Avaya Solution & Interoperability Test Lab

Application Notes for Configuring Trio Enterprise R5.0 from Enghouse Interactive AB with Avaya Communication Server 1000E R7.6 and Avaya Network Routing Server R7.6 using a SIP Connection – Issue 1.0

Abstract

These Application Notes describe how to configure an Avaya Communication Server 1000E R7.6 to interface with Trio Enterprise R5.0, 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.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as the observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect Compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the compliance tested configuration for Avaya Communication Server 1000E R7.6 and Avaya Network Routing Server R7.6 with Trio Enterprise R5.0. Trio Enterprise is a client/server based application running on the Microsoft Windows 2008 Server operating system. 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 Avaya Communication Server 1000E using a SIP connection via Avaya Network Routing Server.

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 and calls are routed based upon a dial plan 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 using the NRS to route the call. 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 simulated PSTN trunks (SIP trunk via Session Manager) were used. Trio Enterprise can perform the usual range of attendant call functions, i.e., centralized answering position: extending PSTN calls to users, placing PSTN calls on behalf of internal users, and performing 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 Digital 3904 Deskphone was used as the attendant's deskphone. When the attendant is called the Trio Enterprise server calls the 3904 phone and bridges the call.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

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 ensure 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 for the compliance test. Trio Enterprise is connected to the CS1000E using a SIP connection via the NRS. The Trio Enterprise Server is configured as a SIP Endpoint. System Manager is used to configure the NRS and Session Manager is used to route simulate PSTN call.

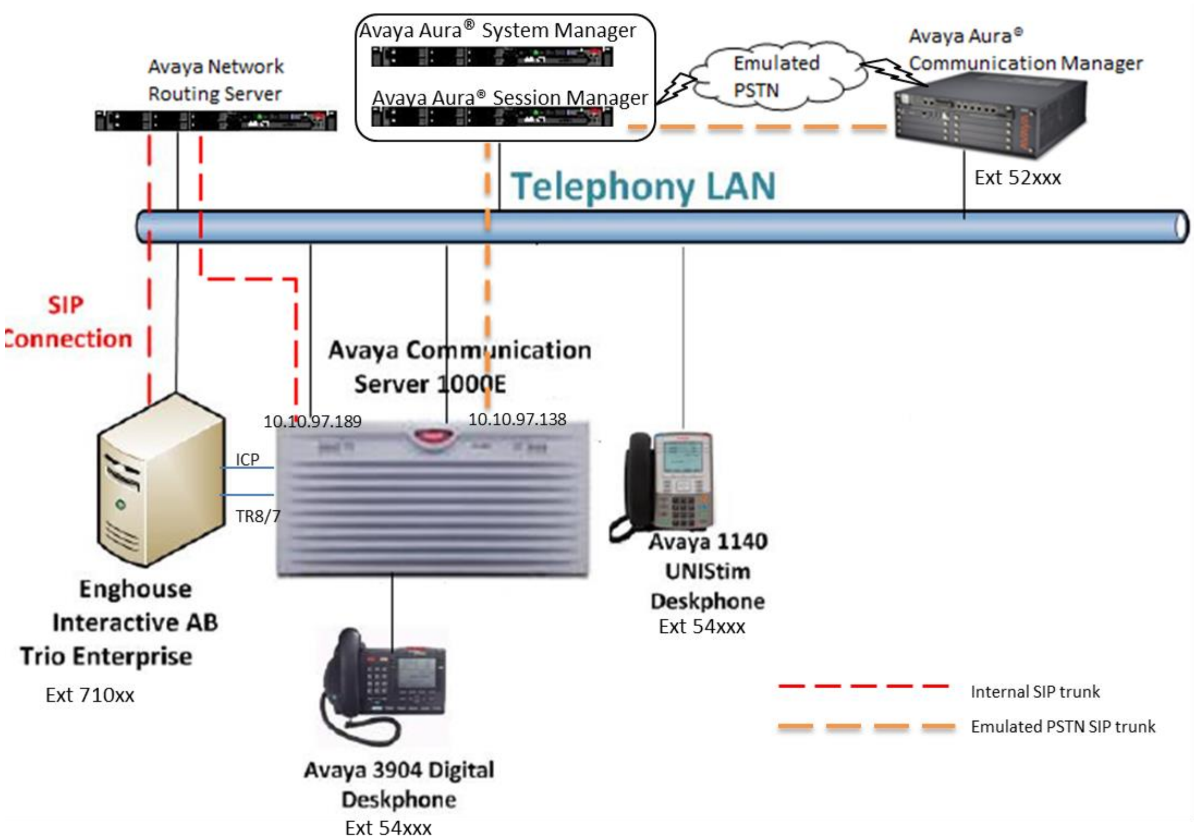


Figure 1: Configuration for Avaya Communication Server 1000E R7.6, Avaya Network Routing Server R7.6 and Trio Enterprise R5.0

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 SP5
Avaya Network Routing Server on CPPM	R7.6 SP5
Avaya Aura® System Manager	6.3 – FP4
Avaya Aura® Session Manager	6.3 – FP4
Avaya 1140 UNIStim Deskphone	UNIStim V0625C8D
Avaya 3904 Digital Deskphone	Core V2.4 Flash V9.4
Trio Enterprise Running on Windows 2008 R2 64-bit server.	Version 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 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. 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).

```
>LD 22              Enter Overlay 22
REQ                 SLT
System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

IPMGs Registered:               2
IPMGs Unregistered:             0
IPMGs Configured/unregistered:  0

TRADITIONAL TELEPHONES 32767    LEFT 32762    USED     5
DECT USERS             32767    LEFT 32767    USED     0
IP USERS               32767    LEFT 32617    USED   150
BASIC IP USERS         32767    LEFT 32766    USED     1
TEMPORARY IP USERS     32767    LEFT 32767    USED     0
DECT VISITOR USER      10000    LEFT 10000    USED     0
ACD AGENTS             32767    LEFT 32743    USED    24
MOBILE EXTENSIONS      32767    LEFT 32767    USED     0
TELEPHONY SERVICES     32767    LEFT 32767    USED     0
CONVERGED MOBILE USERS 32767    LEFT 32767    USED     0
AVAYA SIP LINES        32767    LEFT 32731    USED    36
THIRD PARTY SIP LINES  32767    LEFT 32717    USED    50

PCA                    32767    LEFT 32757    USED    10
ITG ISDN TRUNKS        32767    LEFT 32767    USED     0
H.323 ACCESS PORTS     32767    LEFT 32671    USED    96
AST                    32767    LEFT 32737    USED    30
SIP CONVERGED DESKTOPS 32767    LEFT 32762    USED     5
SIP CTI TR87           32767    LEFT 32757    USED    10
SIP ACCESS PORTS       32767    LEFT 32725    USED    42
RAN CON                32767    LEFT 32767    USED     0
MUS CON                32767    LEFT 32767    USED     0    ACTIVE     0

IP RAN CON             16384    LEFT 16384    USED     0
IP MUS CON             16896    LEFT 16896    USED     0    ACTIVE     0
IP MEDIA SESSIONS      35842    LEFT 35842    USED     0    ACTIVE     0
TNS                    32767    LEFT 32098    USED   669
ACDN                   24000    LEFT 23984    USED    16
AML                       16    LEFT     7    USED     9
IDLE_SET_DISPLAY CS1K_Middle 7.65
LTID                   32760    LEFT 32760    USED     0
RAN RTE                  512    LEFT   512    USED     0
ATTENDANT CONSOLES     32767    LEFT 32767    USED     0
IP ATTENDANT CONSOLES  32767    LEFT 32767    USED     0
BRI DSL                10000    LEFT 10000    USED     0
MPH DSL                  100    LEFT   100    USED     0
DATA PORTS             32767    LEFT 32767    USED     0
PHANTOM PORTS          32767    LEFT 32767    USED     0
TRADITIONAL TRUNKS     32767    LEFT 32638    USED   129
ELC ACCESS PORTS       32767    LEFT 32767    USED     0
DCH                      255    LEFT   245    USED    10
```

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 23 (i.e. **DCH 23**) 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 23**	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**, **PCID** is **SIP** and **TRO** is **YES.**

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	**23**	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
TRO	**YES**	
ACOD	8020	Access Code for trunk route

5.2.3. Add 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	23 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	23	Route list Index number
ENTR	0	First entry for the RLI
ROUT	**23**	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 **71**, **FLEN** is **5** and the **RLI** is **23**.

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	**71**	Distant Steering Code
FLEN	**5**	Flexible Length number of digits
RLI	**23**	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	15	Data number
PORT	0	Port number
DES	ICP2	Description
BPS	1200	Bits per Second
BITL	7	Bit Length
STOP	1	Stop bit
PARY	EVEN	Parity
FLOW	NO	Flow
USER	**ICP**	User type is set to ICP

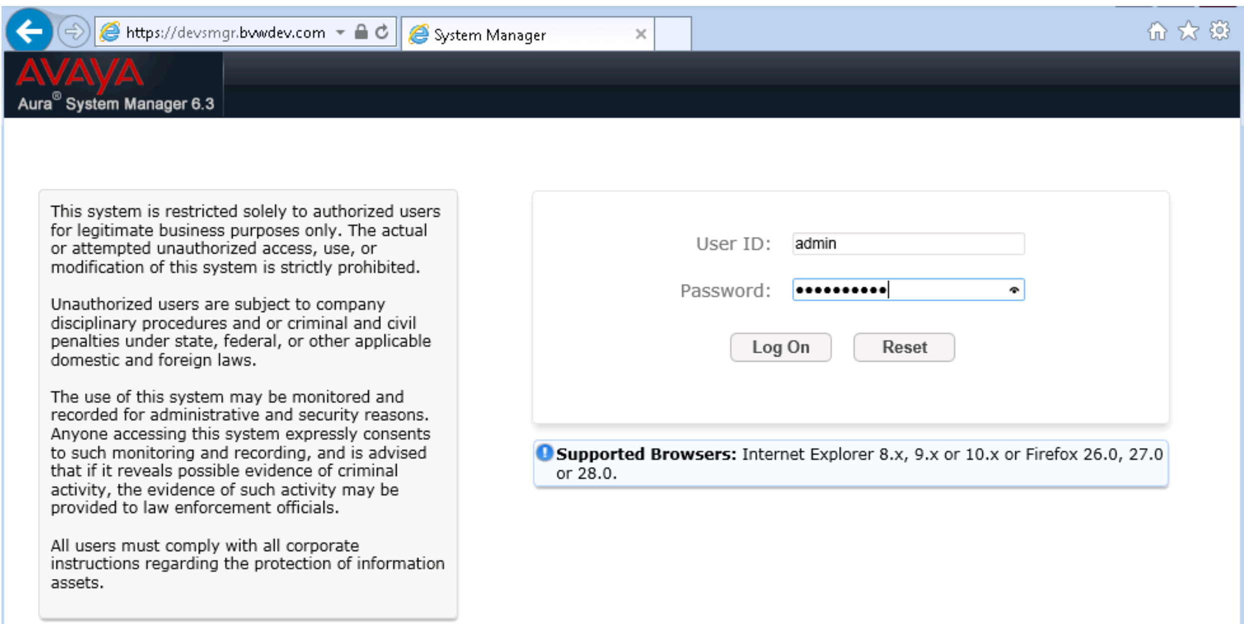
5.5.2. Configuration of ICP in the Customer Data Block

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.

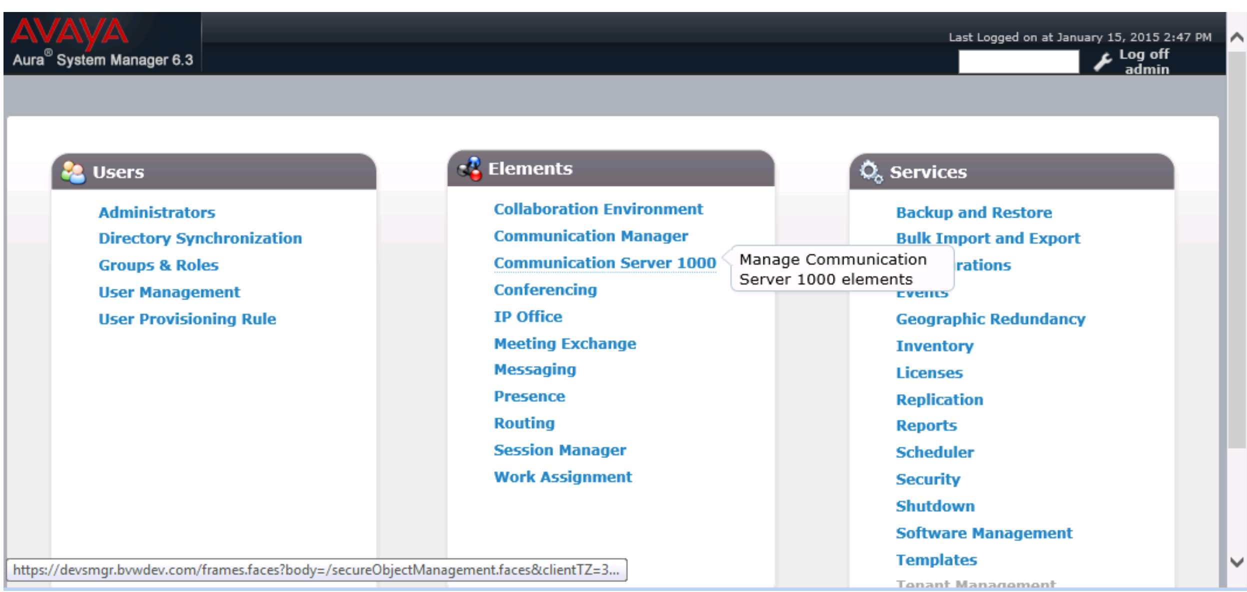
Prompt	**Response**	**Description**
>	**LD 15**	Enter Overlay 15
REQ	**CHG**	Change
TYPE	**icp**	Change the Intercept Computer Update
CUST	0	Customer Number
APL	15	Auxiliary Processor Link used
NIPN	9	Number of Intercept positions
ICCR	NO	Intercept Position Cancelling Reply
ICDN	71003	Trio DN for diversion call treatment
ECDN	71003	Trio DN for diversion call treatment
CPS	CIR	
ICDL	5	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

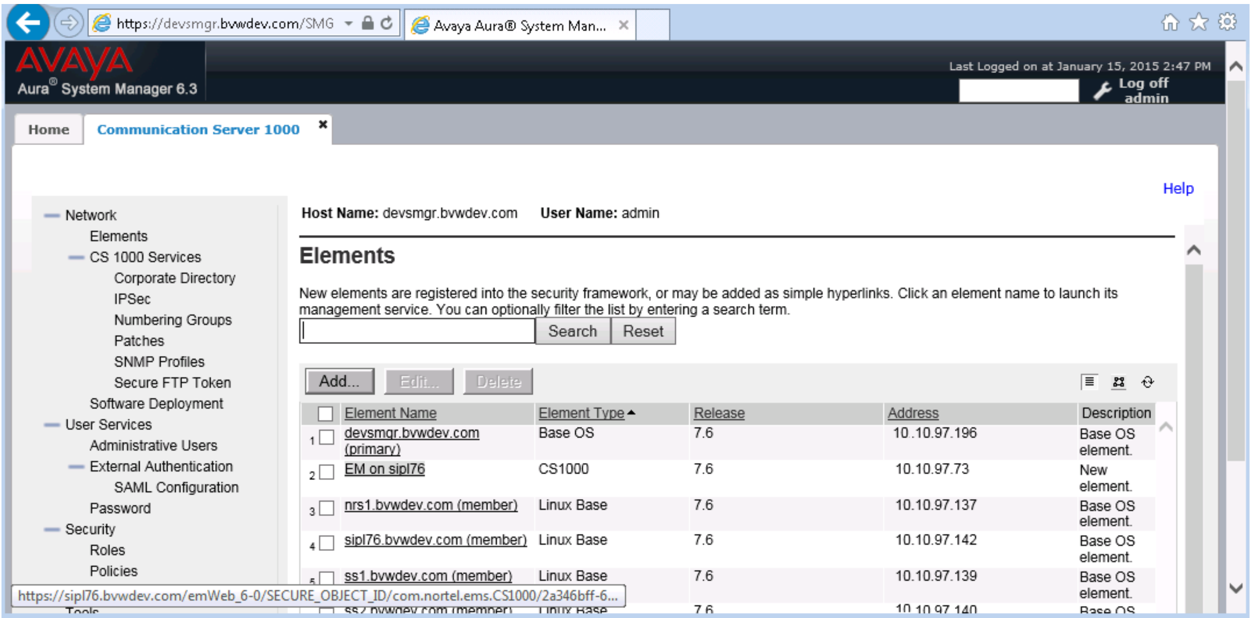
SIP CTI (TR/87) services must be enabled and configured on the CS1000E IP Telephony Node to allow applications obtain presence information or invoke a make-call operation. Changes on the CS1000E Node are performed using Element Manager which is only accessible through the System Manager. To make changes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



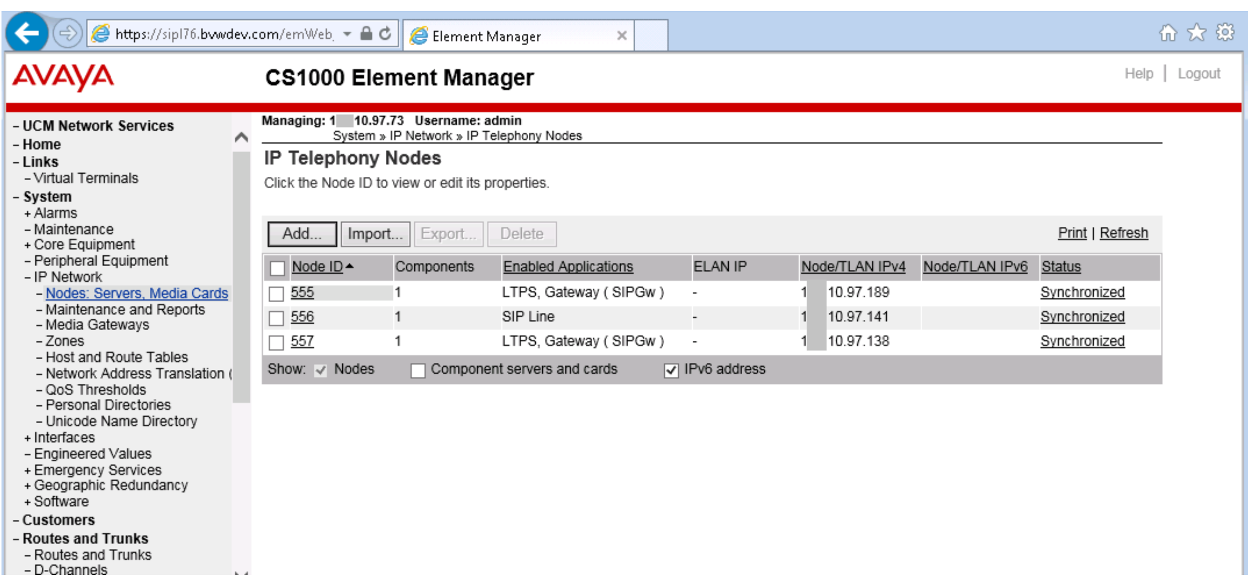
Click on **Communication Server 1000** as shown.



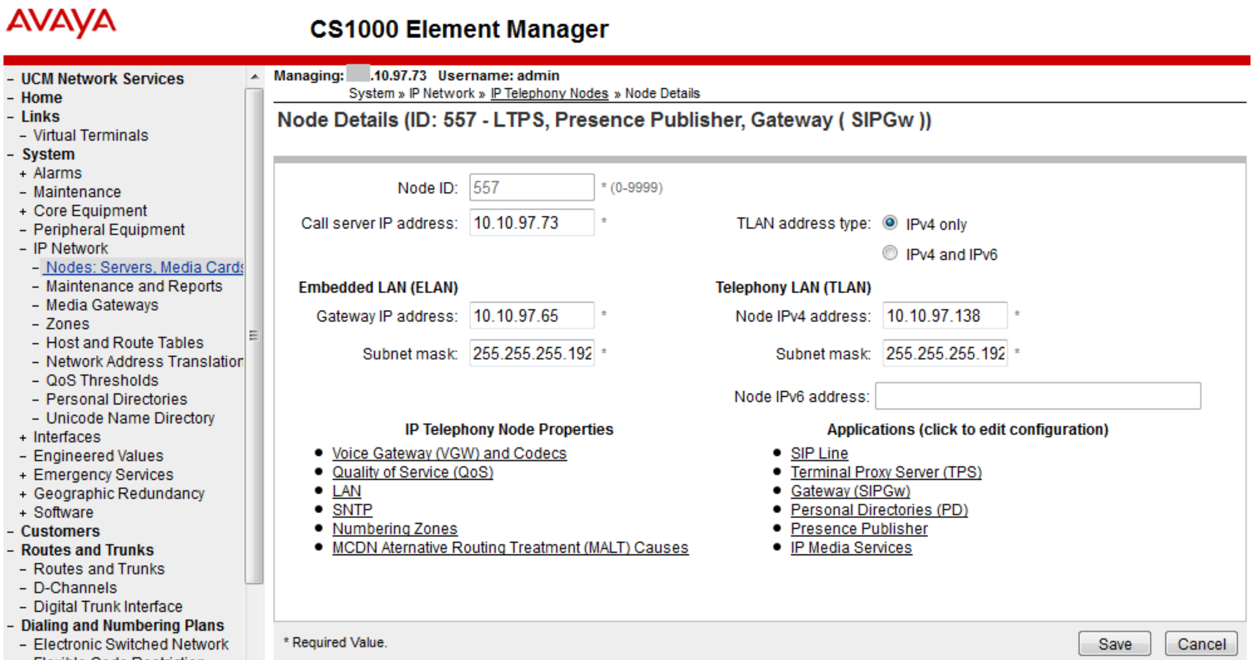
Once **Communication Server 1000** is selected the following screen appears, click on the Element Manager link, in this case click on **EM on sipl76** link.



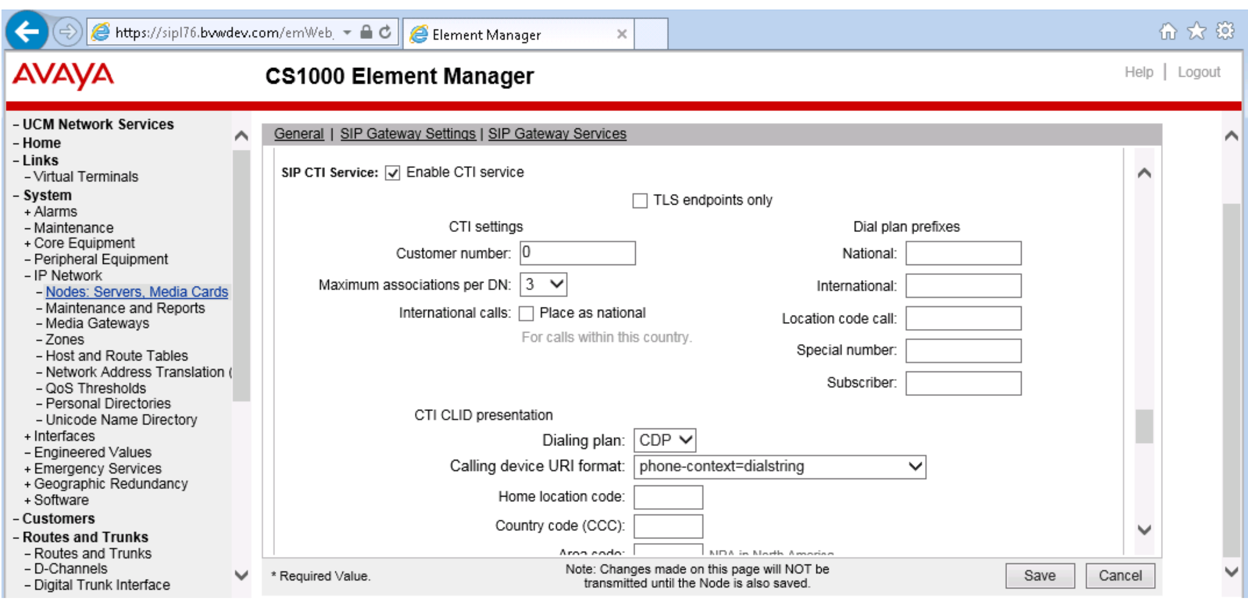
Click on **IP Network → Nodes: Servers, Media Cards** in the left window. **IP Telephony Nodes** page displays the list of available Nodes in CS1000E. During compliance test **Node** 555 is configured to connect to NRS. Note the IP address of this node as it will be required in **Section 7.2** to create CS1000E as an endpoint on NRS. Click on the 557 in **Node ID** column to enable TR87 events on CS1000E. Note this IP address of this node as it will be required in **Section 8.4** Configure TR87 on Trio Enterprise.



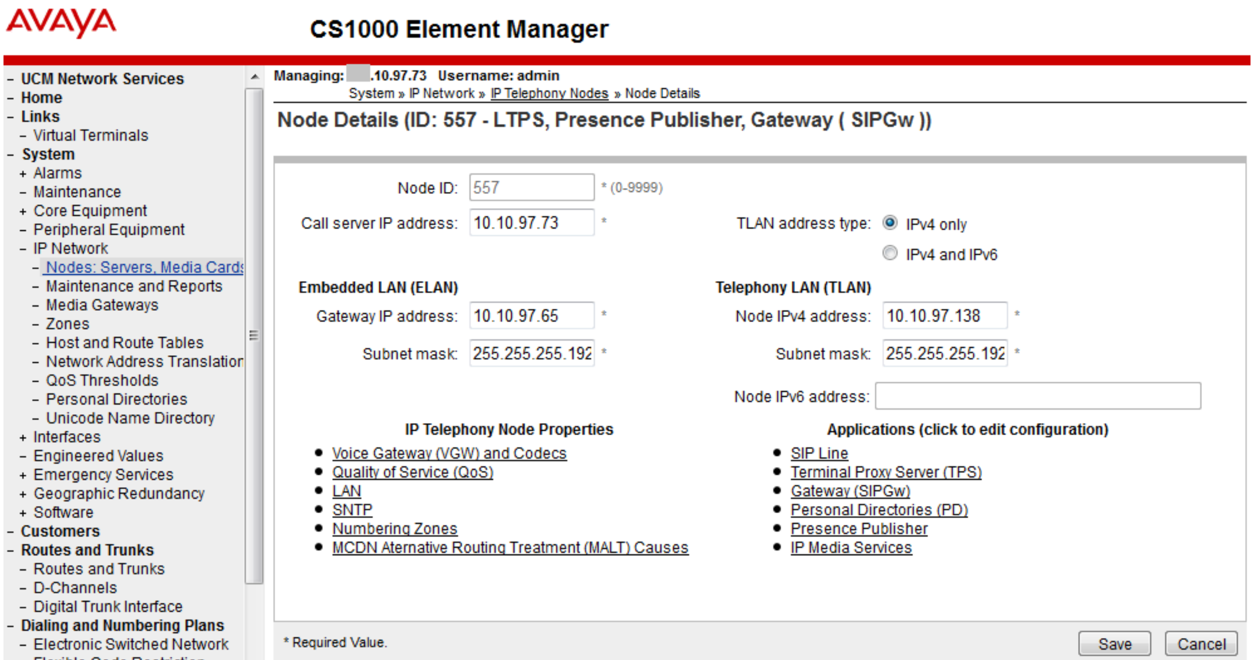
Select **Gateway (SIPGw)** in **Applications (click to edit configuration)** section.



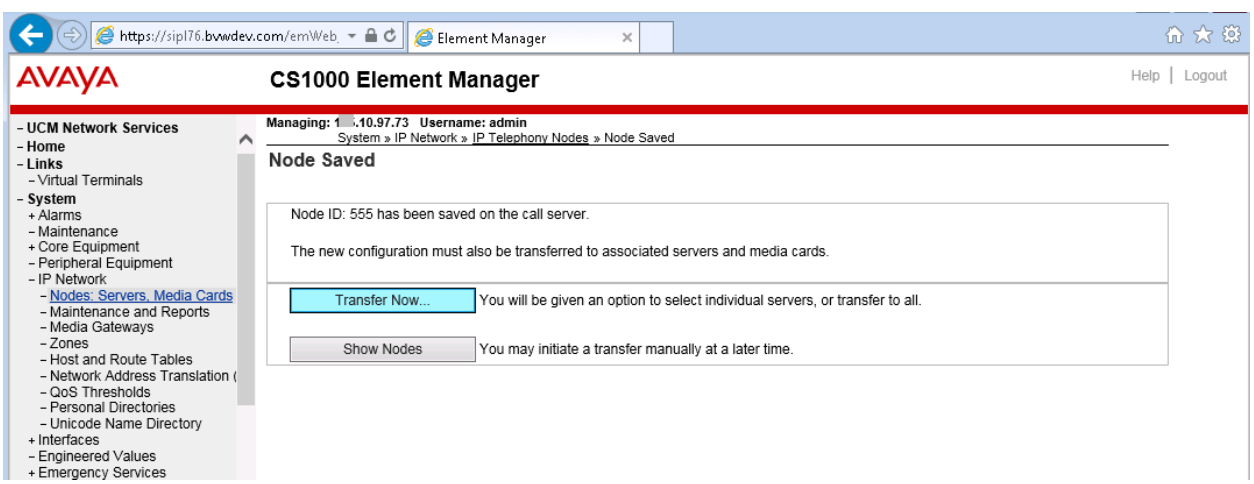
Ensure that **SIP CTI Service** is ticked as shown below and untick the **TLS endpoints only** (if this is ticked); everything else can be left as default. Click on **Save** once finished.



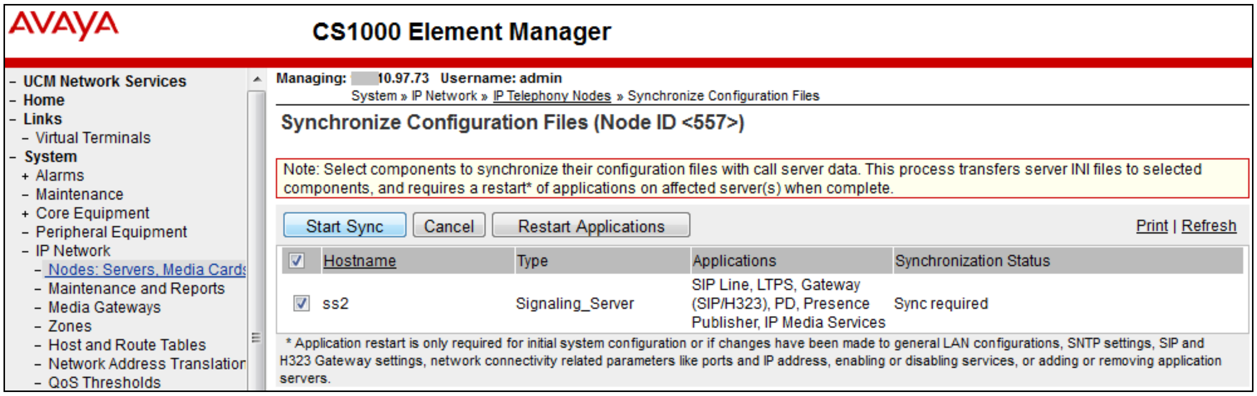
Once **Save** is clicked the following screen appears. Click on **Save** as shown below.



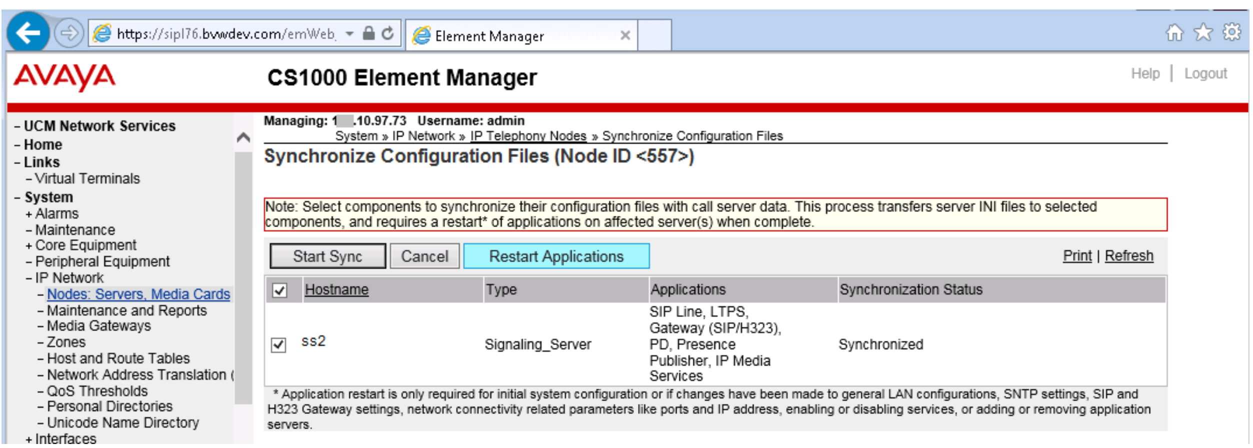
Once Save is clicked the following screen appears. Select **Transfer Now** as highlighted below.



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

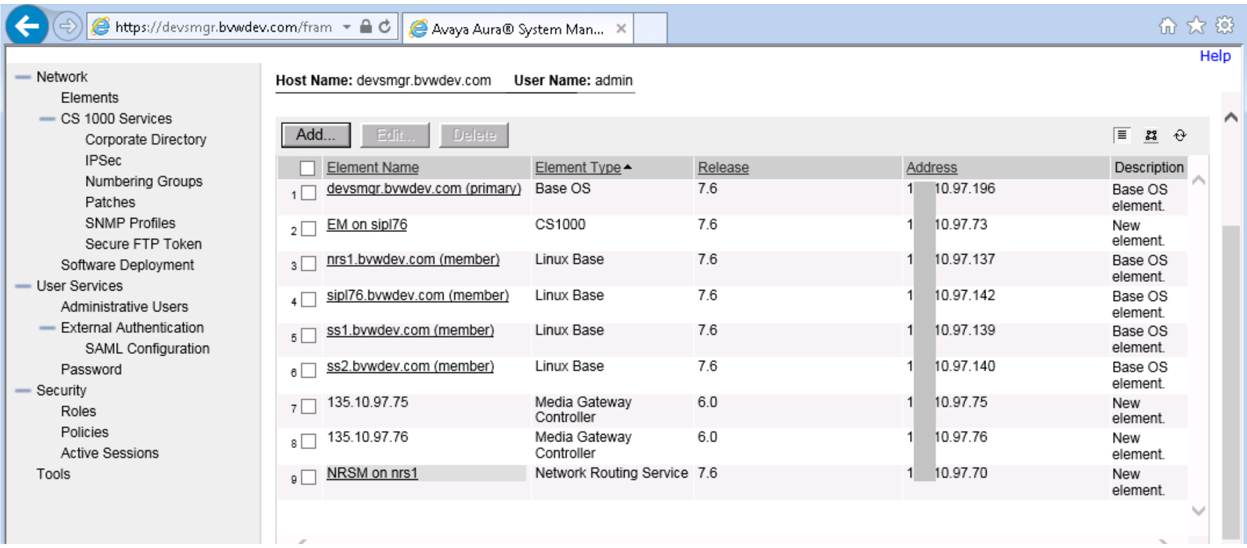


Once the components are synchronized the application will require a restart, select the **Hostname** and click on **Restart Applications** as highlighted below.



7. Configure Avaya Network Routing Server

To make changes on the NRS 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**, click on **Communication Server 1000** as shown in **Section 6**. Click on the NRS **Element Name** or **NRSM on nrs1** as shown in the example below.

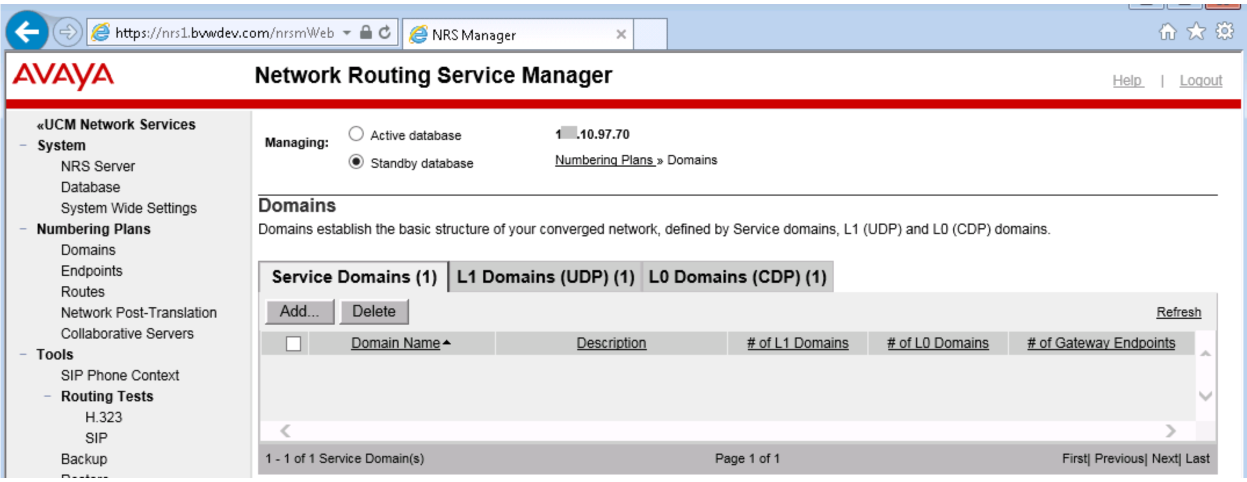


7.1. Add a New Domain

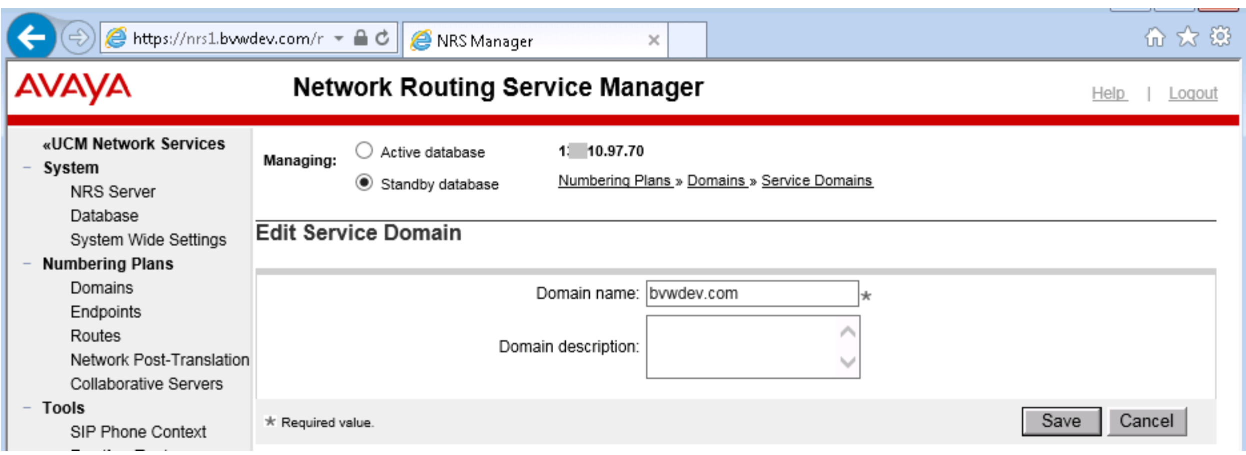
In the event there is no domain present a new one must be added as follows. Note that there are three separate entries for this domain and all three must be added.

- Service Domain
- L1 Domain (UDP)
- L0 Domain (CDP)

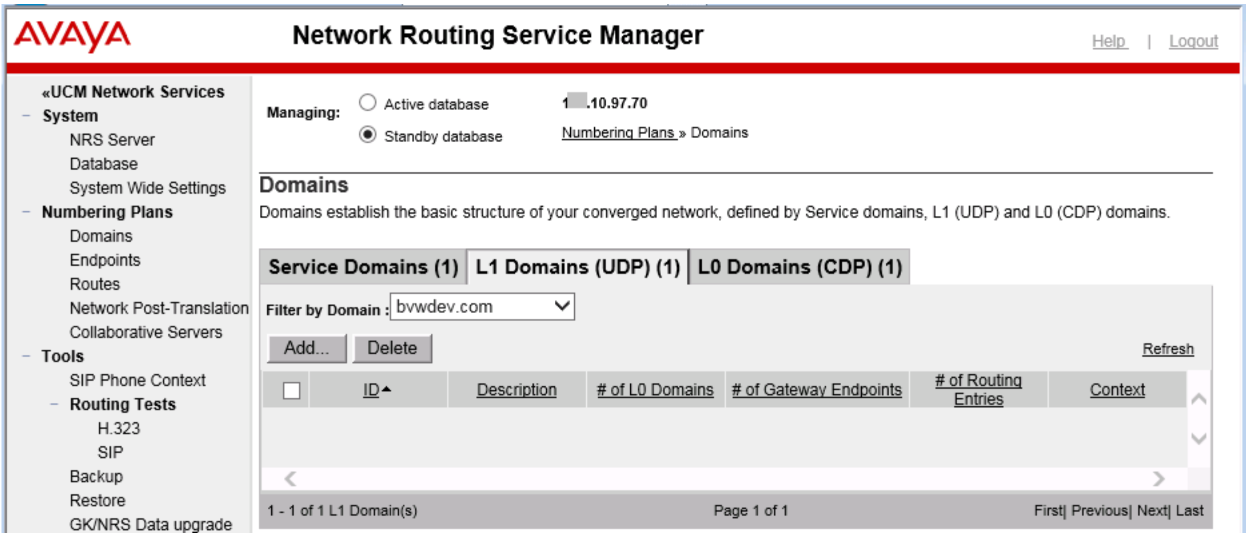
In order to make any change the **Standby database** must be first selected as shown below. Click on **Numbering Plans → Domains** from the left window and under the tab **Service Domains** in the main window click on **Add**.



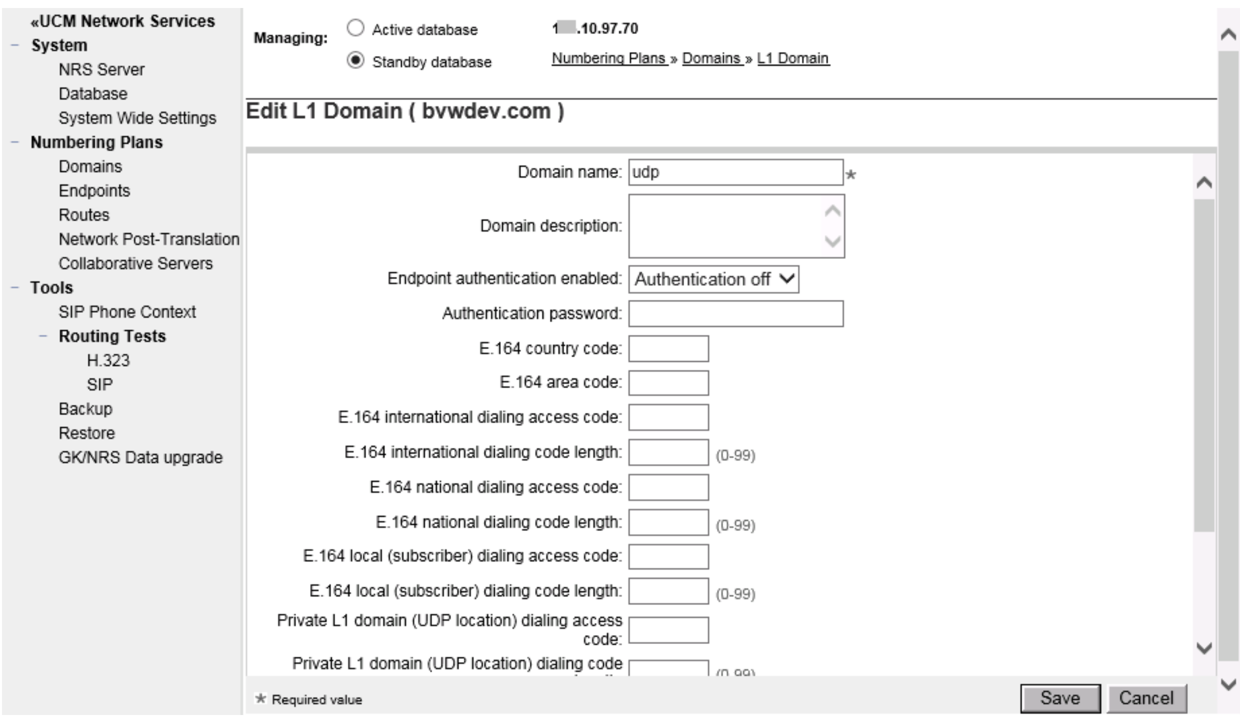
Enter a suitable name for the domain. In the example below **bvwdev.com** is chosen. Click on **Save** once the name is entered.



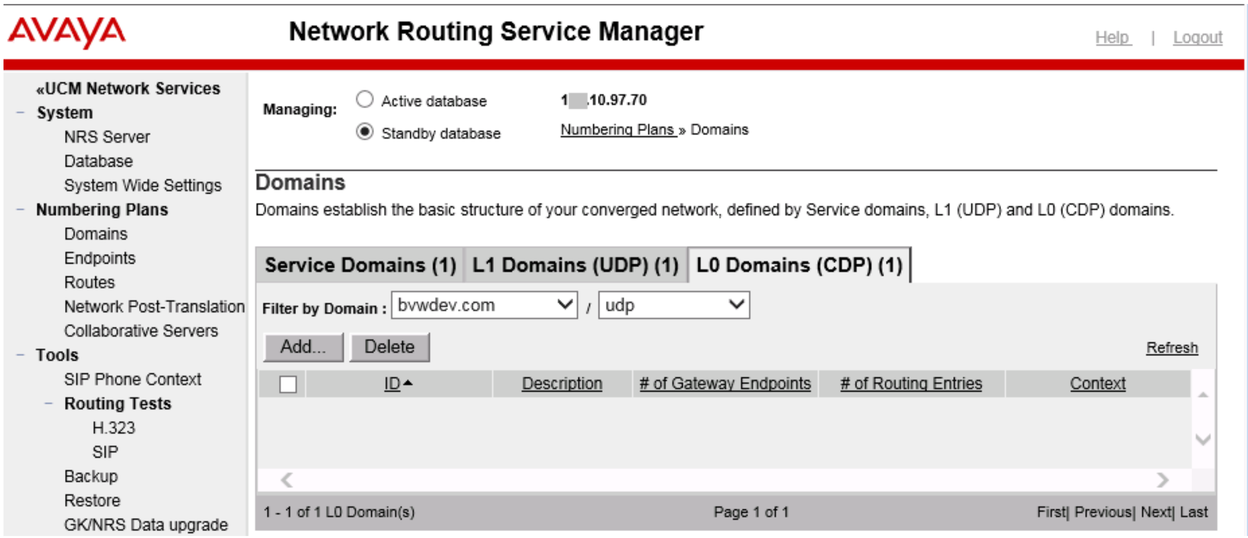
Under the **L1 Domains (UDP)** tab select the domain created above from the drop-down box and click on **Add**.



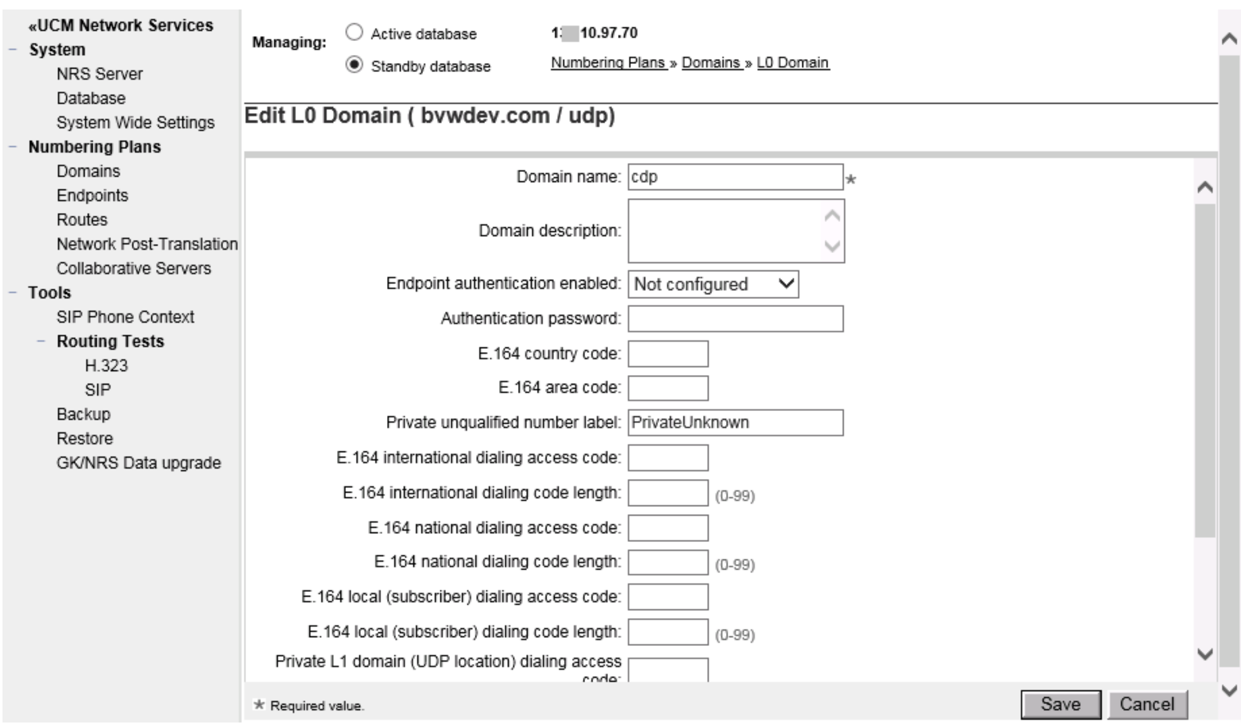
Enter the name **UDP** and click on **Save** at the bottom right of the screen.



Under the **L0 Domains (CDP)** tab select the domains created above from the drop-down boxes and click on **Add**.

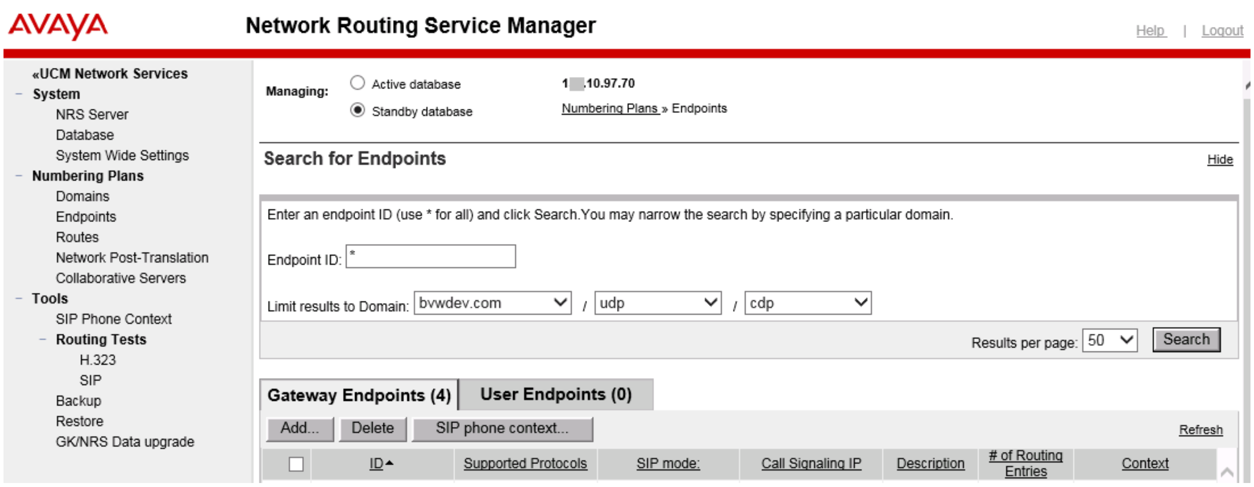


Enter the name **CDP** and click on **Save** at the bottom right of the screen.

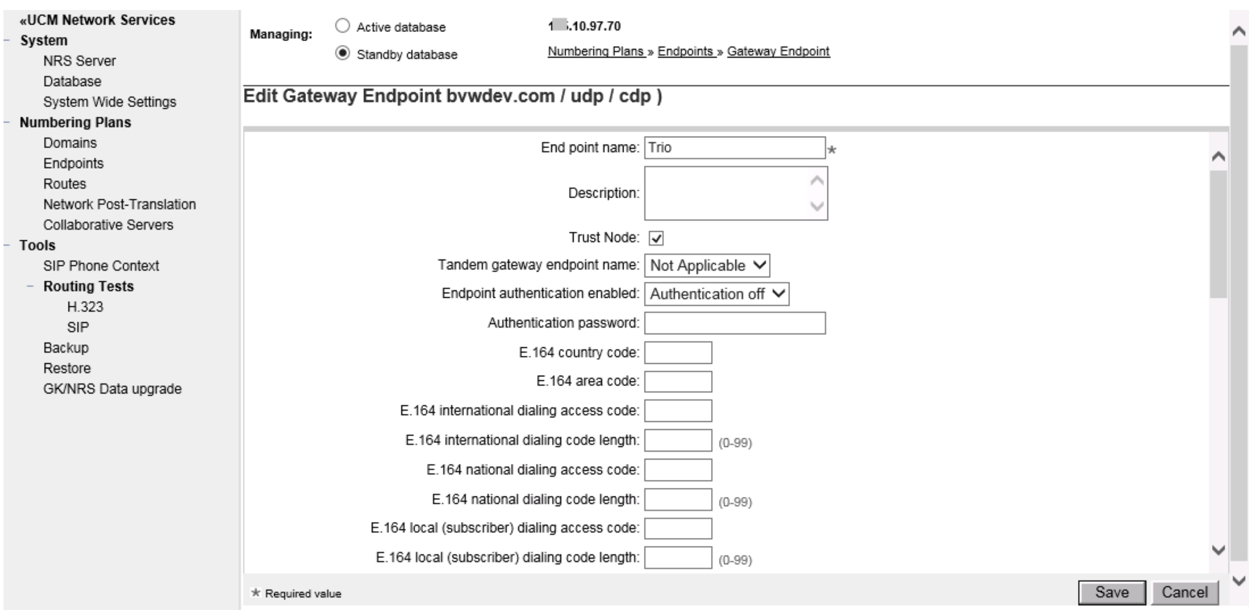


7.2. Add Trio as an Endpoint

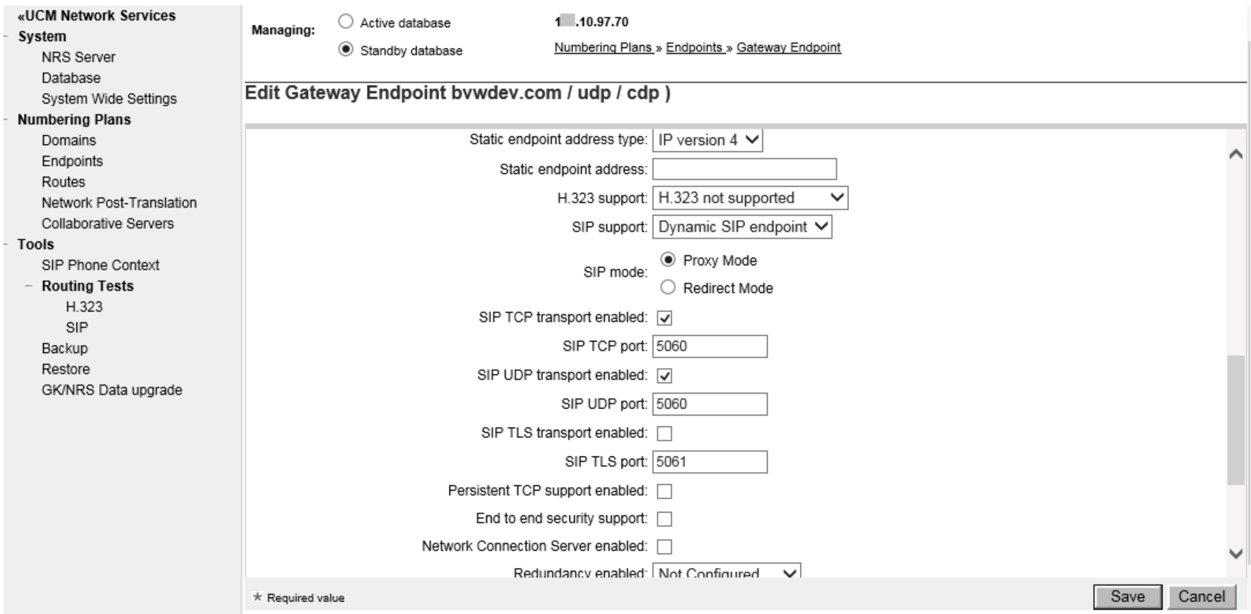
Select **Numbering Plans → Endpoints** in the left window. In the main window choose the newly created domain from **Section 7.1** for the drop-down boxes as highlighted below and under the **Gateway Endpoints** Tab click on **Add**.



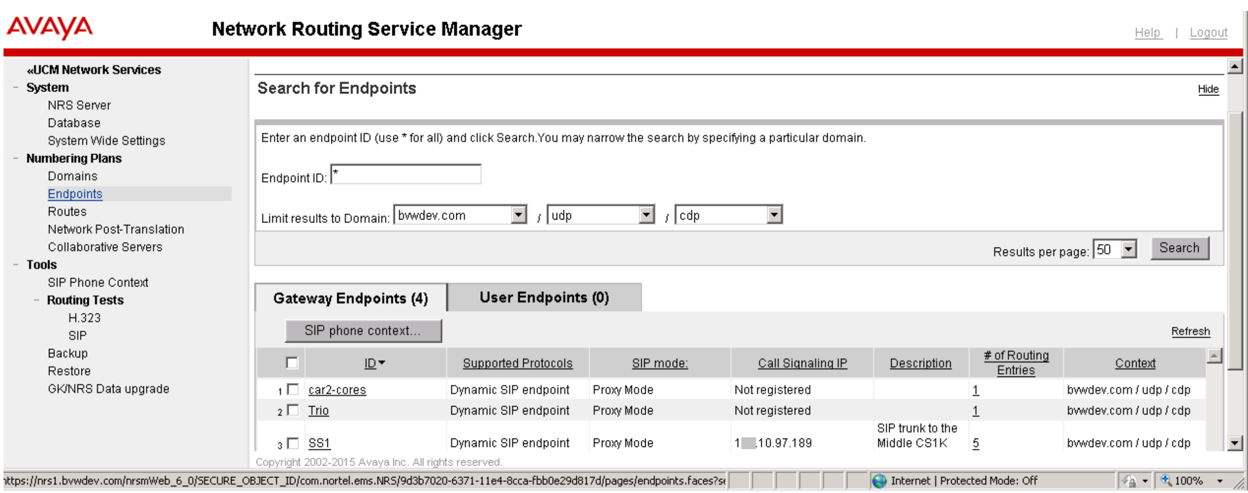
Enter a suitable name for the endpoint and scroll down the page.



This is a SIP endpoint therefore H.323 is not supported and choose **Dynamic SIP endpoint** from the **SIP Support** drop-down box. Ensure that **Proxy Mode** is selected and in the example below both UDP and TCP are selected to allow either transport be used. For Trio ensure that at least **TCP** is chosen. The port number for each is **5060**. Everything else can be left as default and click on **Save** once ready.

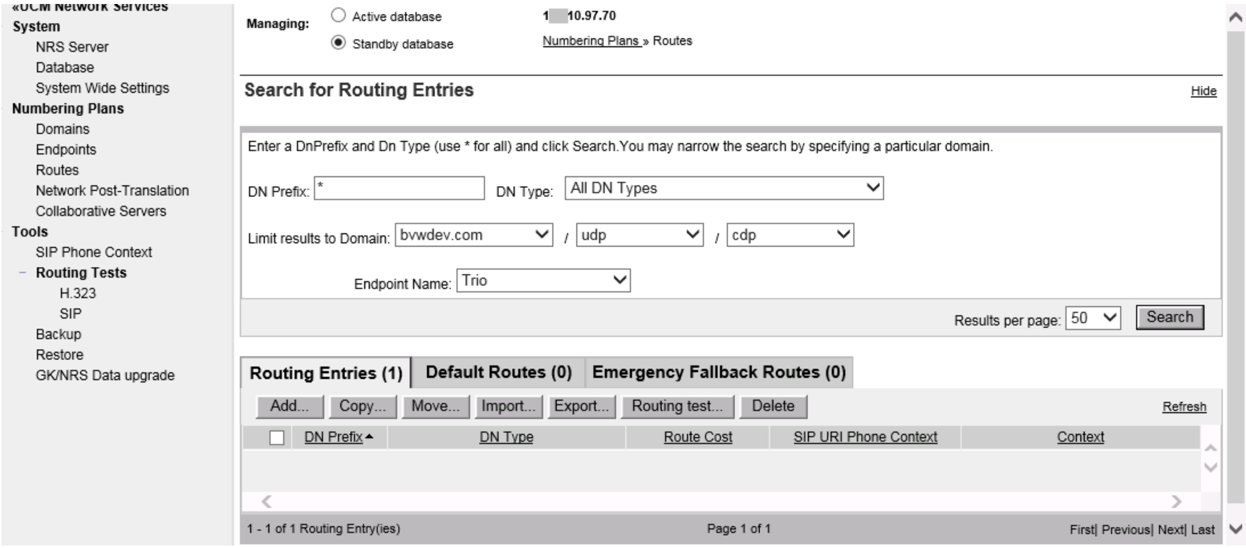


Repeat same steps to create endpoint for **Node** 555 in this case it is **ss1** see detail information of Node 555 in Section **6.**

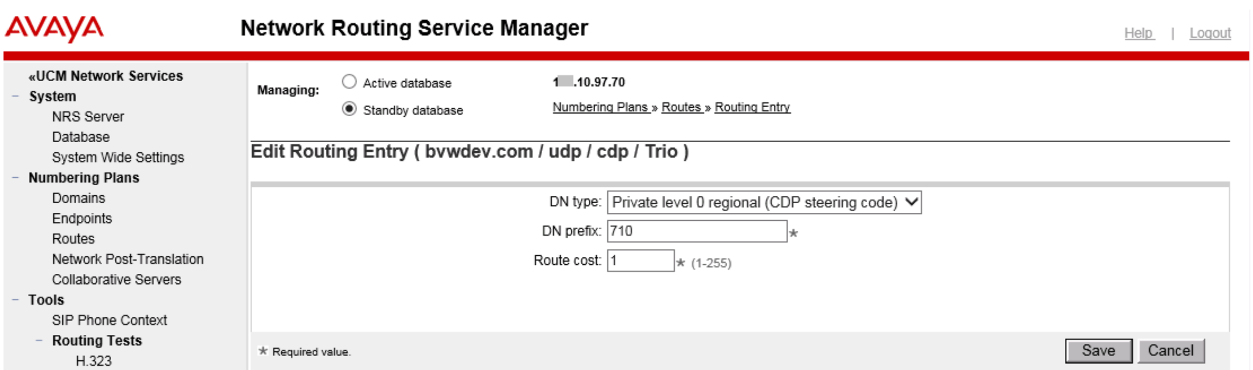


7.3. Add a Routing Entry for Trio

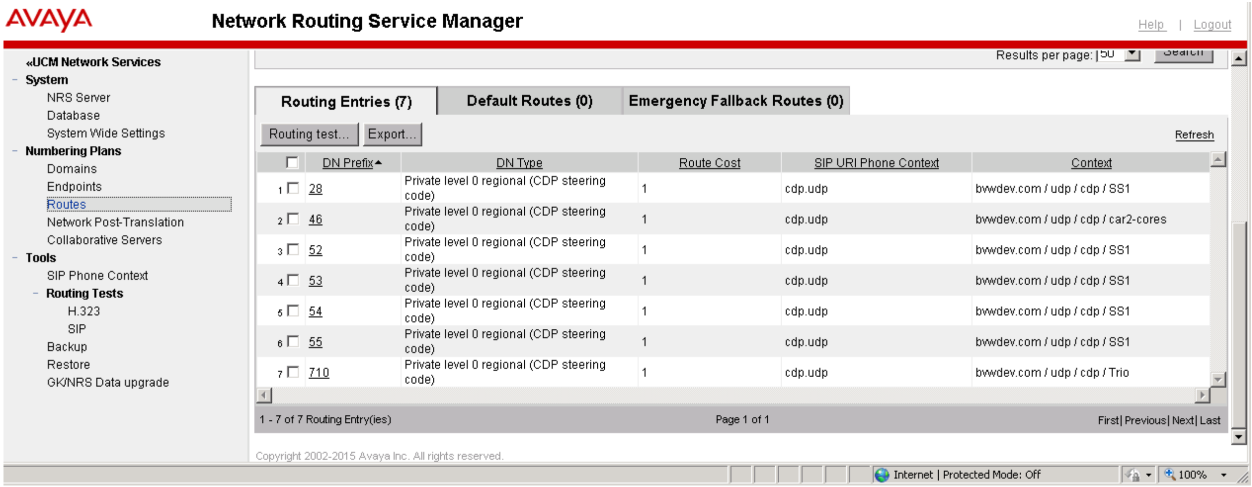
Select **Numbering Plans → Routes** in the left window and choose the domain and Trio endpoint created in **Sections 7.1** and **7.2** for the drop-down choices. Under the **Routing Entries** tab click on **Add**.



Select **Private level 0 regional (CDP steering code)** for **DN type** and enter the correct **DN prefix** with **Route cost** set to **1**. Note that **710** were entered during compliance testing so that numbers **710xx** were routed to the Trio endpoint.

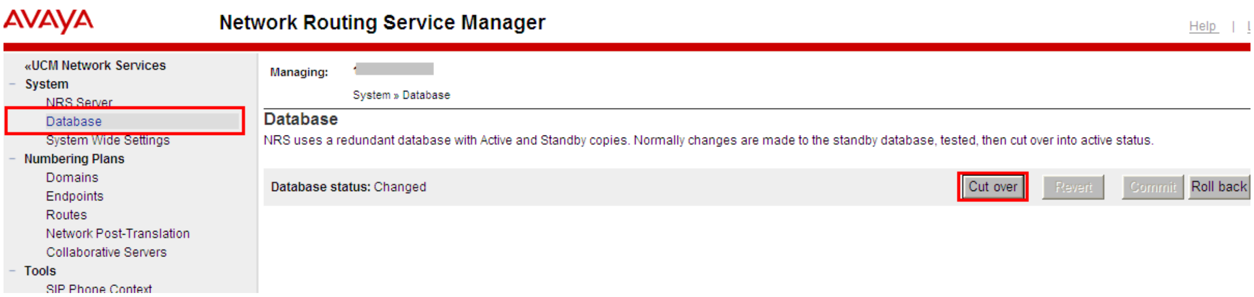


Repeat same step to created routing for 54xxx so that called number 54xxx is routed to CS1000.

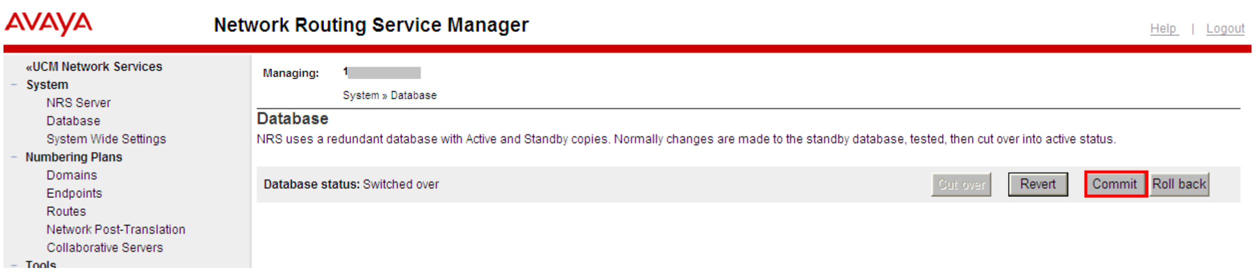


7.4. Save New Changes on NRS

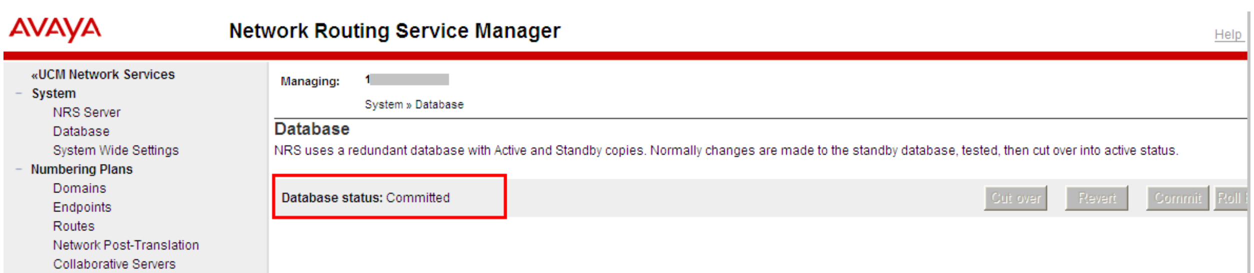
To save the new entries to the database on the NRS the database must be cut over and committed. Select **Database** under **System** in the left window and click on **Cut over** in the right window.



Once the Database is cut over, click on **Commit** in the same window, as shown below.



Once the database is **Committed** as shown below the NRS has been properly configured.



8. Configure Trio Enterprise

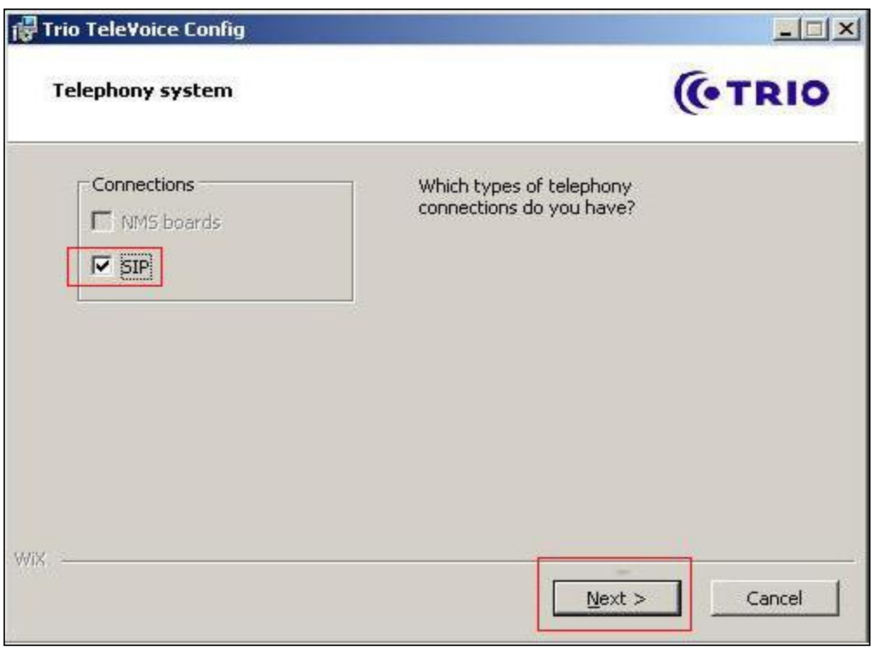
This section describes how to integrate Trio Enterprise with the CS1000E using dynamic SIP. Trio Enterprise is added to the NRS as a Dynamic SIP endpoint and calls are routed to the Trio Enterprise server according to the dialing plan setup in **Section 5.3**. This section shows how to configure Trio 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.

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

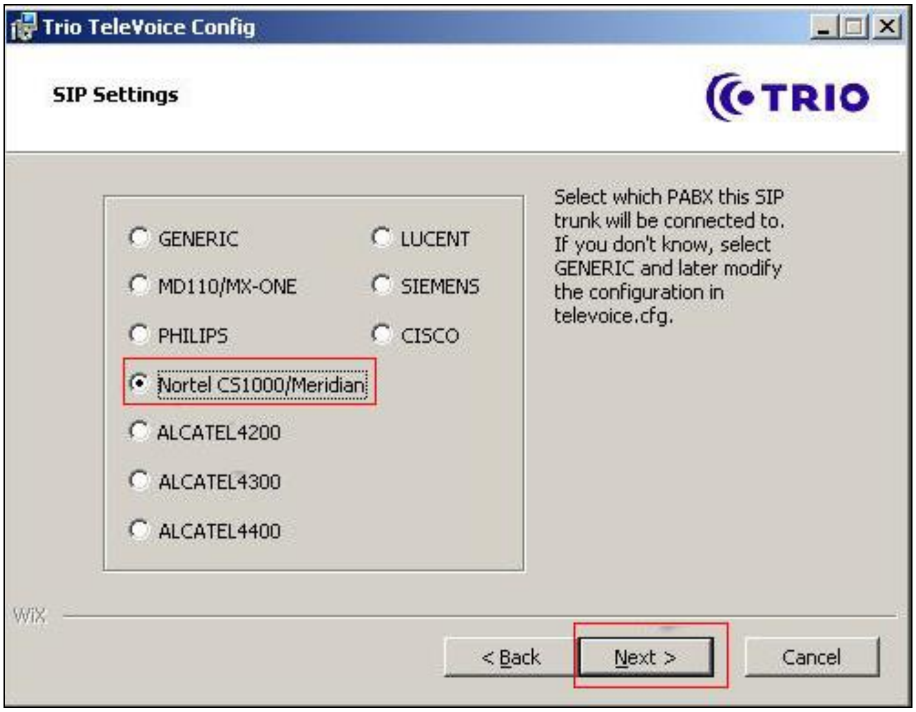
8.1. Configure Trio Enterprise to use SIP Trunks

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

1. Access Windows services. Select **Start → Run**, then type **services.msc** into the **Open** box. 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 screen as shown below.
4. Ensure the **SIP** entry in the **Connections** area is checked.
5. Click **Next** to continue.



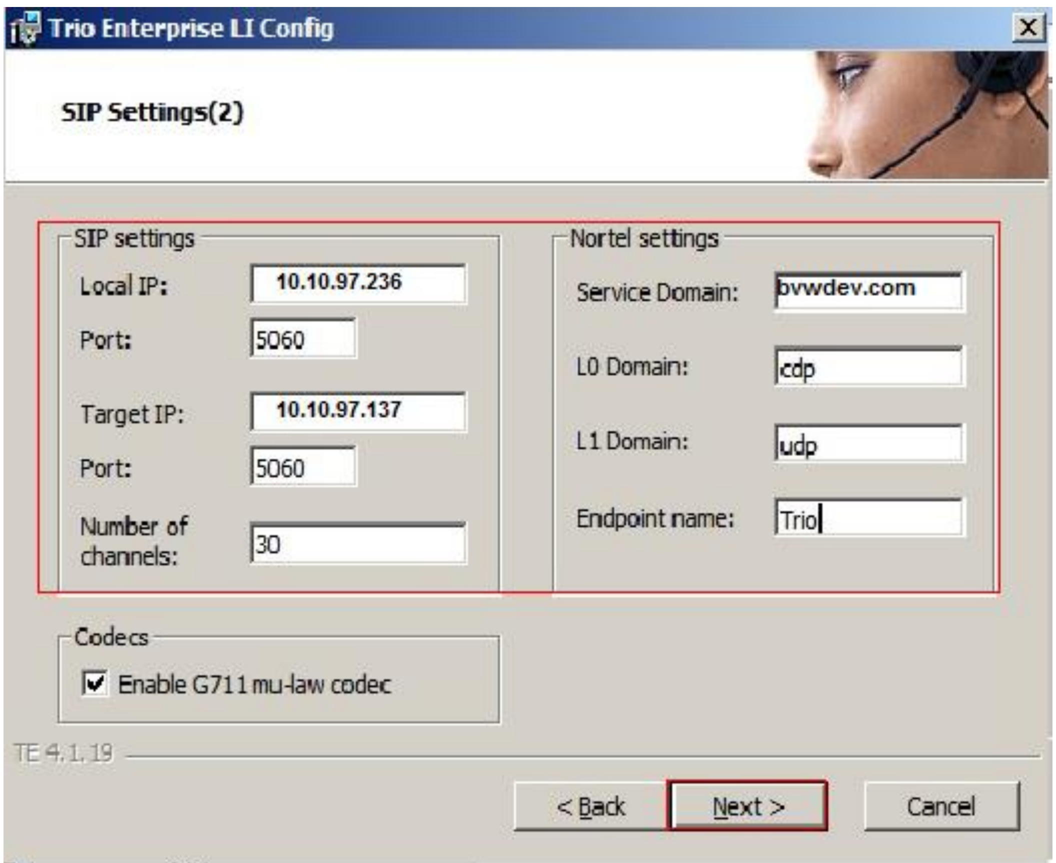
Select **Nortel CS1000/Meridian** under **SIP Settings**. Click **Next** to continue.



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

- **Local IP** The local IP address of the Trio Enterprise server
- **Target IP** The IP address of the Network Routing Server (NRS)
- **Number of channels** The number of channels
- **Service Domain** The Service domain configured in Network Routing Server (**Section 7.1**)
- **L0 Domain** The L0 Domain configured in Network Routing Server (**Section 7.1**)
- **L1 Domain** The L1 Domain configured in Network Routing Server (**Section 7.1**)
- **Endpoint name** Trio endpoint name configured in Network Routing Server (NRS), as configured in **Section 7.2**

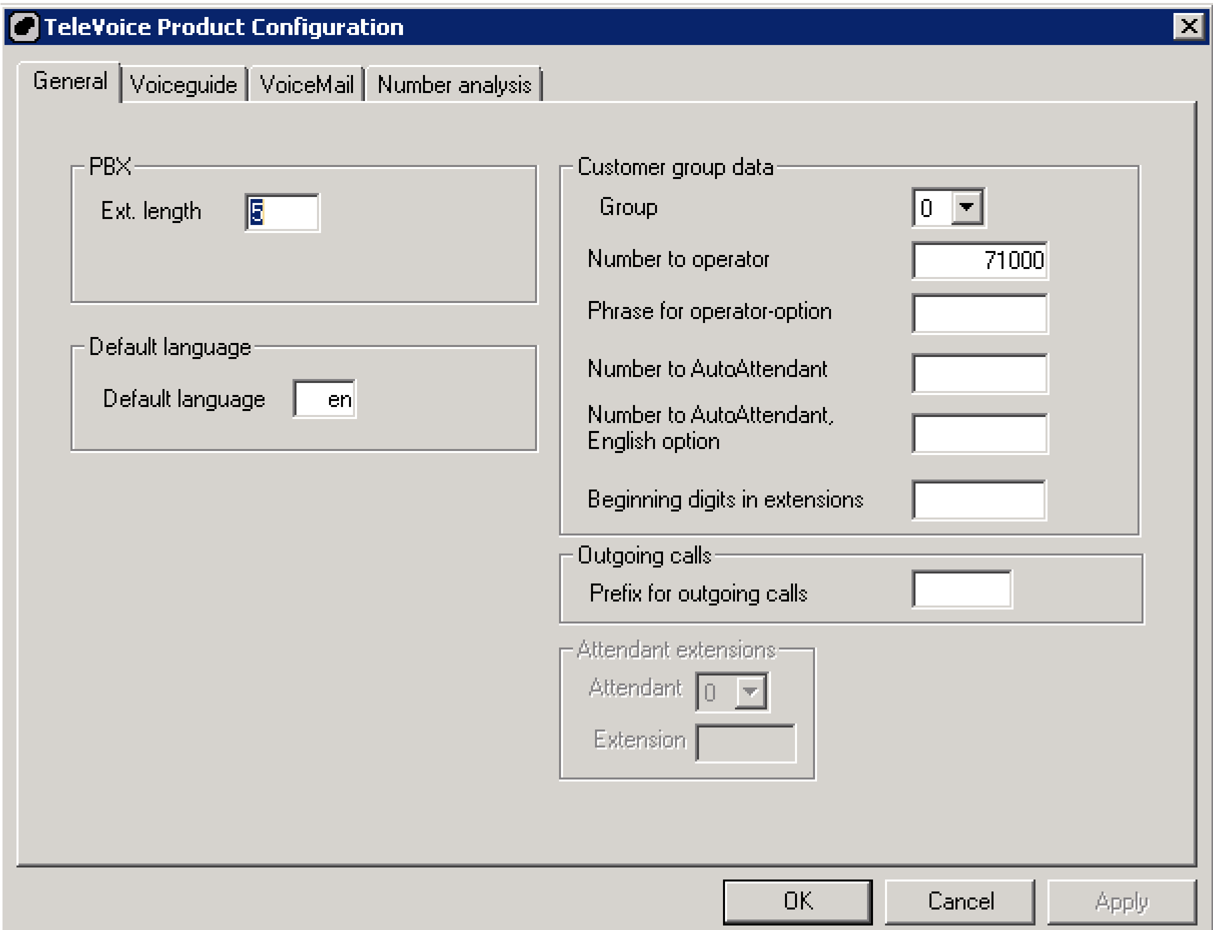
Click **Next** to continue.



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

- **Ext. length** Ext length is **5**
- **Number to operator** Example **71000 (**as was configured in **Section 5.3.2)**

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



Please ensure that the Trio Televoice service is running, and if not please follow these instructions to get this started.

1. Access Windows services. Select **Start → Run**, then type **services.msc** into the **Open** box. Press Return (not shown).
2. When the standard services window opens, locate the Trio Televoice service and start the service (not shown).

8.2. InteractionStudio Configuration

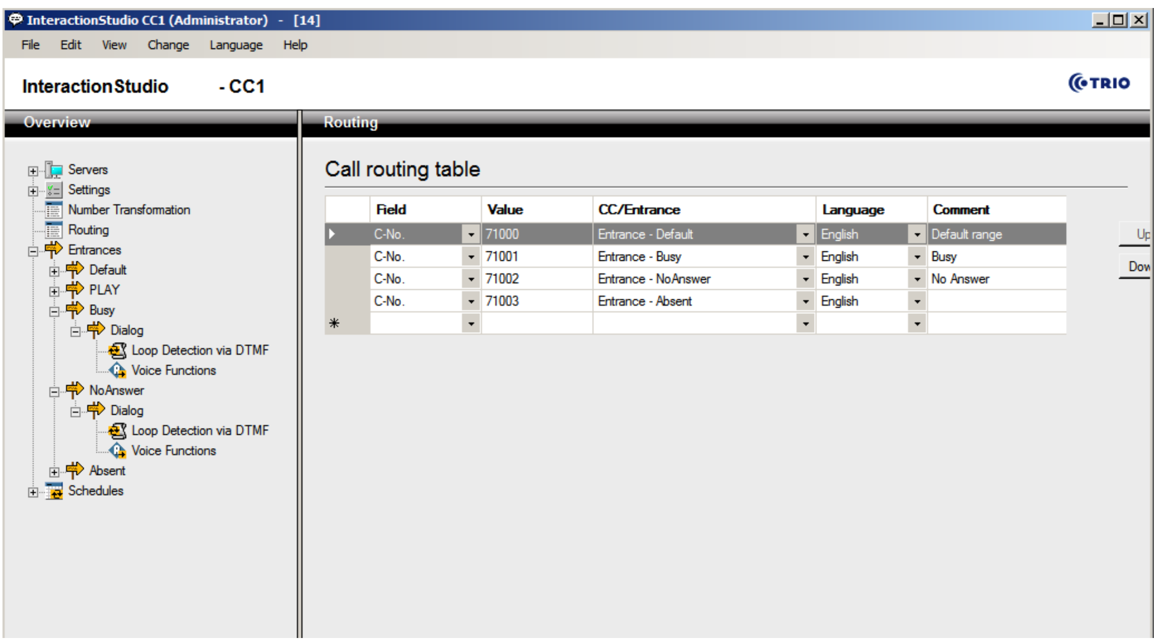
The InteractionStudio is used to configure many features for Trio Enterprise. For compliance testing, the following were configured.

- 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

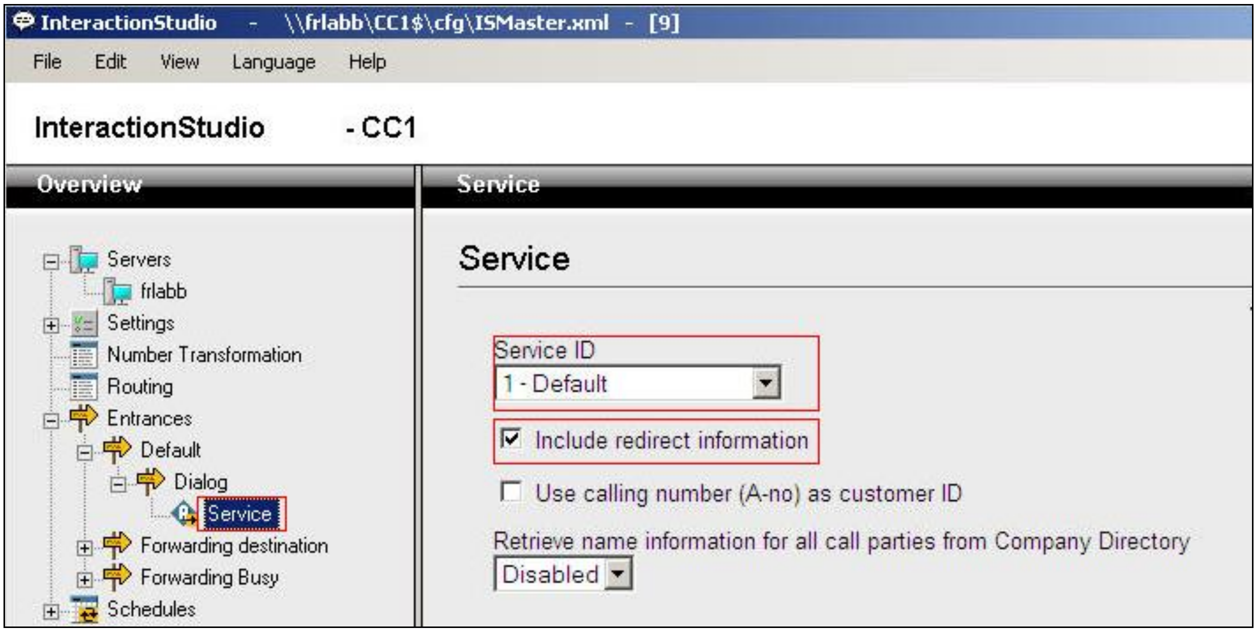
On the Trio Enterprise server, select **Start → Programs → Trio Enterprise → Contact Center → CC1 → Interaction Studio** (not shown). When the InteractionStudio window opens, navigate to **Routing**. A **Call routing table** will open. In the example below:

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



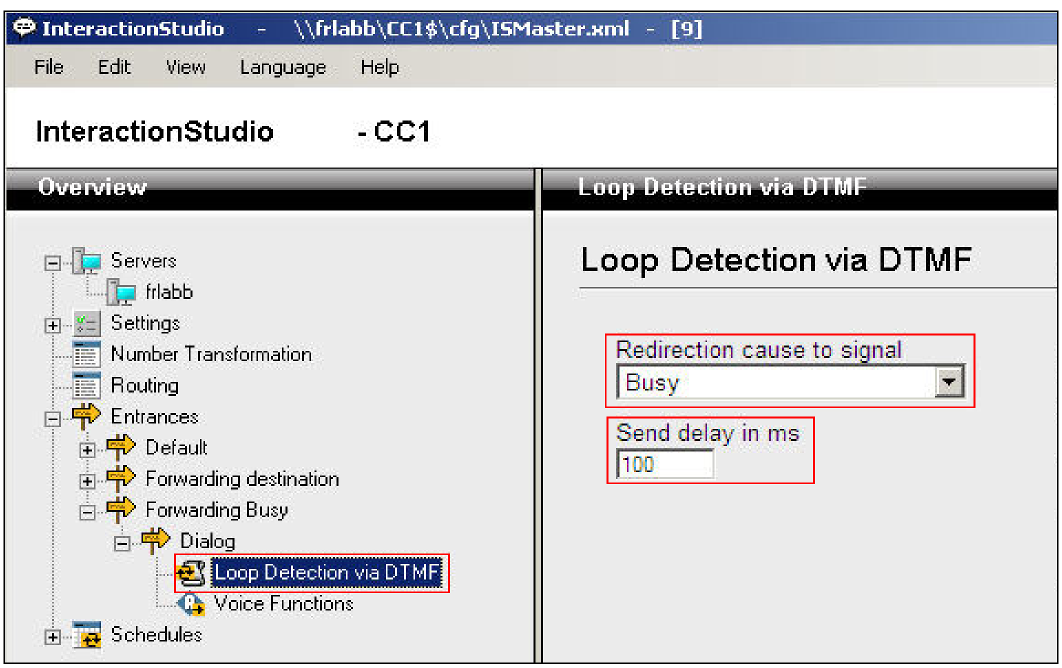
8.2.2. Configure Attendant Service

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



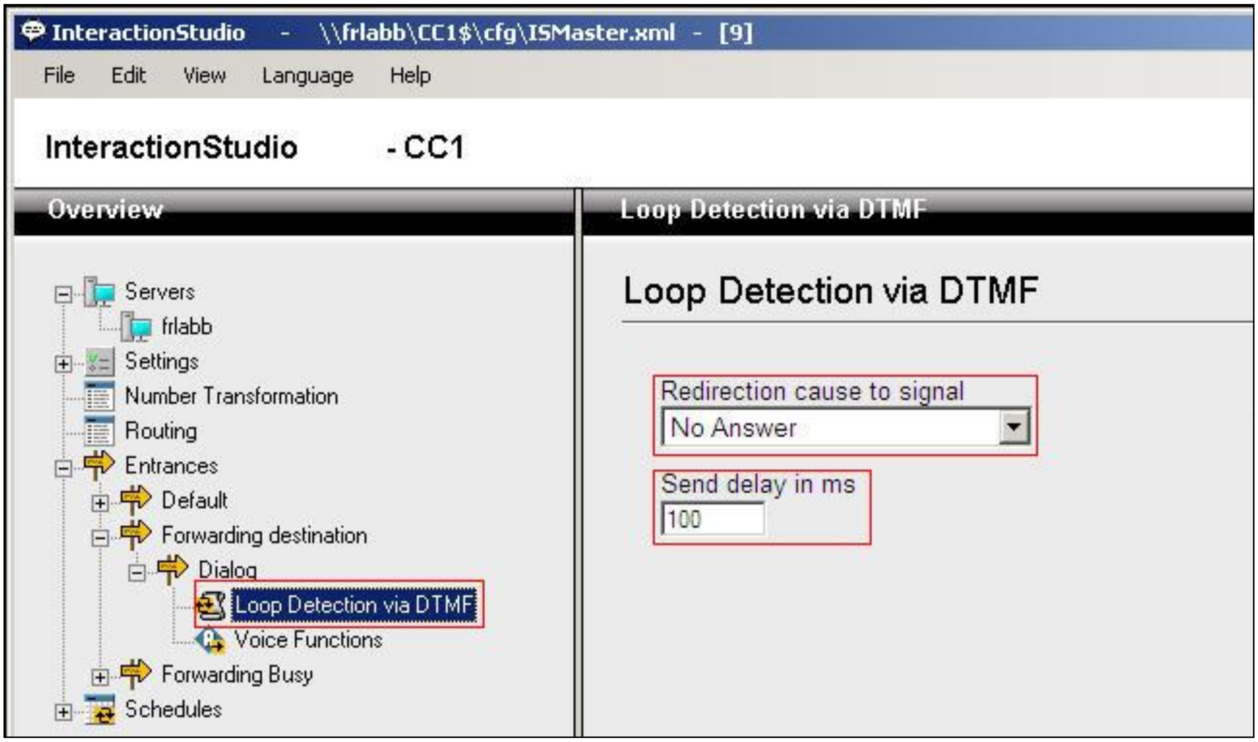
8.2.3. Configure Loop Detection via DTMF for Busy Signal

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

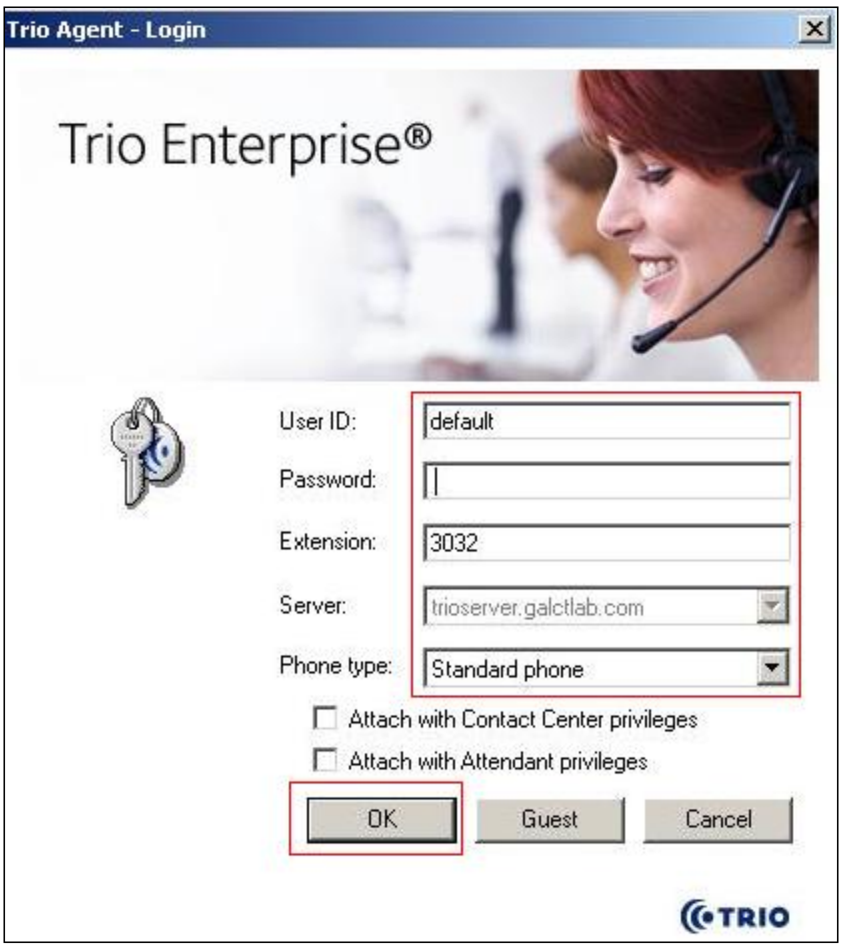
Navigate to **Entrances** → **Forwarding destination** → **Dialog** → **Loop Detection via DTMF**. Choose **No Answer** from the **Redirection cause to signal** drop-down box, and enter **100** in the **Send delay in ms** box.



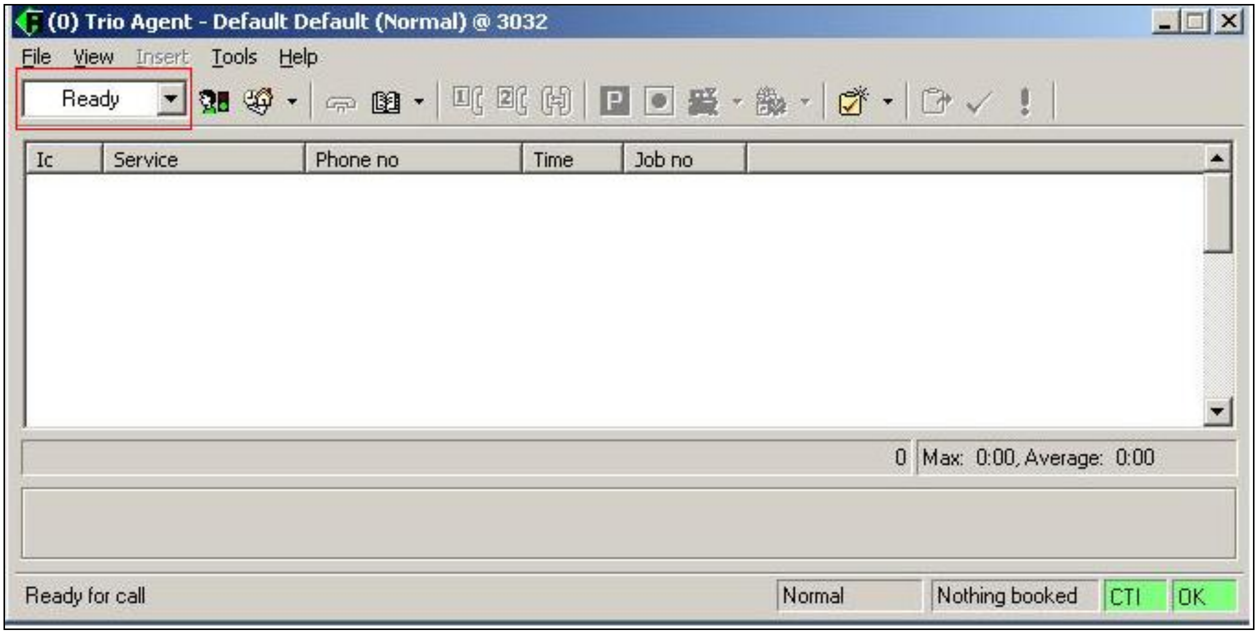
8.3. Configuring Trio Attendant

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 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**. Note this user ID and password is created during the installation of Trio Enterprise server. For **Extension**, enter 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).

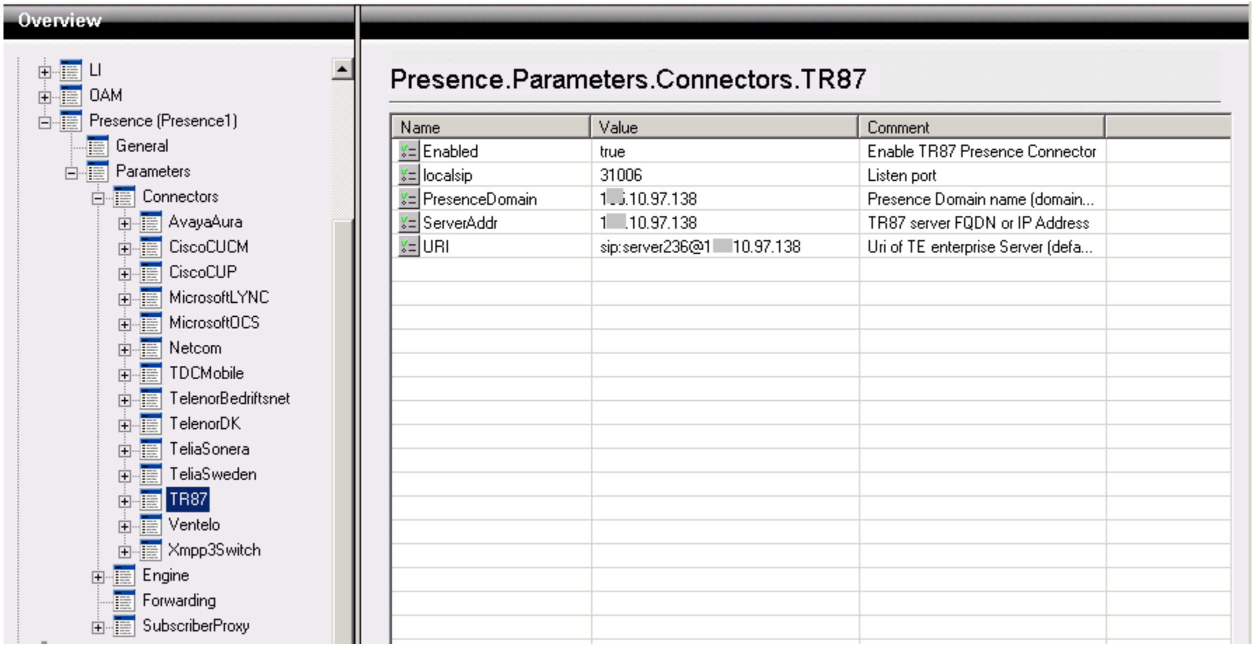


8.4. Configure TR87 on Trio Enterprise

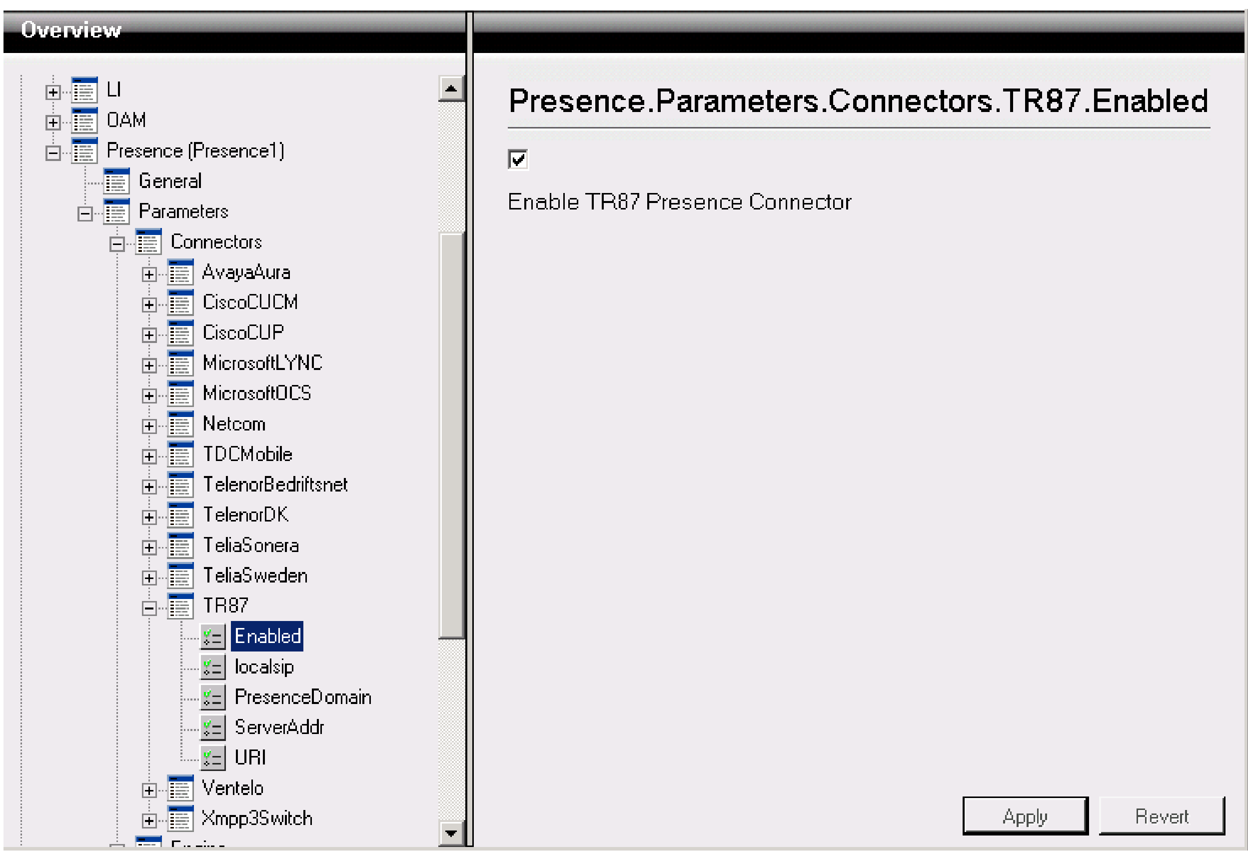
Click on **Start → Programs → Enterprise Management Center**.

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

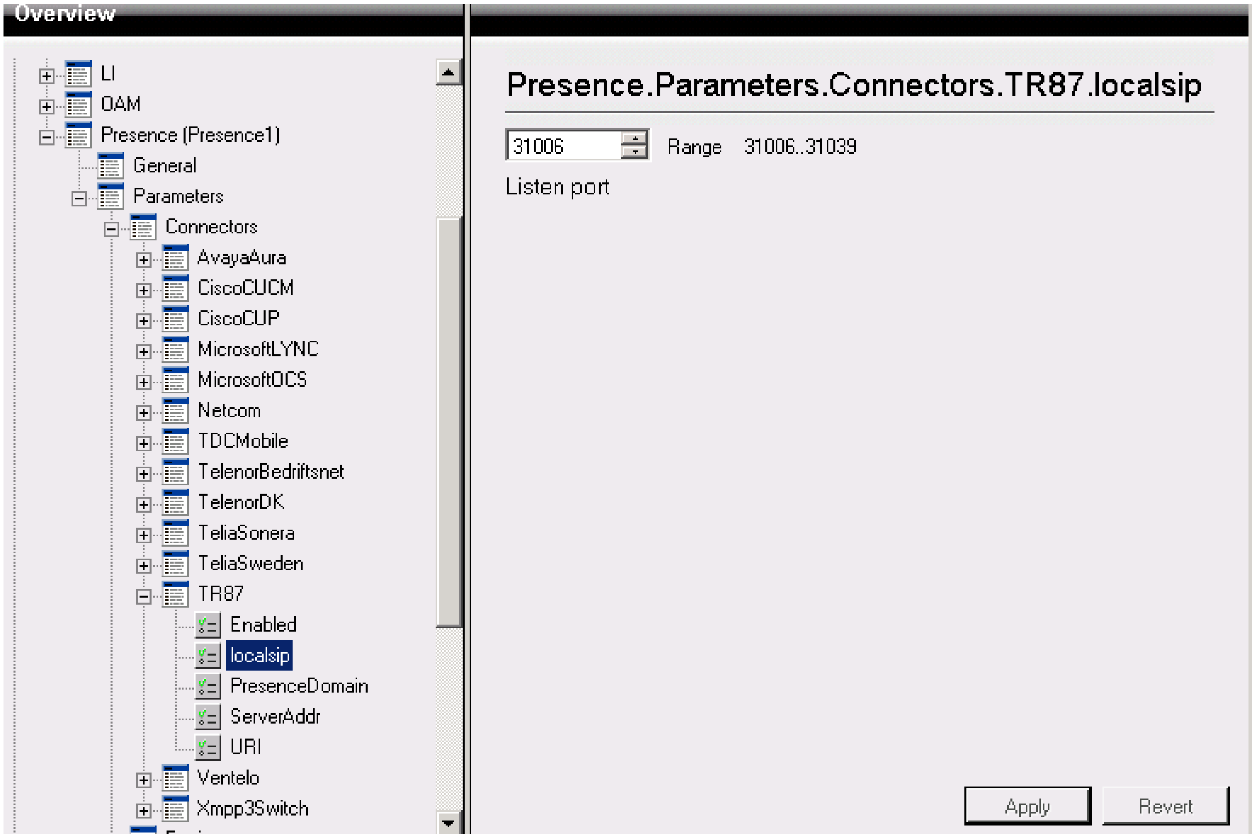
Click on **Parameters → Presence → Parameters → Connectors → TR87** in the left window.



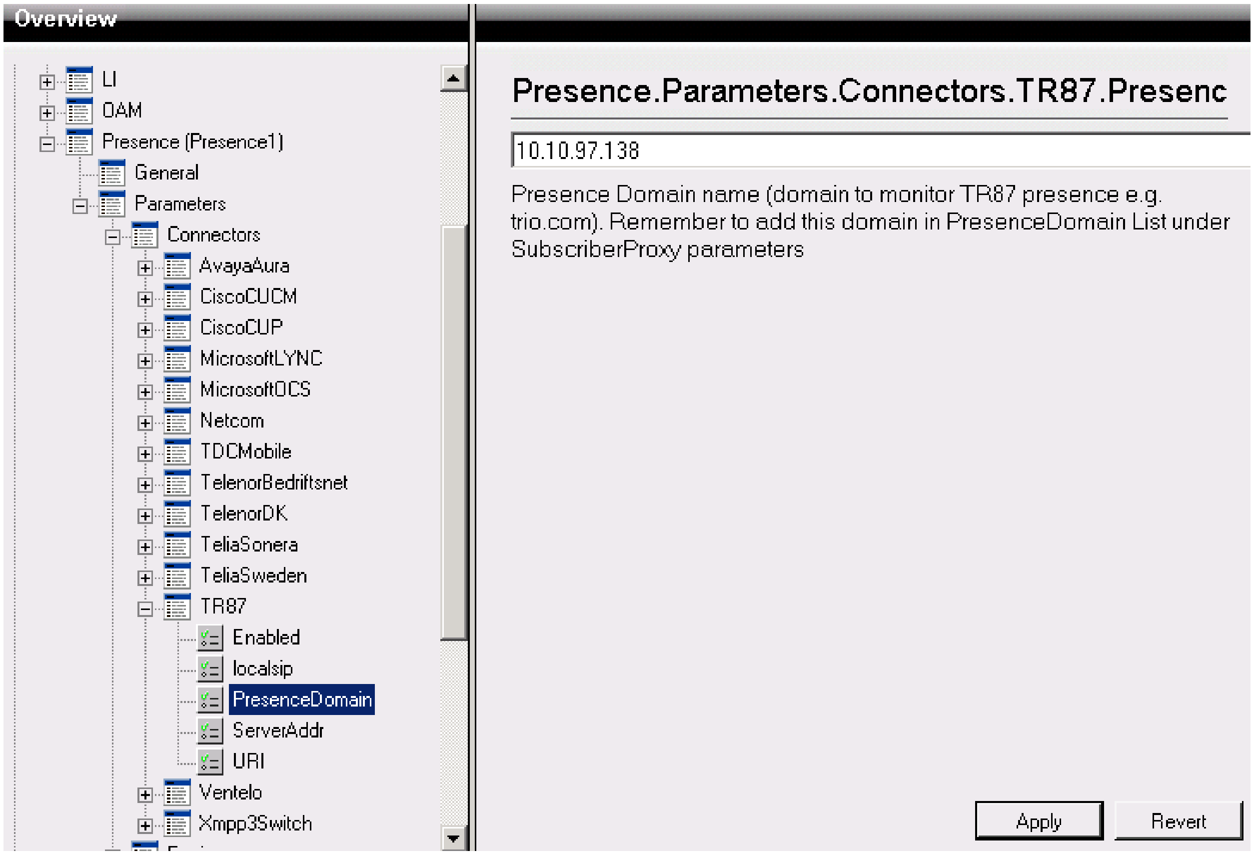
Under **TR87** select **Enabled** in the left window. Ensure that **Enable TR87 Presence Connector** is ticked as shown below. Click **Apply** to continue.



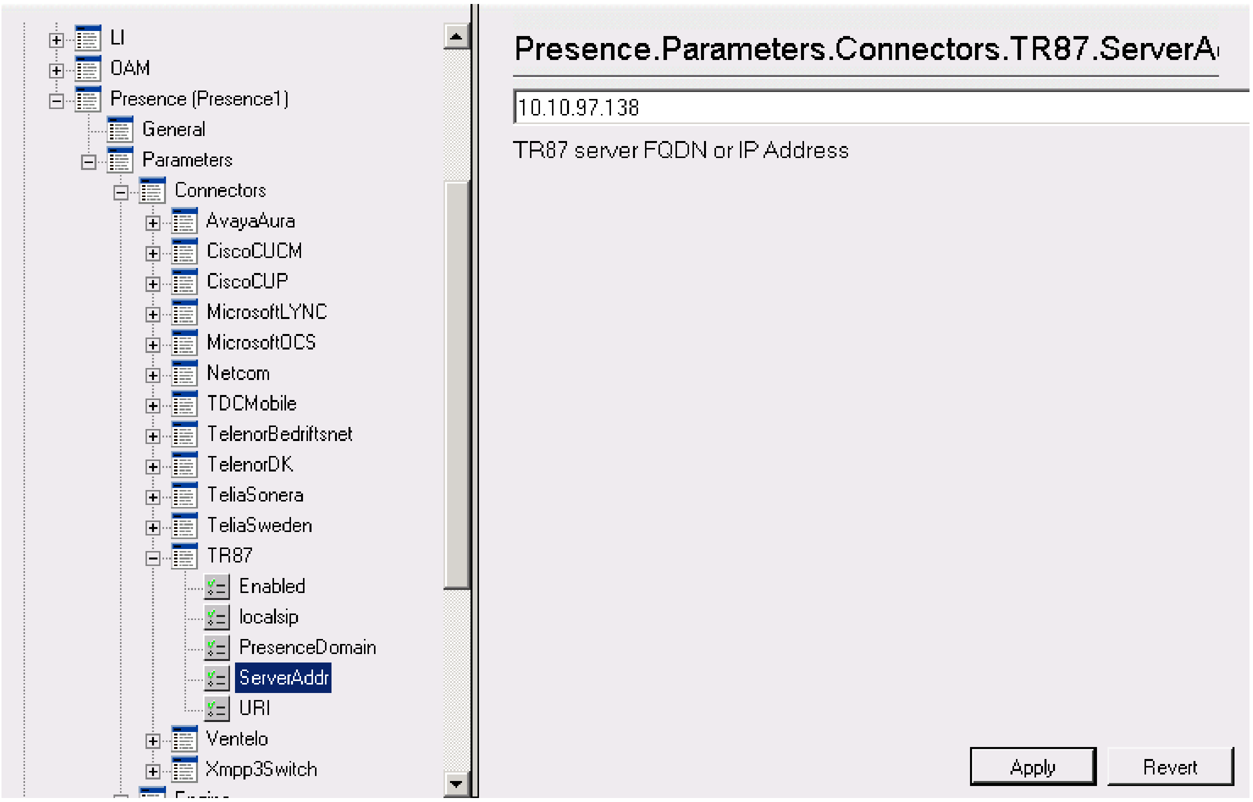
Select **localsip** under **TR87** in the left window and select the **Listen port** for TR87, for compliance testing this was left as default **31006** as shown below. Click **Apply** to continue.



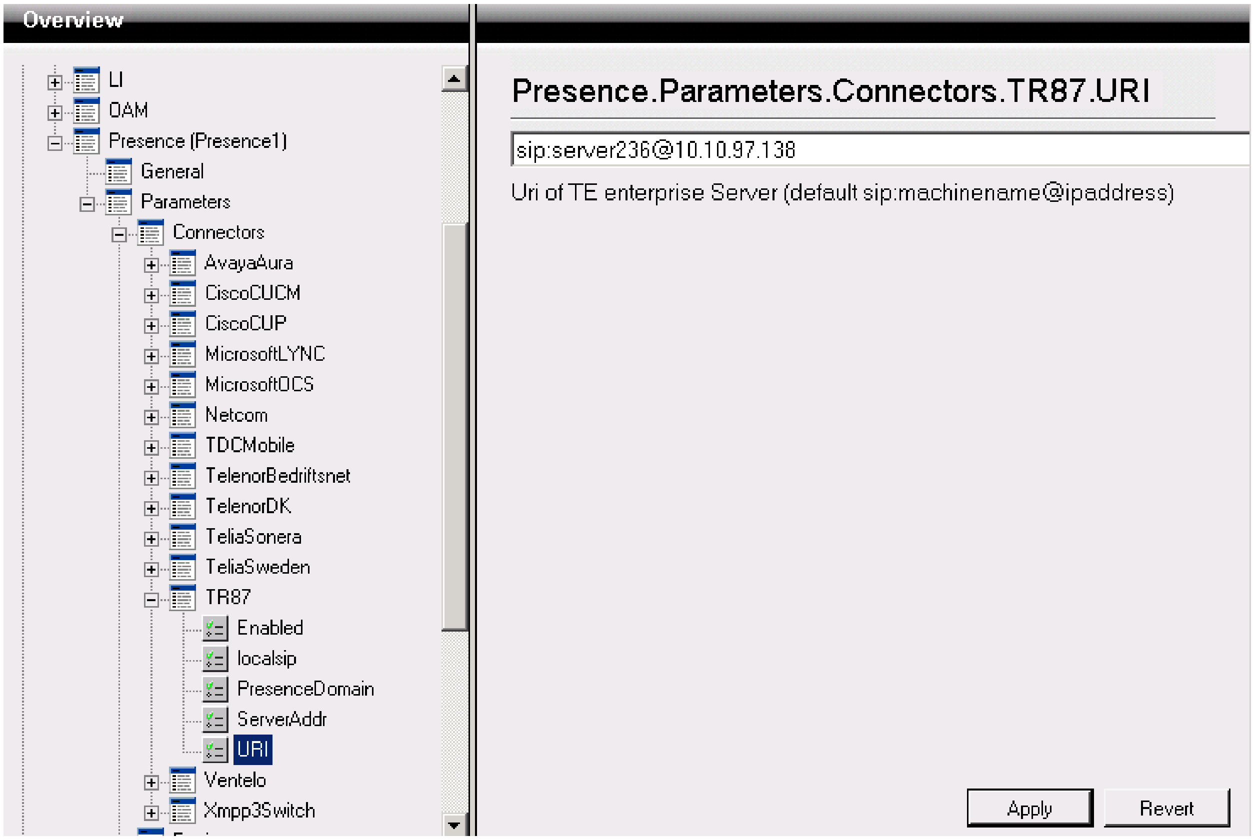
Select **PresenceDomain** under **TR87** in the left window. Enter the Node IP address of the CS1000E as per **Section 6**. Click **Apply** to continue.



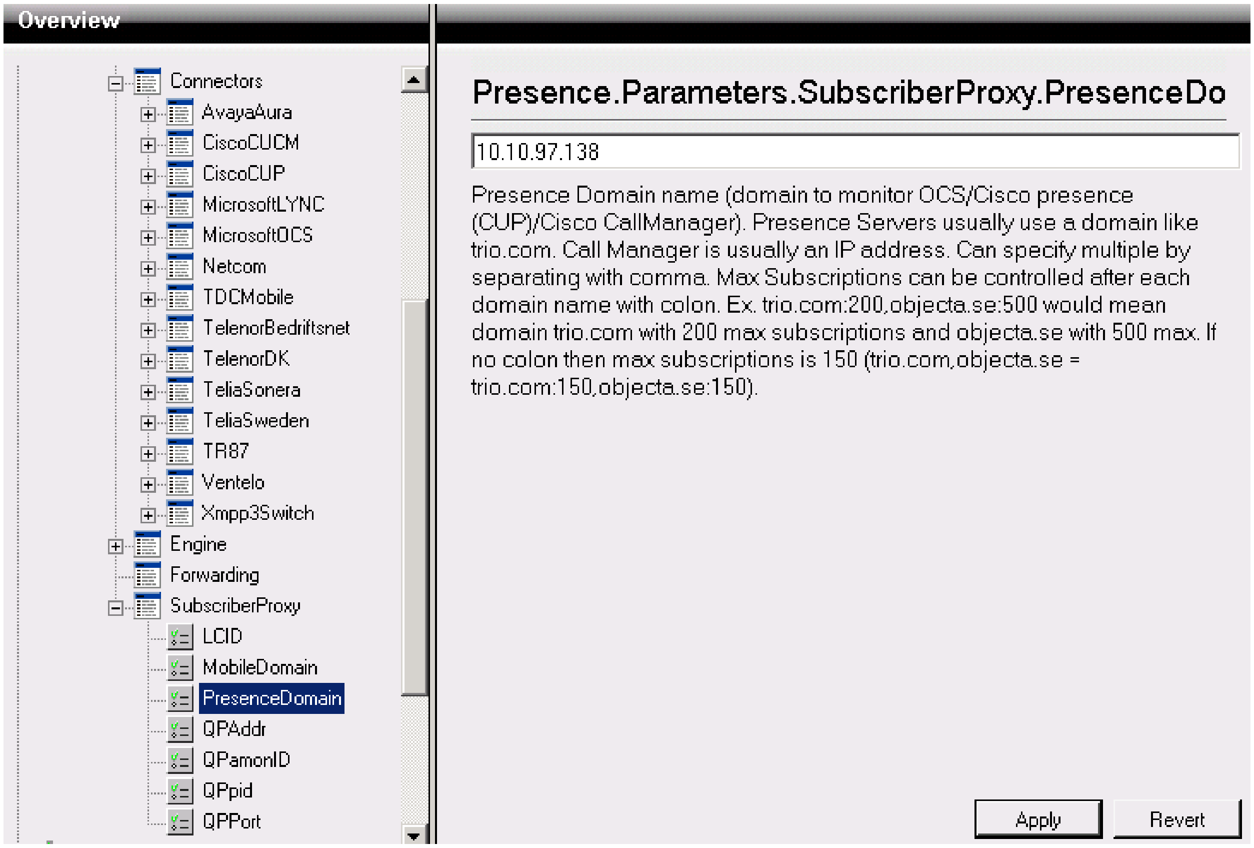
Select **ServerAddr** under **TR87** in the left window and again enter the Node IP address of the CS1000E. Click **Apply** to continue.



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

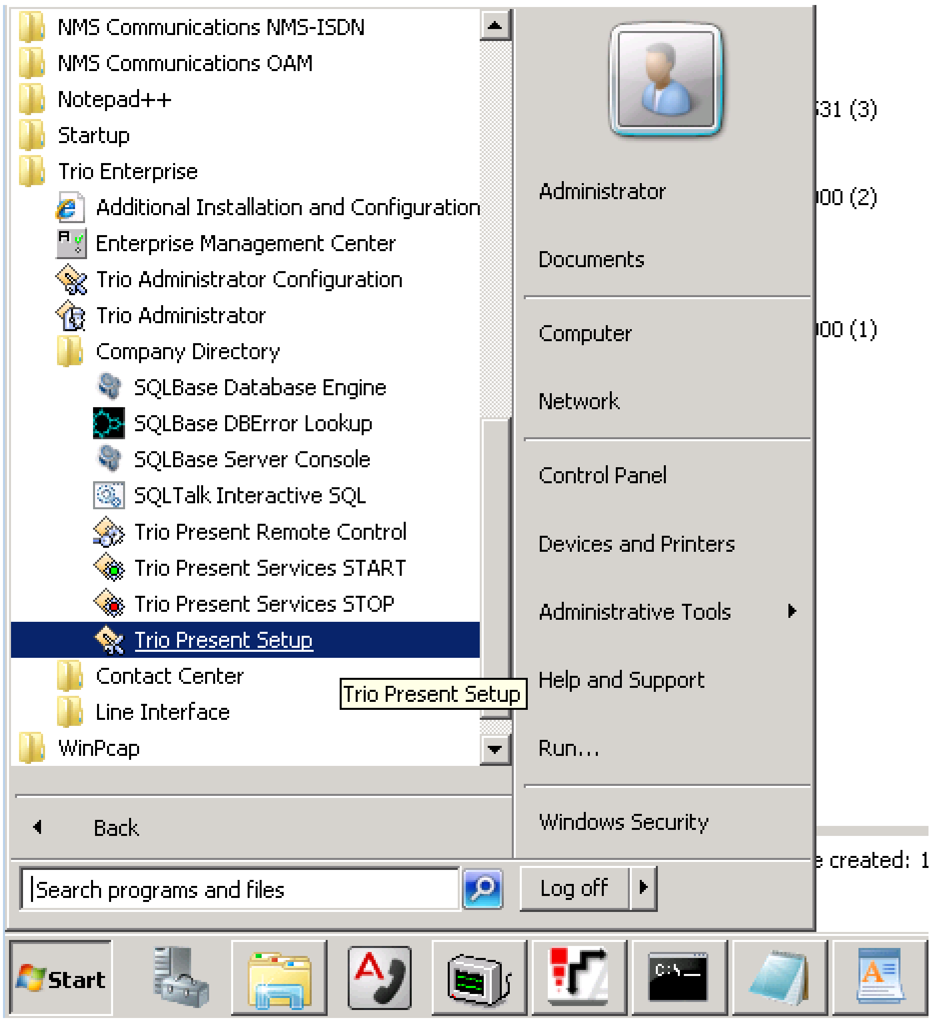


Select **PresenceDomain** under **SubscribeProxy** in the left window. Enter the Node IP address of the CS1000E in the right window. Click **Apply** to continue.

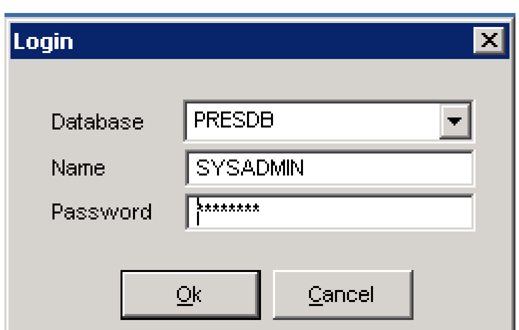


8.5. Configure ICP on Trio

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



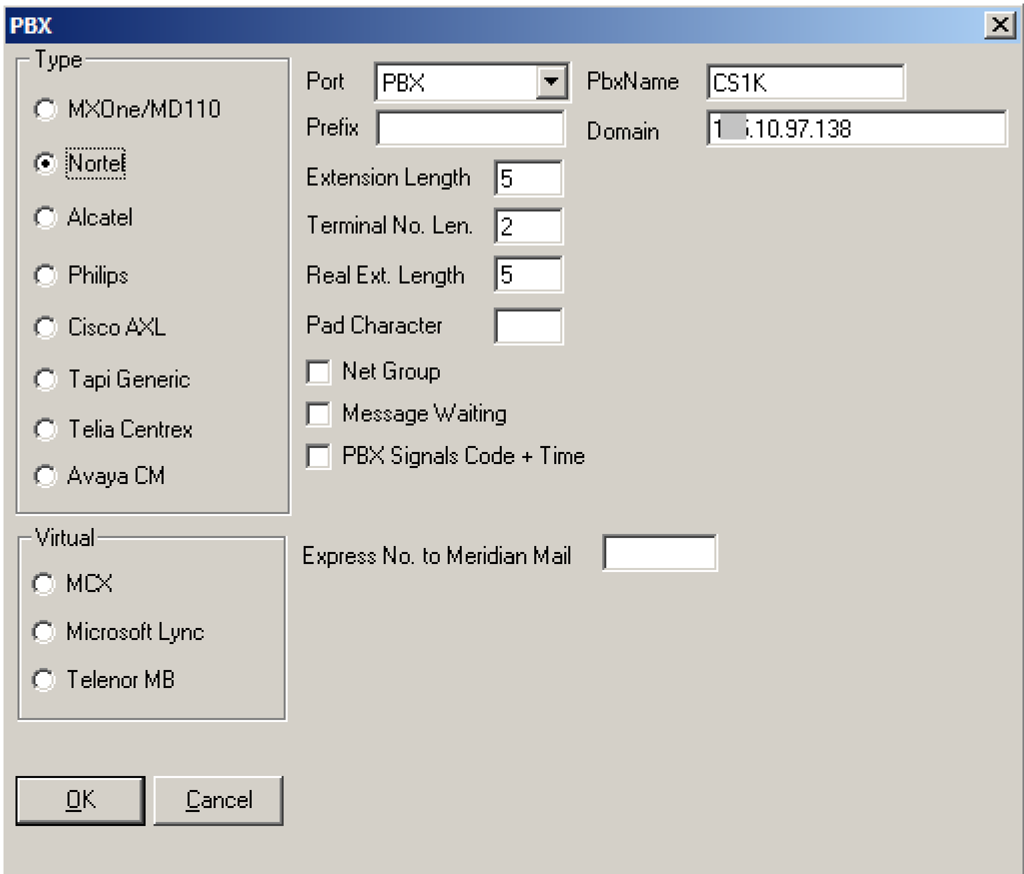
Enter the proper credentials and click **Ok**.



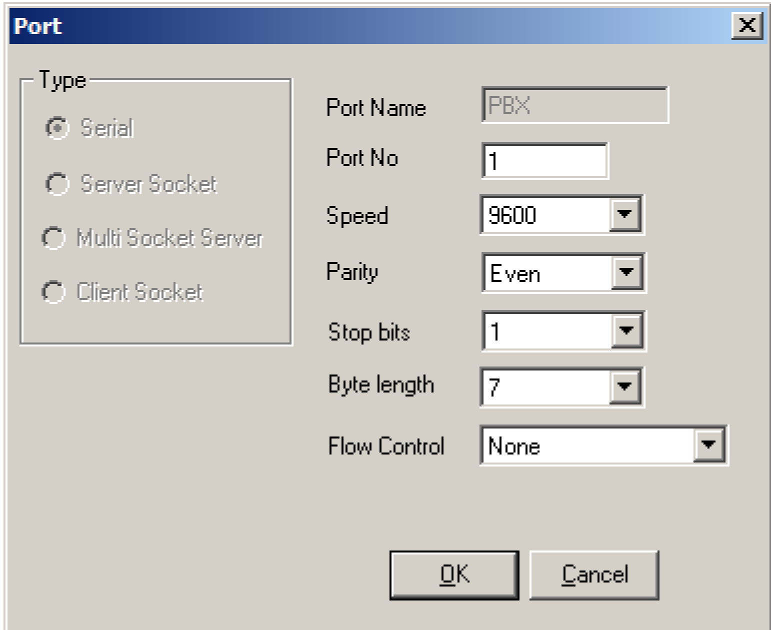
Highlight the selected PBX under the PBX tab and click on **Change** (not shown). This opens the window displayed on the right. Ensure the following are selected.

- **Type** **Nortel**
- **Port** **PBX**
- **Domain** Node IP Address of CS1000E, in this case it is IP address of Node **557**

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



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



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 23** in overlay 96 as shown below. The example below shows that D-Channel **23** is operational and established.

LD 96

```
Prompt                  Response                Description
>
LD 96                   Enter Overlay 96
STAT DCH 23             Check status of D-Channels 23
DCH 023       OPER EST ACTV AUTO
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 R5.0 from Enghouse Interactive AB to successfully interoperate with Avaya Communication Server 1000E R7.6 SP5 and Avaya Network Routing Server R7.6 SP5 using SIP trunks. Trio Enterprise passed all compliance testing successfully. Please see **Section 2.2** 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

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.