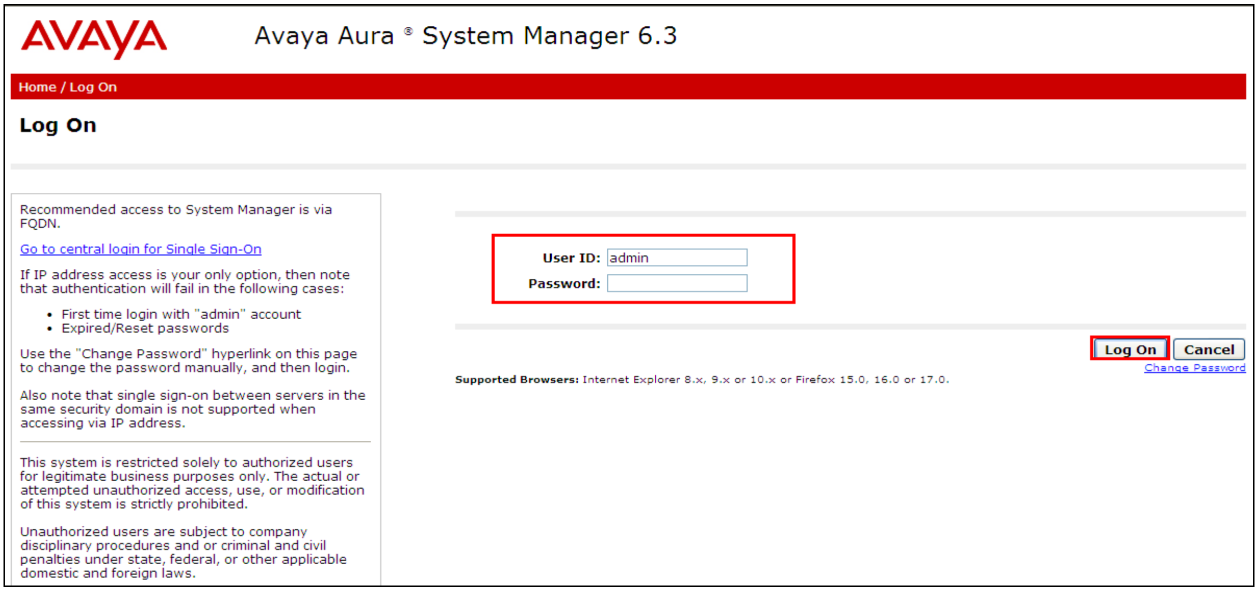
5.5.2. Configuration of ICP in the Customer Data Block

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.

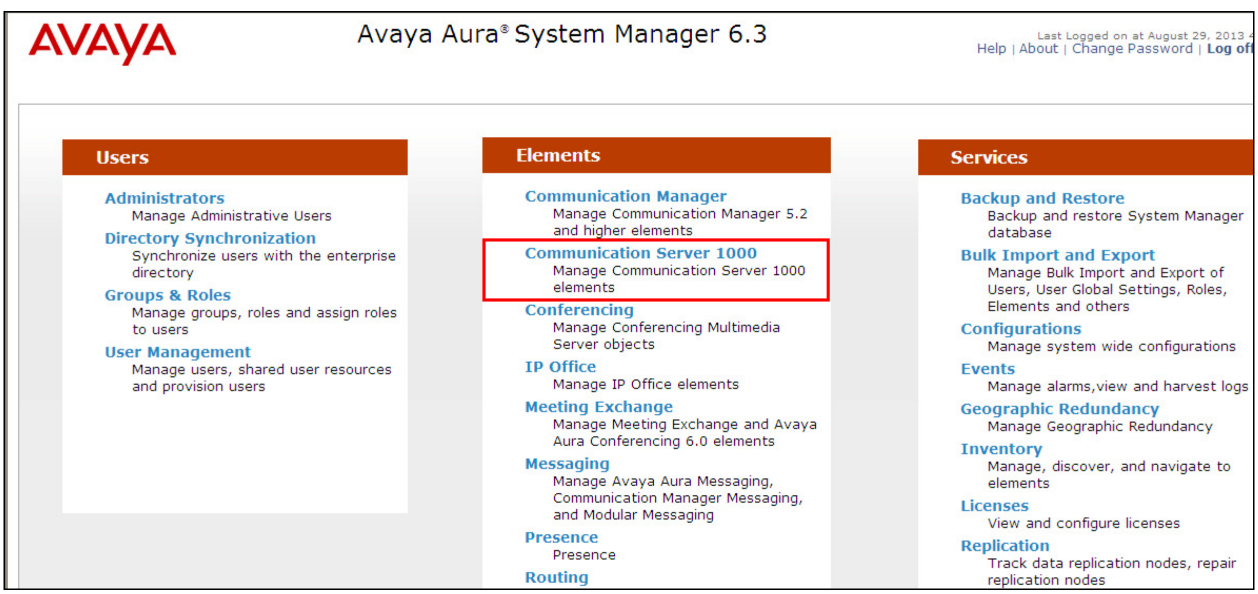
Prompt	Response	Description
>	**LD 15**	Enter Overlay 15
REQ	**CHG**	Change
TYPE	**icp**	Change the Intercept Computer Update
CUST	0	Customer Number
APL	14	Port number configured in **Section 5.5.1**
NIPN	9	Number of Intercept positions
ICCR	NO	Intercept Position Cancelling Reply
ICDN	4002	Internal Call DN
ECDN	4002	External Call DN
ICDL	4	CP DN Length
ICPD	0	ICP Padding Digit
ICTD	YES	Intercept Terminal Dail from Directory

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

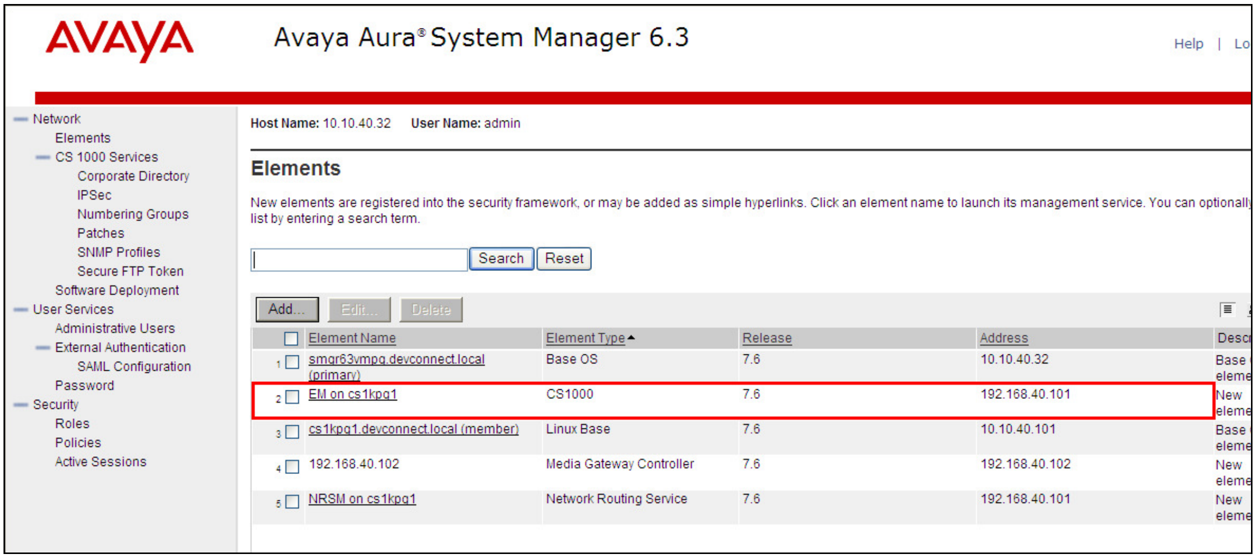
SIP CTI (TR/87) services must be enabled and configured on the CS1000E IP Telephony Node to allow applications obtain presence information or invoke a make call operation. Changes on the CS1000E Node are performed using Element Manager which is only accessible through the System Manager. To make changes 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.



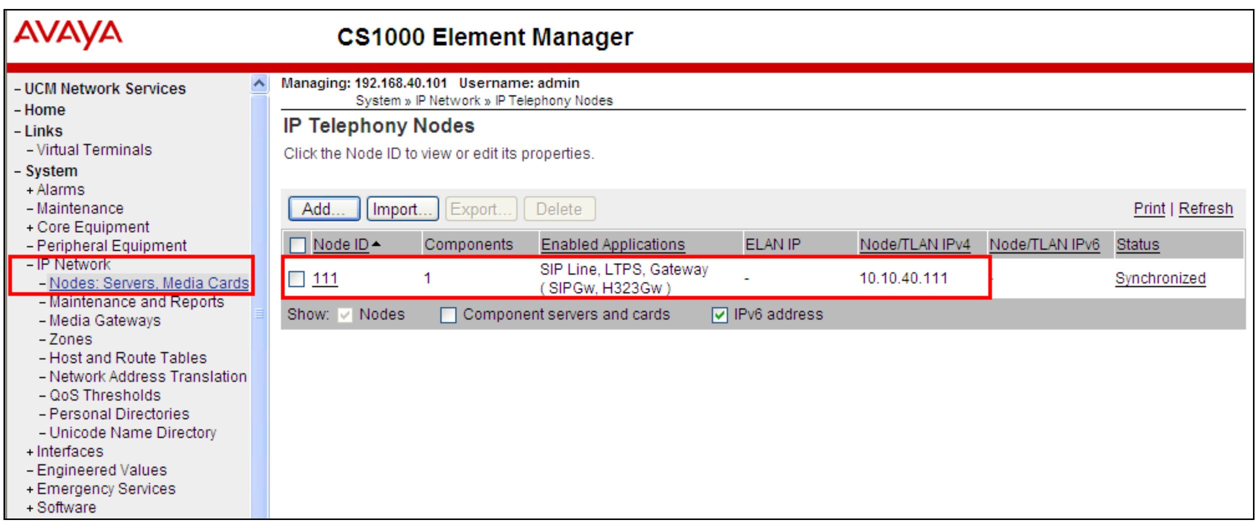
Click on **Communication Server 1000** as shown.



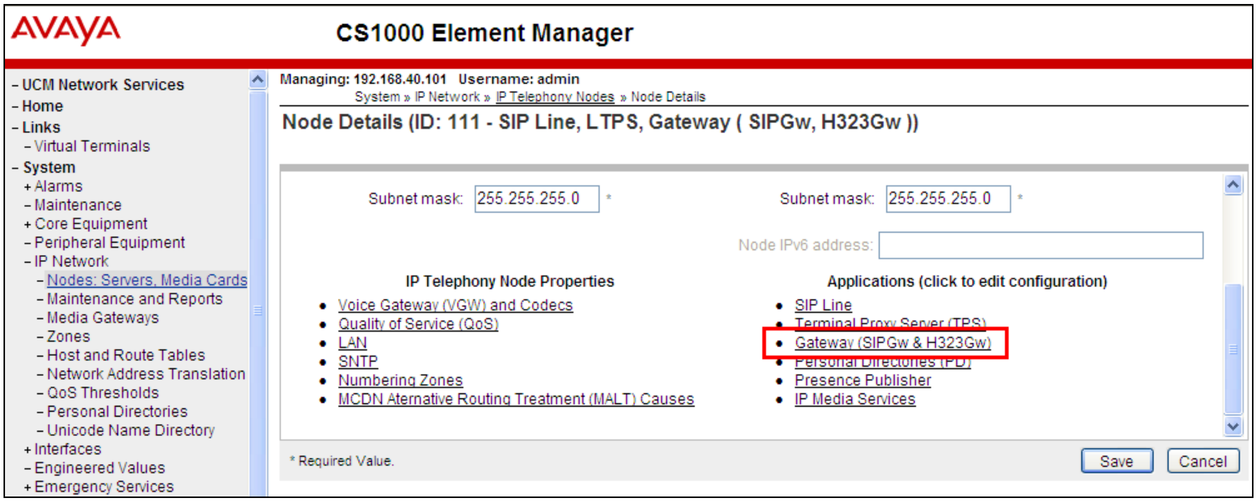
Once **Communication Server 1000** is selected the following screen appears, click on the Element Manager link highlighted below.



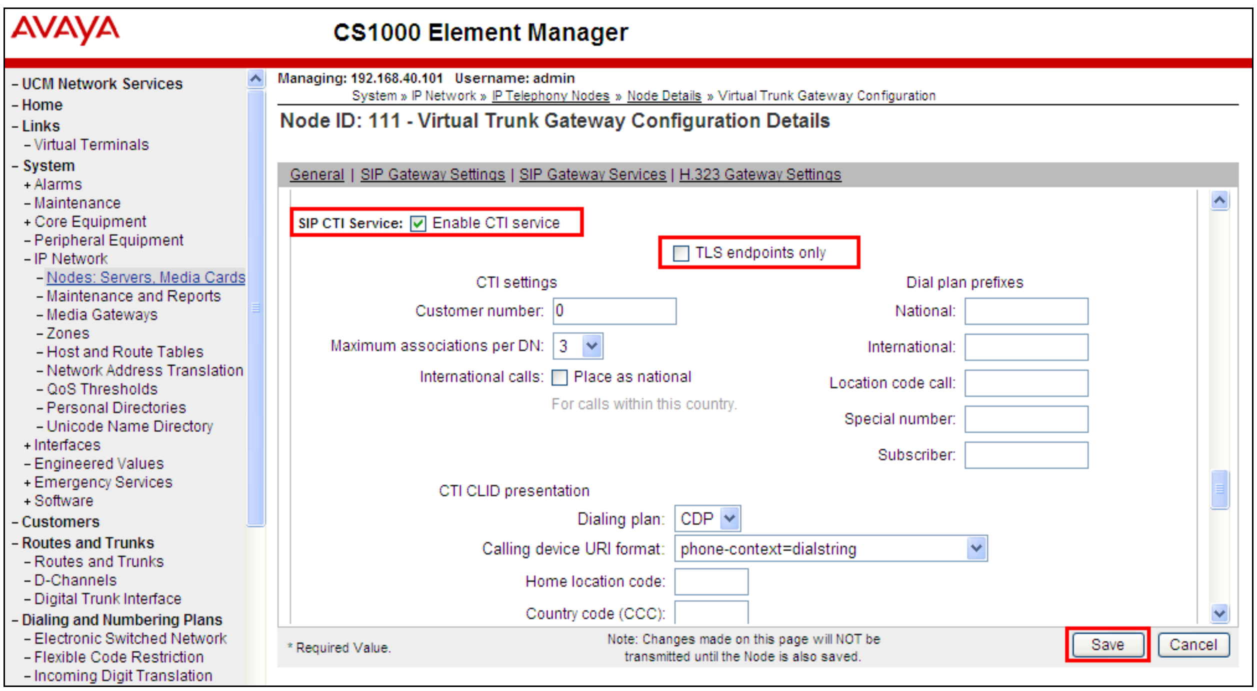
Click on **IP Network →Nodes: Servers, Media Cards** in the left window. Click on the **Node ID** displayed in the right window. Note the IP address of this node as it will be required in **Section 7.7**.



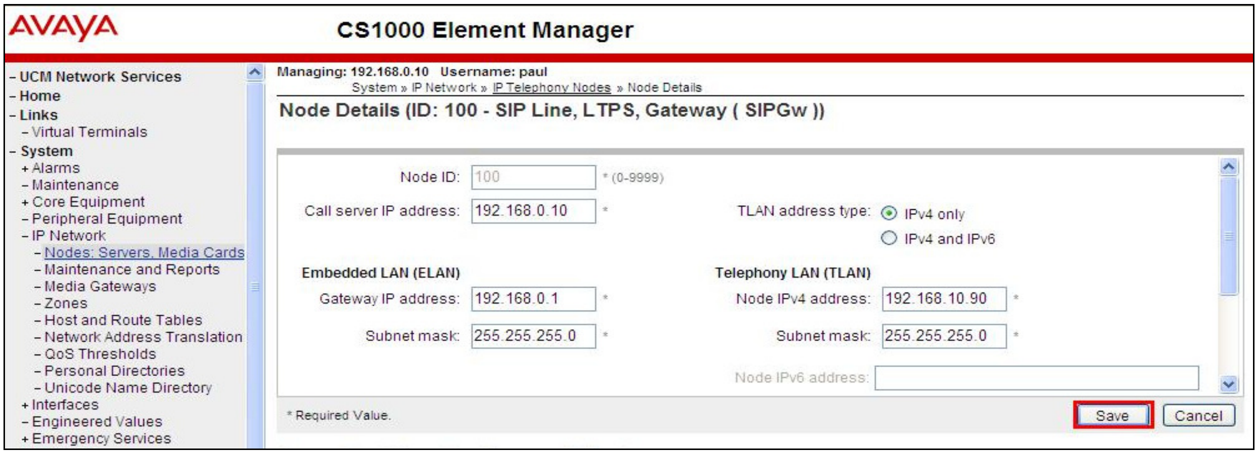
Select **Gateway (SIPGw&H323Gw)** highlighted below.



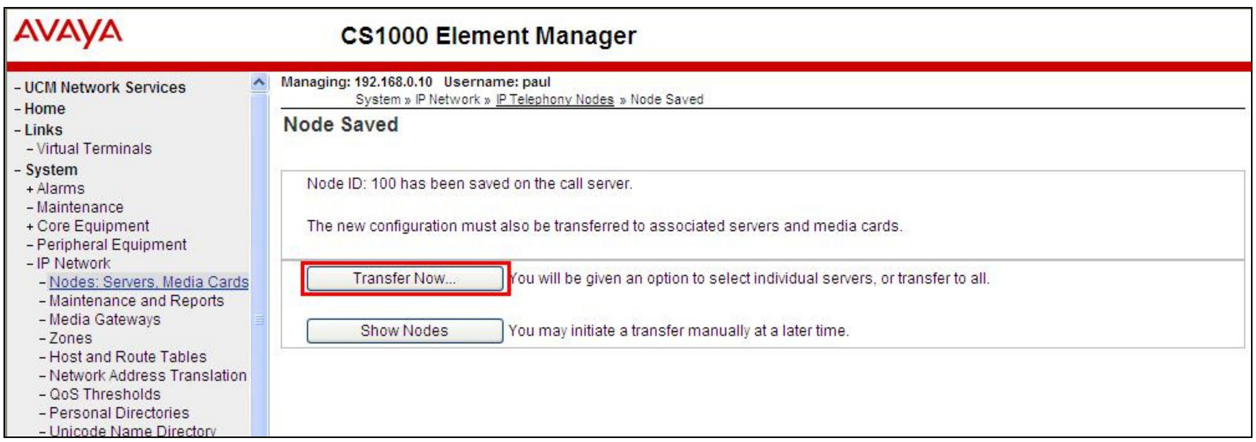
Ensure that **SIP CTI Service** is ticked as shown below and untick the **TLS endpoints only** (if this is ticked), everything else can be left as default. Click on **Save** once finished.



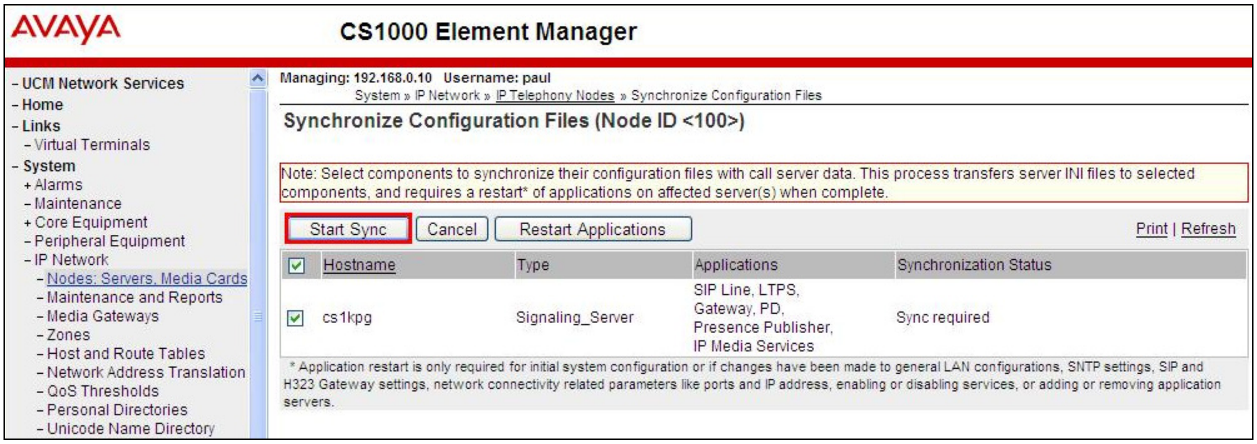
Once **Save** is clicked the following screen appears, click on **Save** as shown below.



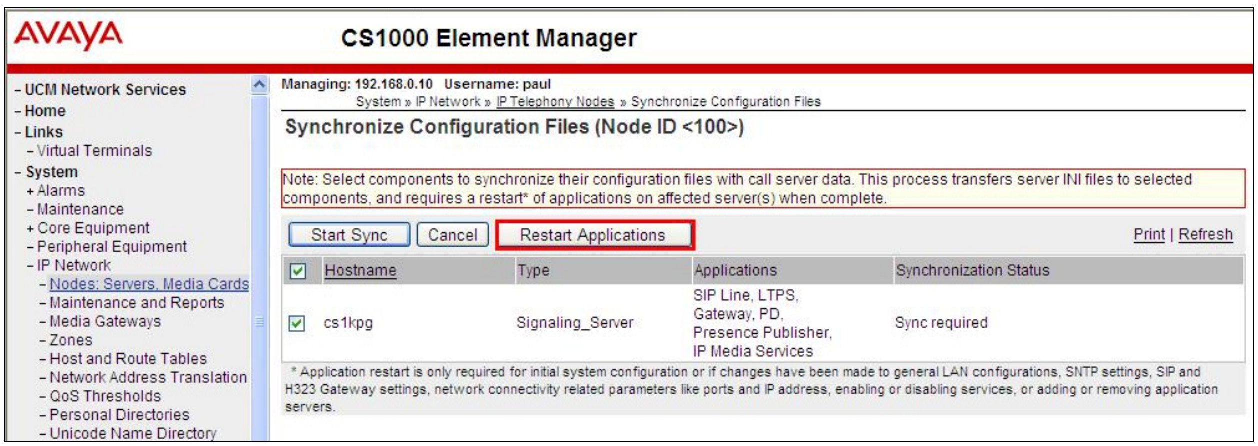
Once **Save** is clicked the following screen appears. Select **Transfer Now** as highlighted below.



Once the information is transferred over then the components need to synchronise their configuration files with the CS1000E call server. Select the **Hostname** as shown below and click on **Start Sync**.



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

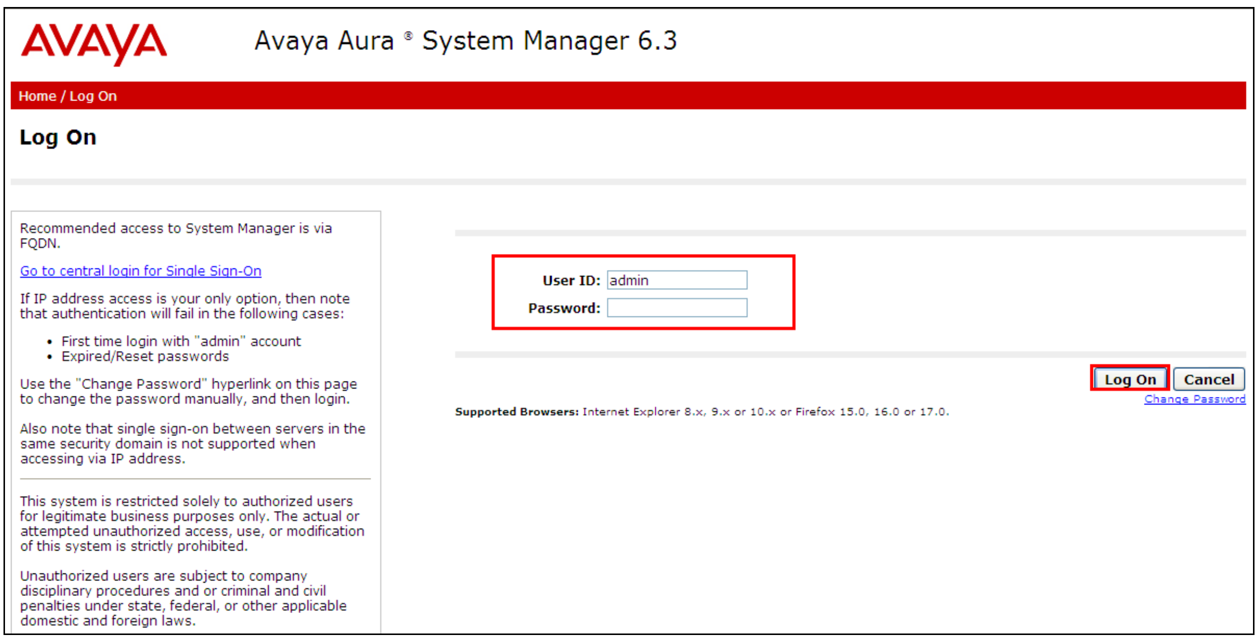


7. Creating SIP Entities on the Avaya Aura® Session Manager

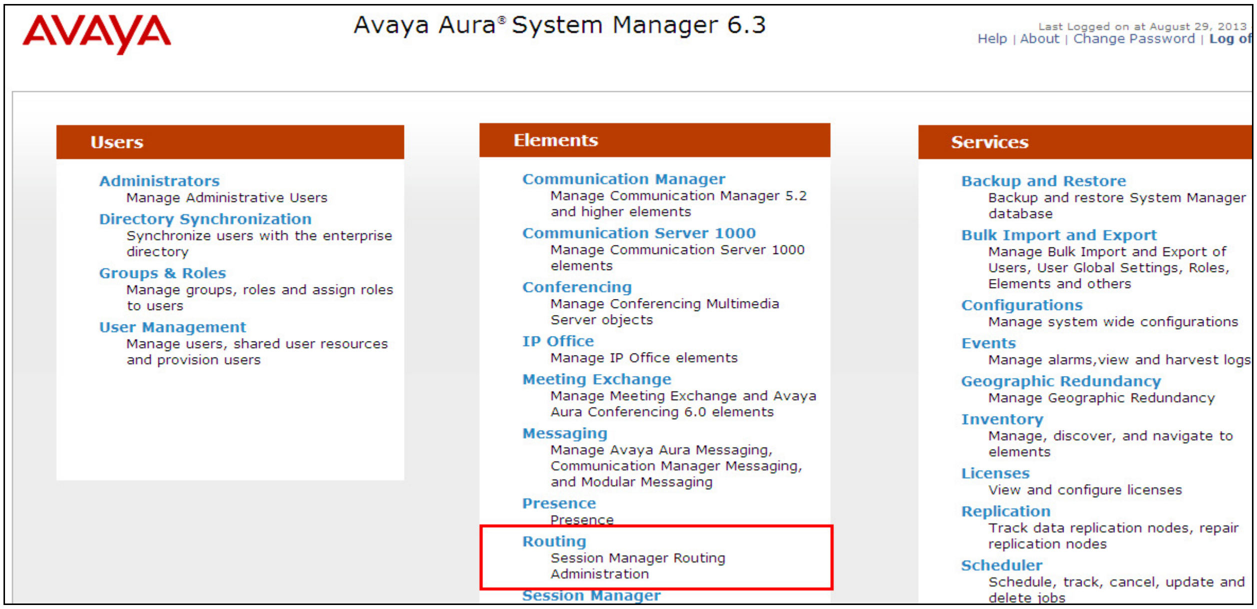
To create the Trio Enterprise Server as a SIP Entity on the Session Manager, the following must be configured.

- Create a SIP Domain
- Create a SIP location
- Create a SIP Entity
- Create an Entity Link
- Create a Routing Policy
- Create a Dial Pattern

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** highlighted below.

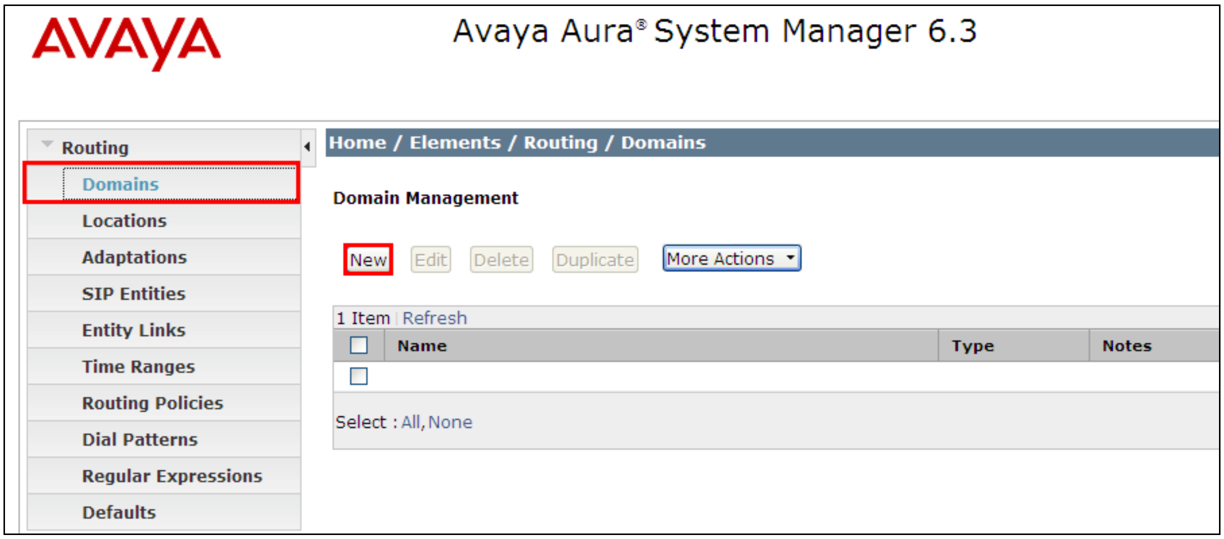


Once logged in select the **Routing** link under the **Elements** column.

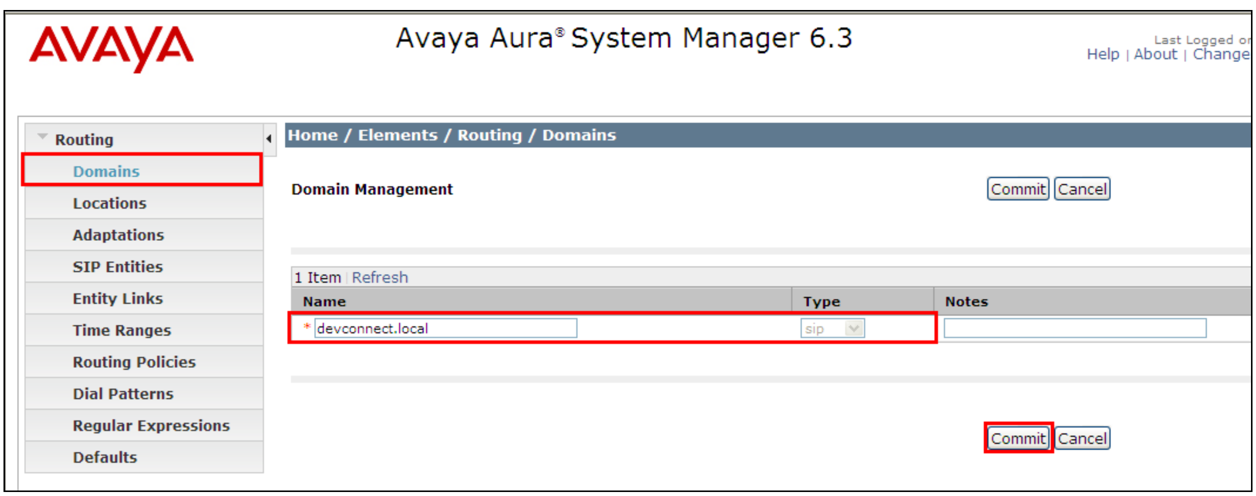


7.1. Administer SIP Domain

SIP domains are created as part of the Session Manager basic configuration. There will be at least one for which System Manager is the authoritative SIP controller. In these sample notes it is **devconnect.local**. In the left column click on **Routing → Domains**, click on **New** in the main window.

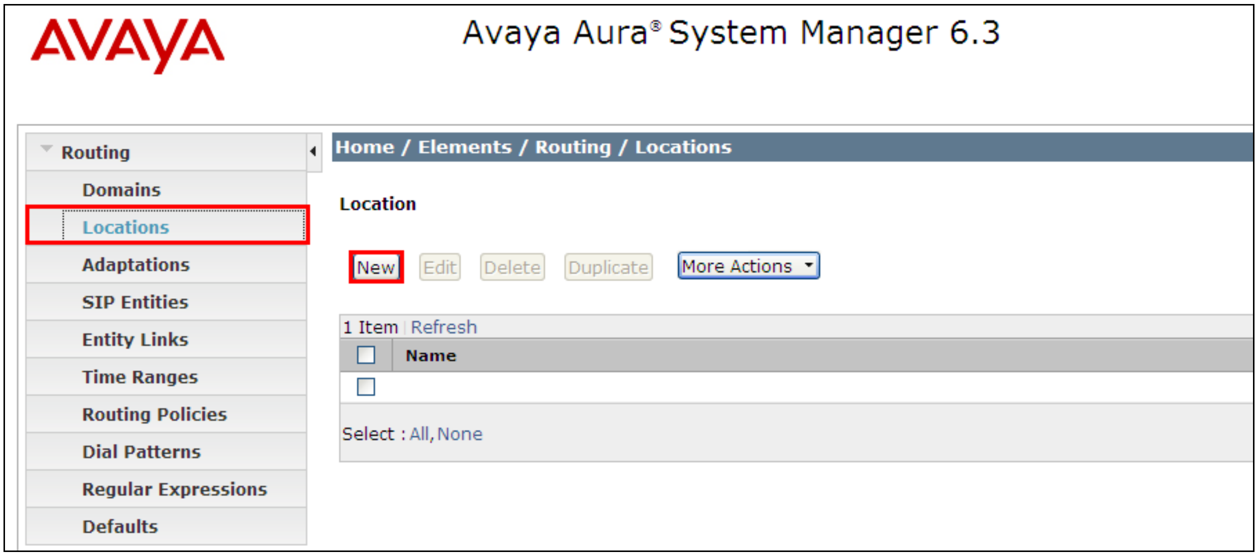


Enter the domain **Name,** set the **Type** as **sip** and click **Commit**.

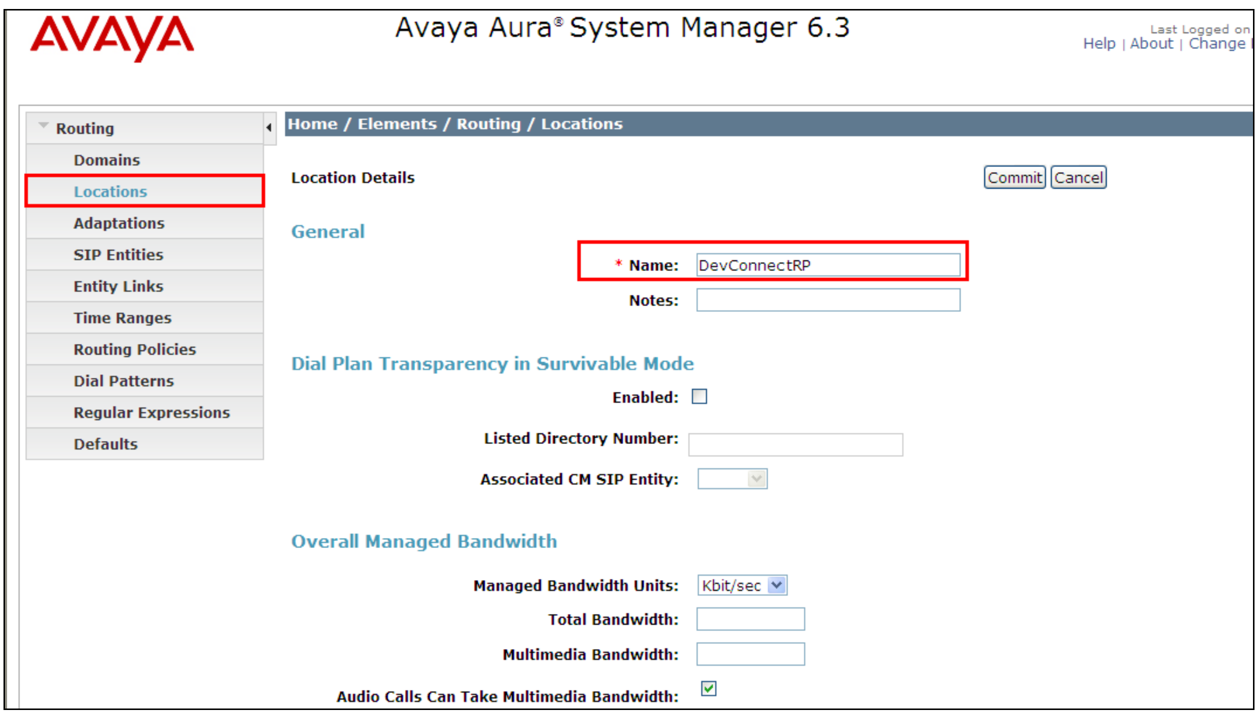


7.2. Administer Location

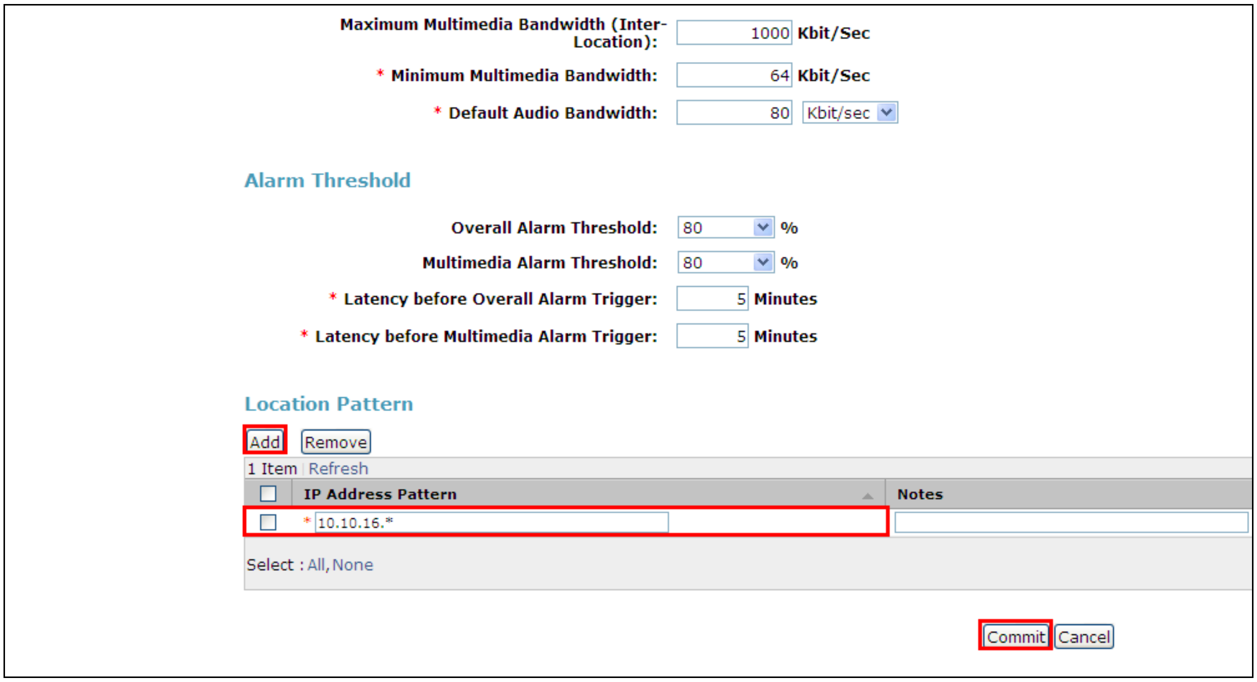
Session Manager uses the origination location to determine which dial patterns to look at when routing a call. In the left window under **Routing** select **Locations**, in the main window click on **New**.



Enter an identifying **Name**, as shown below.

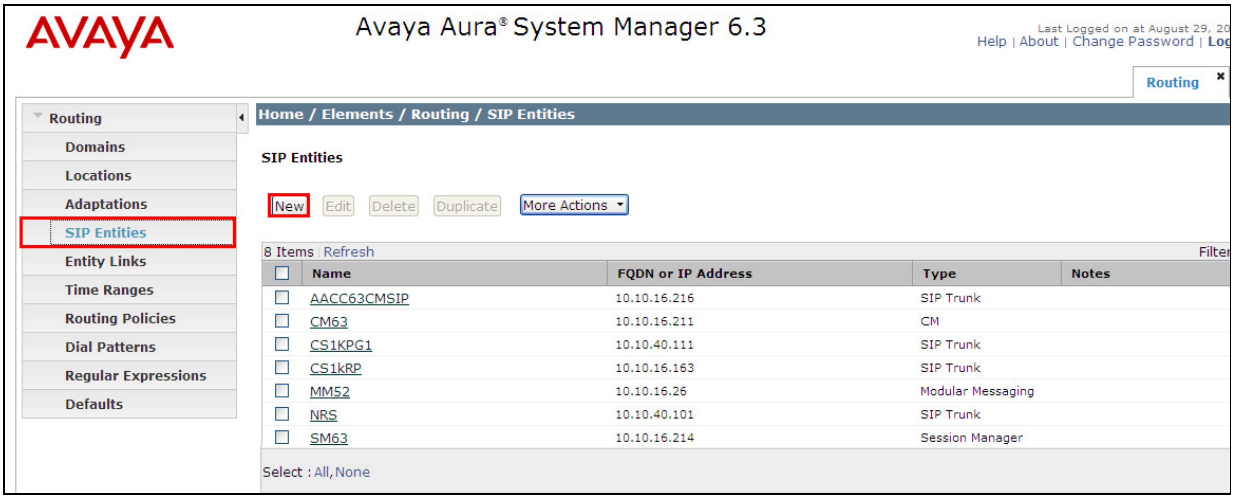


Scroll down to the bottom of the same page where the **Location Pattern** is defined. Click **Add** and enter the IP address range used to logically identify the location. In this case the **IP Address Pattern** is **10.10.16.*** as shown below. Click **Commit** when done.



7.3. Create Trio Enterprise SIP Entity

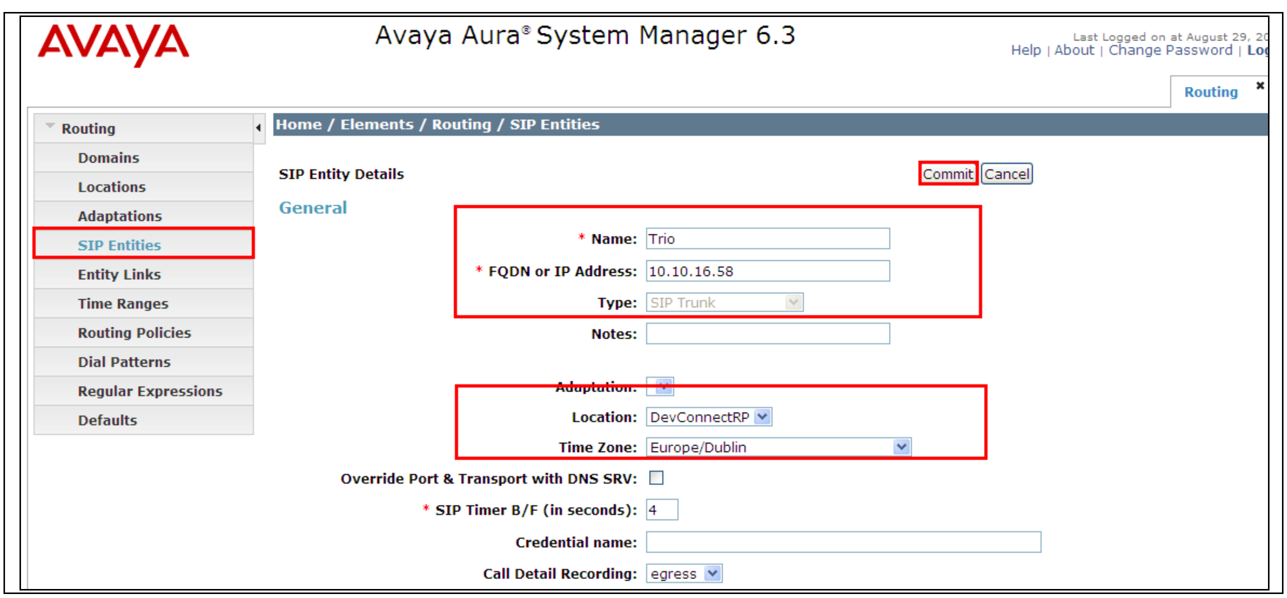
A SIP Entity must be added for Trio Enterprise Endpoint. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button in the main window.



Enter the following for the Trio SIP Entity.

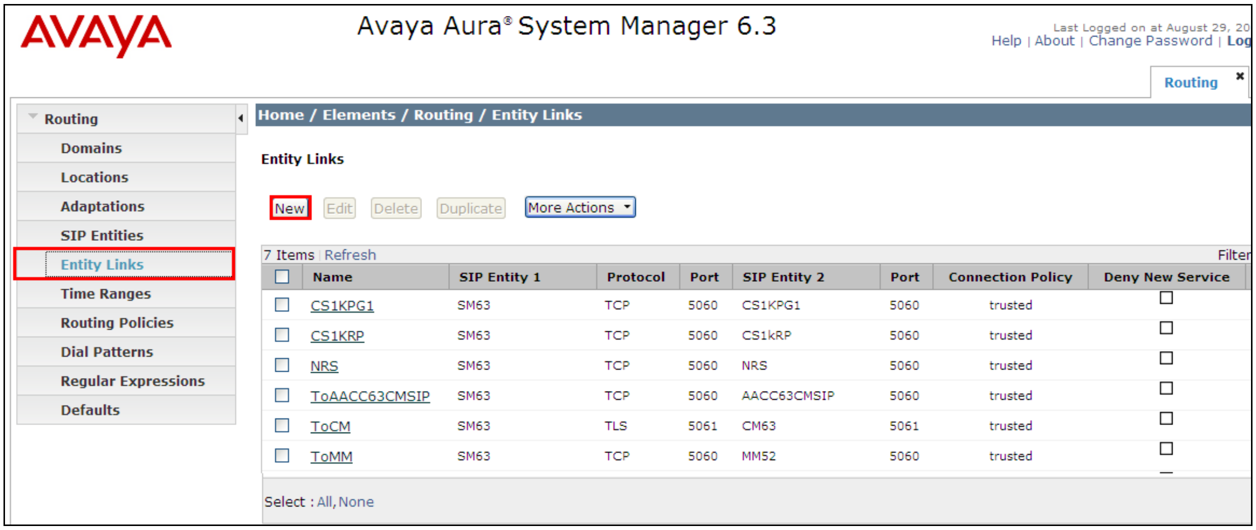
- **Name** An informative name (e.g., **Trio**)
- **FQDN or IP Address** IP address of the signalling interface on the Trio Enterprise
- **Type** **SIP Trunk** for Trio
- **Location** Location created in **Section 7.2**
- **Time Zone** Time zone for this location **Europe/Dublin**

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



7.4. Create Entity Link to Trio Enterprise

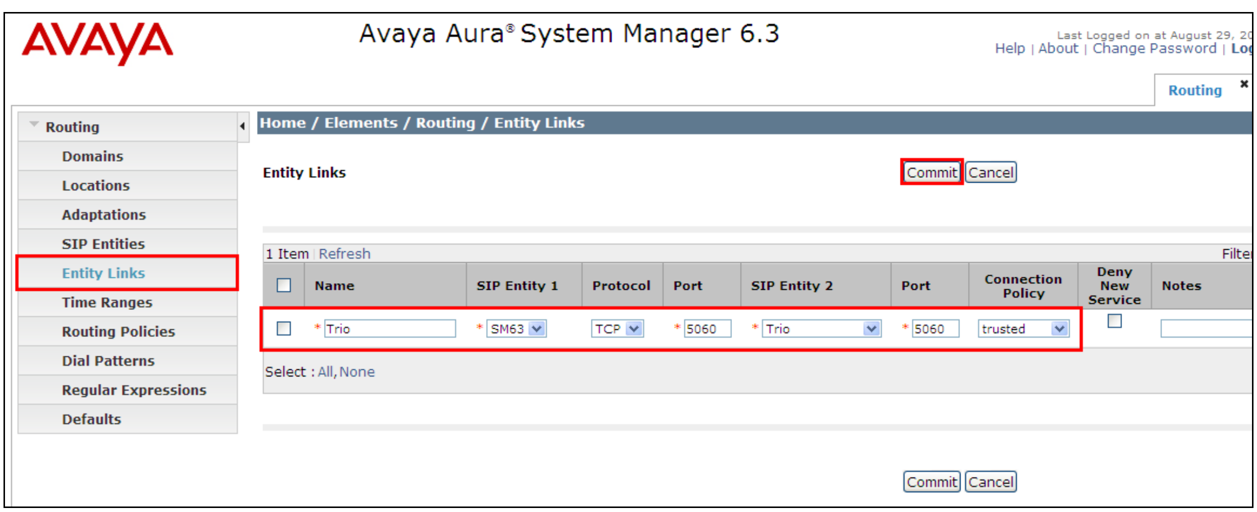
To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button in the main window.



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

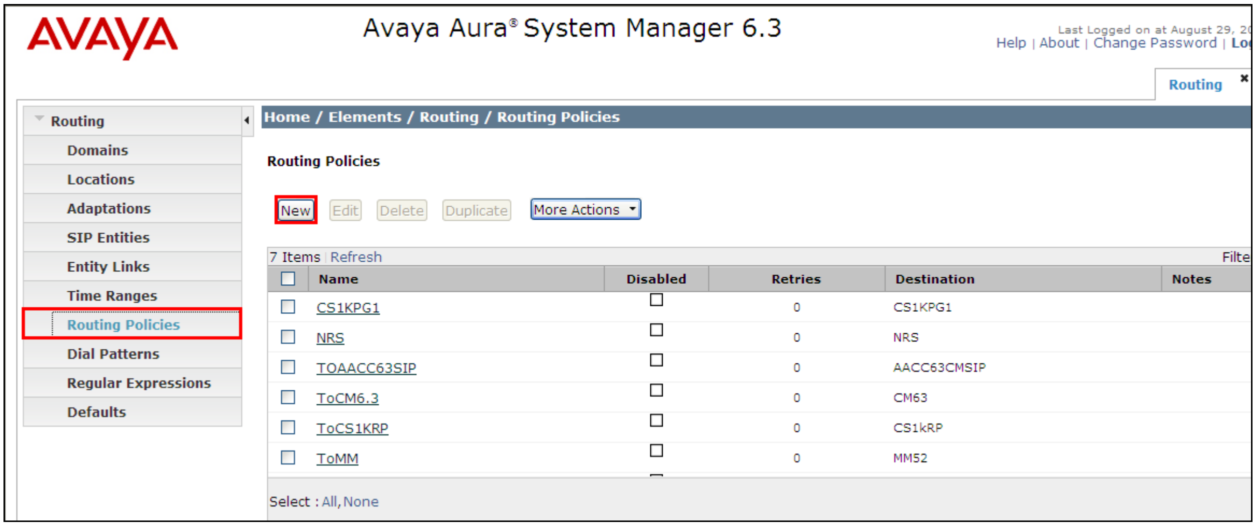
- **Name** An informative name, e.g. **Trio**
- **SIP Entity 1** Select **Session_Manager** from the **SIP Entity 1** dropdown box
- **Protocol** Select **TCP** from the Protocol
- **Port** Enter **5060** as the Port
- **SIP Entity 2** Select **Trio** from the **SIP Entity 2** dropdown box
- **Port**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

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

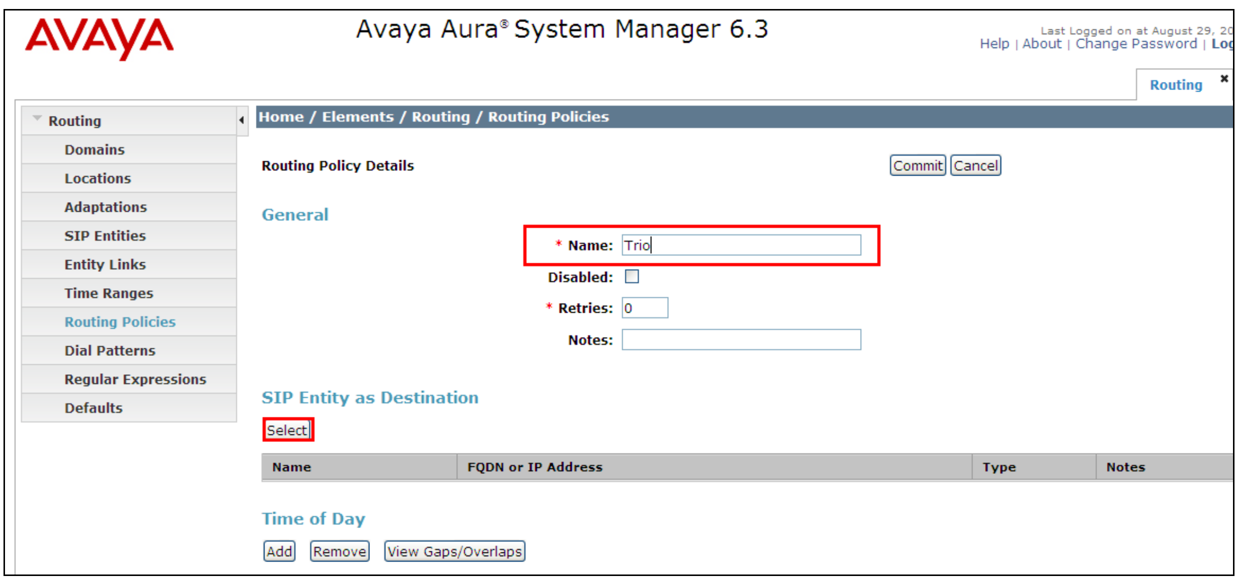


7.5. Create Routing Policy for Trio Enterprise

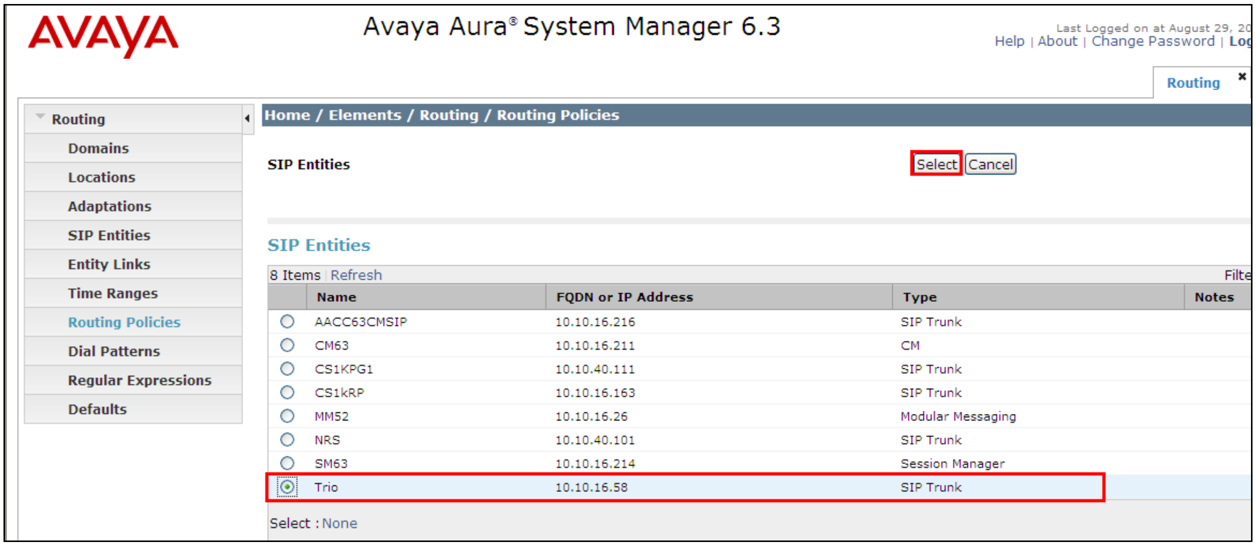
Create routing policies to direct calls to Trio Enterprise. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button in the main window.



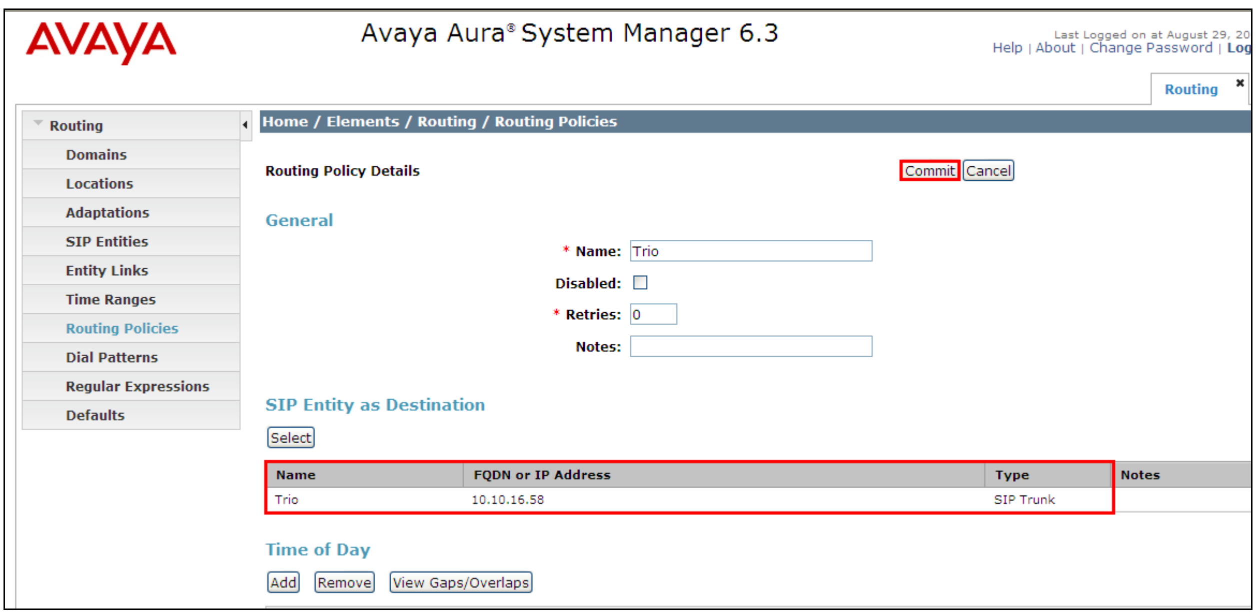
Enter an informative name, (e.g., **Trio**), under **SIP Entity as Destination,** click **Select**.



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.

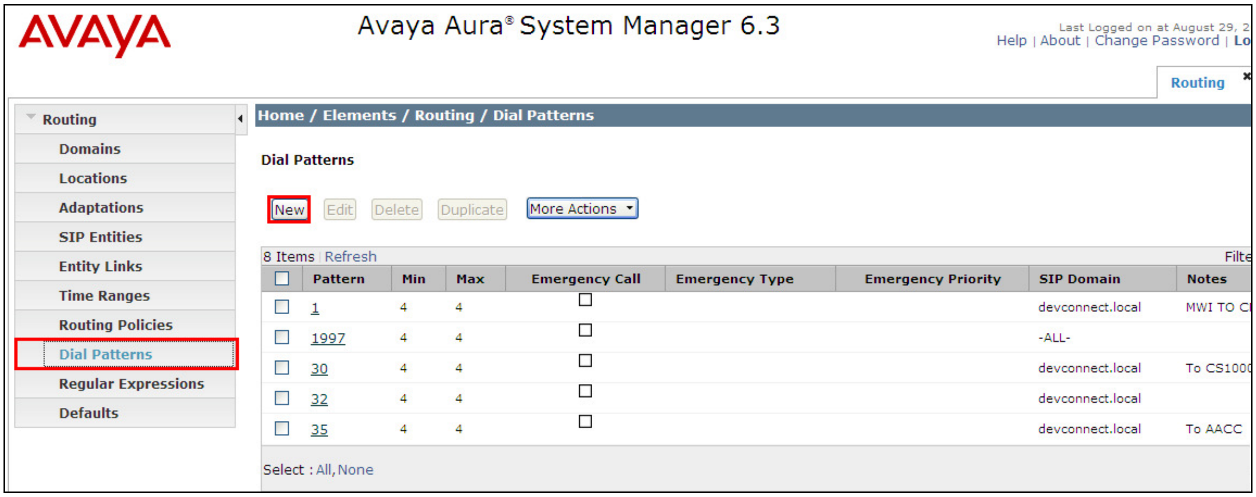


Click the **Commit** button to save. The following screen shows the **Routing Policy Details** for calls to Trio.



7.6. Create Trio Enterprise Dial Pattern

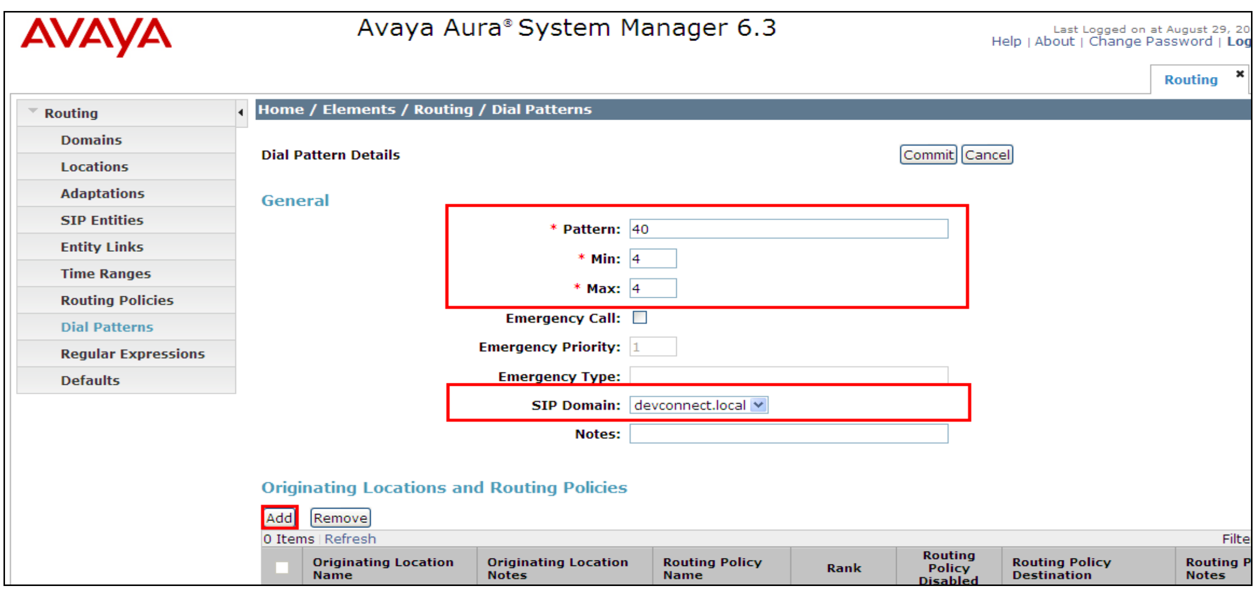
A dial pattern must be defined that will direct calls to Trio Enterprise. During testing there were several numbers used beginning with 40xx in conjunction with the domain name created in **Section 7.1** called devconnect.local. To configure the Trio Enterprise dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button in the main window.



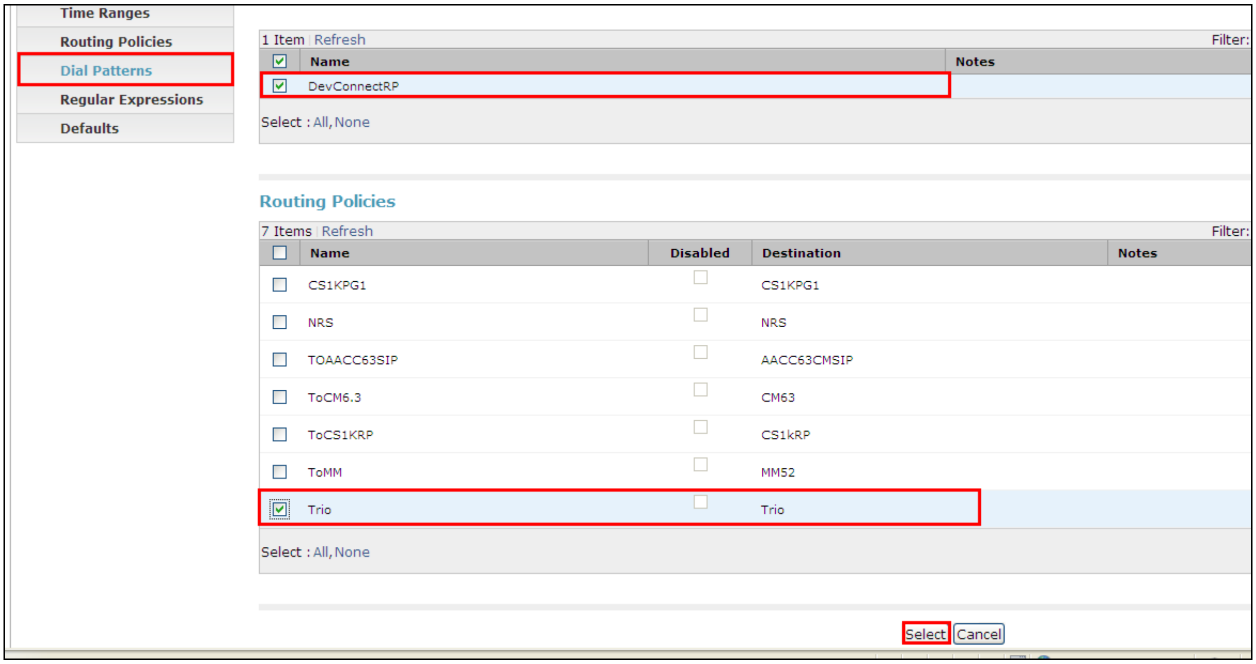
Enter the following information.

- **Pattern** Dialed number or prefix **40**
- **Min** Minimum length of dialed number **4**
- **Max** Maximum length of dialed number **4**
- **SIP Domain** Domain name of **devconnect.local** was used in the compliance testing (See **Section 7.1**)

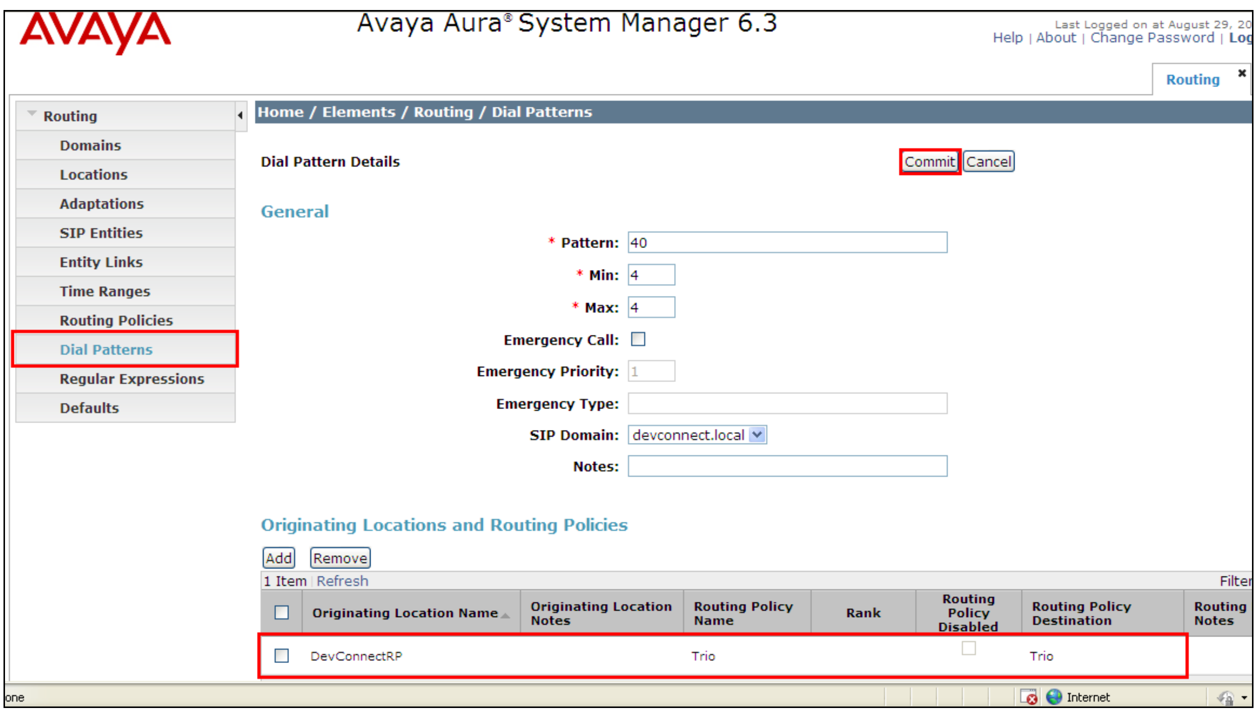
Under **Originating Locations and Routing Policy List,** click on **Add.**



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

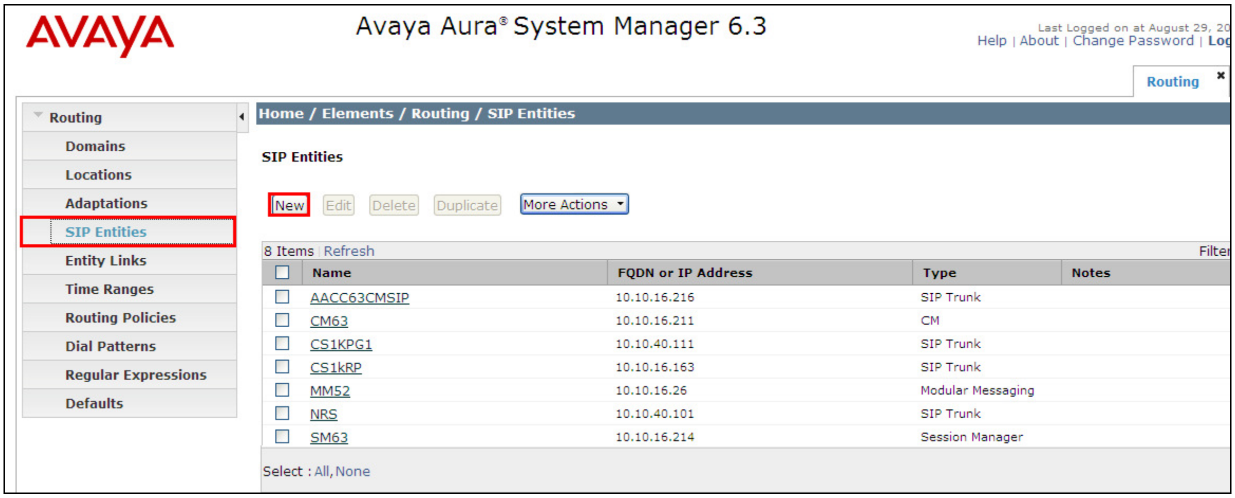


Click the **Commit** button to save.



7.7. Create CS1000E SIP Entity

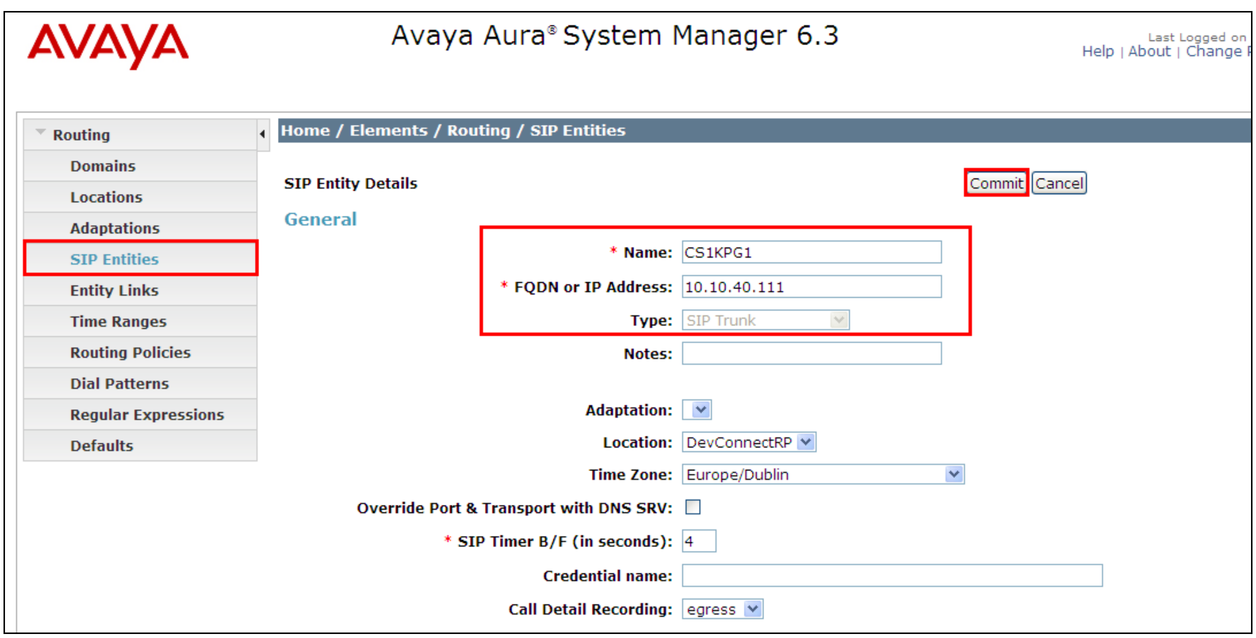
A SIP Entity may already be in place for the CS1000E but in the event there is none present the procedure is the same as that above for the Trio SIP Entity. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button in the main window.



Enter the following for the CS1000E SIP Entity.

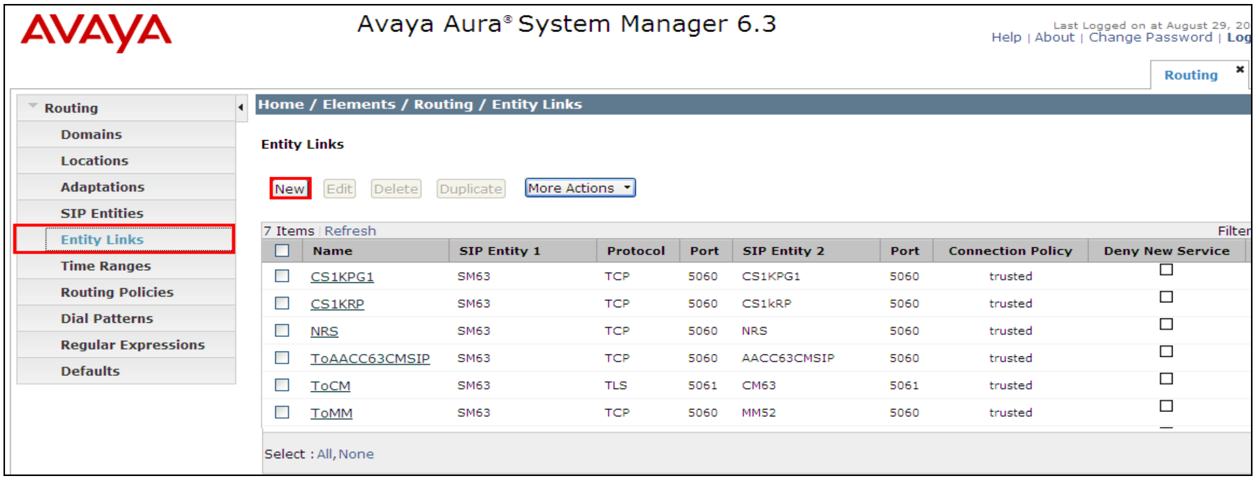
- **Name** An informative name (e.g., **CS1KPG1**)
- **FQDN or IP Address** CS1000E Node IP address as per **Section 6**
- **Type** **SIP Trunk** for the CS1000E
- **Location** Location created in **Section 7.2**
- **Time Zone** Time zone for this location **Europe/Dublin**

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



7.8. Create an Entity link for the CS1000E

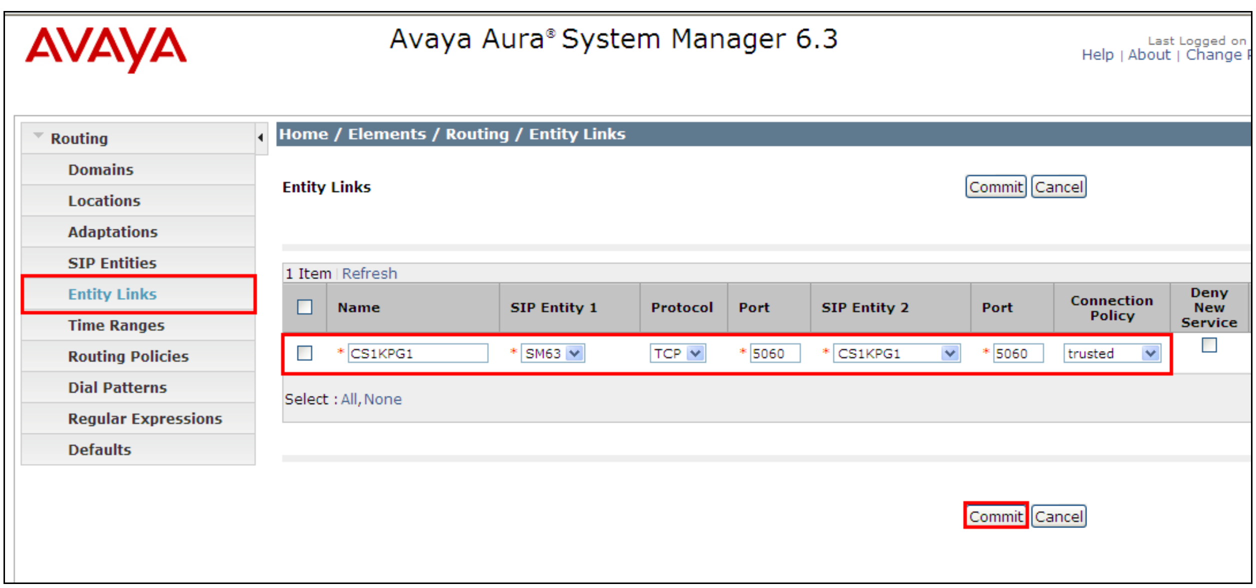
To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button in the main window.



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

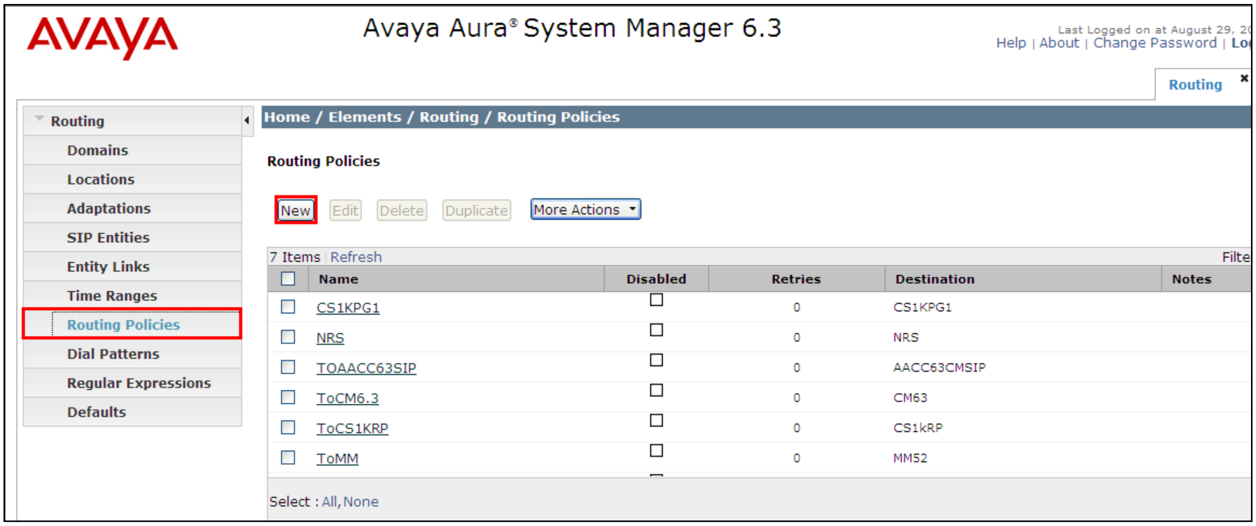
- **Name** An informative name, e.g. **CS1KPG1**
- **SIP Entity 1** Select **Session_Manager** from the **SIP Entity 1** dropdown box
- **Protocol** Select **TCP** from the Protocol
- **Port** Enter **5060** as the Port
- **SIP Entity 2** Select **Trio** from the **SIP Entity 2** dropdown box
- **Port** Enter **5060** as the Port
- **Trusted** Check the **Trusted** check box

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

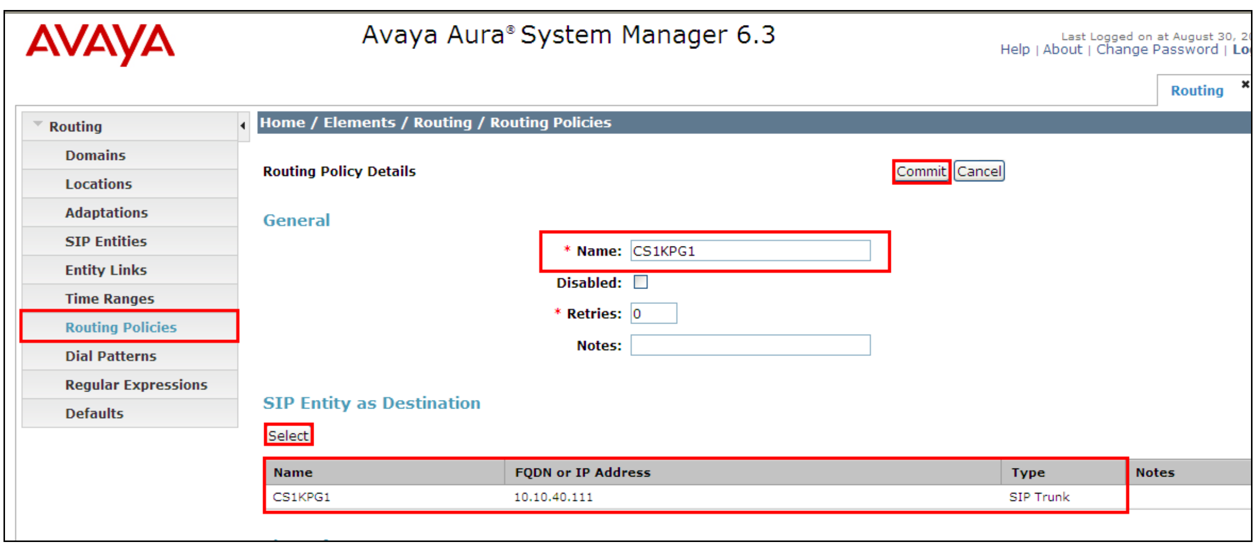


7.9. Create Routing Policy for CS1000E

Create routing policies to direct calls to the CS1000E. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button in the main window.

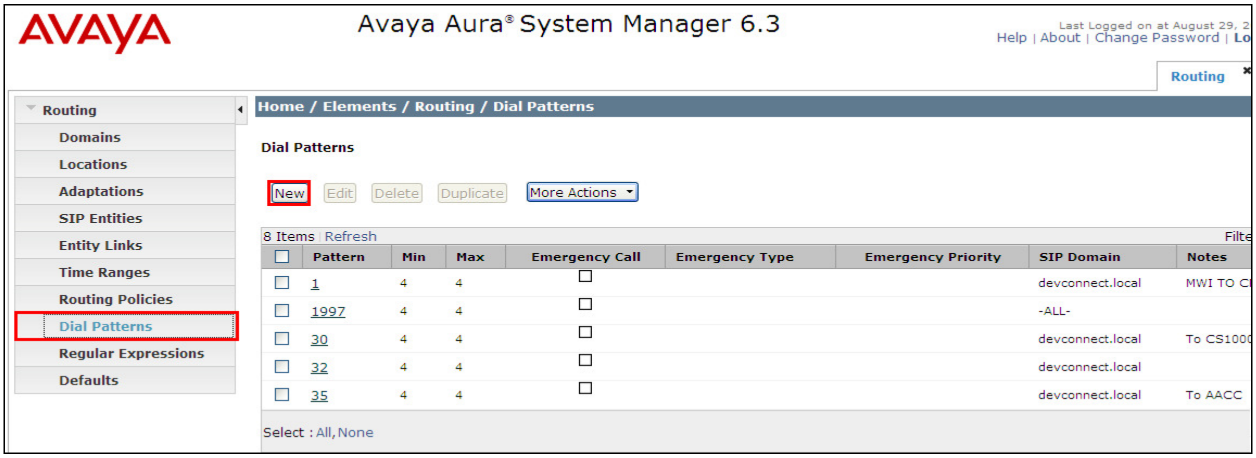


Enter an informative name, (e.g., **CS1KPG1**), under **SIP Entity as Destination,** click **Select**. Check the **CS1000E** radio button and click on the **Select** button to confirm the chosen options (not shown) and then be returned to the **Routing Policies Details** screen as shown below. Click the **Commit** button to save.



7.10. Create CS1000E Dial Pattern

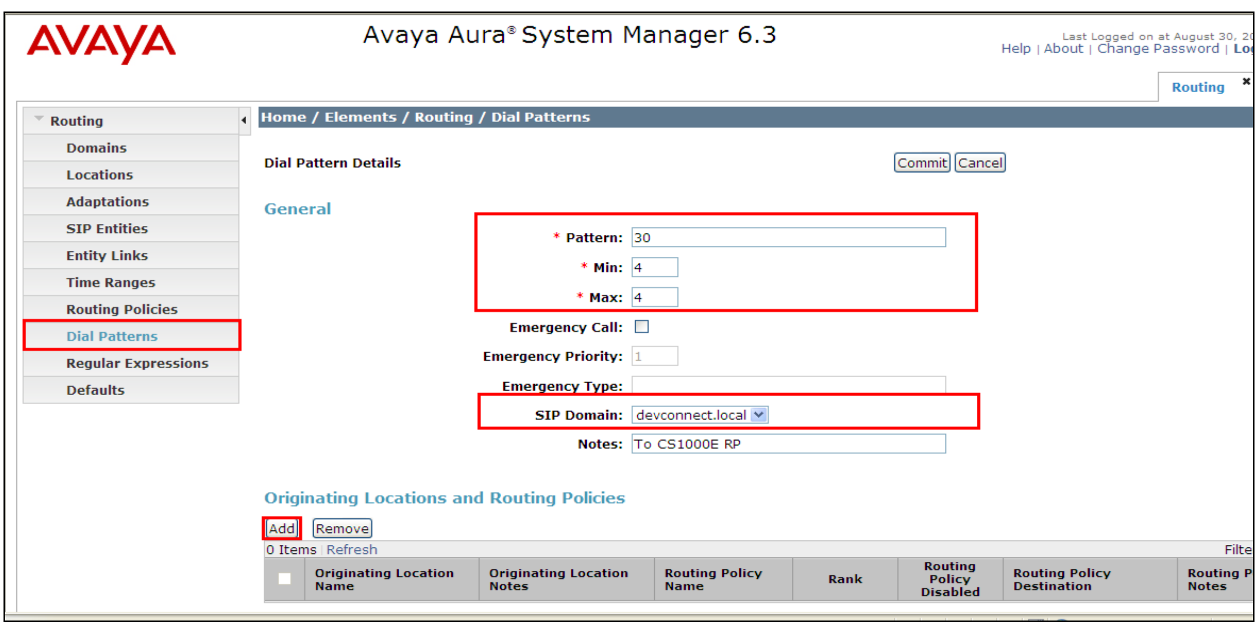
A dial pattern must be defined that will direct calls to the CS1000E. During testing there were several numbers used on the CS1000E beginning with 30xx. To configure the CS1000E dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button in the main window.



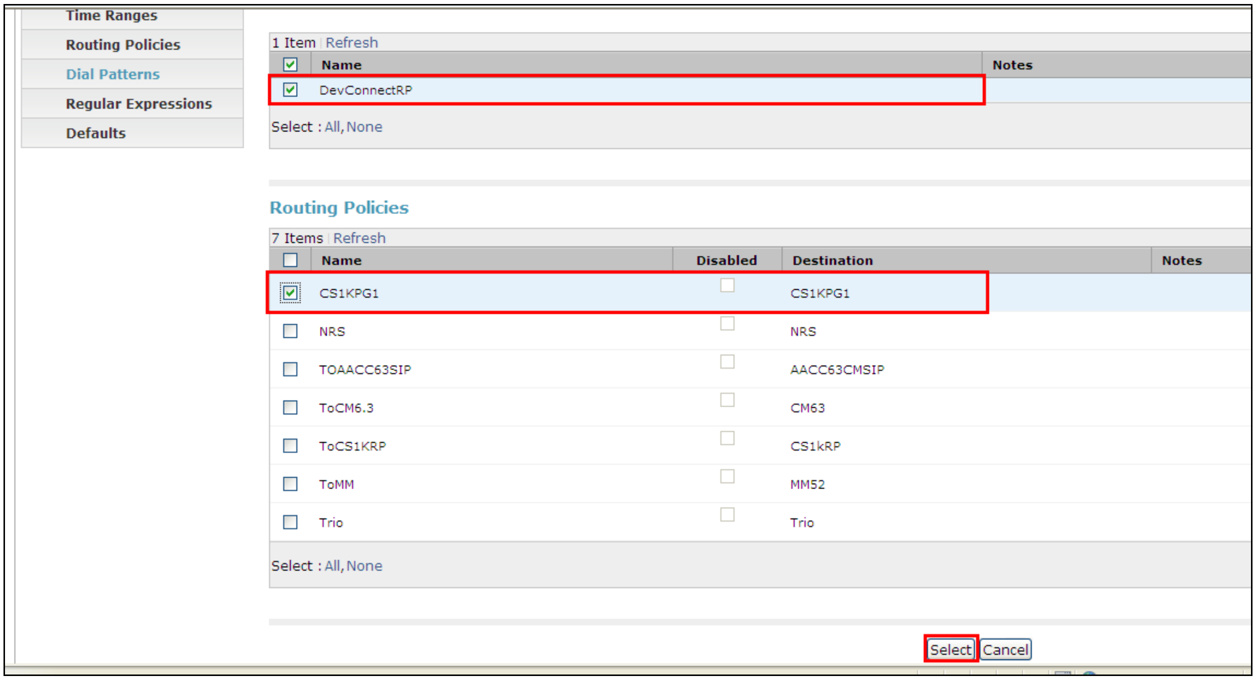
Enter the following information.

- **Pattern** Dialled number or prefix **30**
- **Min** Minimum length of dialled number **4**
- **Max** Maximum length of dialled number **4**
- **SIP Domain** Domain name of **devconnect.local** was used in the compliance testing (See **Section 7.1**)

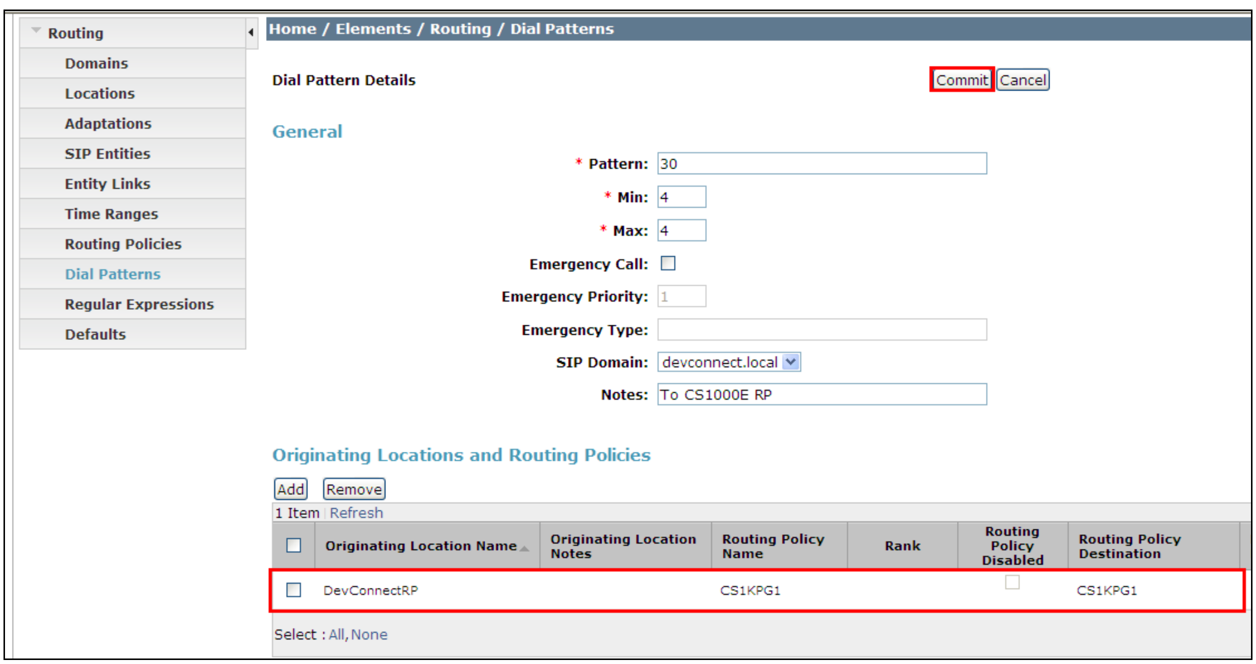
Under **Originating Locations and Routing Policy List,** click on **Add.**



Check the **DevconRP** check box (location setup in **Section 7.2**), and under **Routing Policies** check **CS1KPG1** 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.



8. Configure TRIO Enterprise

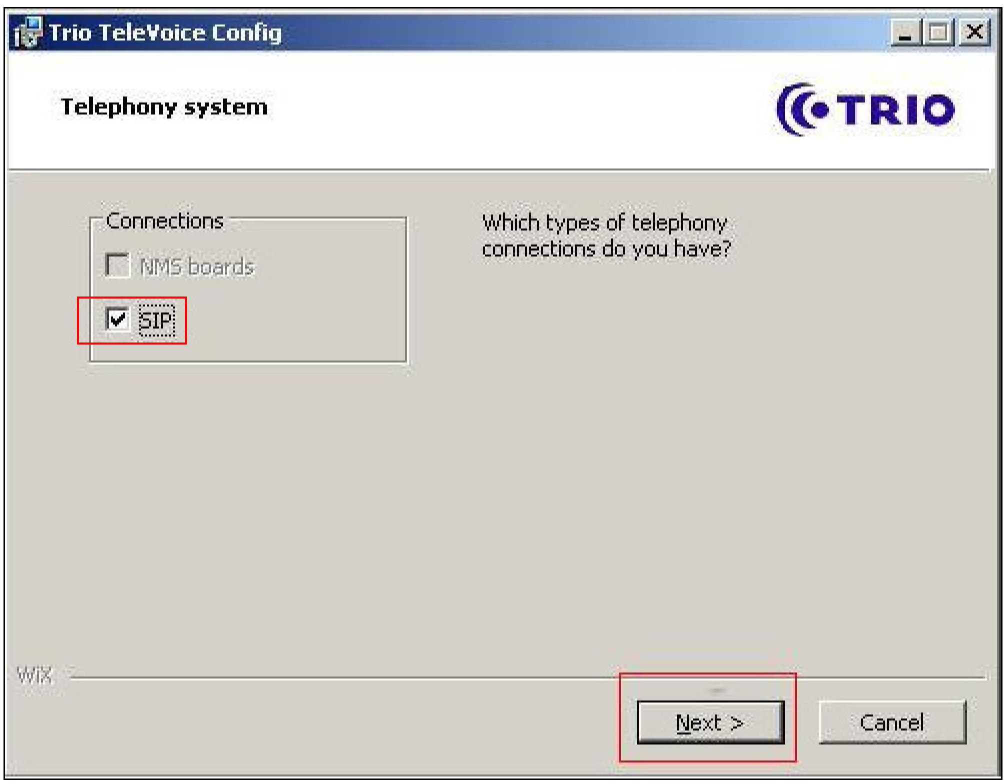
Trio Enterprise connects as a SIP endpoint to the CS1000E through Session Manager. Trio Enterprise is added to Session Manager as a SIP Entity and calls are routed to the Trio Enterprise server according to the Coordinated Dial Plan setup in **Section 5.3**. This section shows how to configure Trio Enterprise to successfully connect to the CS1000E 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

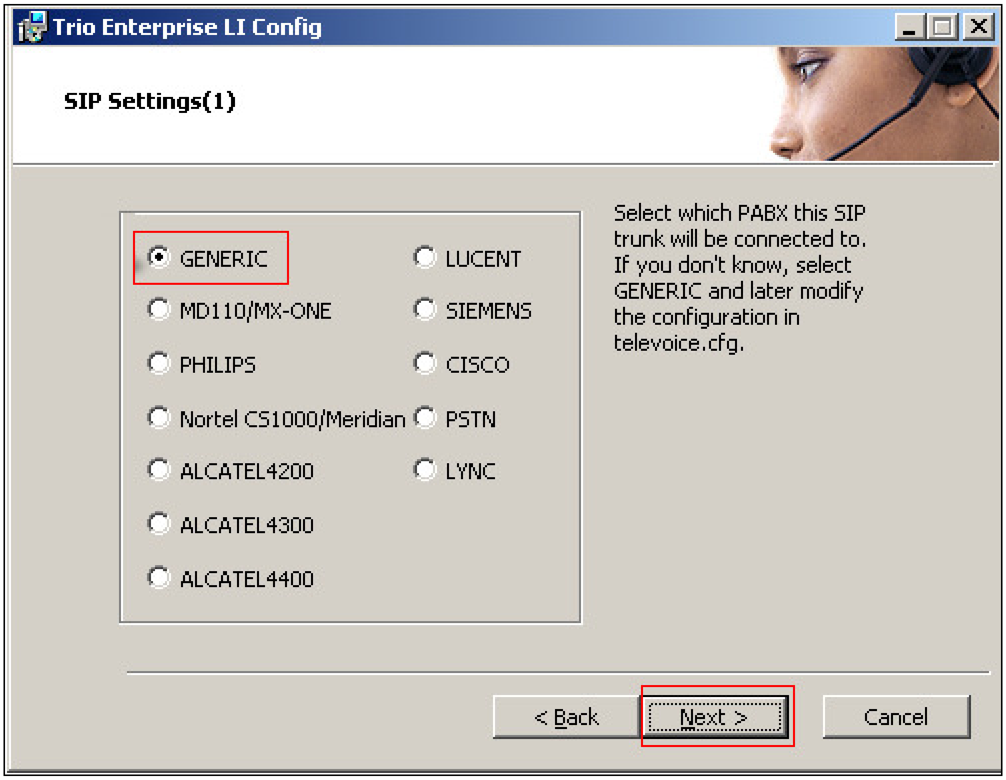
Access Windows services. Select **Start** → **Run**, then type **services.msc** into the command line and press return (not shown). When the services window opens, locate the **Trio Televoice** service, right click and select **stop** to stop the service (not shown).
Launch the Trio configuration application.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



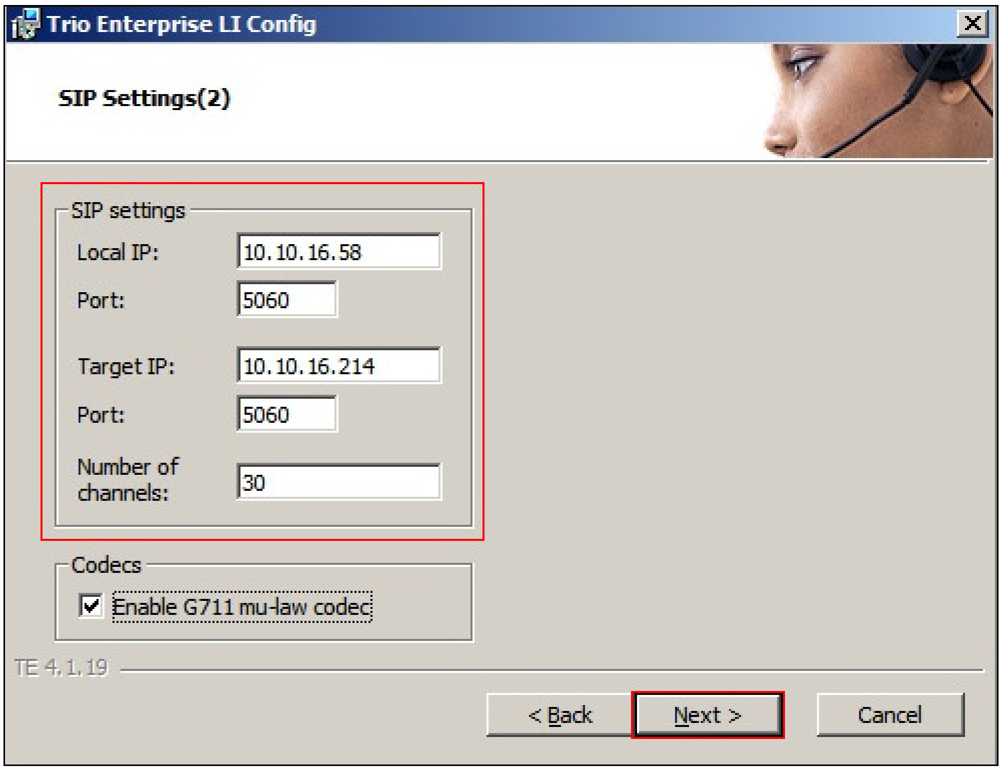
Select **GENERIC** under **SIP Settings**. Click **Next** to continue.



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

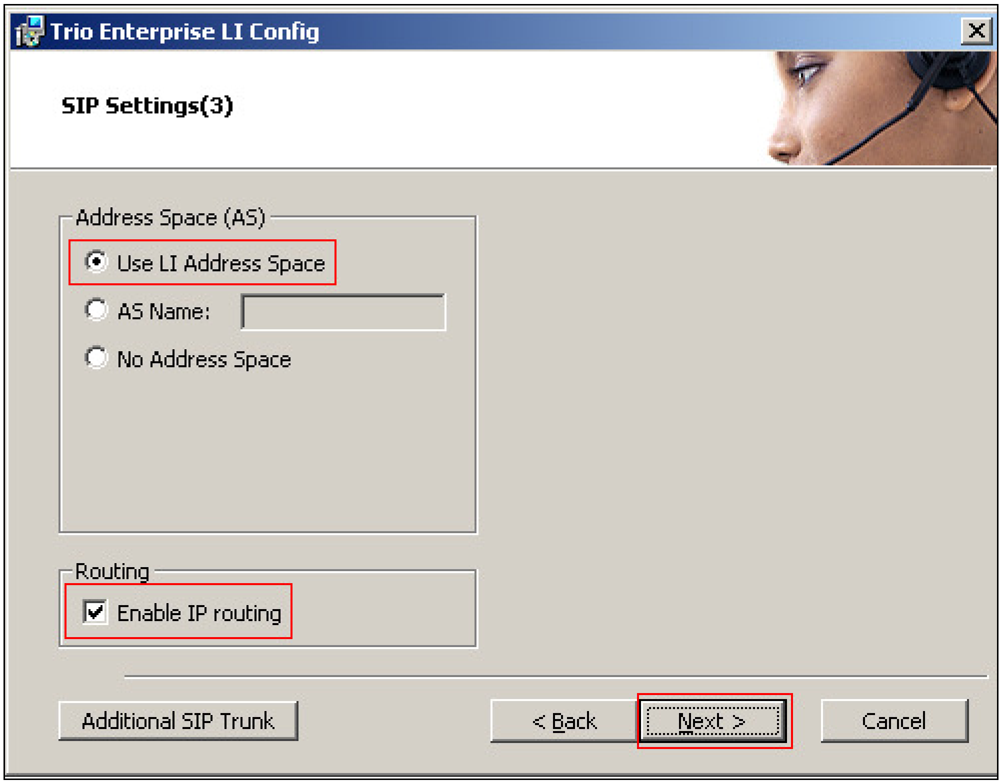
Click **Next** to continue.



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

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

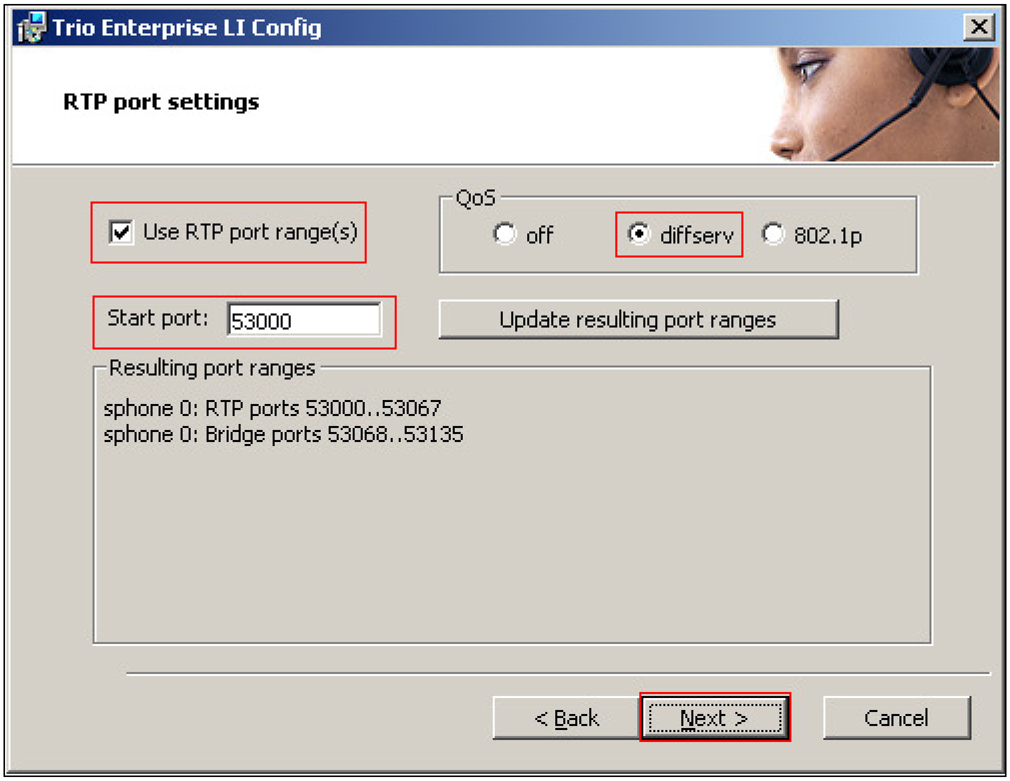
Click **Next** to continue.



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

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

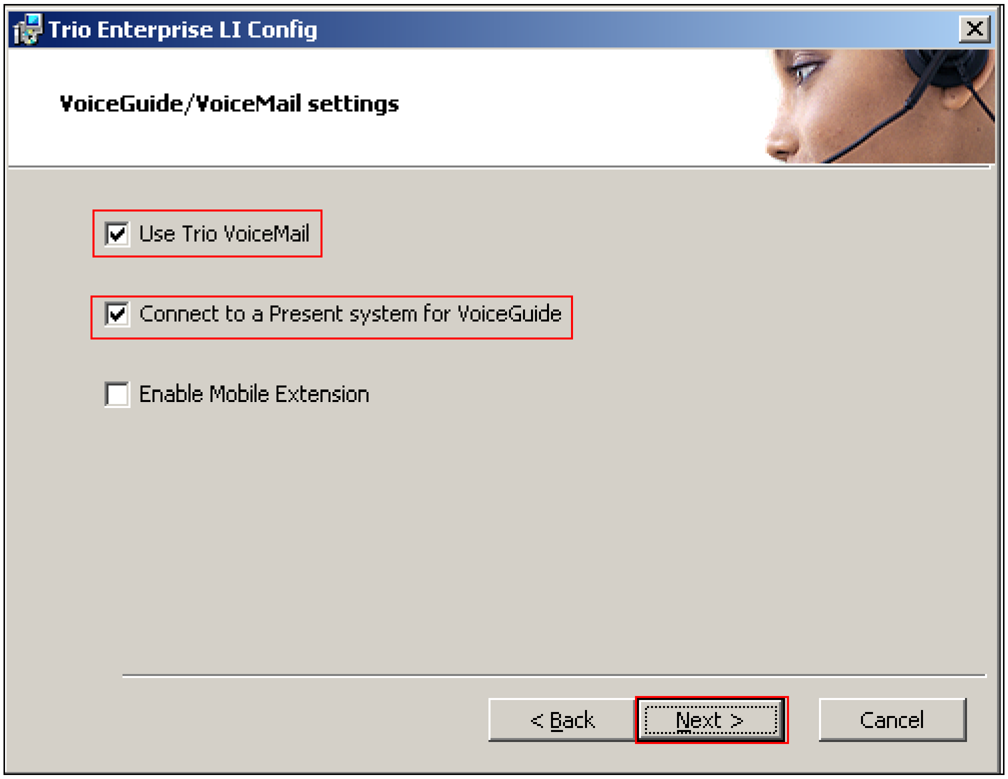
Click **Next** to continue.



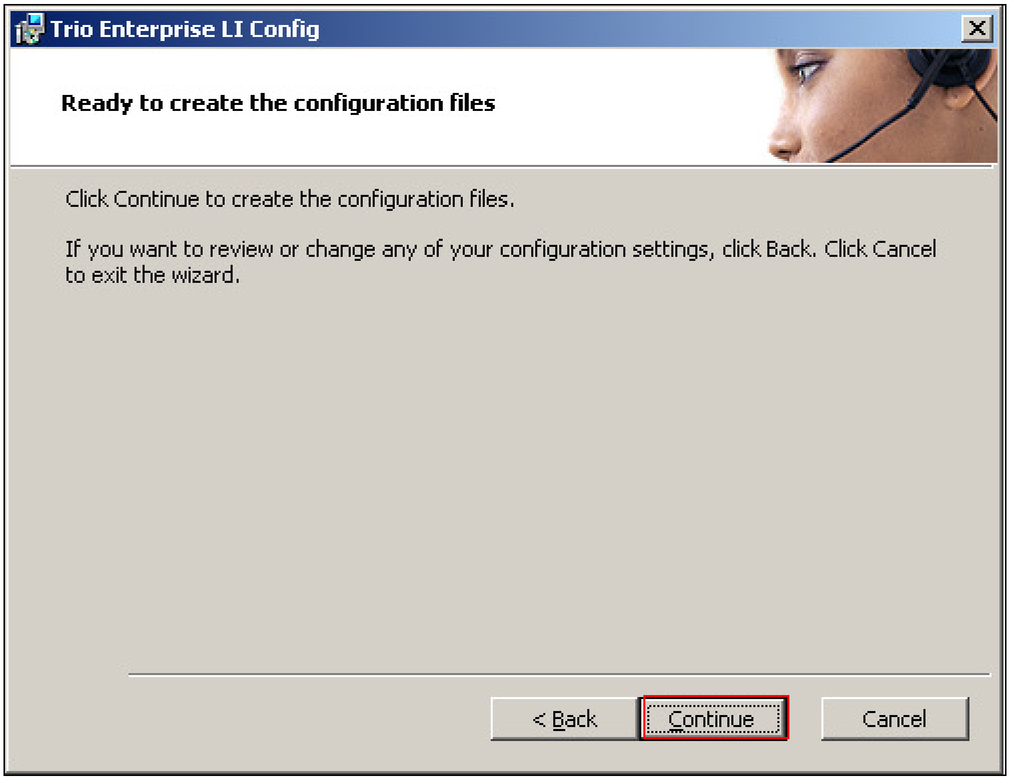
On the **VoiceGuide/VoiceMail settings** page, enter the following settings.

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

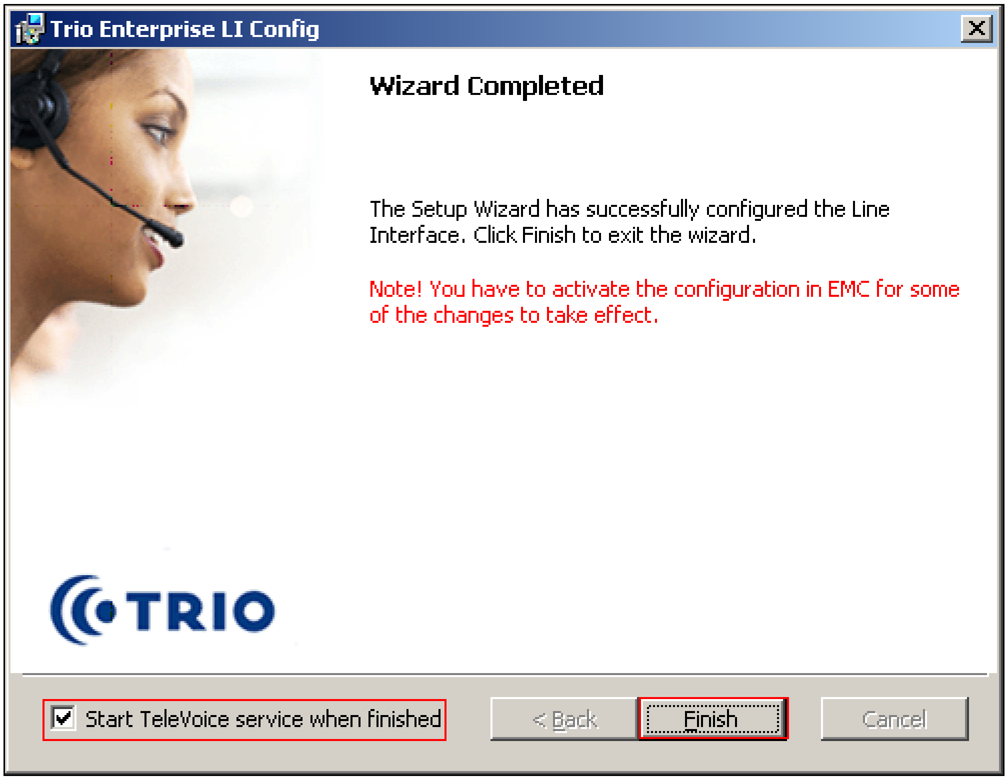
Click **Next** to continue.



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



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



8.2. InteractionStudio Configuration

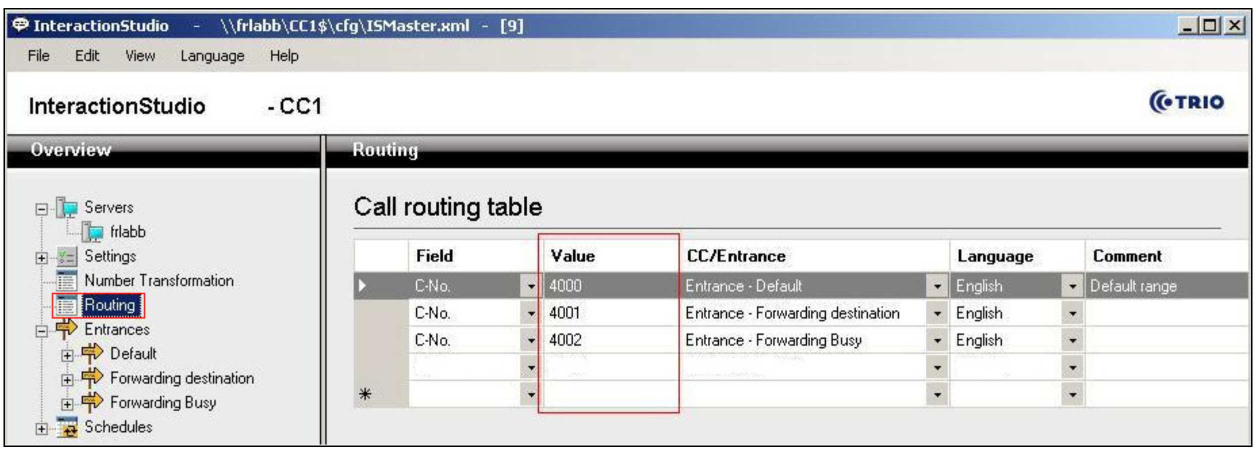
The InteractionStudio is used to configure many features for Trio Enterprise. For compliance testing, the following were configured.

- 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

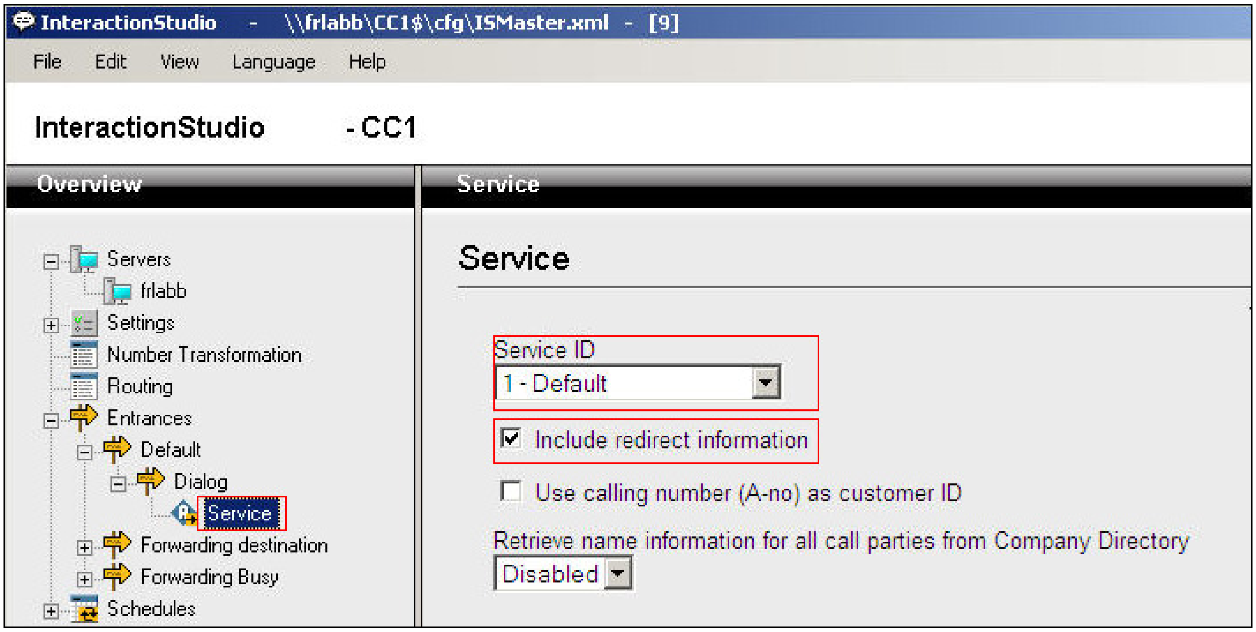
On the Trio Enterprise server, click the **Start** button → **Programs** → **Trio Enterprise** → **Contact Center** → **CC1** → **Interaction Studio** (not shown). When the InteractionStudio window opens, navigate to **Routing**. A **Call routing table** will open. In the example below:

- Extension **4000** is the main queue number
- Extension **4001** is the number that calls go to when Call forward No Answer is activated
- Extension **4002** is the number that calls go to when Call forward Busy is activated



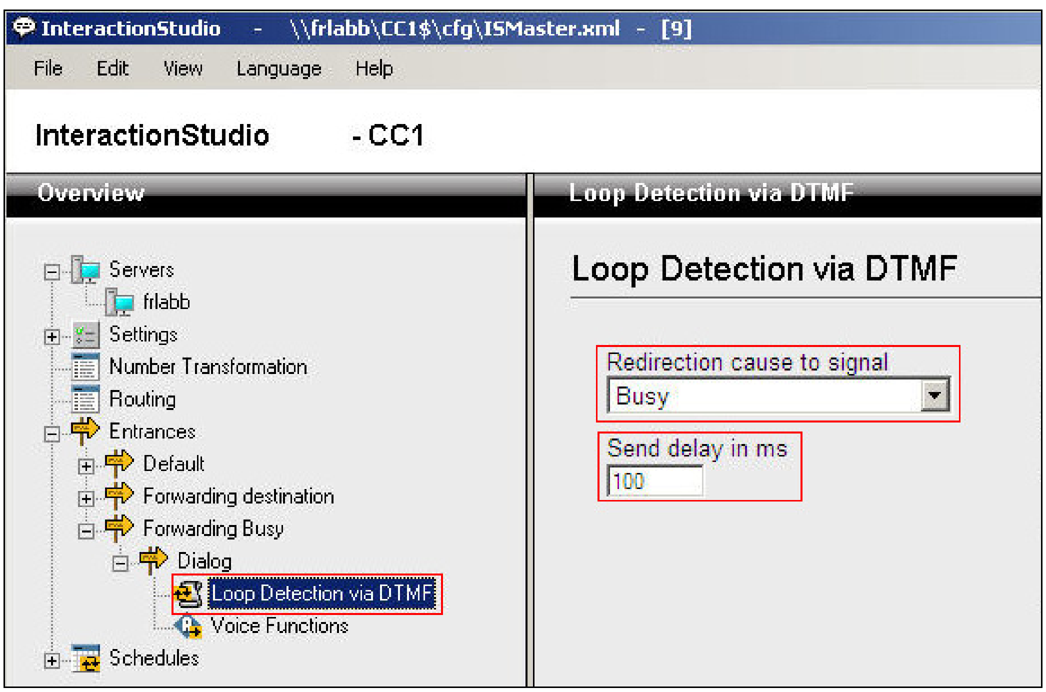
8.2.2. Configure Attendant Service

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



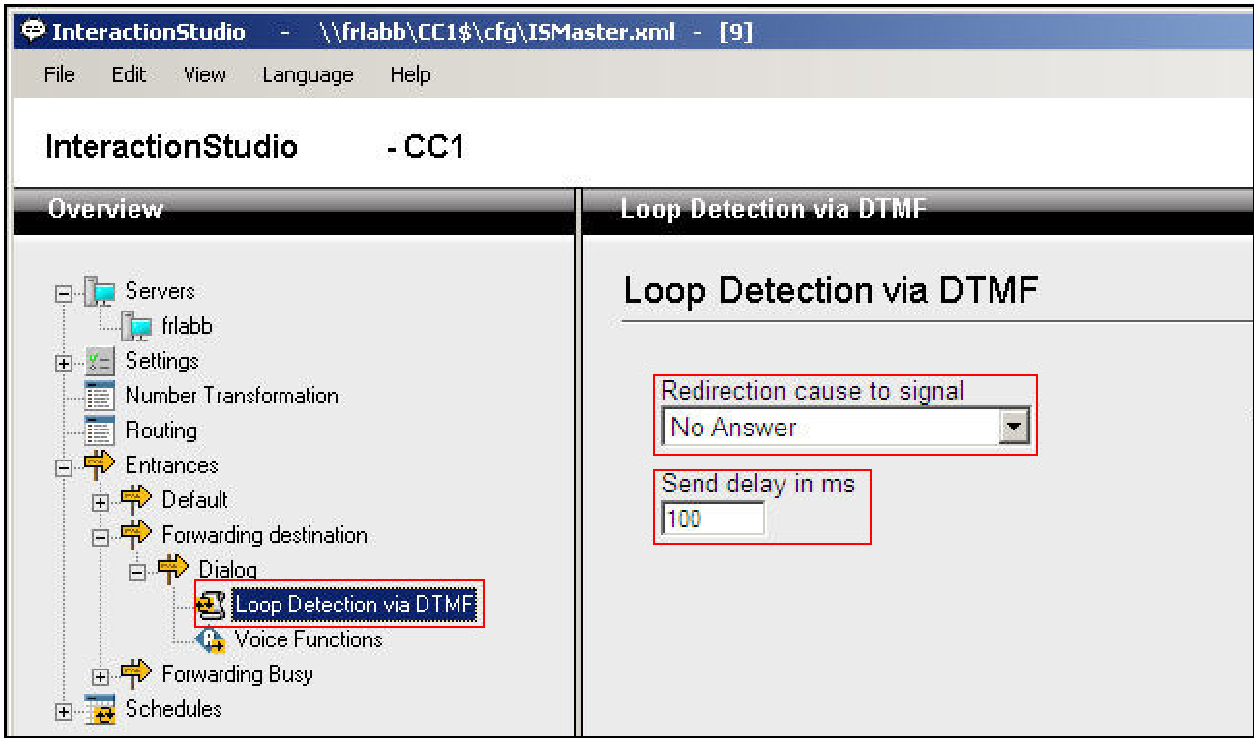
8.2.3. Configure Loop Detection via DTMF for Busy signal

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

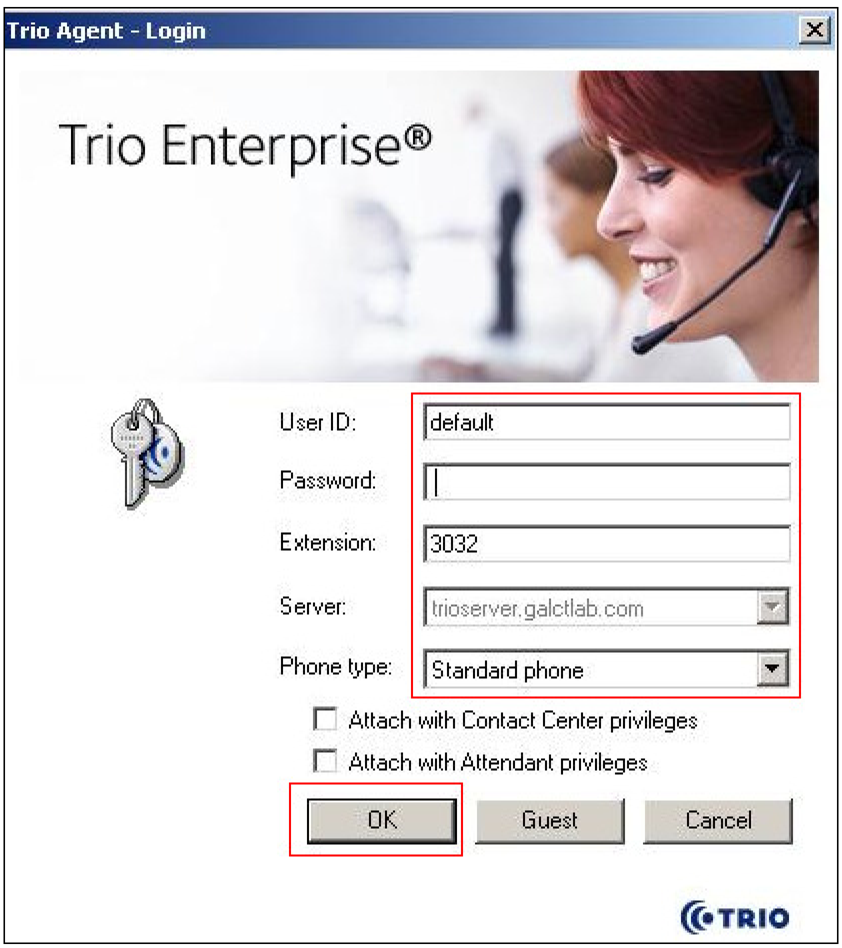
Navigate to **Entrances → Forwarding destination → Dialog → Loop Detection via DTMF.** Choose **No Answer** from the **Redirection cause to signal** drop down box, and enter **100** in the **Send delay in ms** box.



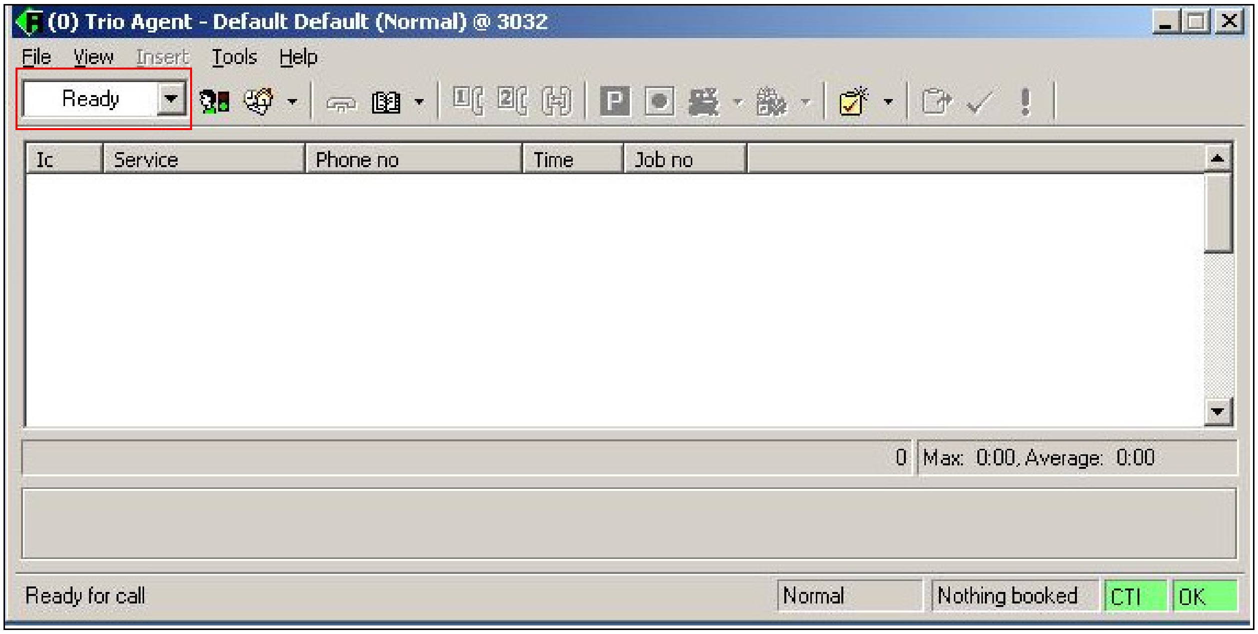
8.3. Configuring Trio Attendant

Trio Attendant is a separate application to Trio Enterprise server and can run concurrently on the same platform. The attendant uses a regular CS1000E 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 CS1000E telephone number that will be used as the agent's audio device (number **3032** 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.

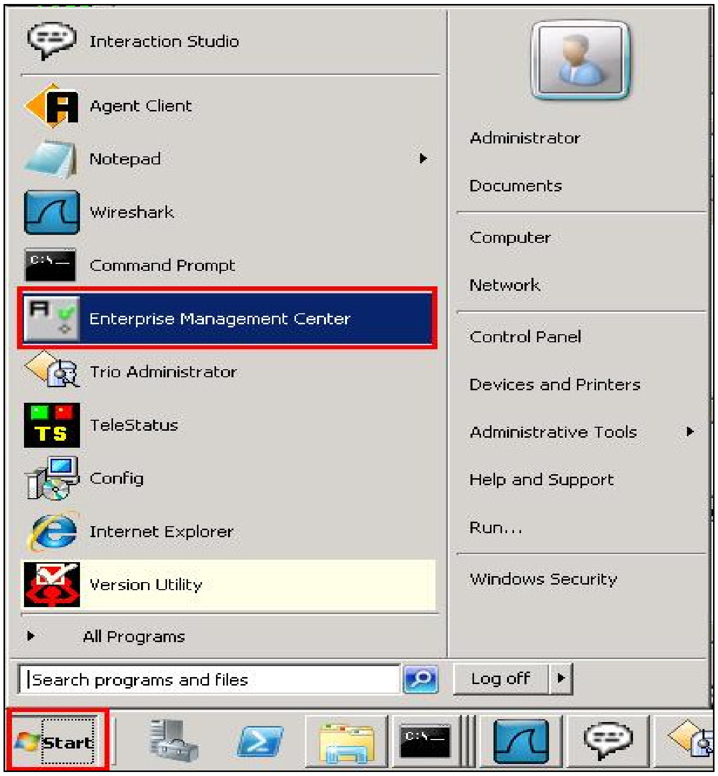


The Trio Agent window appears. Select **Ready** from the drop down box (confirm the traffic light goes green in the small icon to the right of the drop down box).

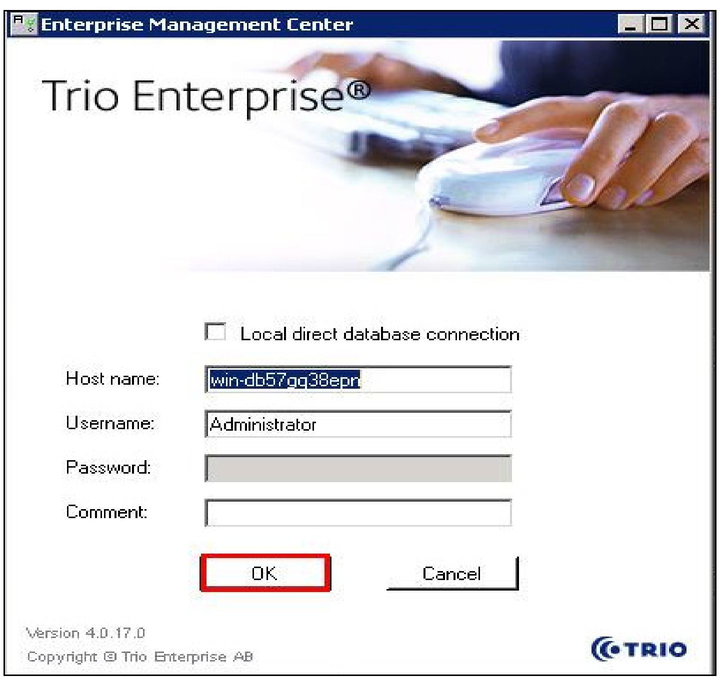


8.4. Configure TR87 on Trio Enterprise

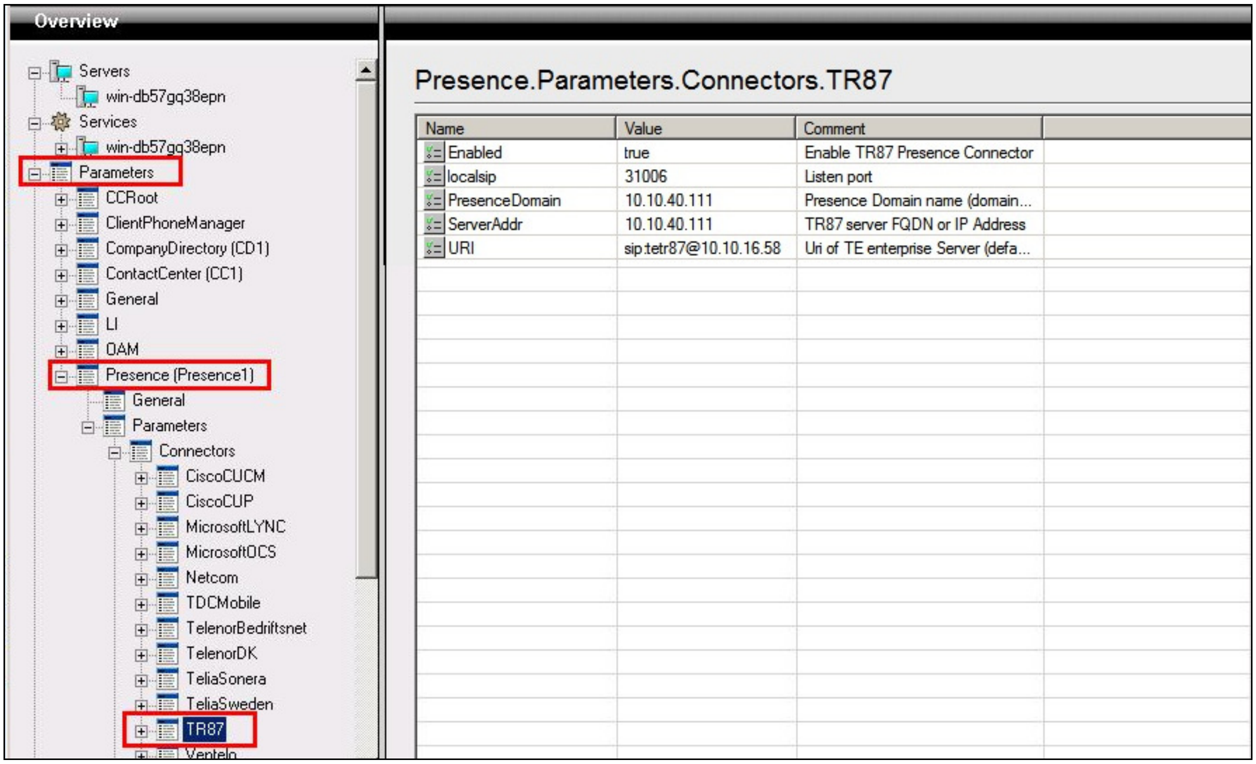
Click on **Start→Programs→Enterprise Management Center**.



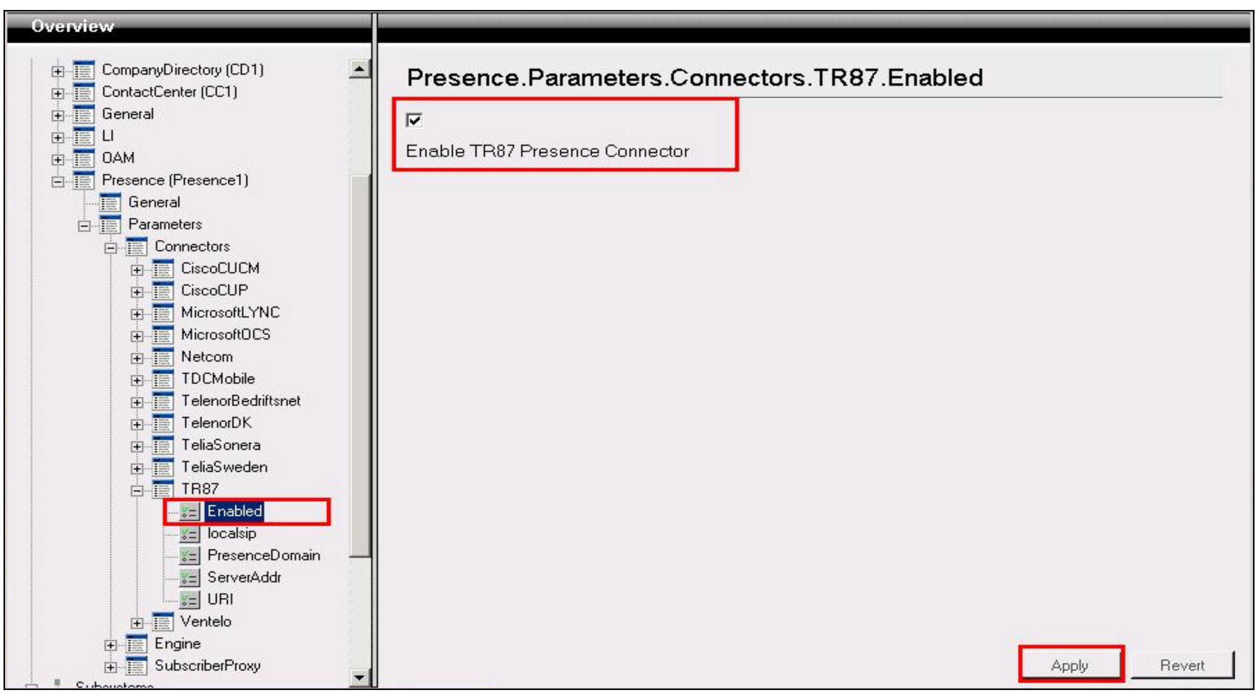
Enter the proper credentials and click on **OK**.



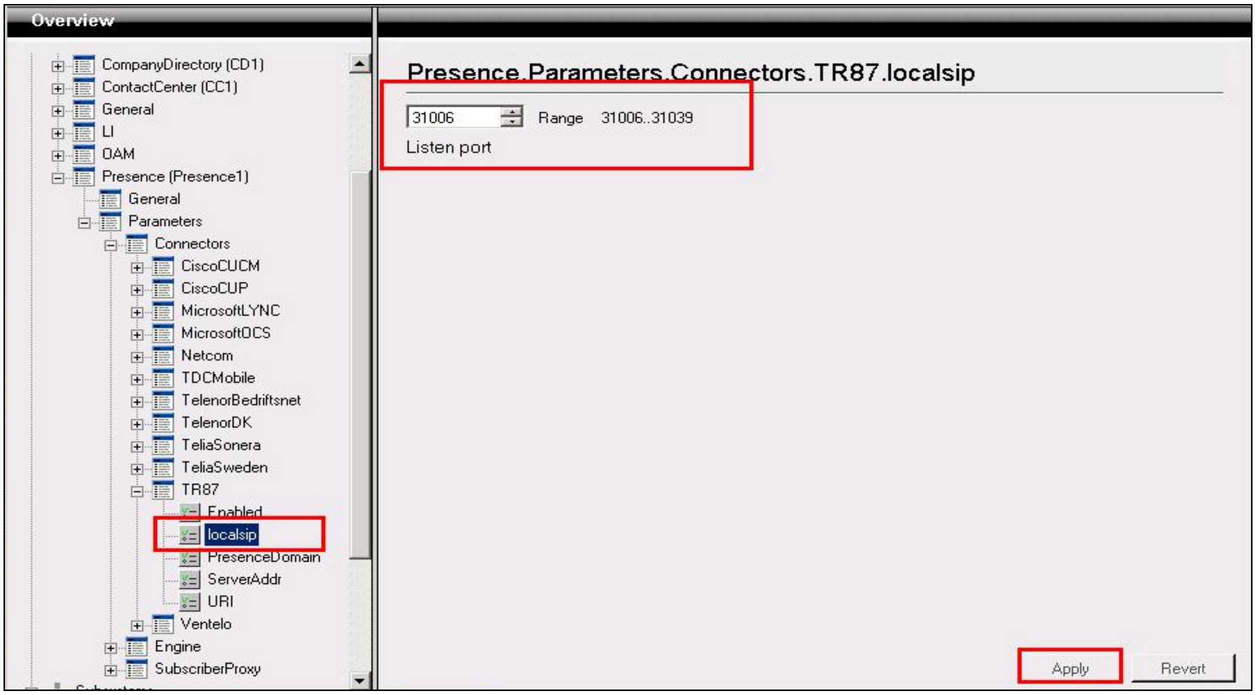
Click on **Parameters→Presence→Parameters→Connectors→TR87** in the left window.



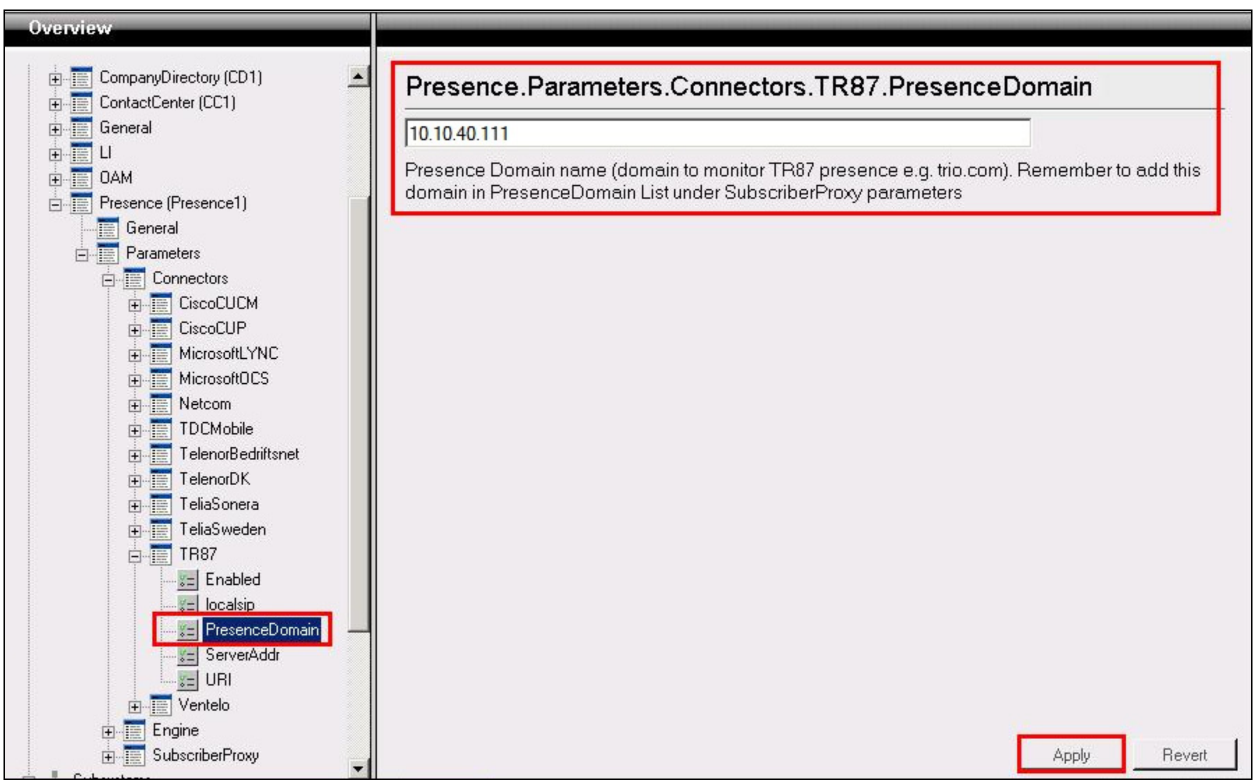
Under **TR87** select **Enabled** in the left window. Ensure that **Enable TR87 Presence Connector** is ticked as shown below. Click **Apply** to continue.



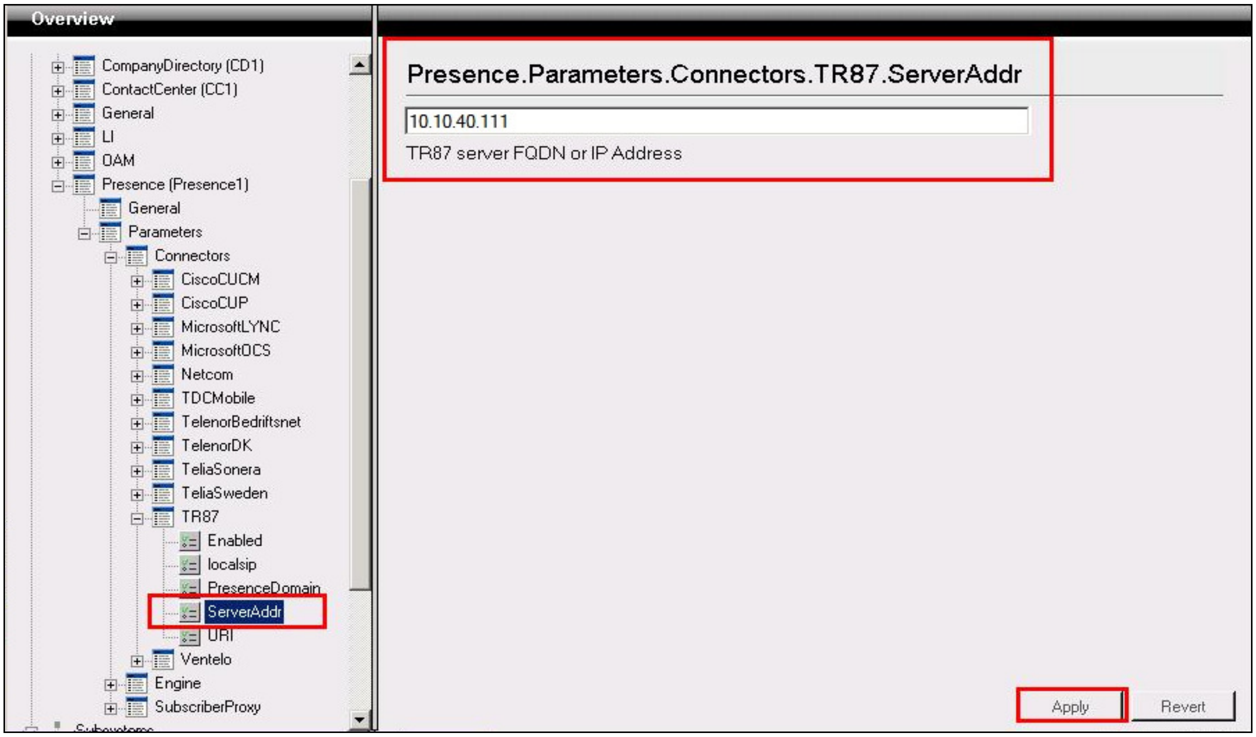
Select **localsip** under **TR87** in the left window and select the **Listen port** for TR87, for compliance testing this was left as default **31006** as shown below. Click **Apply** to continue.



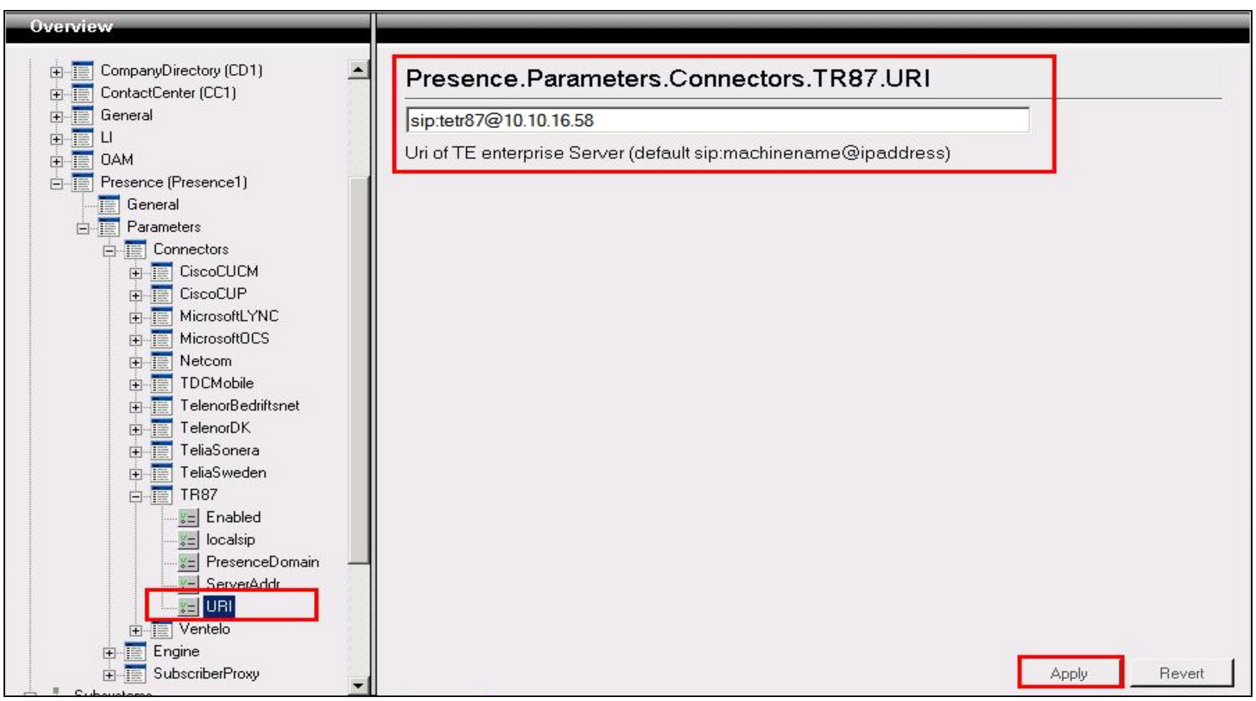
Select **PresenceDomain** under **TR87** in the left window. Enter the Node IP address of the CS1000E as per **Section6**. Click **Apply** to continue.



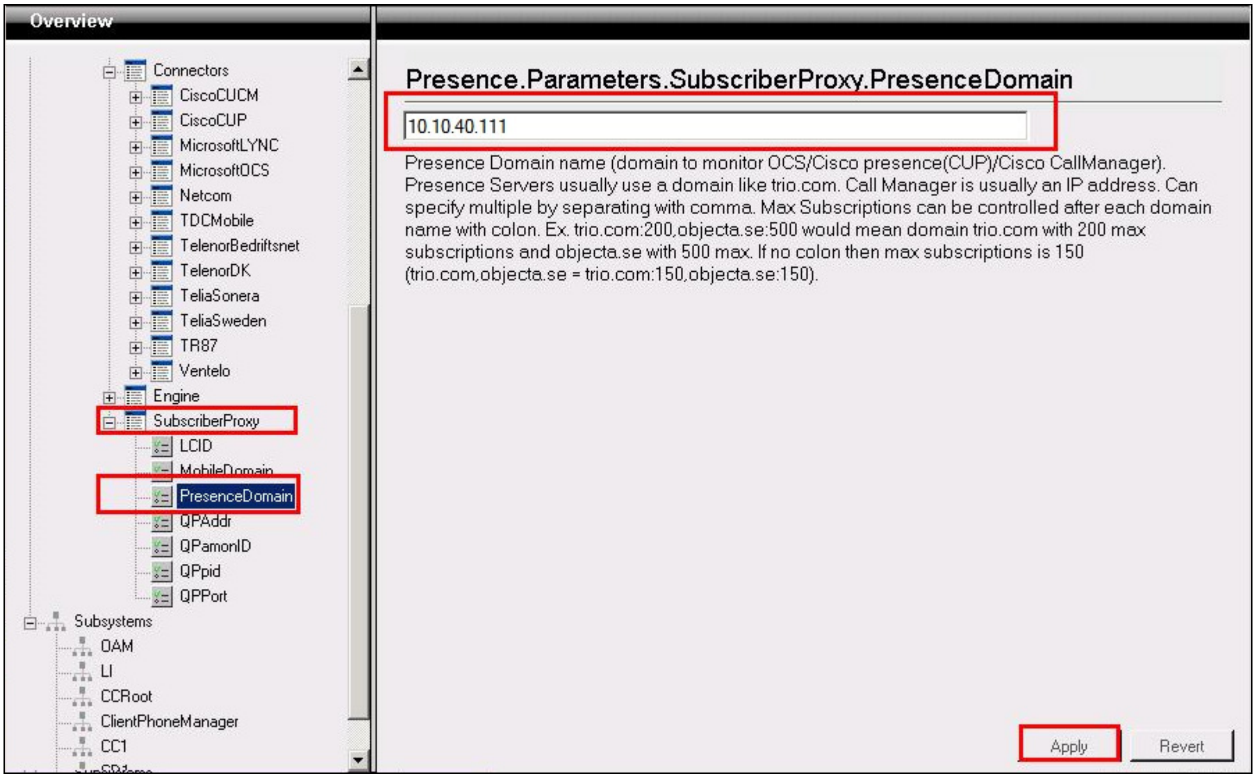
Select **ServerAddr** under **TR87** in the left window and again enter the Node IP address of the CS1000E. Click **Apply** to continue.



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

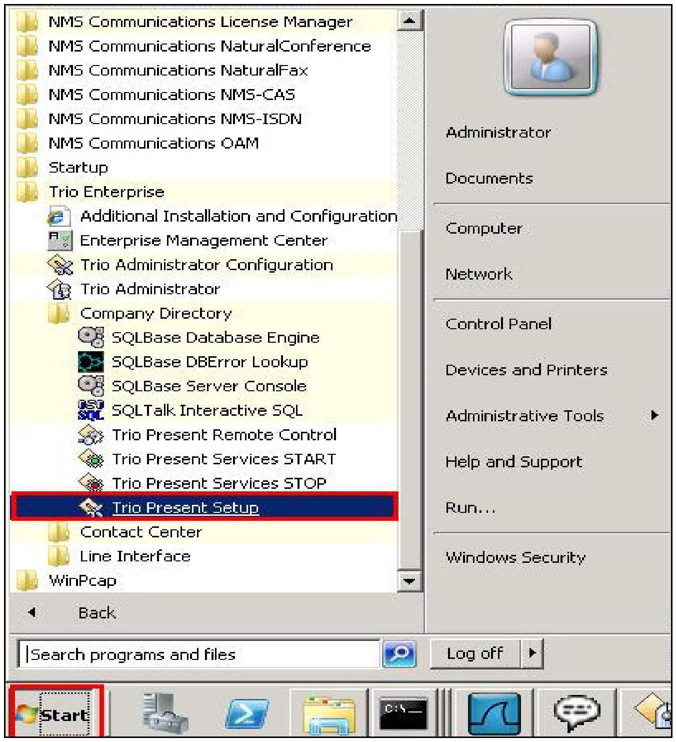


Select **PresenceDomain** under **SubscribeProxy** in the left window. Enter the Node IP address of the CS1000E in the right window. Click **Apply** to continue.

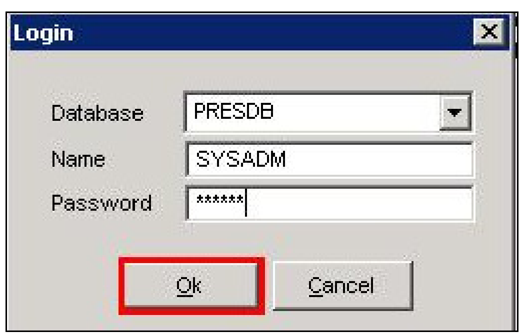


8.5. Configure ICP on TRIO

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



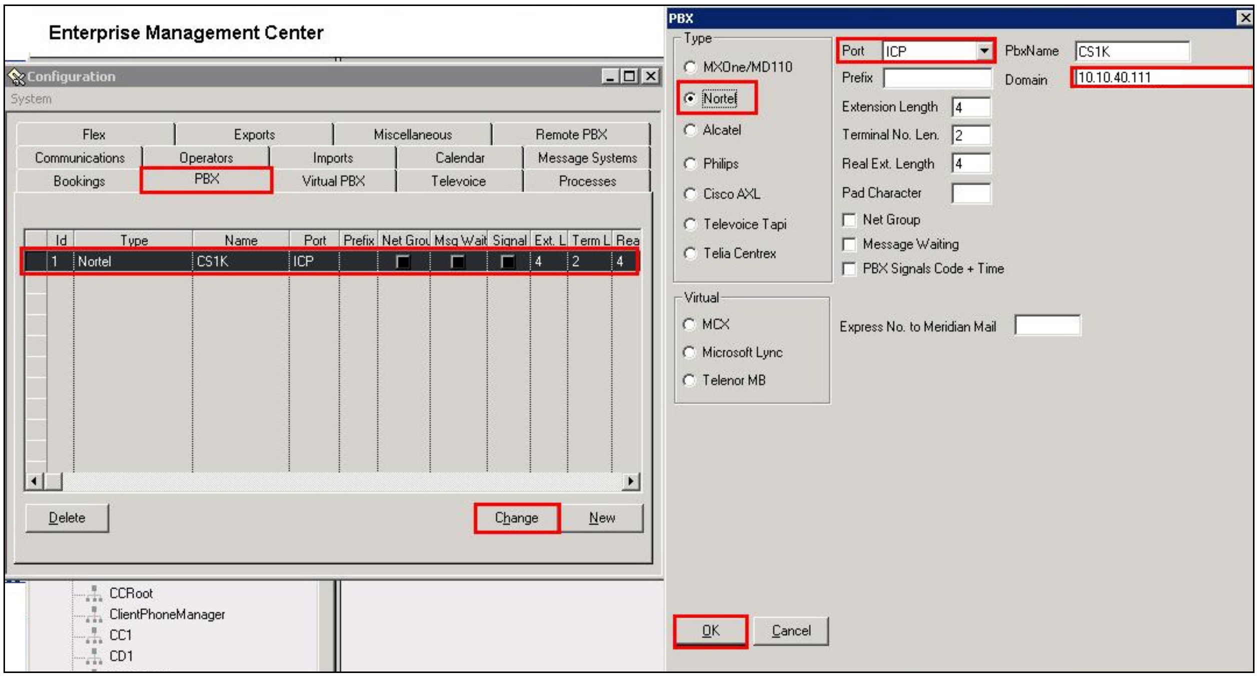
Enter the proper credentials and click **Ok**.



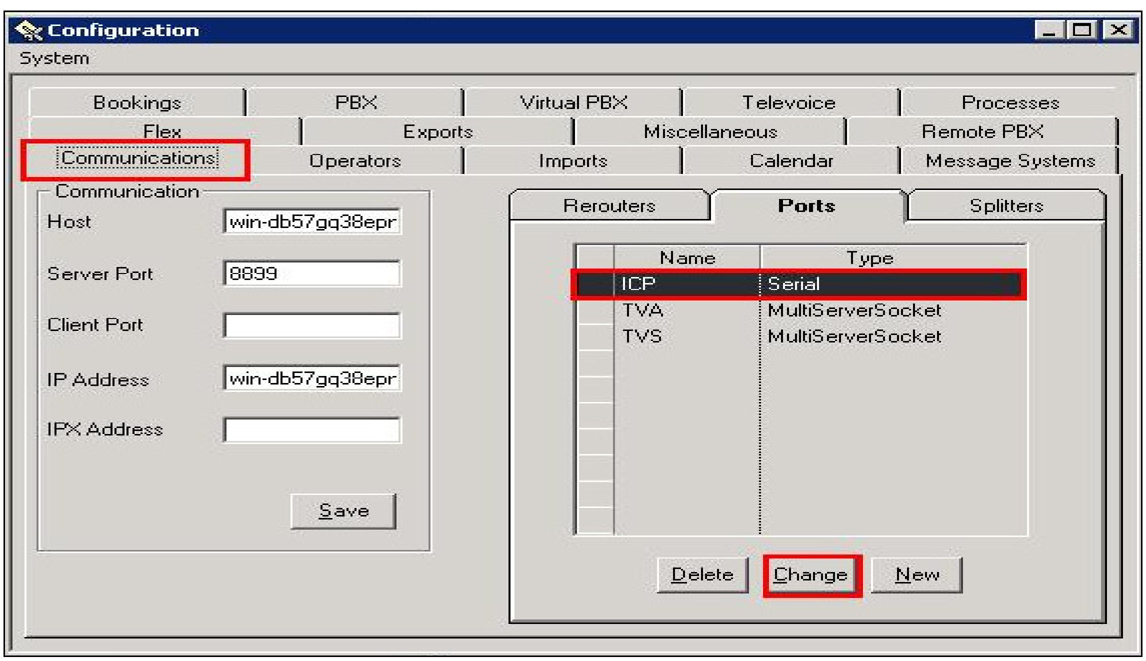
Highlight the selected PBX under the PBX tab and click on change. This opens the window displayed on the right. Ensure the following are selected.

- **Type** **Nortel**
- **Port** **ICP**
- **Domain** **Node IP Address** of CS1000E

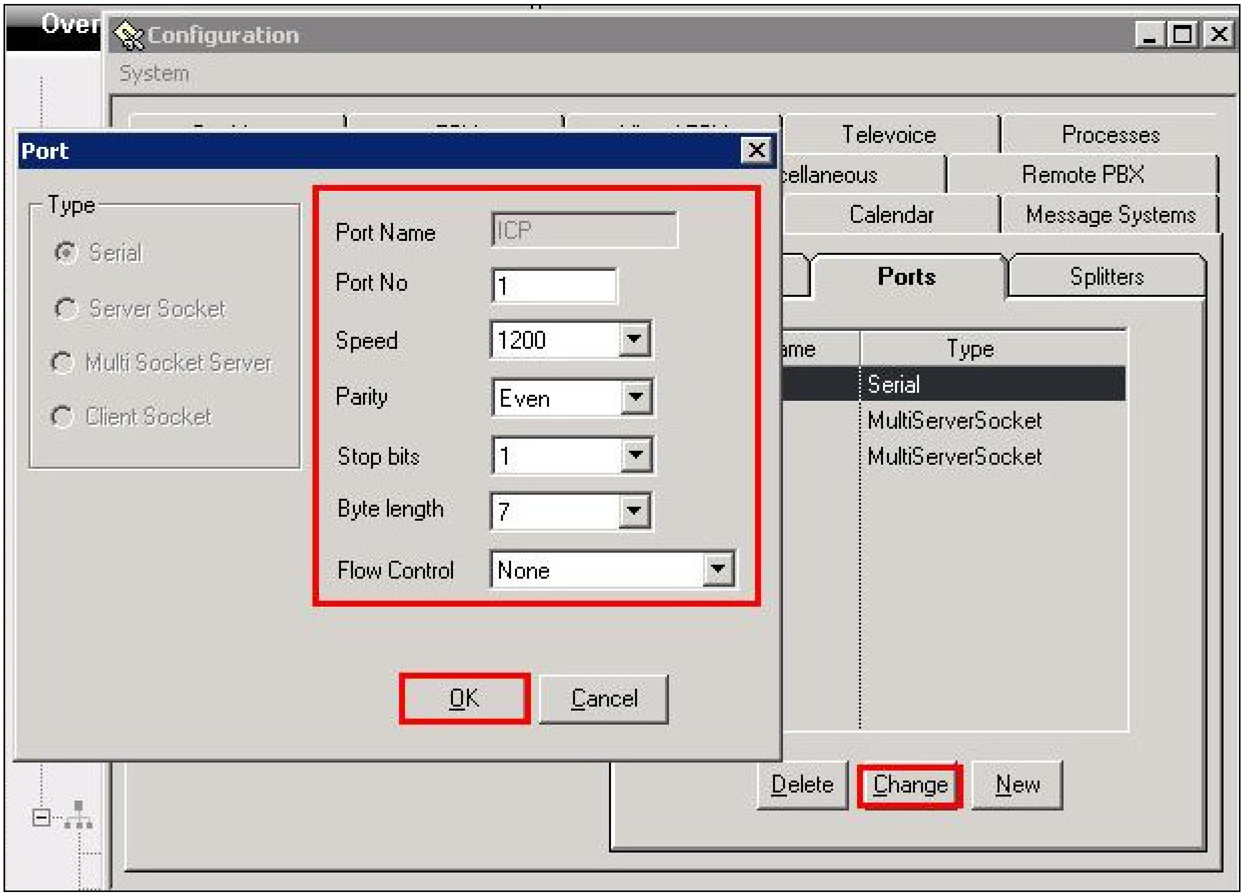
Select **OK** once the correct information is entered.



Under the **Communications** tab select **ICP** as highlighted below and click **Change**.



Enter the information that was entered in **Section 5.5.1** previous, click **OK** once all correct information is added.



Solution & Interoperability Test Lab Application Notes

9. Verification Steps

This section provides the tests that can be performed to verify correct configuration of CS1000E and Session Manager with TRIO Enterprise.

9.1. Status of D-Channel on Avaya Communication Server 1000E

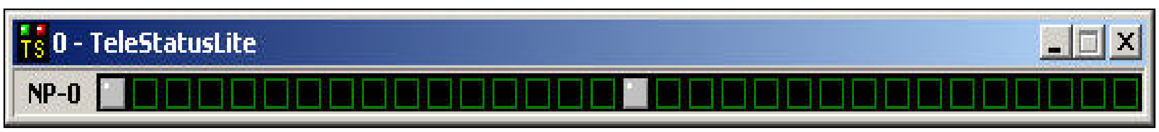
Check the status of the D-channel setup in **Section 5.2.1** by running the command **STAT DCH** in overlay 96 as shown below. The example below shows that D-Channel **66** is operational and established.

LD 96

Prompt	Response	Description
>	LD 96	Enter Overlay 96
.	STAT DCH	Check status of all D-Channels
DCH 066	OPER EST	DES :to_Trio

9.2. Status of SIP Channels on Trio Enterprise

To confirm a successful Trio Enterprise connection with the CS1000E, 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).



10. Conclusion

These Application Notes describe the configuration steps required for Trio Enterprise R4.1from Enghouse Interactive AB to successfully interoperate with Avaya Communication Server 1000E 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

Appendix A

Installed CS1000E Dependency List

CS1000E Linux Service Pack 2

In System service updates: 26

PATCH#	IN_SERVICE	DATE	SPECINS	REMOVABLE	NAME
0	Yes	27/08/13	NO	yes	cs1000-linuxbase-7.65.16.21-04.i386.000
1	Yes	27/08/13	NO	YES	cs1000-patchWeb-7.65.16.21-04.i386.000
2	Yes	27/08/13	NO	YES	cs1000-dmWeb-7.65.16.21-01.i386.000
3	Yes	28/08/13	NO	yes	cs1000-snmp-7.65.16.00-01.i686.000
4	Yes	28/08/13	NO	YES	cs1000-nrsm-7.65.16.00-03.i386.000
5	Yes	28/08/13	NO	YES	cs1000-oam-logging-7.65.16.01-01.i386.000
6	Yes	28/08/13	NO	yes	cs1000-cs1000WebService_6-0-7.65.16.21-00.i386.000
7	Yes	28/08/13	NO	YES	cs1000-sps-7.65.16.21-01.i386.000
8	Yes	28/08/13	NO	YES	cs1000-pd-7.65.16.21-00.i386.000
9	Yes	28/08/13	NO	YES	cs1000-shared-carrdtct-7.65.16.21-01.i386.000
10	Yes	28/08/13	NO	YES	cs1000-shared-tpselect-7.65.16.21-01.i386.000
11	Yes	28/08/13	NO	YES	cs1000-emWebLocal_6-0-7.65.16.21-01.i386.000
12	Yes	28/08/13	NO	yes	cs1000-dbcom-7.65.16.21-00.i386.000
13	Yes	28/08/13	NO	YES	cs1000-csmWeb-7.65.16.21-05.i386.000
14	Yes	28/08/13	NO	YES	cs1000-shared-xmsg-7.65.16.21-00.i386.000
15	Yes	28/08/13	NO	YES	cs1000-vtrk-7.65.16.21-29.i386.000
16	Yes	28/08/13	NO	YES	cs1000-tps-7.65.16.21-05.i386.000
17	Yes	28/08/13	NO	YES	cs1000-mscAnnc-7.65.16.21-02.i386.001
18	Yes	28/08/13	NO	YES	cs1000-mscAttn-7.65.16.21-04.i386.001
19	Yes	28/08/13	NO	YES	cs1000-mscConf-7.65.16.21-02.i386.001
20	Yes	28/08/13	NO	YES	cs1000-mscMusc-7.65.16.21-02.i386.001
21	Yes	28/08/13	NO	YES	cs1000-mscTone-7.65.16.21-03.i386.001
22	Yes	28/08/13	NO	YES	cs1000-bcc-7.65.16.21-21.i386.000
23	Yes	28/08/13	NO	YES	cs1000-Jboss-Quantum-7.65.16.21-3.i386.000
24	Yes	28/08/13	NO	YES	cs1000-emWeb_6-0-7.65.16.21-06.i386.000
25	Yes	28/08/13	NO	yes	cs1000-cs-7.65.P.100-01.i386.001

CS1000E Call Server Patches

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VERSION 4121
RELEASE 7
ISSUE 65 P +
DepList 1: core Issue: 01 (created: 2013-06-14 03:54:33 (est))

IN-SERVICE PEPS
PAT# CR #          PATCH REF #     NAME      DATE        FILENAME       SPECINS
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001  wi01045058    ISS1:1OF1       p32214_1  28/08/2013  p32214_1.cpl   NO
002  wi01085855    ISS1:1OF1       p32658_1  28/08/2013  p32658_1.cpl   NO
003  wi01053314    ISS1:1OF1       p32555_1  28/08/2013  p32555_1.cpl   NO
004  wi01060382    iss1:1of1       p32623_1  28/08/2013  p32623_1.cpl   YES
005  wi01070580    ISS1:1OF1       p32380_1  28/08/2013  p32380_1.cpl   NO
006  wi01067822    ISS1:1OF1       p32466_1  28/08/2013  p32466_1.cpl   YES
007  wi01061481    ISS1:1OF1       p32382_1  28/08/2013  p32382_1.cpl   NO
008  wi01072032    ISS1:1OF1       p32448_1  28/08/2013  p32448_1.cpl   NO
009  wi01022599    ISS1:1OF1       p32080_1  28/08/2013  p32080_1.cpl   NO
010  wi01035976    ISS1:1OF1       p32173_1  28/08/2013  p32173_1.cpl   NO
011  wi01065922    ISS1:1OF1       p32516_1  28/08/2013  p32516_1.cpl   NO
012  wi01055480    ISS1:1OF1       p32712_1  28/08/2013  p32712_1.cpl   NO
013  wi01041453    ISS1:1OF1       p32587_1  28/08/2013  p32587_1.cpl   NO
014  wi01078723    ISS1:1OF1       p32532_1  28/08/2013  p32532_1.cpl   NO
015  WI0110261     ISS1:1OF1       p32758_1  28/08/2013  p32758_1.cpl   NO
016  wi01064599    iss1:1of1       p32580_1  28/08/2013  p32580_1.cpl   NO
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018  wi01072027    ISS1:1OF1       p32689_1  28/08/2013  p32689_1.cpl   NO
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024  wi01063864    ISS1:1OF1       p32410_1  28/08/2013  p32410_1.cpl   YES
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MDP>USING DEPLIST ZIP FILE DOWNLOADED :2013-08-27 09:21:58(est)
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