Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Trio Enterprise R3.2 with Avaya Communication Server 1000E R7.5 and Avaya Network Routing Server 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 Network Routing Server 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 Network Routing Server R7.5. 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 1140E was 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 an Avaya Communication Server 1000E (CS1000E). The Trio Enterprise server connects to the CS1000E via SIP trunks configured on an Avaya Network Routing Server (NRS), 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 the NRS. The Trio Enterprise Server is configured as a SIP Endpoint. Avaya Unified Communications Management is used to configure the NRS.

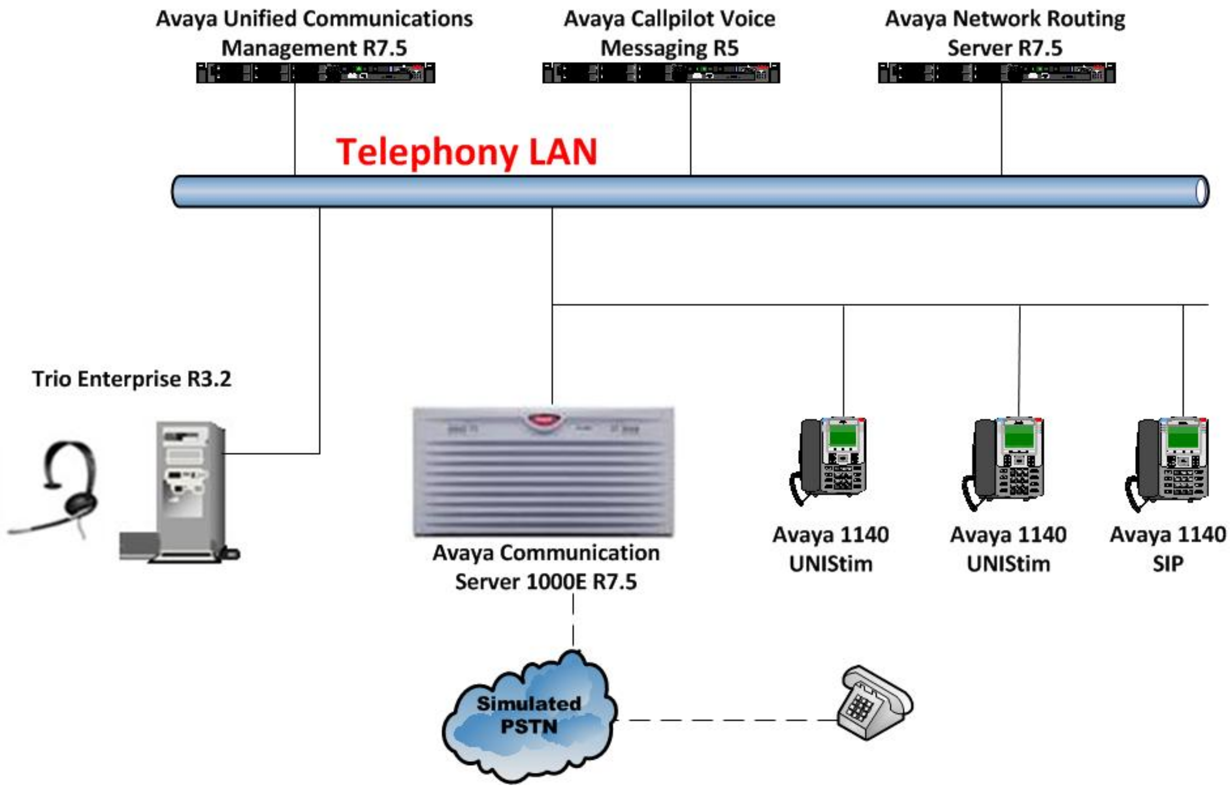


Figure 1: Configuration for Avaya Communication Server 1000E, Avaya Network Routing Server and Trio Enterprise R3.2

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 Unified Communications Management running on Avaya S8800 Server	R7.5
Avaya Network Routing Server running on Avaya S8800 Server	R7.5
Avaya 1140 UNIStim Deskphone	UNIStim V0625C8D
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 for Trio the following steps are used.

- 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 Communication Server 1000E 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	**CHG**	Change
TYPE	NIT	Night Service
CUST	**0**	Customer Number
NIT1	5000	XXXXXXXXXXXXXXXXX

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 was created. At the **CTYP** prompt, enter **DCIP.** This signifies the SIP D-Channel.

Note: In the Telnet screenshots below, only the unique prompt inputs are shown. Enter a carriage return (CR) for all other prompts to set default values.

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 5.1.2.** 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 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 an **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.**

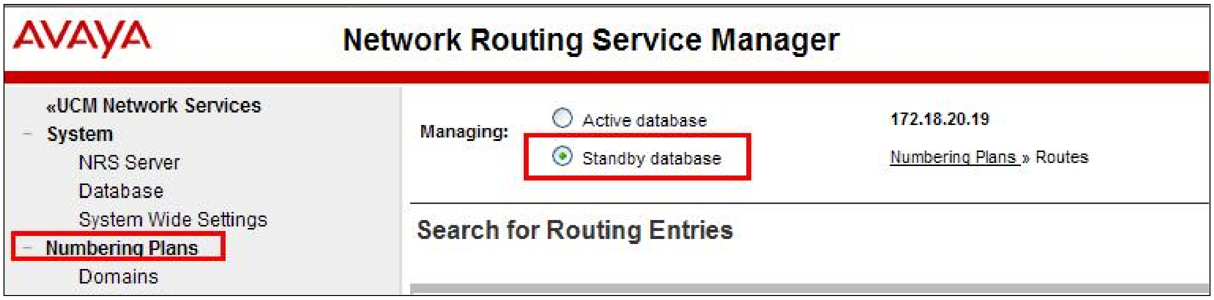
Note: The RLI number used is the one created in **Section 5.2.1.**

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

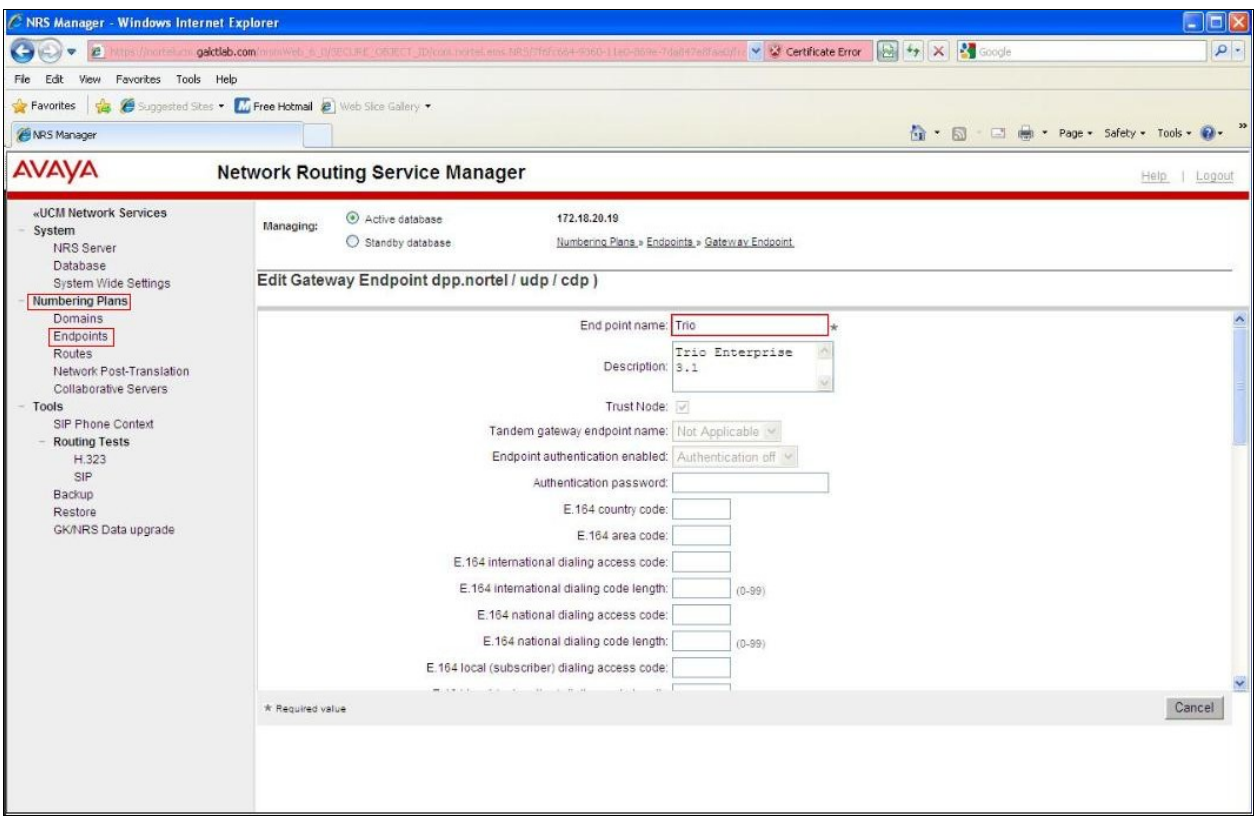
6. Configuration of Trio Enterprise on Avaya Network Routing Server

In order to make any changes on the NRS the standby database must be active, ensure that the **Standby database** button is selected as shown below.

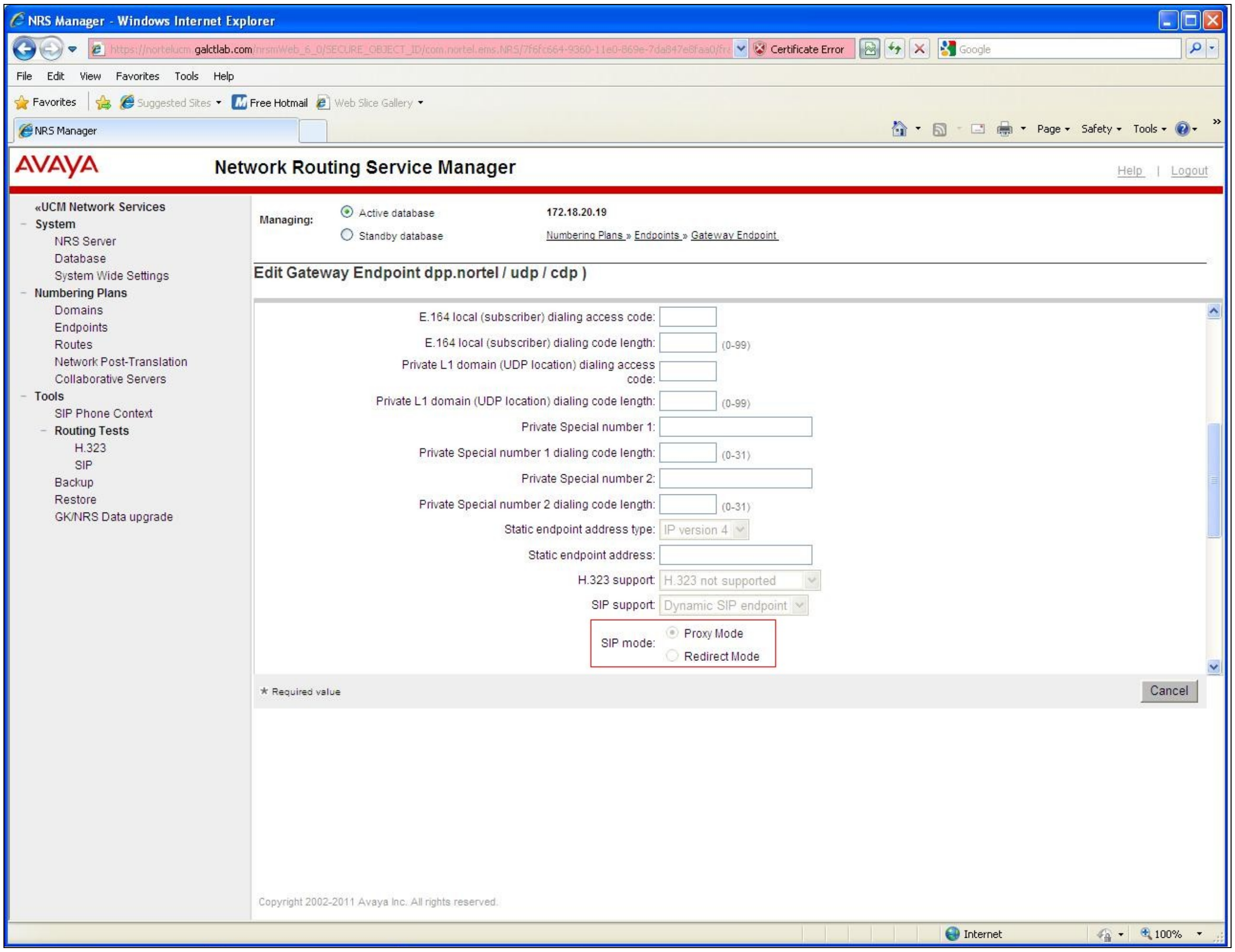


6.1. Configure Trio Enterprise as a Gateway Endpoint

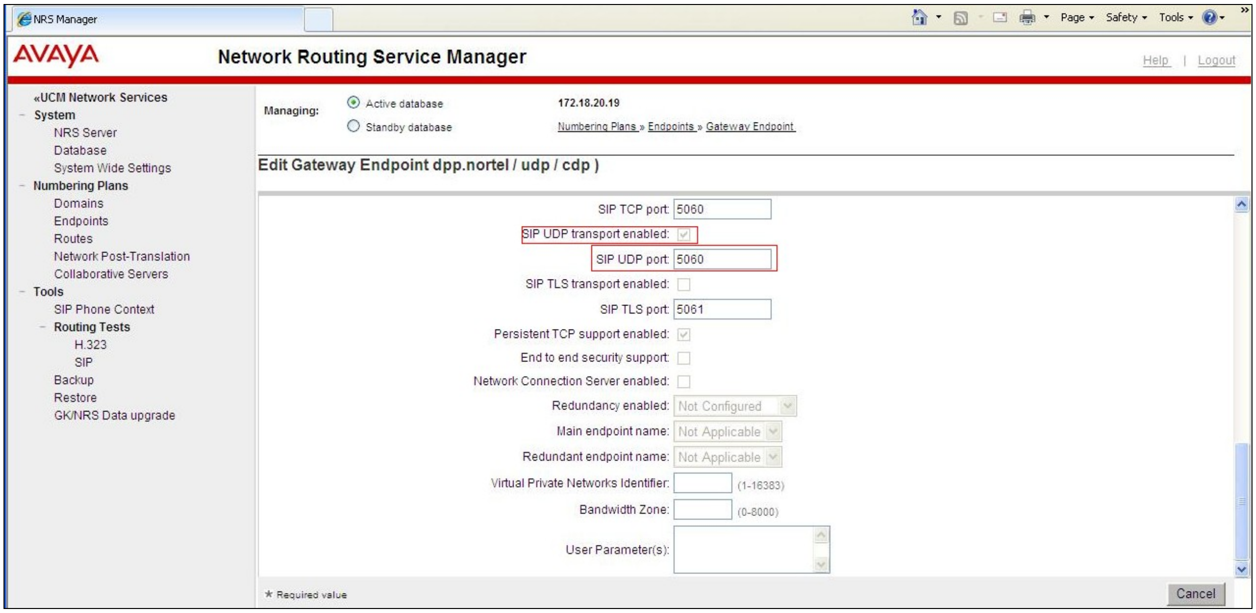
The Trio Enterprise is configured as a Dynamic Endpoint on the NRS. From the NRS Manager Page, navigate to **Numbering Plans → Endpoints,** enter **End point name** Trio.



Scroll down to **SIP Mode** and select the **Proxy Mode** radio button.



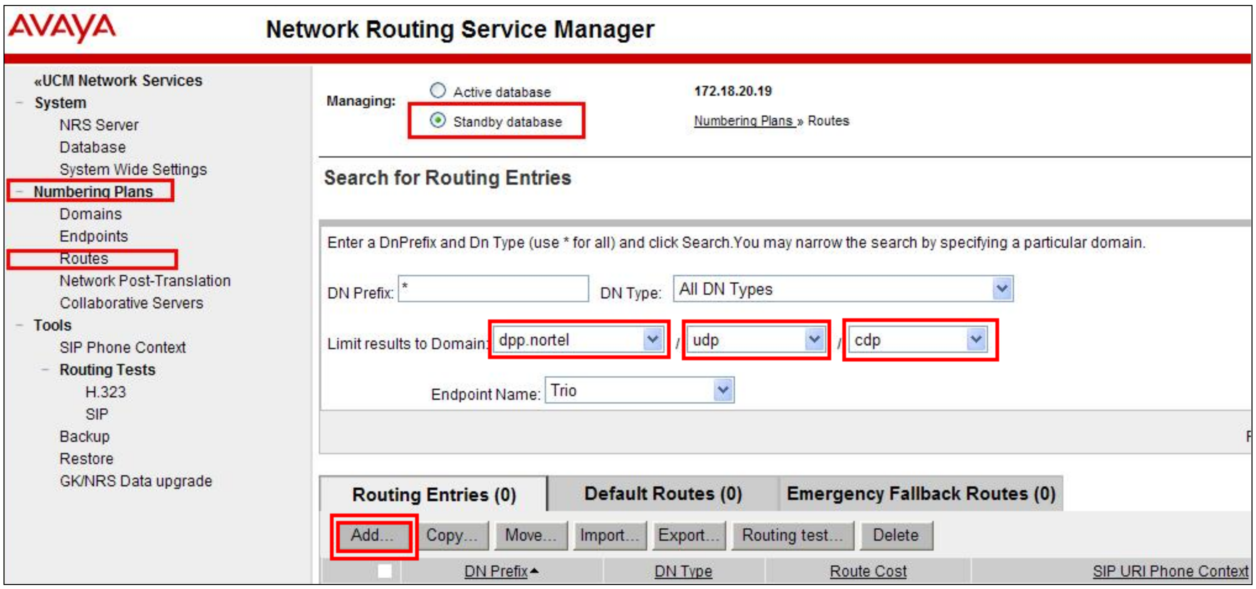
Scroll down to **SIP UDP Transport enabled** checkbox. Enable **SIP UDP Transport** and enter **5060** for **SIP UDP Port.**



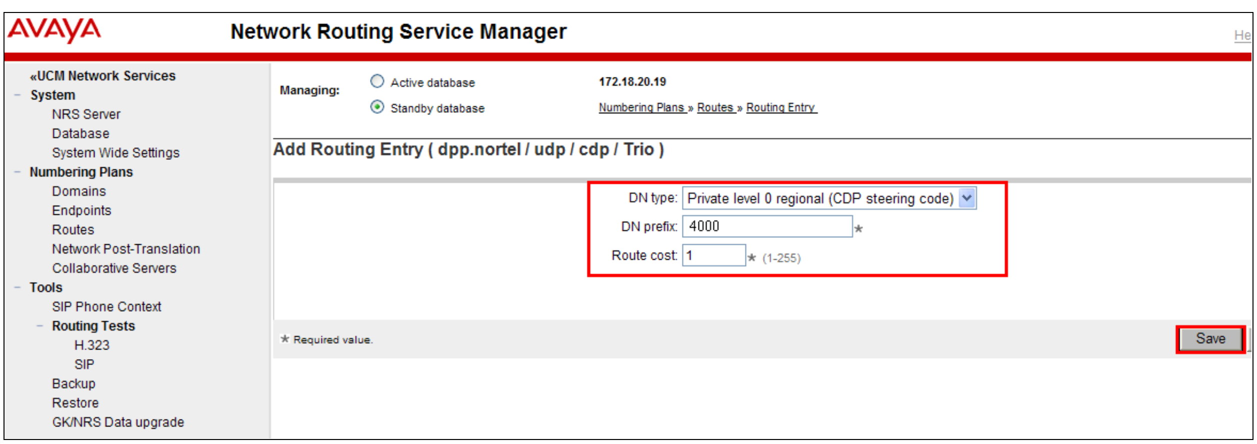
6.2. Configure Routing Entries for Trio Enterprise Gateway Endpoint

From the **NRS Manager** page, navigate to **Numbering Plans → Routes.**

- Select the appropriate **Domain** from the drop down box highlighted below
- Select the L0 Domain as **UDP** and L1 Domain as **CDP** as shown below
- Select **TRIO** as **Endpoint Name (configured in** and click on **Add** button to add a routing entry



- Select **Private level 0 regional (CDP steering code)** for **DN type**
- Enter the DN. (Example 4000) for **DN Prefix**
- Enter **1** for **Route cost**
- Click **Save**



Once the configuration is complete the Database must be cutover to become **Active** (not shown).

7. Configure TRIO Enterprise Server

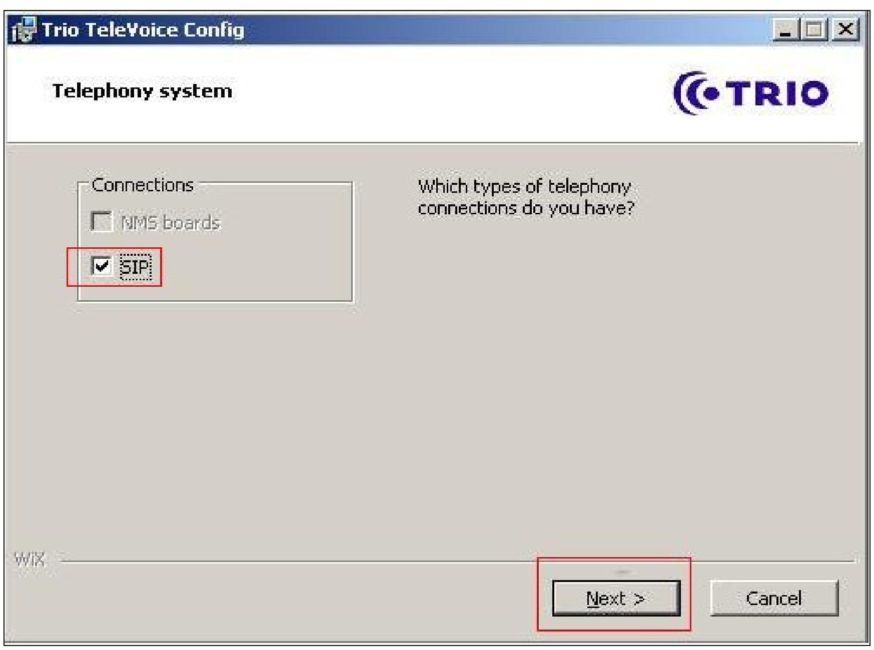
This section describes how to integrate Trio Enterprise with the CS1000E by registering to the NRS. Trio Enterprise is added to the NRS as a dynamic SIP Endpoint, see configuration setup in **Section 6.1,** in order to successfully route calls to that endpoint. . The installation of the Trio Enterprise software is assumed to be completed and the Trio services are up and running

Note: During the configuration of Trio Enterprise in some windows shows **Nortel CS1000/Meridian**, this relates to the **Avaya Communication Server 1000E.**

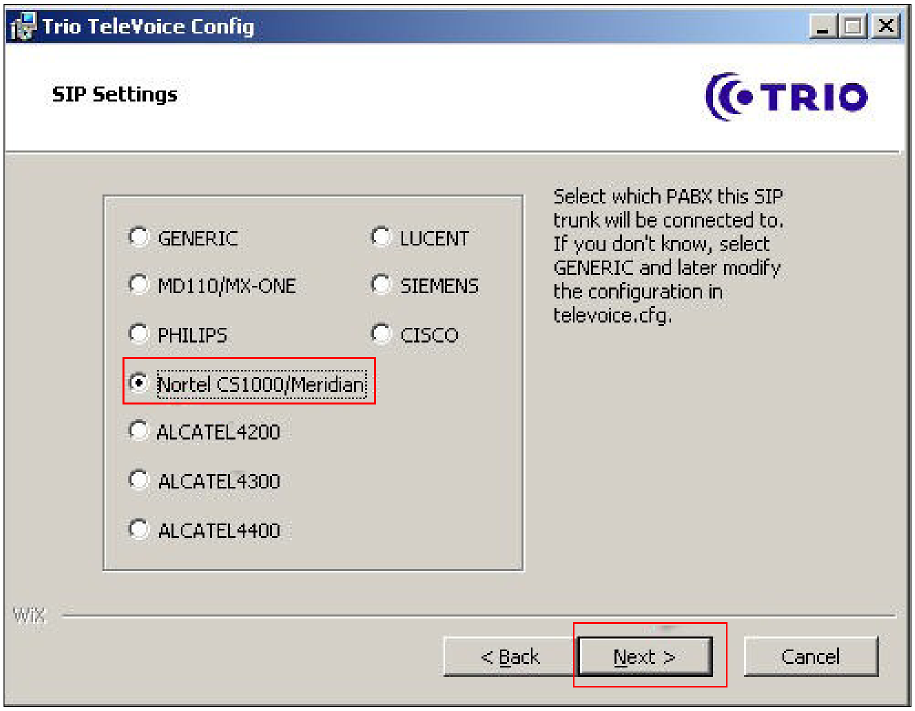
7.1. Configure Trio Enterprise to use SIP Trunks

Trio Enterprise must be connected to Communication Server 1000E before it can process calls. This section shows how to configure Trio Enterprise with a SIP connection to the CS1000E.. The steps to configure SIP Trunks are as follows:

1. Access Windows services. Select **Start** → **Run**, then type **services.msc** into the command line. Press return (not shown)
2. When the standard services window opens, locate the Trio Televoice service and stop the service (not shown)
3. 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
4. Ensure the **SIP** entry in the **Connections** area is checked
5. Click **Next** to continue



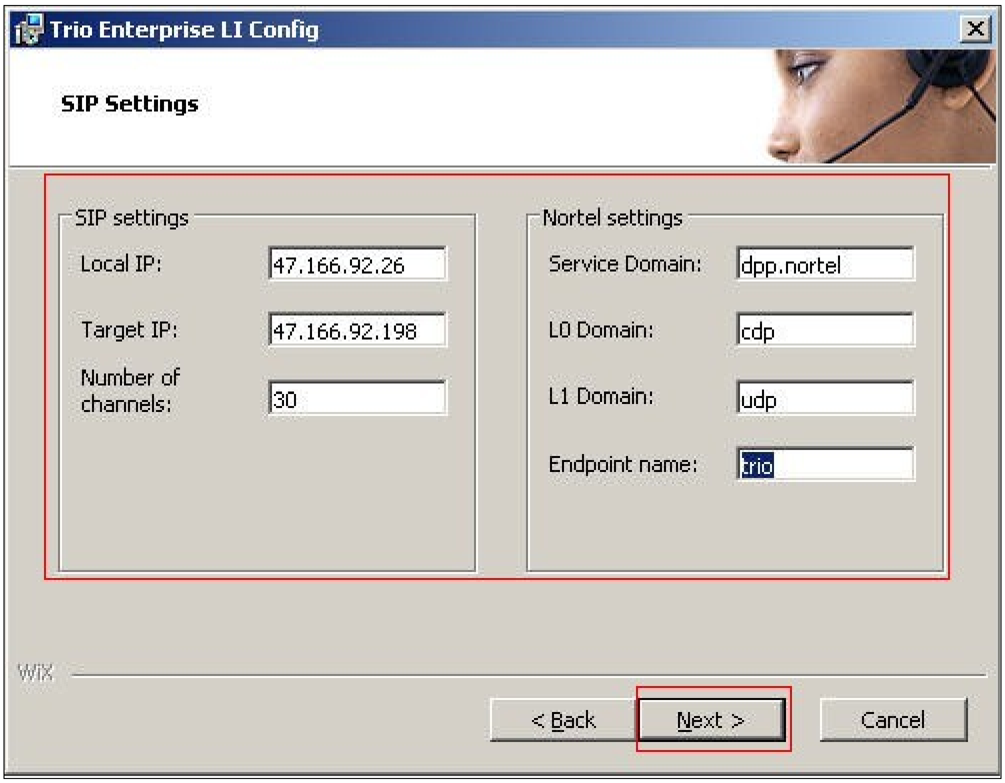
Select **Nortel CS1000/Meridian** under **SIP Settings**. Click **Next** to continue.



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

1. **Local IP** The local IP address of the Trio Enterprise server
2. **Target IP** The IP address of the Network Routing Server (NRS)
3. **Number of channels** The number of channels
4. **Service Domain** The Service domain configured in Network Routing Server (NRS)
5. **L0 Domain** The L0 Domain configured in Network Routing Server (NRS)
6. **L1 Domain** The L1 Domain configured in Network Routing Server (NRS)
7. **Endpoint name** TRIO endpoint name configured in Network Routing Server (NRS), as configured in **Section 6.1**

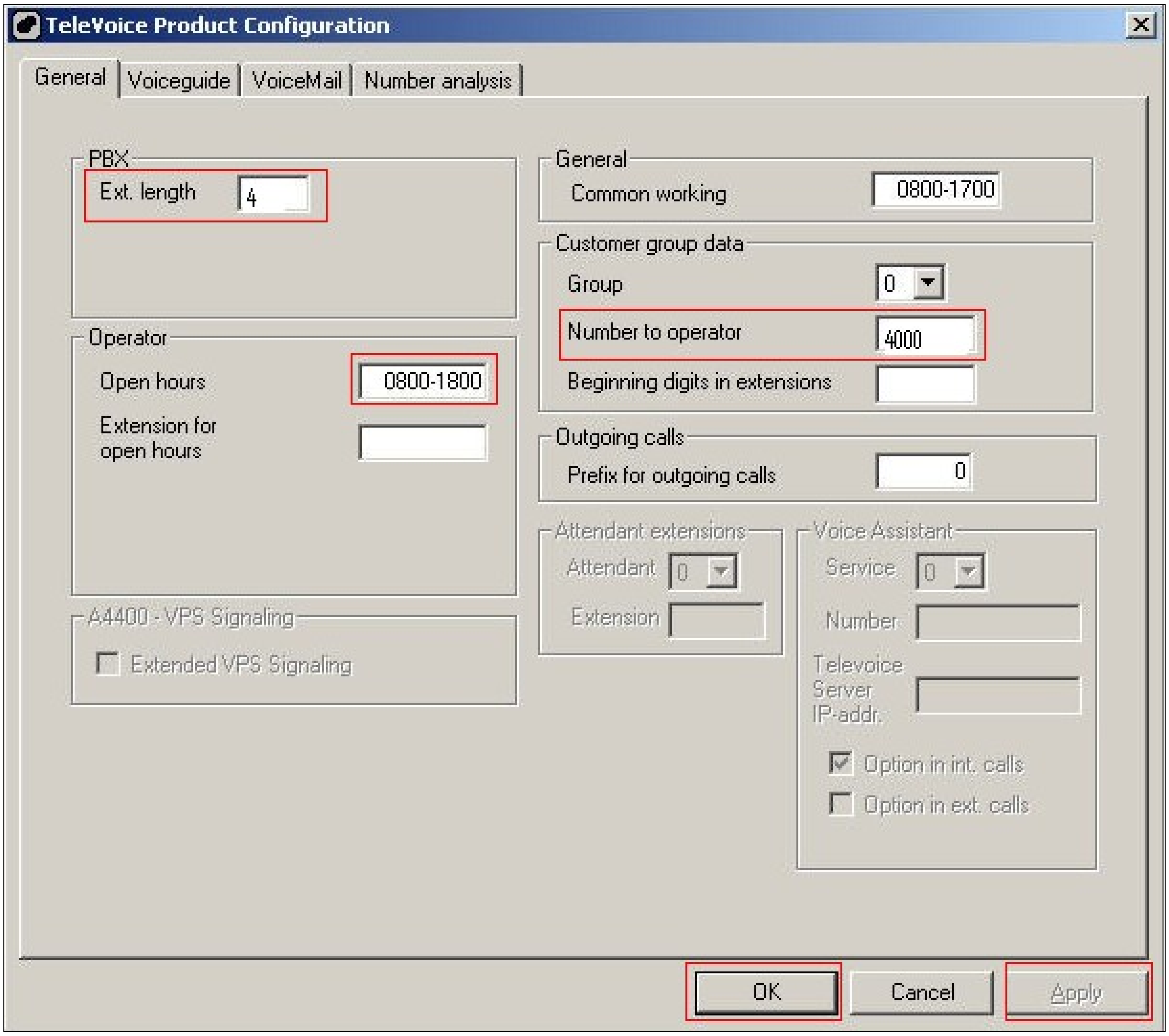
Click **Next** to continue.



In the **General** tab on the **TeleVoice Product Configuration** page, enter the following:

- **Ext. length** Ext length is **4**
- **Operator Open hours** Example **0800-1800**
- **Number of operator** Note **4000** is the DN routing to Trio setup in Section 6.2

Click on **Apply** button followed by the **OK** button.



7.2. InteractionStudio Configuration

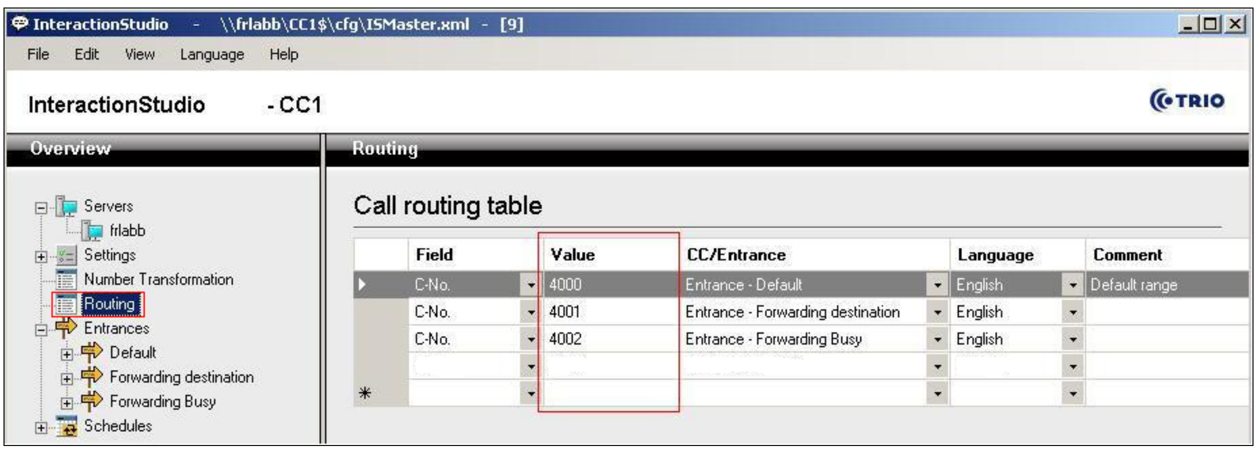
The InteractionStudio is used to configure many features for Trio Enterprise. For compliance testing, the following were configured.

- 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

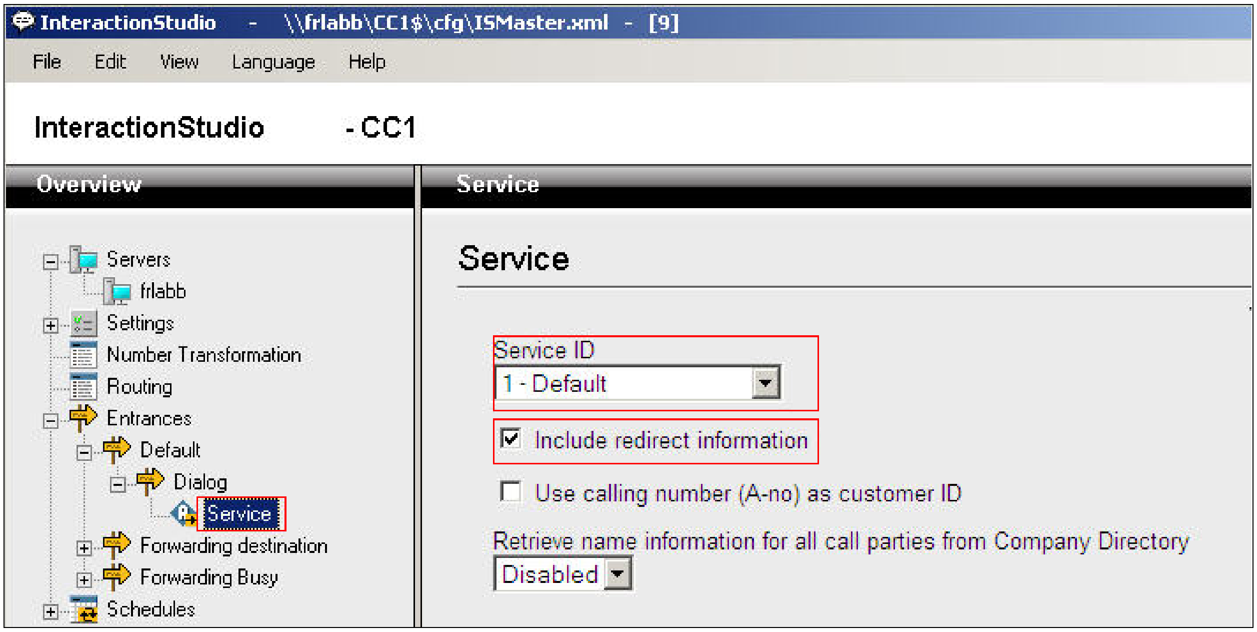
On the Trio Enterprise server, double click on the **InteractionStudio** executable file (not shown). When the InteractionStudio window opens, navigate to **Routing**. A **Call routing table** will open. In the example below:

- 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.



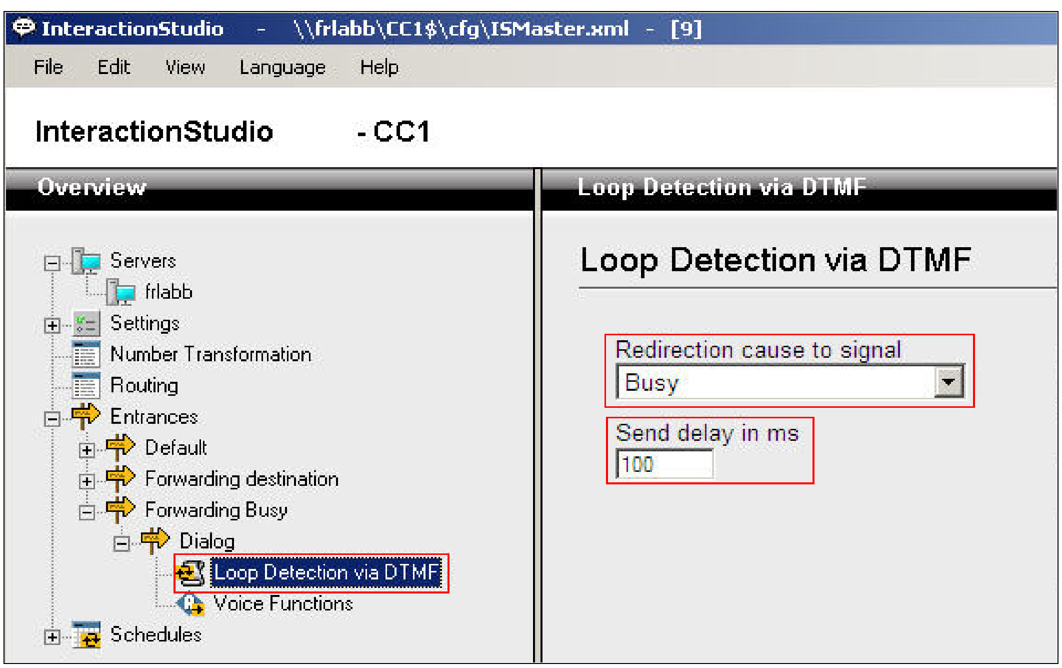
7.2.2. Configure Attendant Service

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



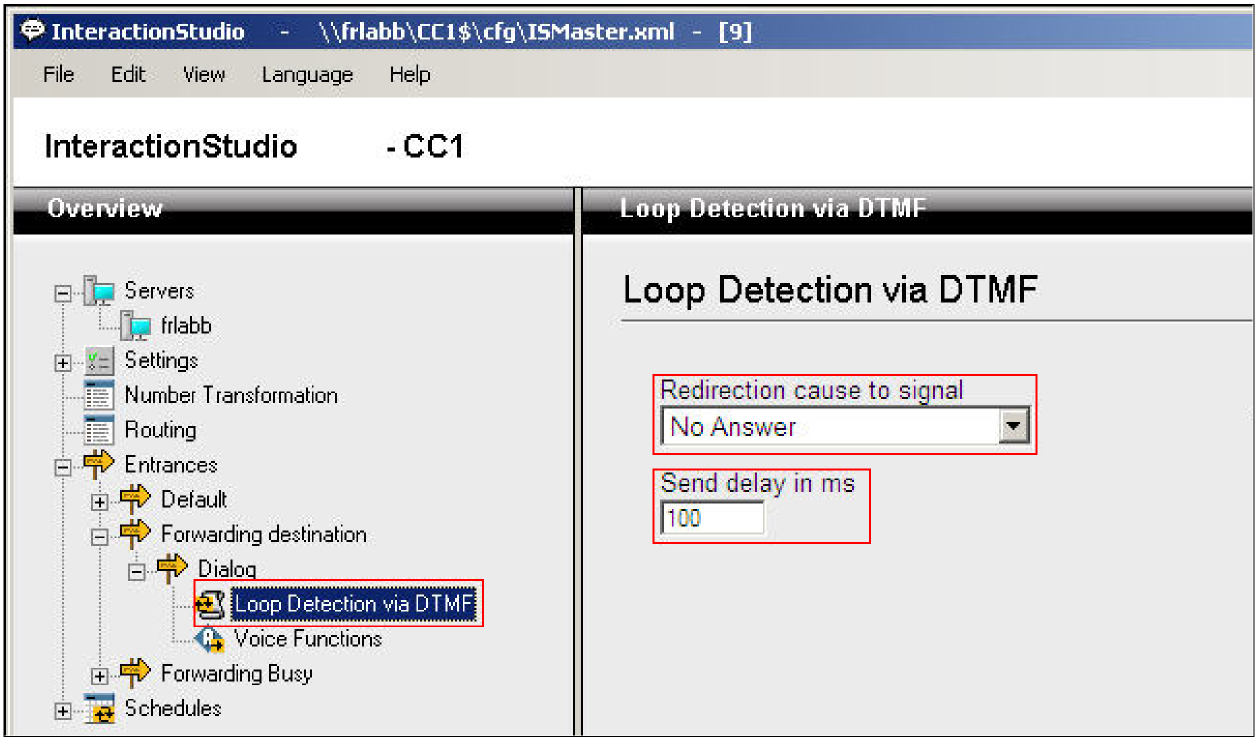
7.2.3. Configure Loop Detection via DTMF for Busy signal

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

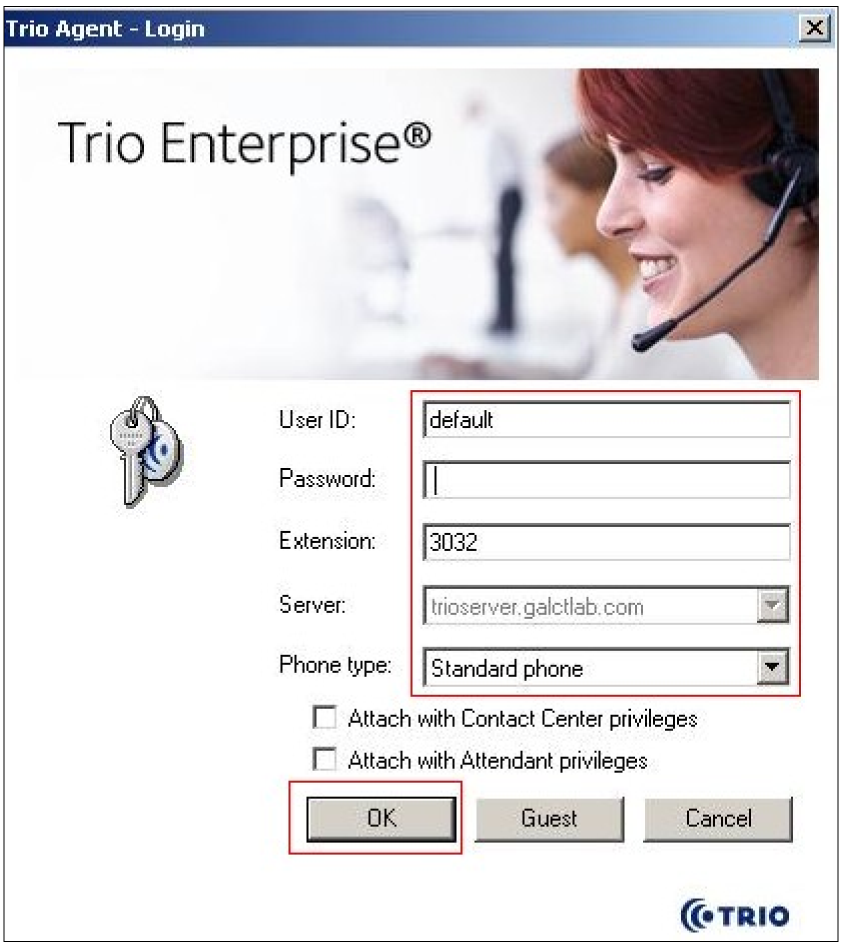
Navigate to **Entrances → Forwarding destination → Dialog → Loop Detection via DTMF.** Choose **No Answer** from the **Redirection cause to signal** drop down box, and enter **100** in the **Send delay in ms** box.



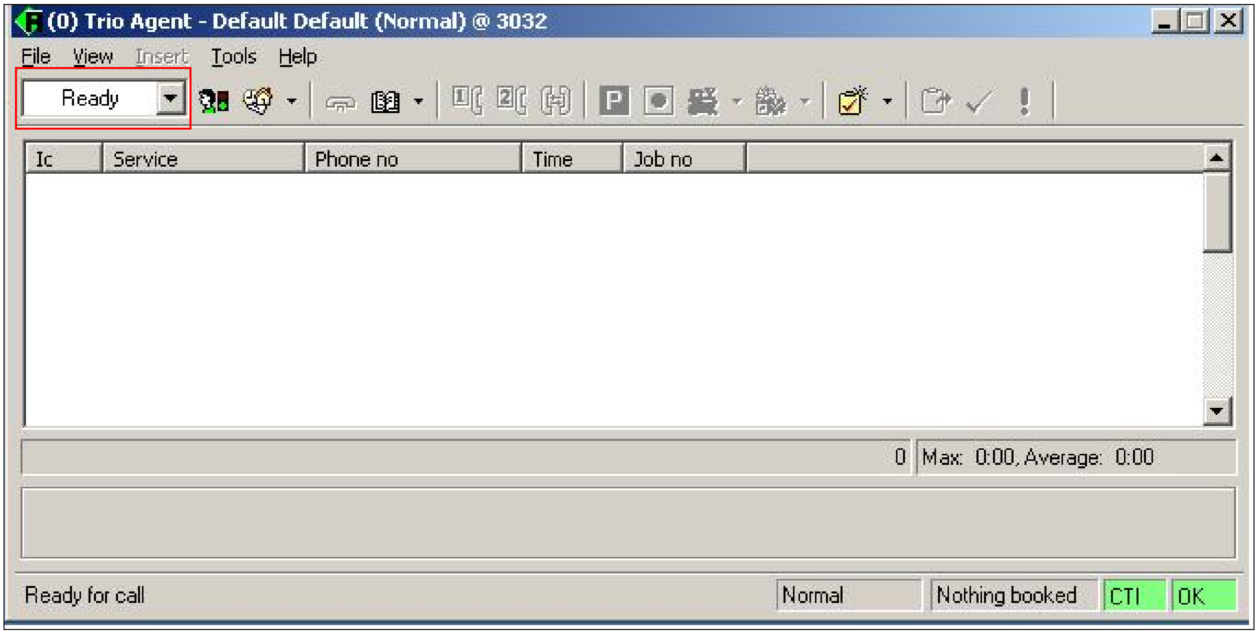
7.3. Configuring Trio Attendant

Trio attendant is a separate application to Trio Enterprise server and can run concurrently on the same platform. The attendant uses a regular Communication Server 1000E telephone to make and receive calls, which are directed to the phone by Trio Enterprise server. The steps to configure Trio Attendant are as follows. 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 Communication Server 1000E 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 system with TRIO Enterprise.

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

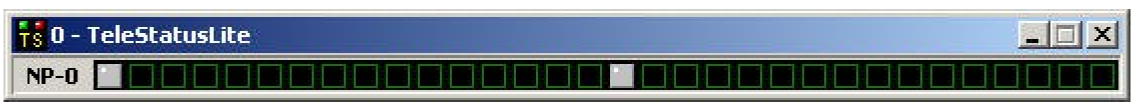
Check the status of the D-channel setup in **Section 5.1.2** by running the command **STAT DCH** in overlay 96 as shown below. The example below shows that D-Channel **66** is operational and established.

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 (not shown). A new window opens, showing the SIP trunk channel status as a series of green squares with the first and sixteenth squares grayed 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 Network Routing Server R7.5 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*

[2] *Network Routing Service Fundamentals Avaya Communication Server 1000, Release 7.5, Document No, NN43001-130 03.10*

[3] *Unified Communications Management Common Services Fundamentals Avaya Communication Server 1000 Document No, NN43001-116 05.17*

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
```

Appendix B
Installed Network Routing Service Services Pack

```
Service_Pack_Linux_7.50_17_20110301.ntl
```

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.