Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Trio Enterprise R3.2 with Avaya Communication Server 1000E R7.5 and Avaya Aura® Session Manager R6.1 using a SIP connection – Issue 1.0

Abstract

These Application Notes describe how to configure an Avaya Communication Server 1000E R7.5 to interface with Trio Enterprise R3.2, which is operating as an attendant answering position. Trio Enterprise is a software application installed on a Windows server that interfaces with Avaya Communication Server 1000E using a SIP connection via Avaya Aura® Session Manager R6.1 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 R7.5 with Trio Enterprise R3.2. Trio Enterprise is a client/server based application running on Microsoft Windows operating systems. Trio Enterprise provides users with an attendant answering position for Avaya Communication Server 1000E that does not need 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 R6.1. Calls are made over trunks to PSTN destinations as well as internal Avaya Communication Server 1000E users. 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.

Note: The Trio Enterprise server places a call to the attendant's deskphone, for compliance testing an Avaya 1140Ewas used. When the attendant is called the Trio Enterprise server calls the 1140E and bridges the call.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using a 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.

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.

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 call
- 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

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.

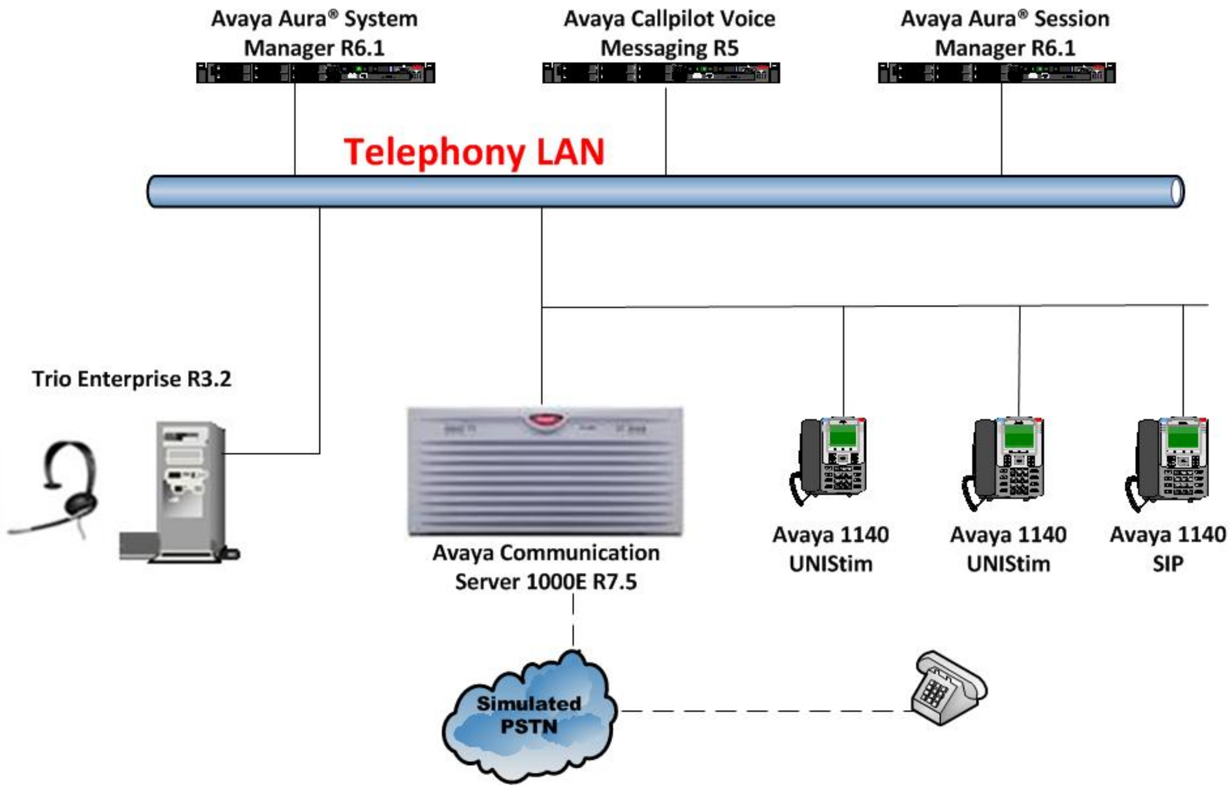


Figure 1: Configuration for Avaya Communication Server 1000E, Avaya Aura® Session Manager and Trio Enterprise R3.2

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	R7.5 SP1 (See Appendix A for the installed dependency list used during compliance testing)
Avaya Aura® System Manager running on Avaya S8800 Server	R6.1 SP4
Avaya Aura® Session Manager running on Avaya S8800 Server	R6.1 SP4
Avaya 1140 UNIStim Deskphone	UNIStim V0625C8D
Avaya 1140 SIP Deskphone	SIP V04.00.04.00
Avaya Call Pilot running on Avaya Callpilot 600r Server	Version 5.00.41 Patch Line-up:CP50041SU08S CP500508G09C
Trio Enterprise Running on Desktop PC (Minimum Specification Pentium IV, 3 GHz, 1 GB Ram, 1 USB Hand/Headset	Version 3.2

5. Configure Avaya Communication Server 1000E

The configuration operations illustrated in this section were performed using terminal access to the CS1000E over a telnet session. 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 10**.

Note: The configuration of the PRI interface to the PSTN is 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. Configuring Avaya Communication Server 1000E

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

- Configure Network Attendant Service (NAS) and Night (NIT) Data
- Create a D-channel for the SIP trunk
- Create Route Data Block
- Adding TIE Trunks

5.1.1. Configure NAS and NIT Data

The CS1000E is configured with attendant groups where the NAS and NIT functions route the calls between the nodes and out to Trio Enterprise. Use the **NEW** command in **LD 86** to configure **NAS**.

LD 86

Prompt	Response	Description
>	**LD 86**	Enter Overlay 86
REQ	**NEW**	New Data
CUST	**0**	Customer Number
FEAT	**NAS**	Network Attendant Service
TBL	**0**	NAS routing Table 0

Use the **NEW** command in **LD 15** to configure **NIT_DATA.**

LD 15

Prompt	Response	Description
>	**LD 15**	Enter Overlay 15
REQ	**NEW**	Change
TYPE	NIT	Night Service
CUST	**0**	Customer Number
NIT1	5000	Night number

5.1.2. 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

5.1.3. 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.

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

5.1.4. Adding TIE Trunks

Use the **NEW** command in **LD 14** to add (**IPTI**) **TIE** trunks to the new route created in **Section 5.1.3.** 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 New
TYPE	IPTI	IP TIE trunk
TN	096 0 3 1	Loop Shelf Card Unit
CUST	0	Customer Number as defined in LD15
RTMB	20 1	Route number and Member number

5.2. Configure a Coordinated Dialing Plan

There are a number of ways to setup a dialing plan to call the Trio Enterprise. For the compliance testing a Coordinated Dialing Plan (CDP) was used.

5.2.1. Create a Route List Index

In order to create a CDP, a Route List Index (RLI) in overlay 86 is required. Use the **NEW** command in **LD 86** to create a **RLI**.

Note: Enter the route (**ROUT)** that was created in **Section 5.1.3.**

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

5.2.2. 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 5.2.1.**

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

6. Creating a SIP Endpoint on the Avaya Aura® Session Manager for Trio Enterprise

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

- SIP Entity
- Entity Link
- Route Policy
- Dail Pattern

Note: To get more information for any input field you can press the **Help** link at anytime.

Configuration of Session Manager is achieved by accessing the browser-based GUI of System Manager, using the URL **http://<fqdn>/SMGR** or **http://<ip-address>/SMGR**, where "<fqdn> is the fully qualified domain name of System Manager or the **<ipaddress>** is the IP address of System Manager. Log in with the appropriate credentials. Once logged in select the **Routing** link under the **Elements** column.

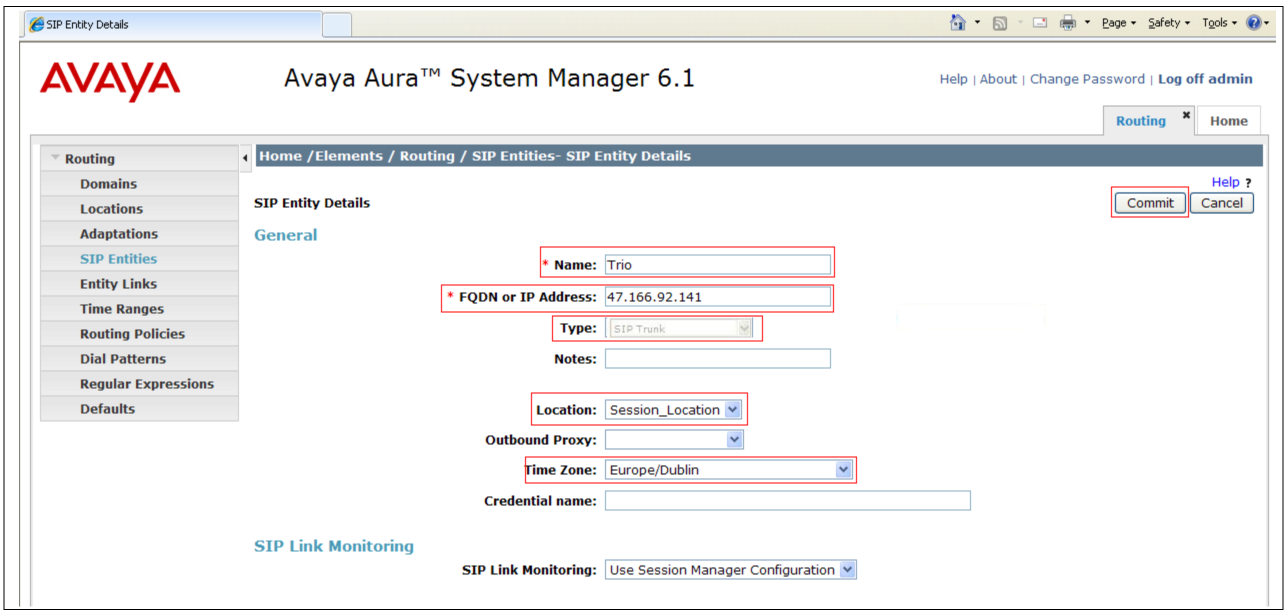
6.1. Adding Trio Enterprise SIP Entity

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 (not shown). Enter the following for the Trio SIP Entity:

Under **General:**

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

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



6.2. Adding Communication Server 1000E SIP Entity

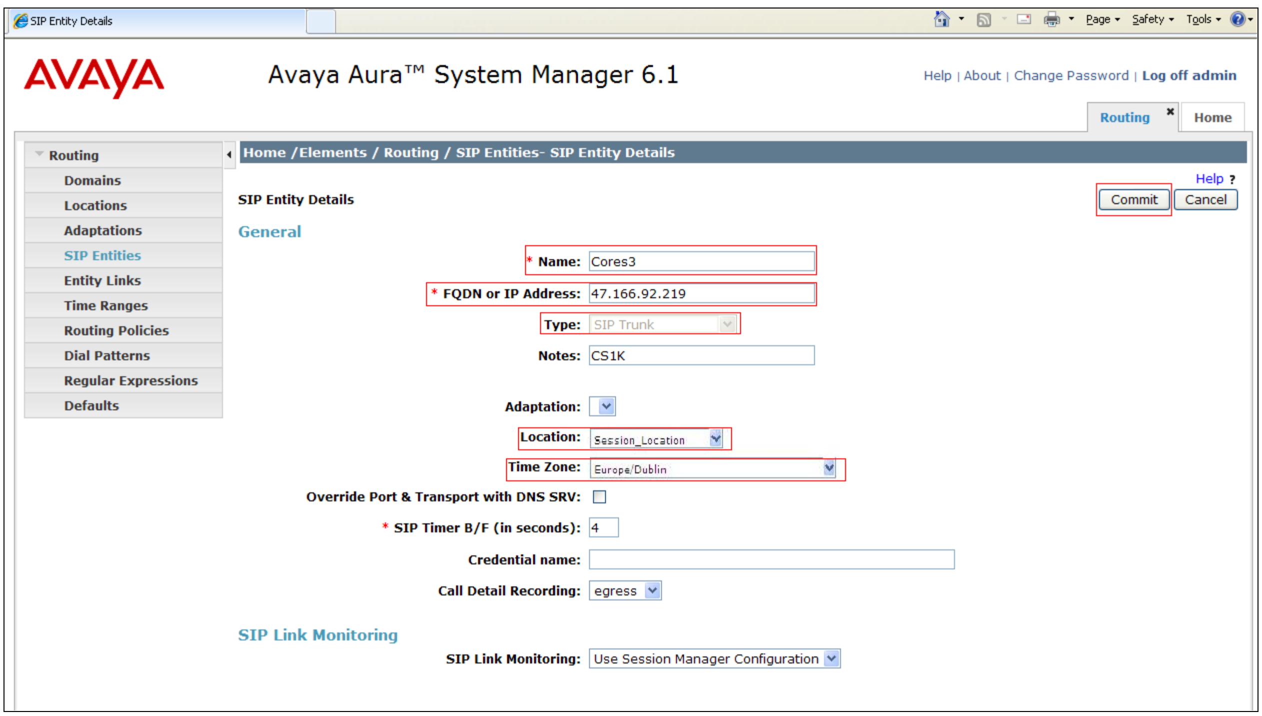
A SIP Entity must be added for CS1000E. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown).

Notes: A SIP Entity was already configured for the CS1000E and was called **cores3**.

Enter the following for the Trio SIP Entity:
Under **General:**

- **Name** An informative name (e.g., **Cores3**)
- **FQDN or IP Address** Node IP address of the CS1000E
- **Type** **SIP Trunk** for Cores3
- **Location** **Session_Location**
- **Time Zone** Time zone for this location **Europe/Dublin**

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



6.3. Create an Entity Link to Trio Enterprise

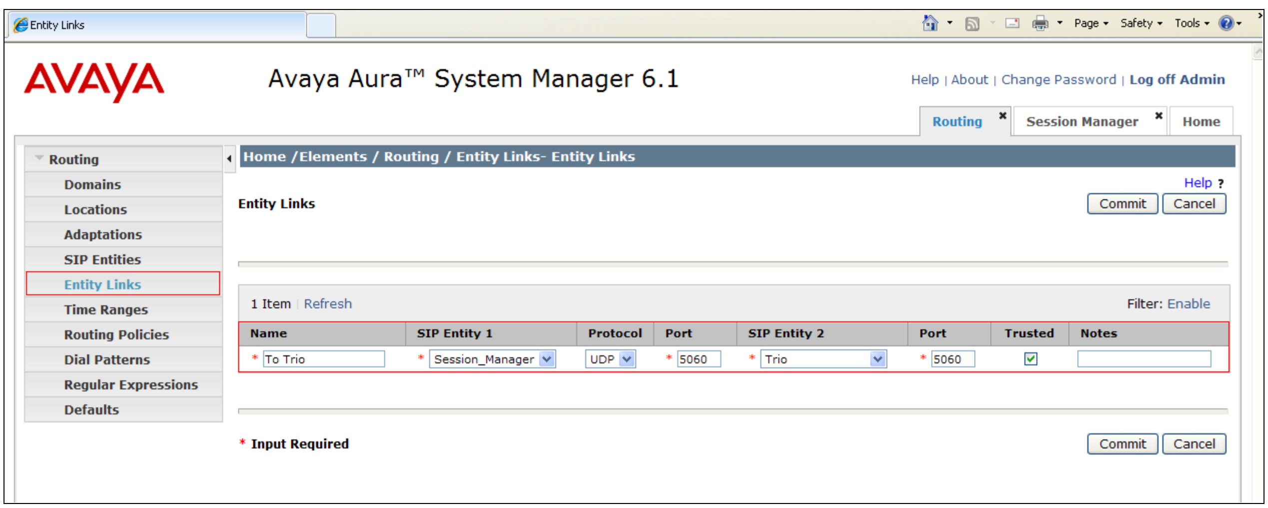
A SIP trunk between a Session Manager and the Trio Enterprise is required.

Notes: An Entity Link was already configured between Session Manager and the CS1000E and is called **to cores3**.

To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown). Fill in the following fields in the new row that is displayed.

- **Name** An informative name, e.g. **To Trio**
- **SIP Entity 1** Select **Session_Manager** from the **SIP Entity 1** dropdown box
- **Protocol** Select **UDP** 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.

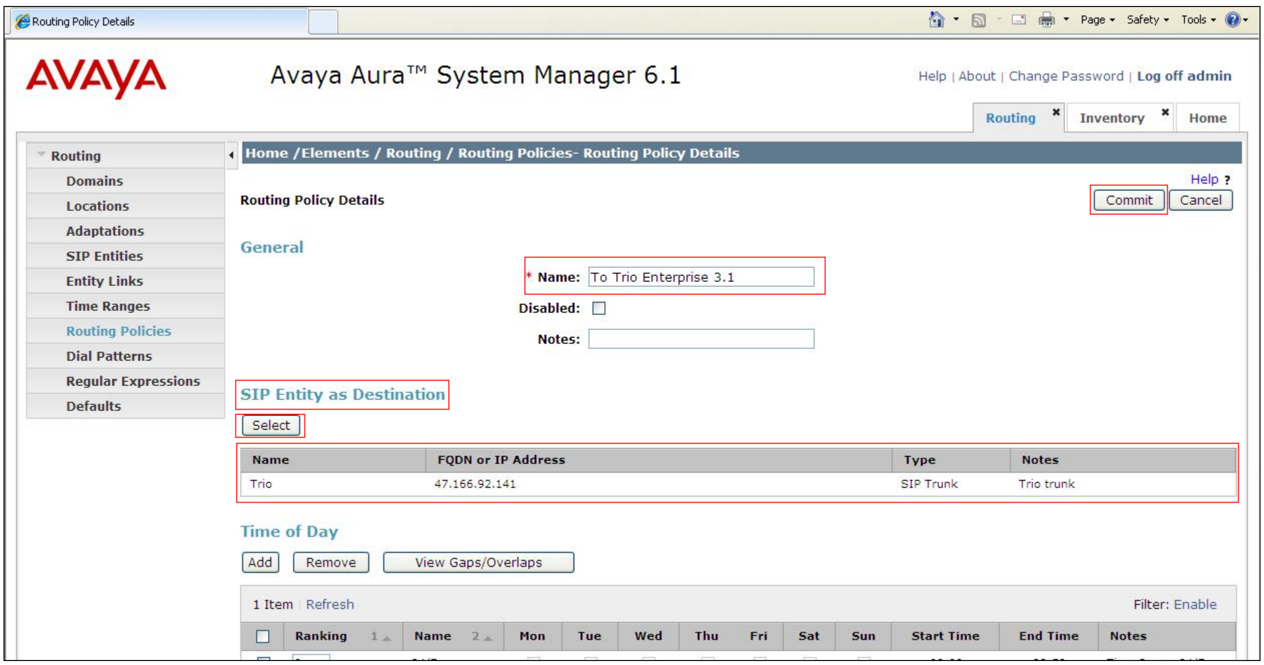


6.4. Configure Routing Policy for Trio Enterprise

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 (not shown).
Under **General** enter the following:

- **Name** Enter an informative name , (e.g., **To Trio Enterprise 3.1**)

Under **SIP Entity as Destination,** click **Select**, and then check the **Trio** radio button (not shown). Click on the **Select** button to confirm the chosen options and then be returned to the **Routing Policies Details** screen and click the **Commit** button to save. The following screen shows the **Routing Policy Details** for calls to Trio.



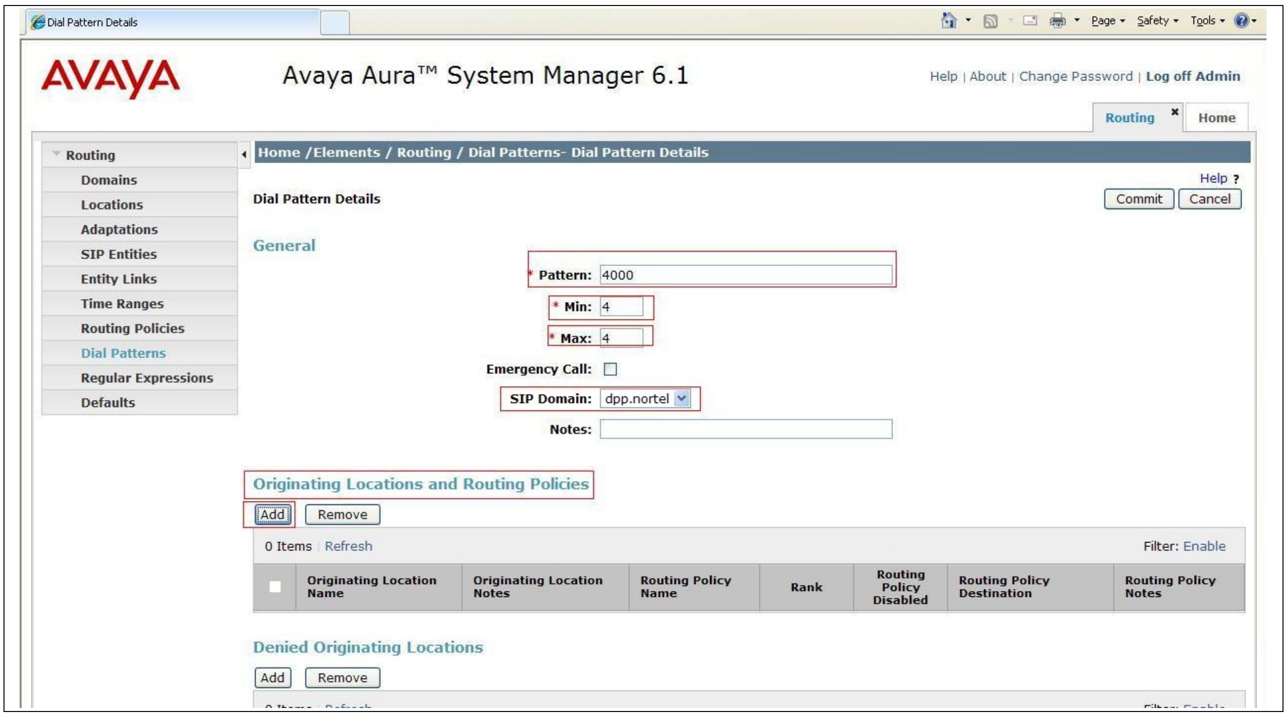
6.5. Create Trio Enterprise Dial Pattern

A dial pattern must be defined that will direct calls to the Trio Enterprise. During testing 4 numbers were used 4000, 4001, 4002, and 5030, the Domain name was **dpp.nortel**. To configure the Trio Enterprise Pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

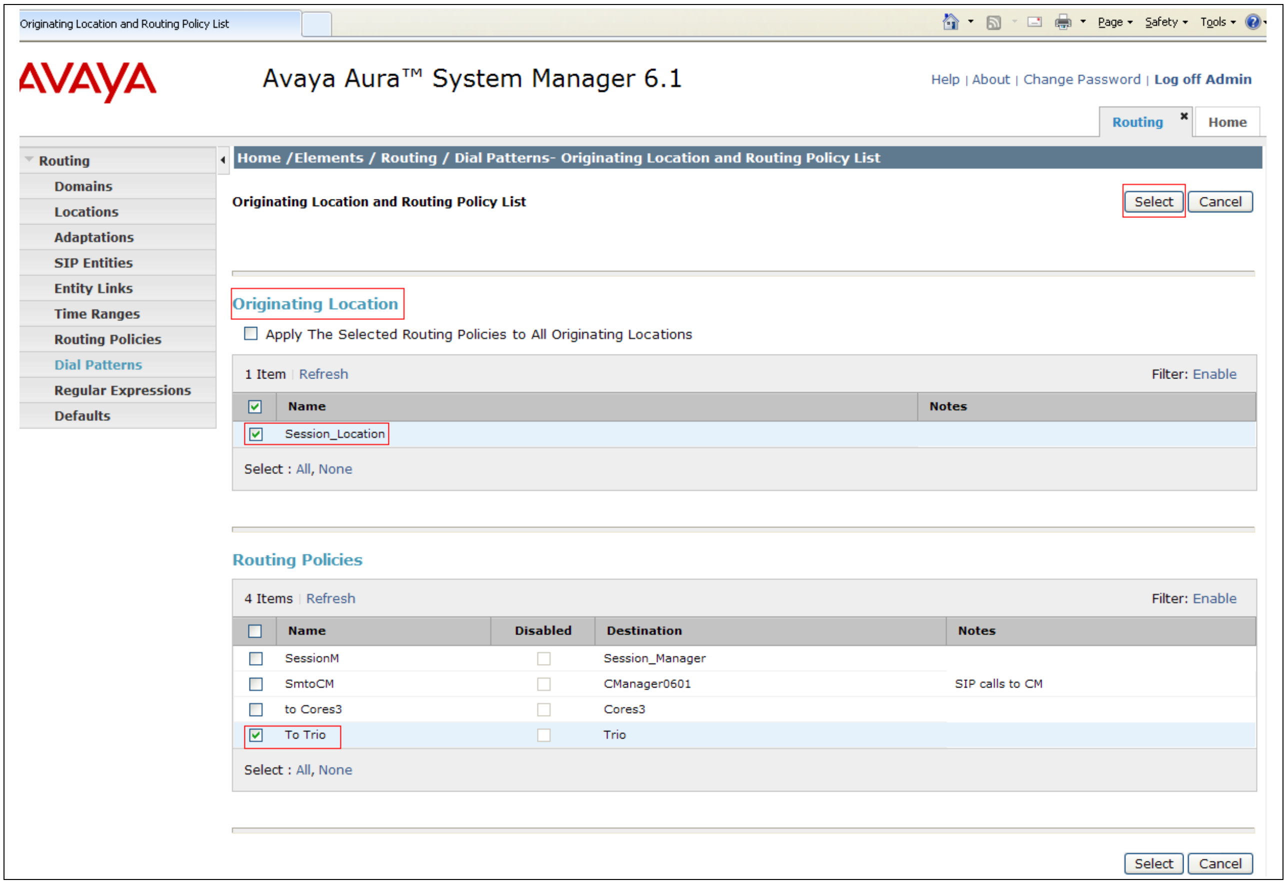
Under **General** carry out the following for each number.

- **Pattern** Dialed number or prefix **4000**
- **Min** Minimum length of dialed number **4**
- **Max** Maximum length of dialed number **4**
- **SIP Domain** Selection of **dpp.nortel** was used in the compliance testing

Continue to **Originating Locations and Routing Policy List.**



Select **Add** (not shown). Under **Originating Location** check **Session_Location** check box, and under **Routing Policies** check **ToTrio** 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.

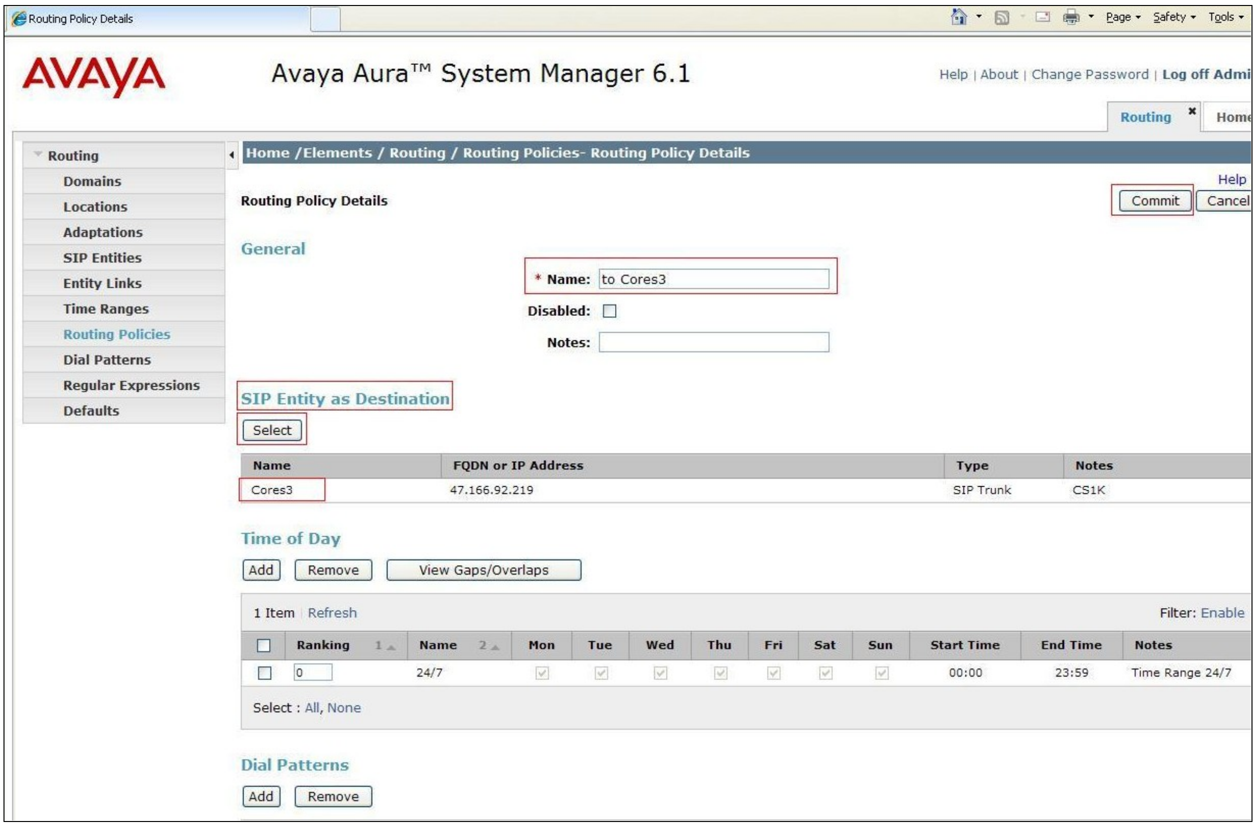


6.6. Configure Routing Policies for Communication Server 1000E

Create routing policies to direct calls to CS1000E. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown).
Under **General** enter the following:

- **Name** Enter an informative name, (e.g. **to Cores3**)

Under **SIP Entity as Destination,** click **Select**, and then check the **cores3** radio button (not shown). Click on the **Select** button to confirm the chosen options and then be returned to the **Routing Policies Details** screen and click the **Commit** button to save. The following screen shows the **Routing Policy Details** for calls to CS1000E.



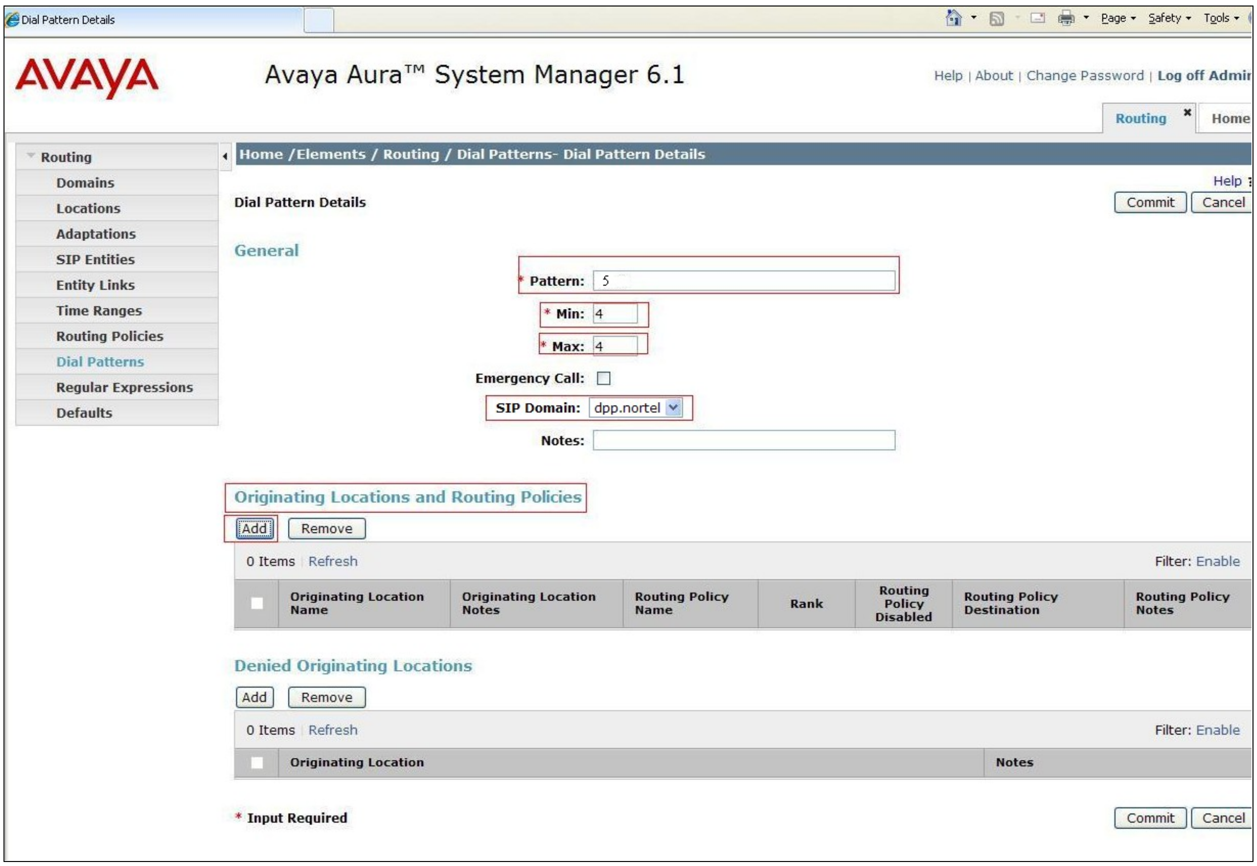
6.7. Create Avaya Communication Server 1000E Dial Pattern

A dial pattern is defined that directs calls to the CS1000E. All numbers beginning with 5 and are 4 digits in length (for example 5180) are routed to the CS1000E. Select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

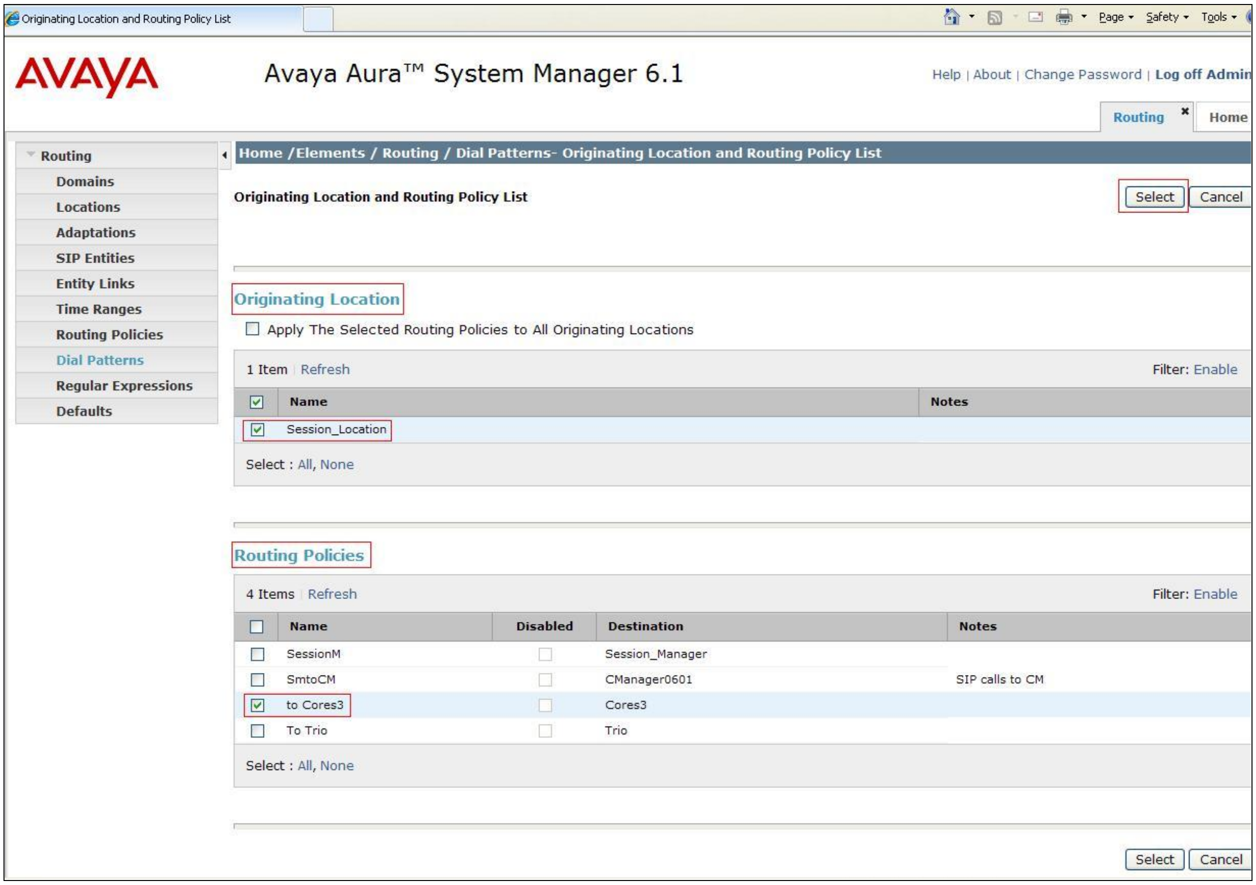
Under **General** carry out the following for each number.

- **Pattern** Dialed number or prefix **5**
- **Min** Minimum length of dialed number **4**
- **Max** Maximum length of dialed number **4**
- **SIP Domain** Select **dpp.nortel**

Continue to **Originating Locations and Routing Policy List.**



Select **Add** (not shown). Under **Originating Location**, check **Session Location** check box and under **Routing Policies** select **to Cores3.** Click **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. Configure TRIO Enterprise Server

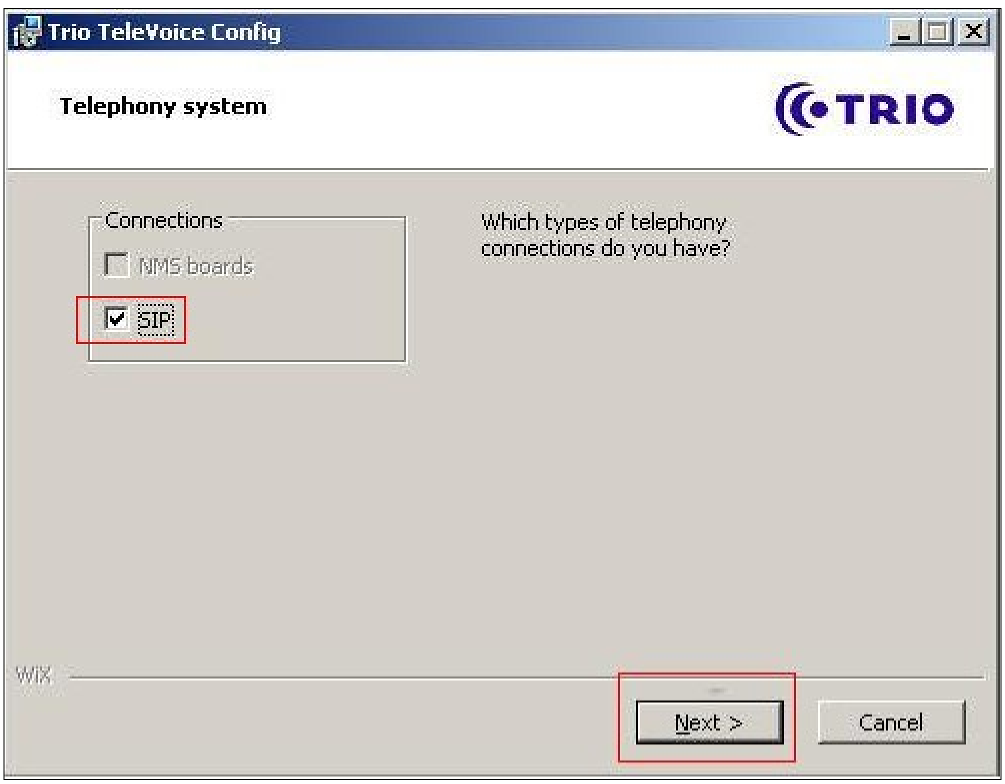
Trio Enterprise connects to the CS1000E through Session Manager. Trio Enterprise is added to Session Manager as a SIP endpoint and calls are routed to the Trio Enterprise server according to the dial Plan setup in **Sections 5.2 and 6.5**. This section shows how to configure Trio Enterprise to successfully 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.

7.1. Configure Trio Enterprise to use SIP Trunks

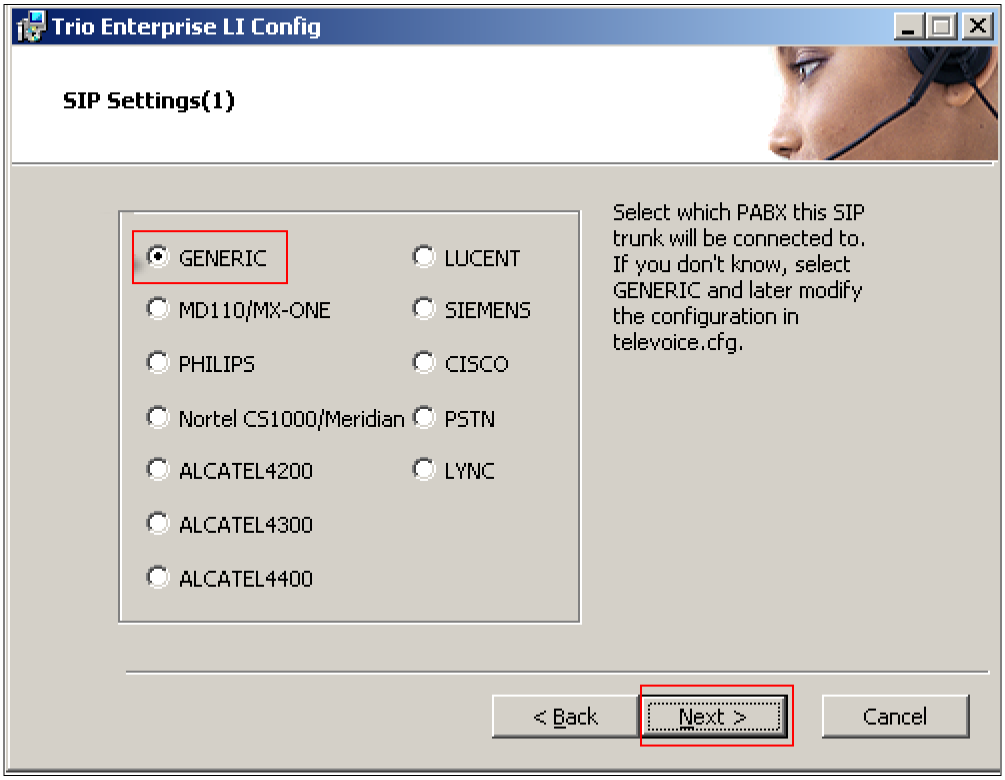
Access Windows services, select **Start** → **Run**, then type **services.msc** into the command line and press return (not shown). When the services window opens, locate the **Trio Televoice s**ervice right click and select **stop** to stop the service (not shown). Launch the Trio configuration application.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



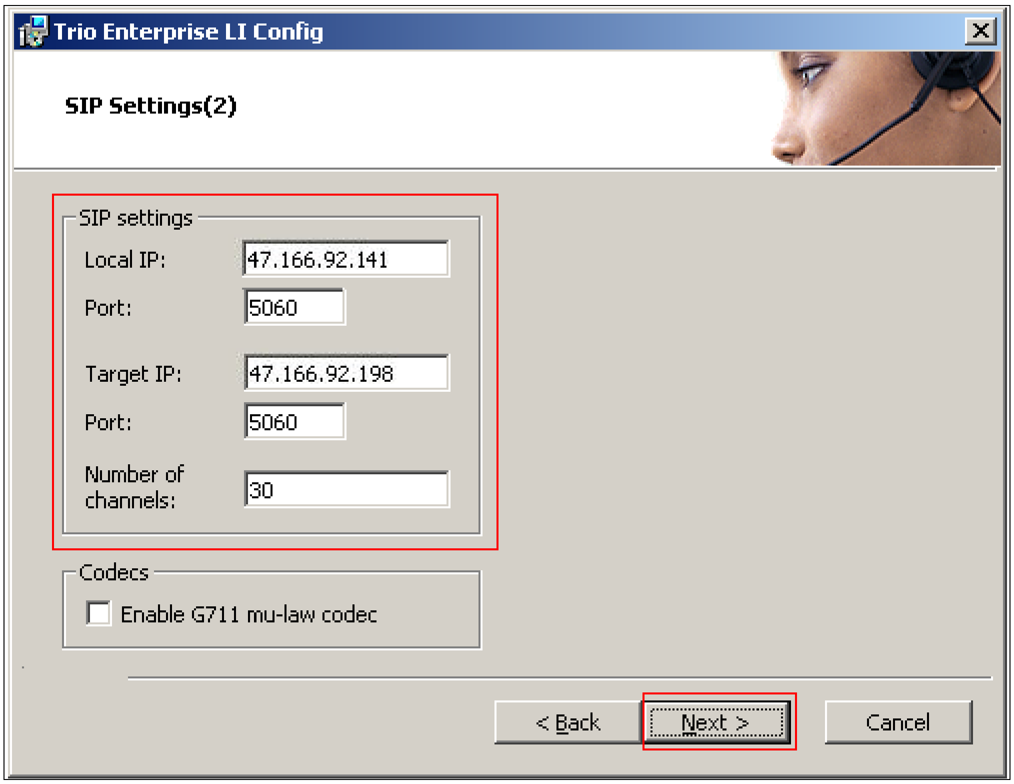
Select **GENERIC** under **SIP Settings**. Click **Next** to continue.



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

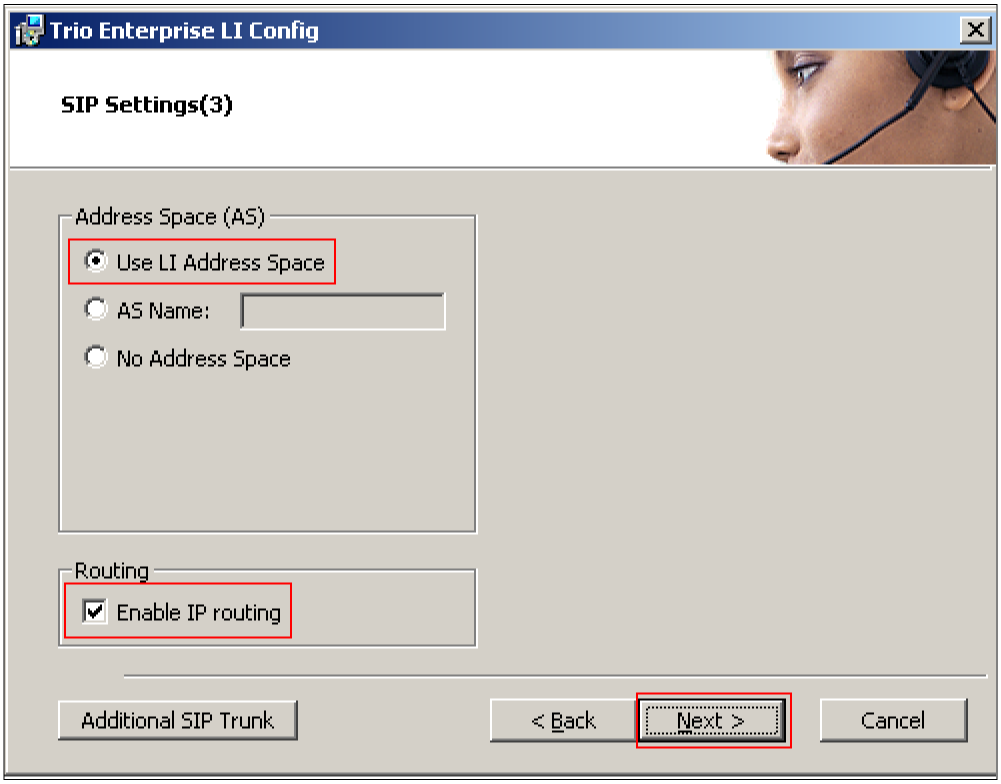
Click **Next** to continue.



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

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

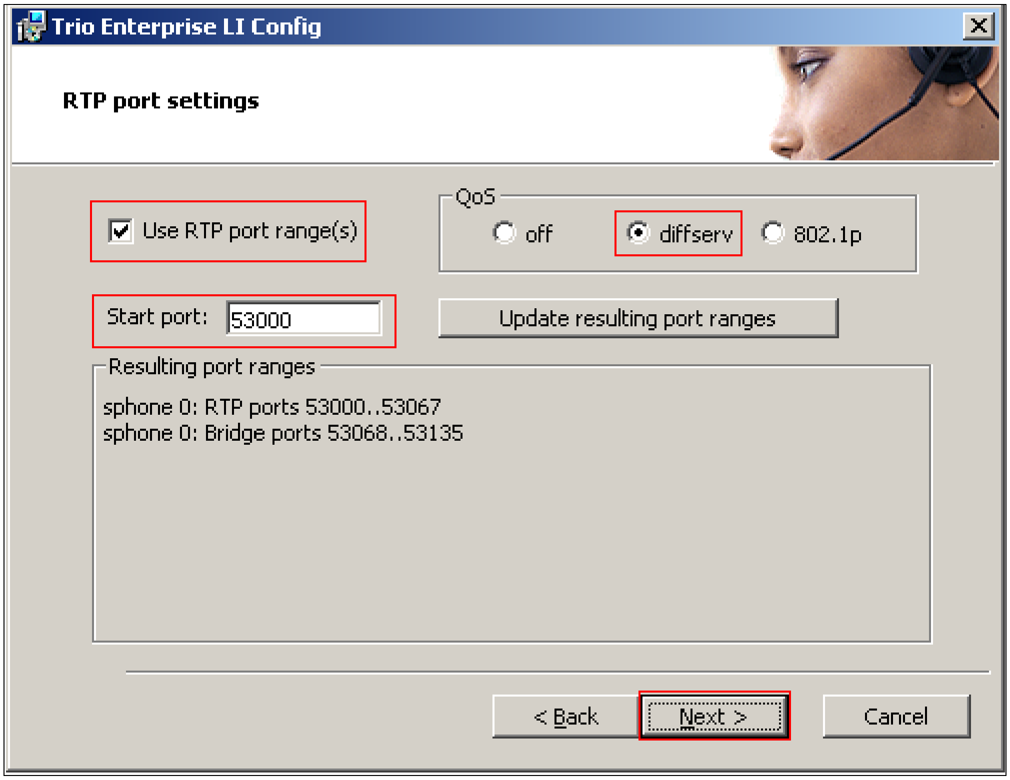
Click **Next** to continue.



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

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

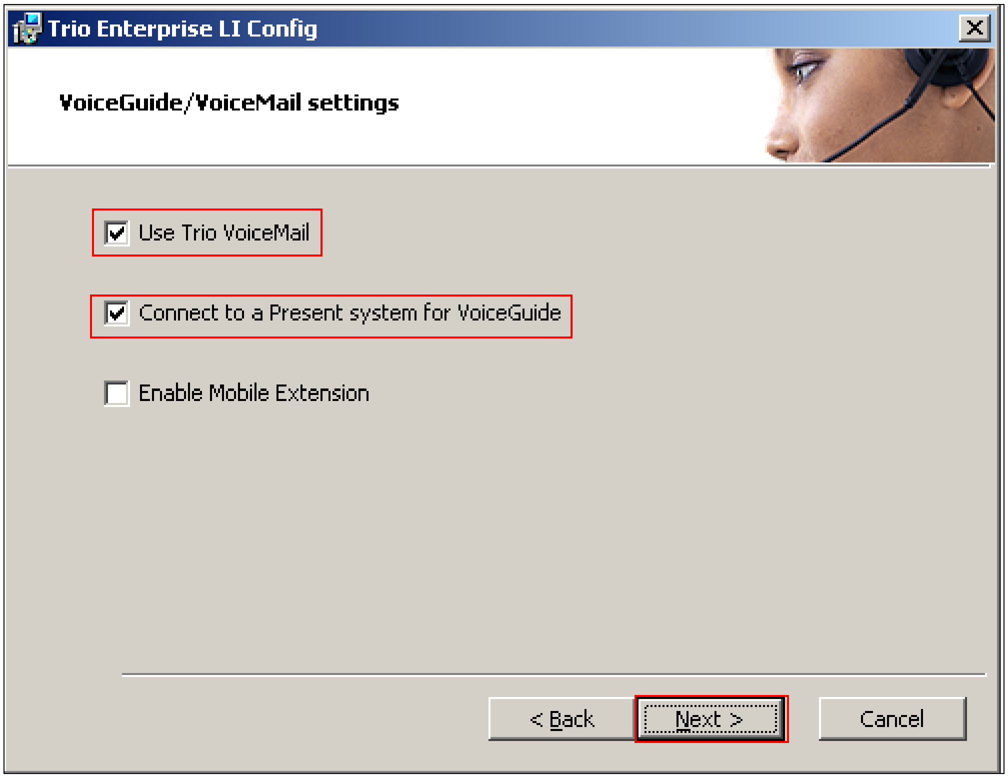
Click **Next** to continue.



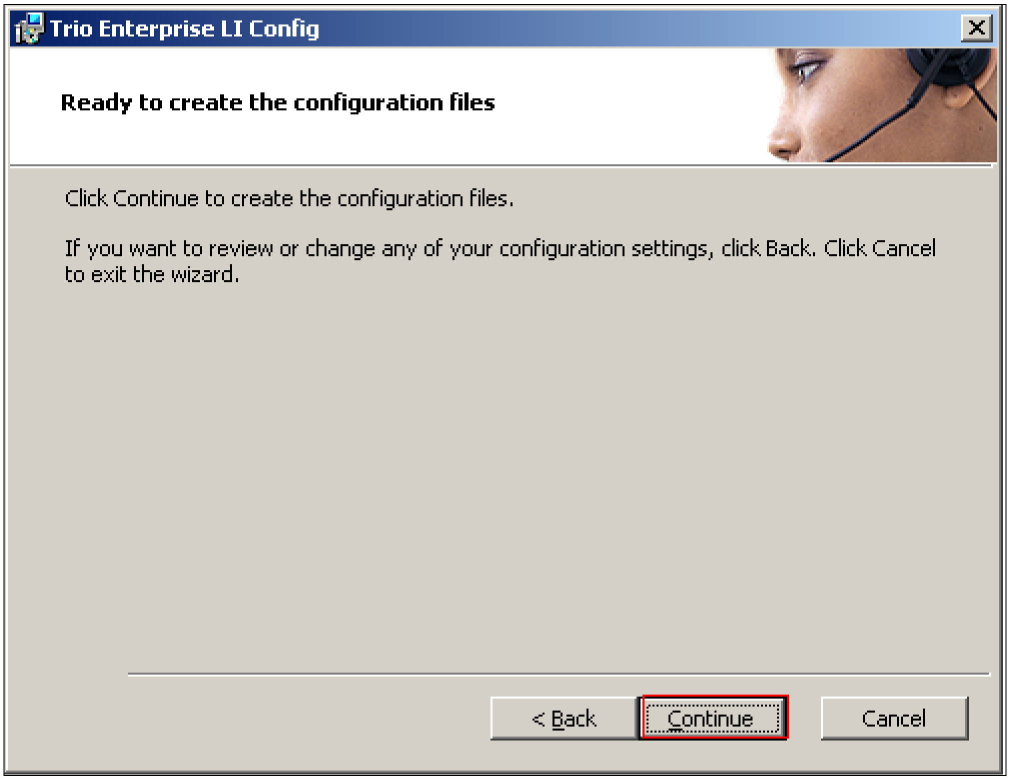
On the **VoiceGuide/VoiceMail settings** page, enter the following settings.

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

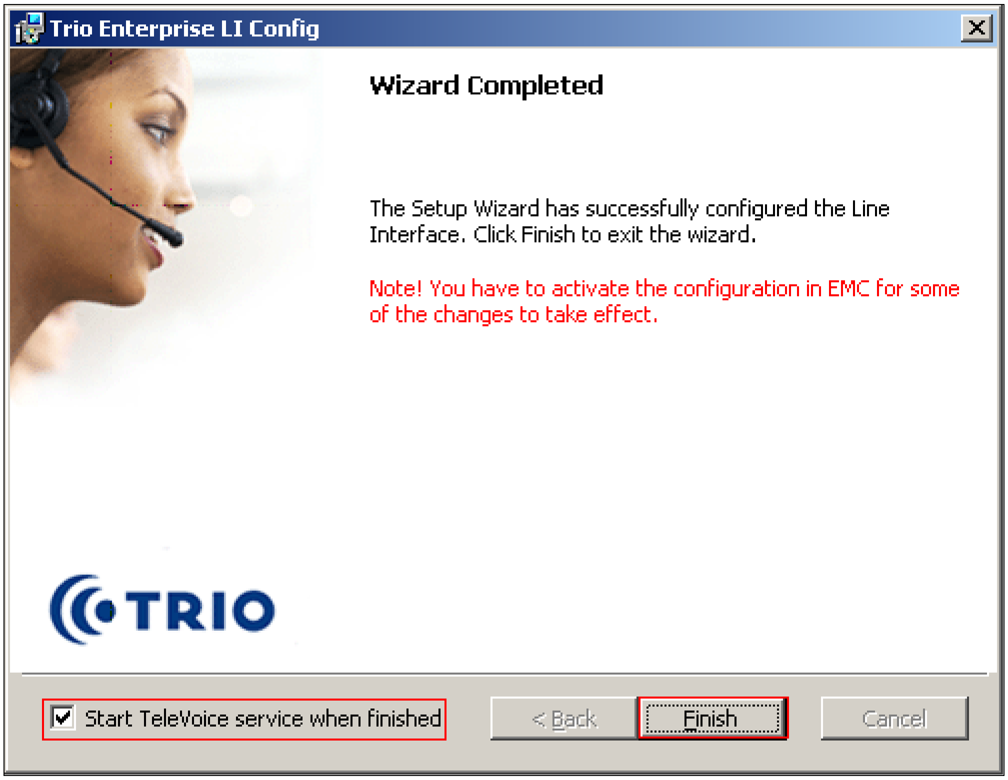
Click **Next** to continue.



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.



7.2. InteractionStudio Configuration

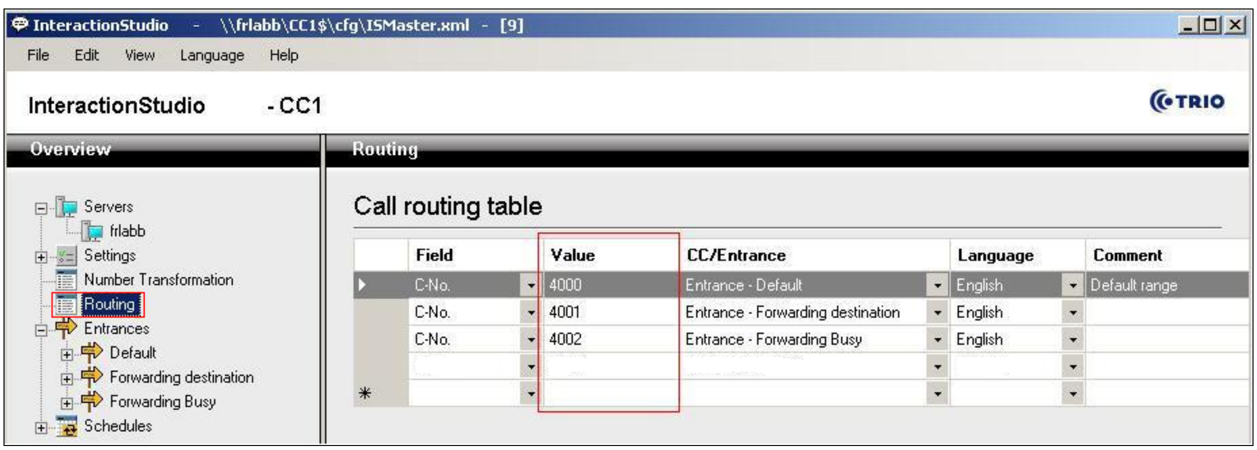
The InteractionStudio is used to configure many features for Trio Enterprise. For compliance testing, the following were configured.

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

7.2.1. Configure Call Routing table

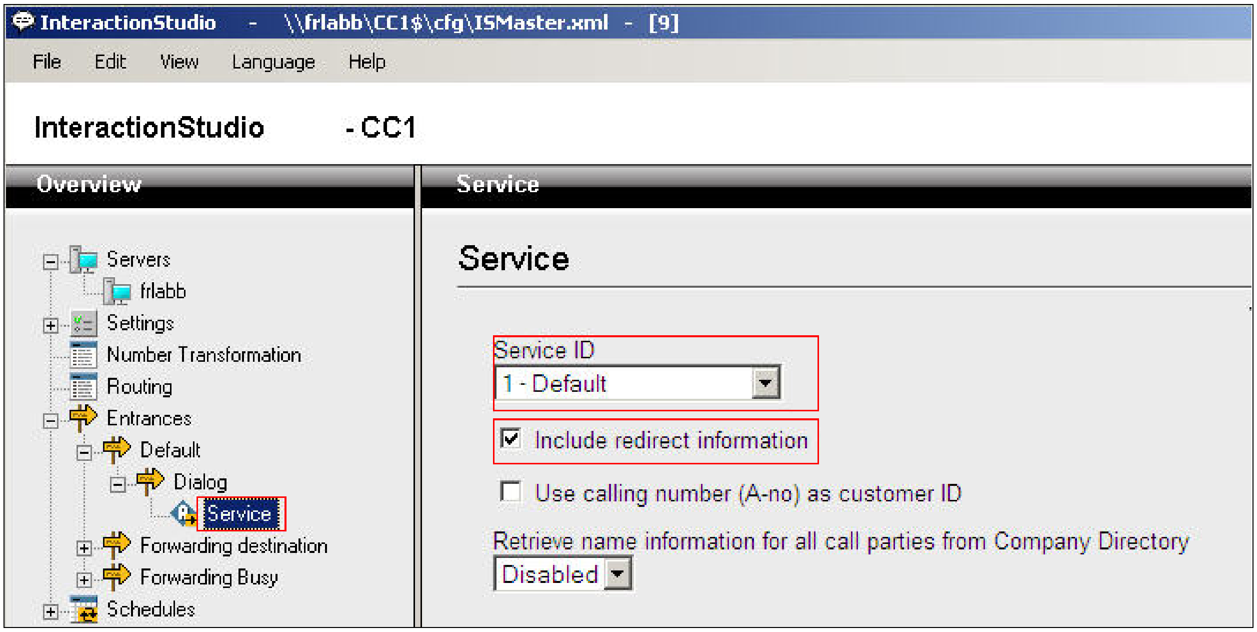
On the Trio Enterprise server, double click on the **InteractionStudio** executable file (not shown). When the InteractionStudio window opens, navigate to **Routing**. A **Call routing table** will open. In the example below:

- 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



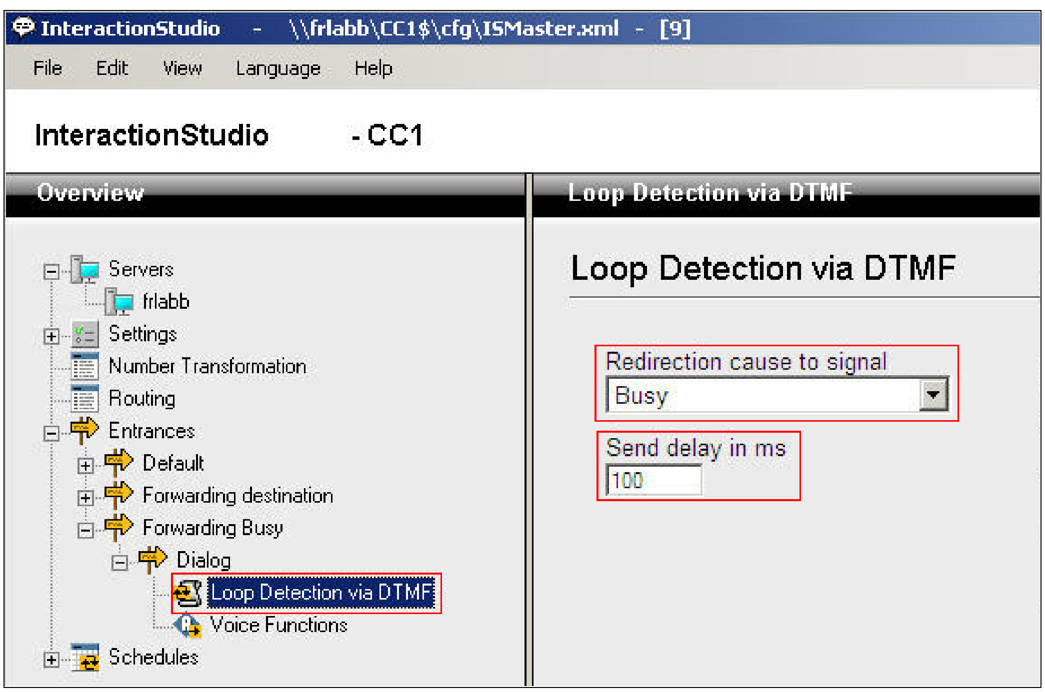
7.2.2. Configure Attendant Service

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



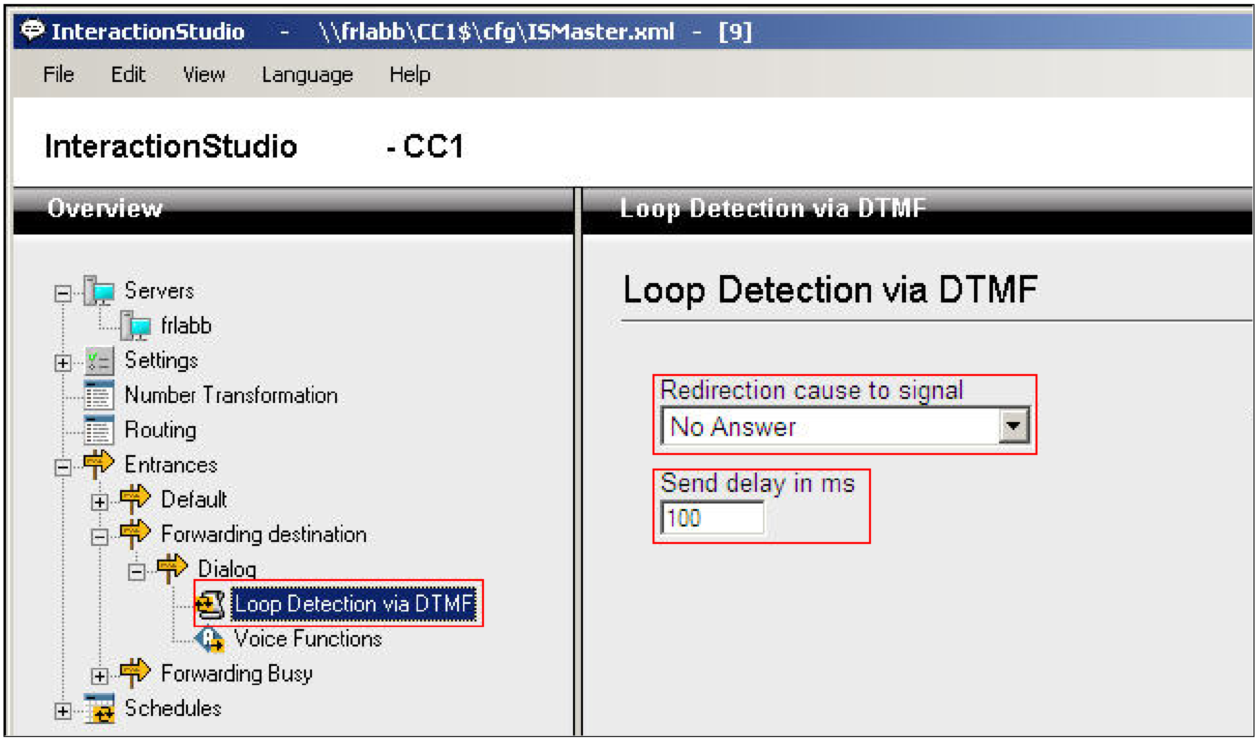
7.2.3. Configure Loop Detection via DTMF for Busy signal

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.



7.2.4. Configure Loop Detection via DTMF for No Answer signal

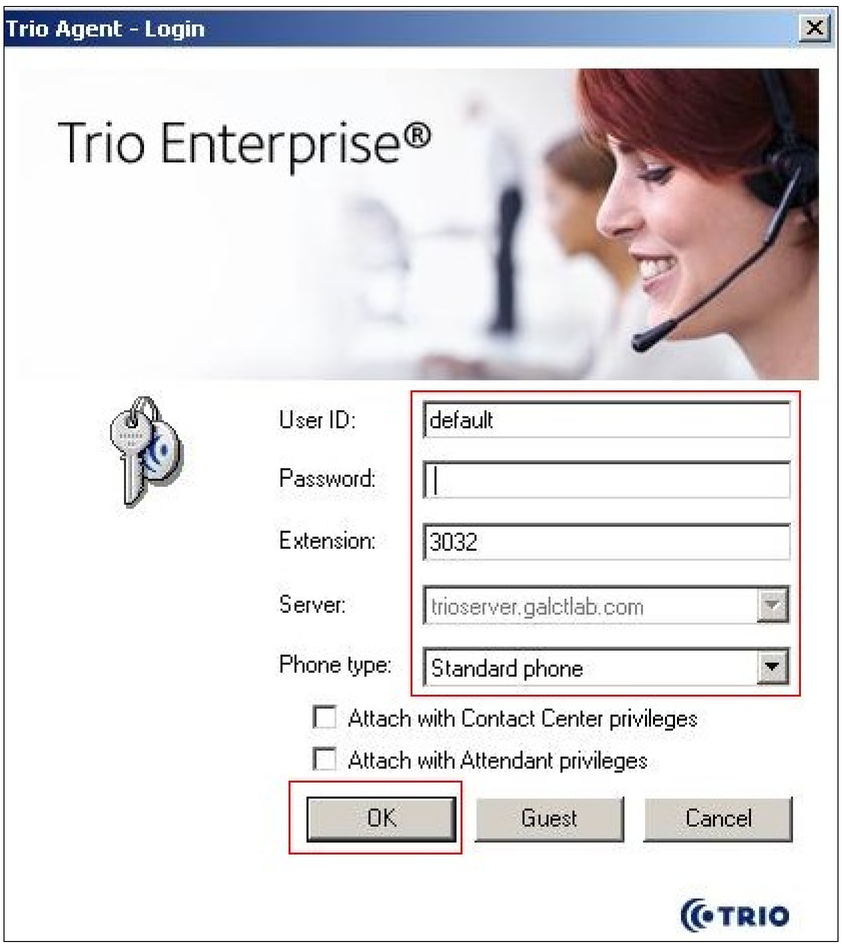
Navigate to **Entrances → Forwarding destination → Dialog → Loop Detection via DTMF.** Choose **No Answer** from the **Redirection cause to signal** drop down box, and enter **100** in the **Send delay in ms** box.



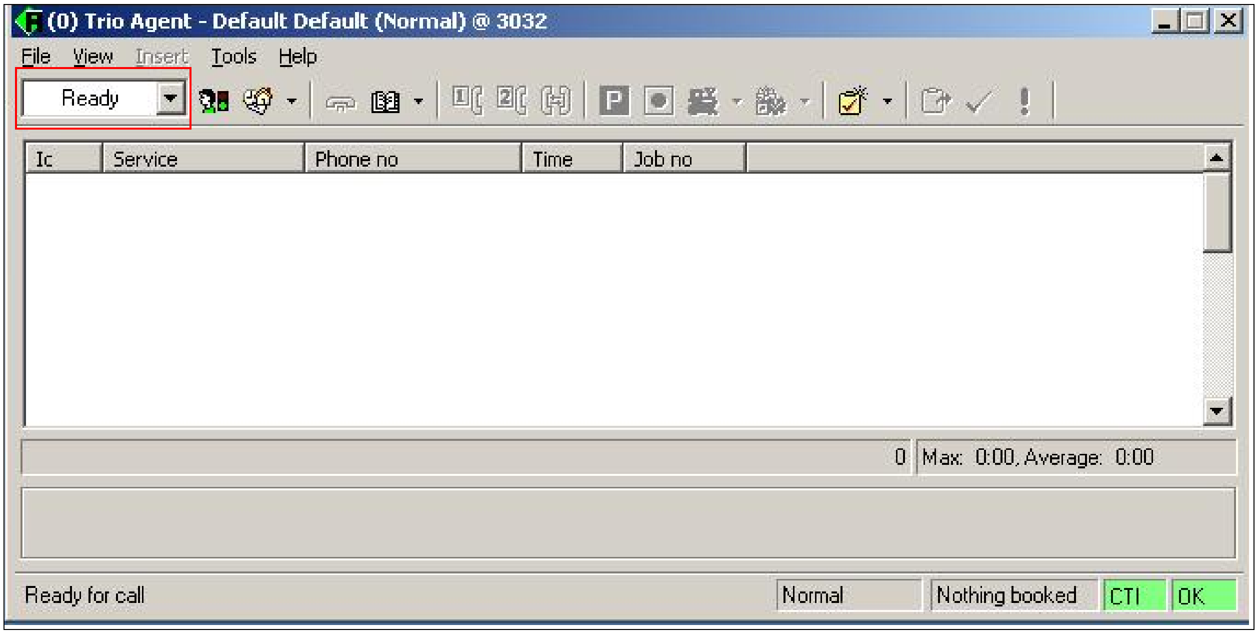
7.3. Configuring Trio Attendant

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**. 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. Verification Steps

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

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

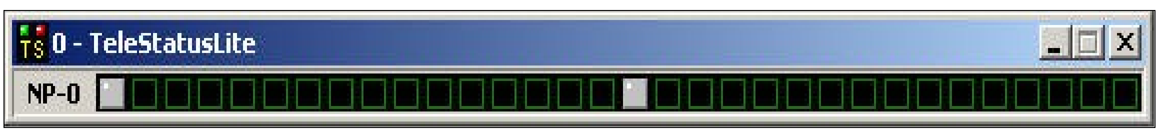
Check the status of the D-channel setup in **Section 5.1.2** by running the command **STAT DCH** in overlay 96 as shown below. The example below shows that D-Channel **66** is operational and established.

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**

8.2. Status of D-Channel 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 with the first and seventeenth squares greyed out (these are the D-Channel and resync timeslots). Confirm the trunks are all in the idle state (unfilled green squares).



9. Conclusion

These Application Notes describe the configuration steps required for Trio Enterprise R3.2 to successfully interoperate with Avaya Communication Server 1000E R7.5 and Avaya Aura® Session Manager R6.1 using SIP trunks. Trio Enterprise passed all compliance testing successfully; please see **Section 2.2** of these Application Notes for results and observations.

10. 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 7.5; Document No. NN43001-611_05.02, Dec 2010*

[2] *Administering Avaya Aura® Session Manager, Doc # 03603324, Issue 1 Release 6.1*

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

```
VERSION 4121
RELEASE 7
ISSUE 50 Q +
DepList 1: core Issue: 01 (created: 2011-03-15 10:26:33 (est))

IN-SERVICE PEPS
PAT# CR #            PATCH REF #     NAME      DATE        FILENAME       SPECINS
000  wi00688505      ISS1:1OF1       p30595_1  14/06/2011  p30595_1.cpl   NO
001  wi00835294      ISS1:1OF1       p30565_1  14/06/2011  p30565_1.cpl   NO
002  wi00832106      ISS1:1OF1       p30550_1  14/06/2011  p30550_1.cpl   NO
003  wi00837618      ISS1:1OF1       p30594_1  14/06/2011  p30594_1.cpl   NO
004  wi00852365      ISS1:1OF1       p30707_1  14/06/2011  p30707_1.cpl   NO
005  wi00843623      ISS1:1OF1       p30731_1  14/06/2011  p30731_1.cpl   YES
006  wi00839255      ISS1:1OF1       p30591_1  14/06/2011  p30591_1.cpl   NO
007  wi00832626      ISS2:1OF1       p30560_2  14/06/2011  p30560_2.cpl   NO
008  wi00857566      ISS1:1OF1       p30766_1  14/06/2011  p30766_1.cpl   NO
009  wi00841980      ISS1:1OF1       p30618_1  14/06/2011  p30618_1.cpl   NO
010  wi00837461      ISS1:1OF1       p30597_1  14/06/2011  p30597_1.cpl   NO
011  wi00839821      ISS1:1OF1       p30619_1  14/06/2011  p30619_1.cpl   NO
012  wi00842409      ISS1:1OF1       p30621_1  14/06/2011  p30621_1.cpl   NO
013  wi00838073      ISS1:1OF1       p30588_1  14/06/2011  p30588_1.cpl   NO
014  wi00850521      ISS1:1OF1       p30709_1  14/06/2011  p30709_1.cpl   YES
015  wi00860722      ISS1:1OF1       p30784_1  14/06/2011  p30784_1.cpl   YES
016  wi00839134      ISS1:1OF1       p30698_1  14/06/2011  p30698_1.cpl   YES
017  wi00836981      ISS1:1OF1       p30613_1  14/06/2011  p30613_1.cpl   NO
```

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.