



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.

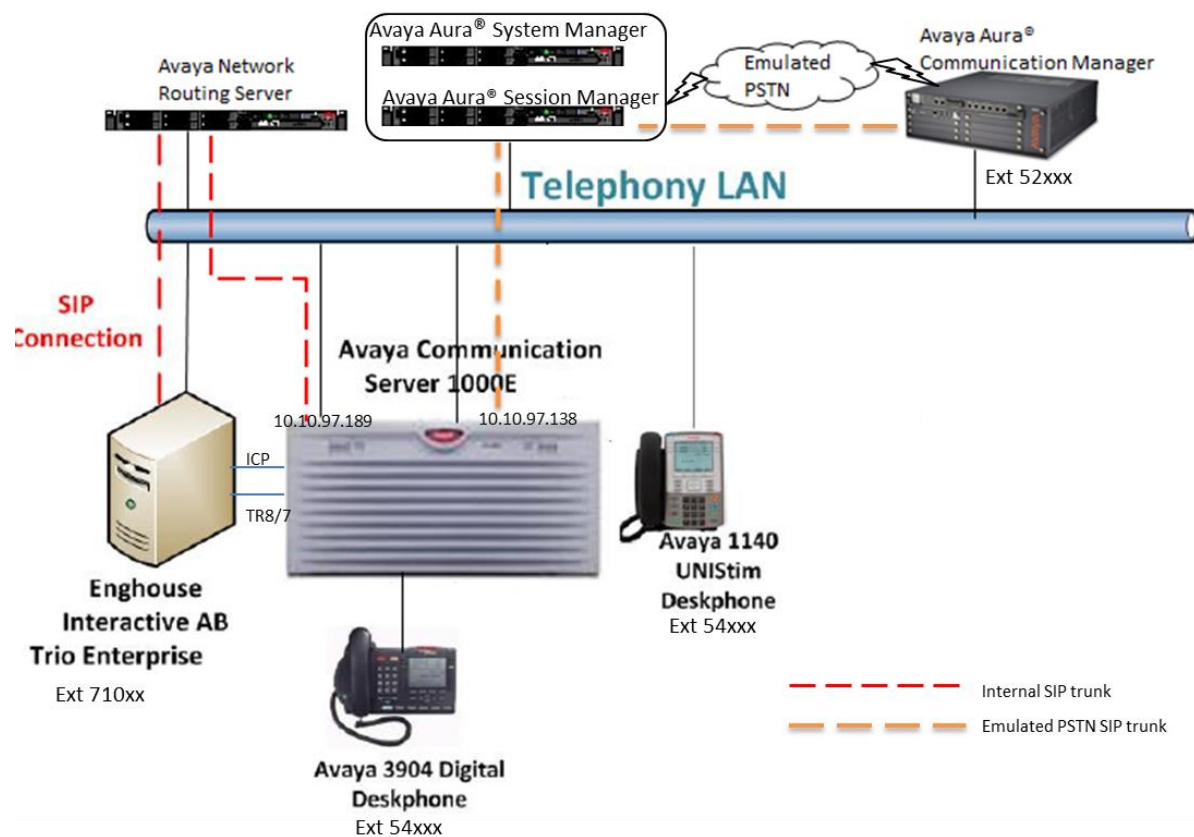


Figure 1: Configuration for Avaya Communication Server 1000E R7.6, Avaya Network Routing Server R7.6 and Trio Enterprise R5.0

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 UNISlim Deskphone	UNISlim 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.

AVAYA
Aura® System Manager 6.3

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to law enforcement officials.

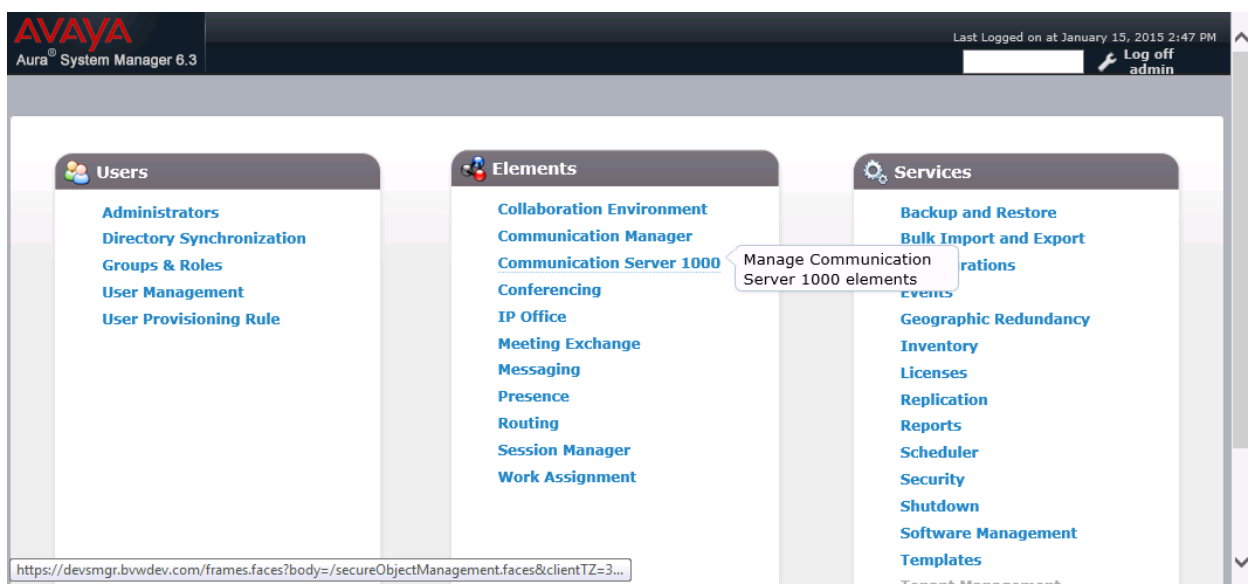
All users must comply with all corporate instructions regarding the protection of information assets.

User ID:

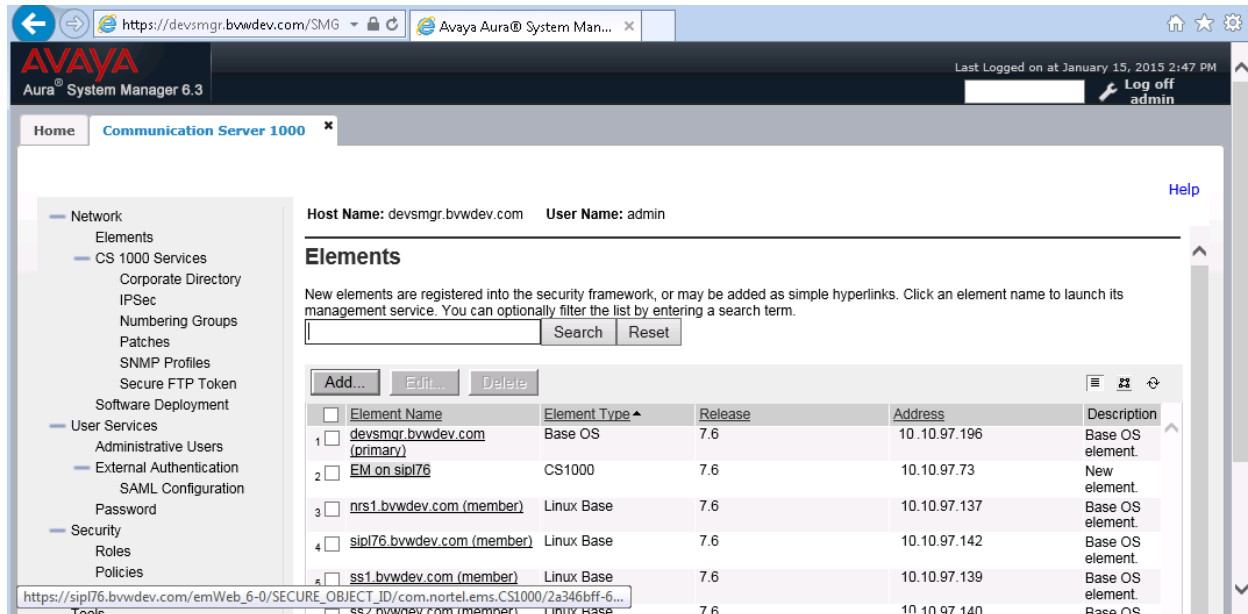
Password:

Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 26.0, 27.0 or 28.0.

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 sip176** link.



Host Name: devsmgr.bvwdev.com User Name: admin

Elements

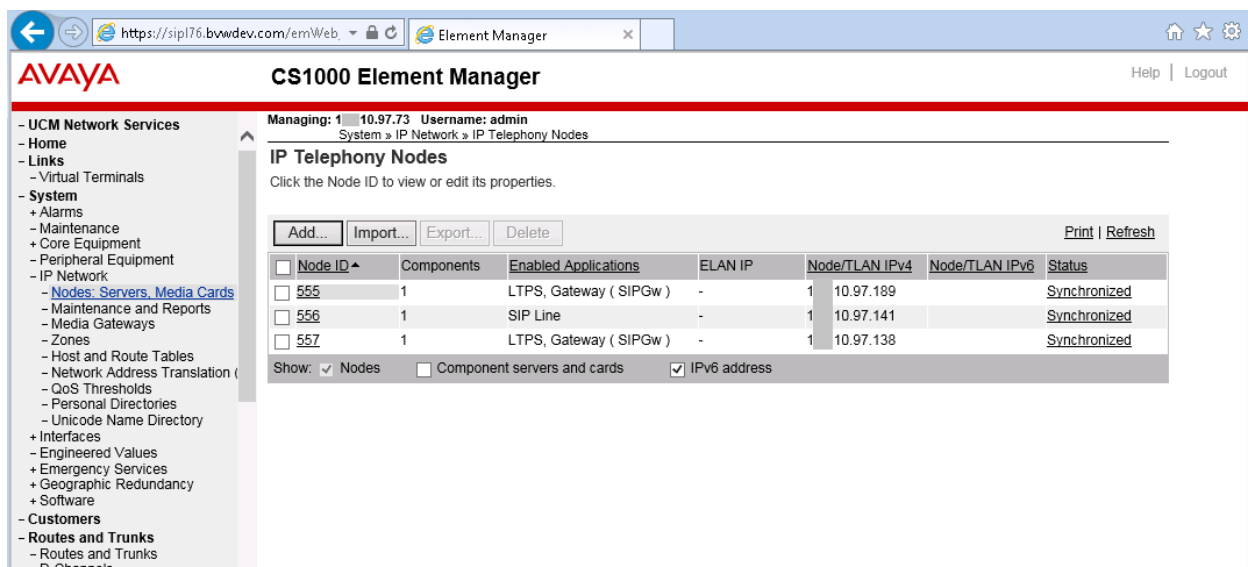
New elements are registered into the security framework, or may be added as simple hyperlinks. Click an element name to launch its management service. You can optionally filter the list by entering a search term.

Search Reset

Add... Edit... Delete

	Element Name	Element Type	Release	Address	Description
1	devsmgr.bvwdev.com (primary)	Base OS	7.6	10.10.97.196	Base OS element.
2	EM on sip176	CS1000	7.6	10.10.97.73	New element.
3	nrs1.bvwdev.com (member)	Linux Base	7.6	10.10.97.137	Base OS element.
4	sip176.bvwdev.com (member)	Linux Base	7.6	10.10.97.142	Base OS element.
5	ss1.bvwdev.com (member)	Linux Base	7.6	10.10.97.139	Base OS element.
6	ss2.bvwdev.com (member)	Linux Base	7.6	10.10.97.140	Base OS element.

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.



Managing: 10.97.73 Username: admin
System » IP Network » IP Telephony Nodes

IP Telephony Nodes

Click the Node ID to view or edit its properties.

Add... Import... Export... Delete Print | Refresh

Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
555	1	LTPS, Gateway (SIPGw)	-	10.97.189		Synchronized
556	1	SIP Line	-	10.97.141		Synchronized
557	1	LTPS, Gateway (SIPGw)	-	10.97.138		Synchronized

Show: ☒ Nodes ☐ Component servers and cards ☒ IPv6 address

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

AVAYA CS1000 Element Manager

Managing: 10.97.73 Username: admin
System » IP Network » IP Telephony Nodes » Node Details

Node Details (ID: 557 - LTPS, Presence Publisher, Gateway (SIPGw))

Node ID: 557 * (0-9999)

Call server IP address: 10.10.97.73 * TLAN address type: ☒ IPv4 only
☐ IPv4 and IPv6

Embedded LAN (ELAN) **Telephony LAN (TLAN)**

Gateway IP address: 10.10.97.65 * Node IPv4 address: 10.10.97.138 *

Subnet mask: 255.255.255.192 * Subnet mask: 255.255.255.192 *

Node IPv6 address:

IP Telephony Node Properties

- Voice Gateway (VGW) and Codecs
- Quality of Service (QoS)
- LAN
- SNTP
- Numbering Zones
- MCDN Alternative Routing Treatment (MALT) Causes

Applications (click to edit configuration)

- SIP Line
- Terminal Proxy Server (TPS)
- Gateway (SIPGw)
- Personal Directories (PD)
- Presence Publisher
- IP Media Services

* Required Value. Save Cancel

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.

AVAYA CS1000 Element Manager

Help | Logout

General | SIP Gateway Settings | SIP Gateway Services

SIP CTI Service: ☒ Enable CTI service ☐ TLS endpoints only

CTI settings

Customer number: 0

Maximum associations per DN: 3

International calls: ☐ Place as national
For calls within this country.

CTI CLID presentation

Dialing plan: CDP

Calling device URI format: phone-context=dialstring

Home location code:

Country code (CCC):

Area code:

Dial plan prefixes

National:

International:

Location code call:

Special number:

Subscriber:

* Required Value. Save Cancel

Note: Changes made on this page will NOT be transmitted until the Node is also saved.

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

The screenshot shows the AVAYA CS1000 Element Manager interface. The left sidebar contains a navigation tree with categories like UCM Network Services, Home, Links, System, and Customers. The main content area is titled 'Node Details (ID: 557 - LTPS, Presence Publisher, Gateway (SIPGw))'. It displays various configuration fields for the node, including Node ID (557), Call server IP address (10.10.97.73), Embedded LAN (ELAN) Gateway IP address (10.10.97.65), and Subnet mask (255.255.255.192). There are also fields for Telephony LAN (TLAN) Node IPv4 address (10.10.97.138) and Subnet mask (255.255.255.192). A section titled 'IP Telephony Node Properties' lists several links like Voice Gateway (VGW) and Codecs, Quality of Service (QoS), LAN, SNTP, Numbering Zones, and MCDN Alternative Routing Treatment (MALT) Causes. Another section titled 'Applications (click to edit configuration)' lists links like SIP Line, Terminal Proxy Server (TPS), Gateway (SIPGw), Personal Directories (PD), Presence Publisher, and IP Media Services. At the bottom right, there are 'Save' and 'Cancel' buttons.

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

The screenshot shows the AVAYA CS1000 Element Manager interface after saving the node configuration. The left sidebar is the same as the previous screenshot. The main content area is titled 'Node Saved'. It displays a message: 'Node ID: 555 has been saved on the call server. The new configuration must also be transferred to associated servers and media cards.' Below this message, there are two buttons: 'Transfer Now...' and 'Show Nodes'. The 'Transfer Now...' button is highlighted in blue. To the right of the buttons, there is explanatory text: 'You will be given an option to select individual servers, or transfer to all.' and 'You may initiate a transfer manually at a later time.'

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**.

AVAYA CS1000 Element Manager

Managing: 10.97.73 Username: admin
System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

Synchronize Configuration Files (Node ID <557>)

Note: Select components to synchronize their configuration files with call server data. This process transfers server INI files to selected components, and requires a restart* of applications on affected server(s) when complete.

Start Sync Cancel Restart Applications [Print](#) [Refresh](#)

<input checked="" type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input checked="" type="checkbox"/>	ss2	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	Sync required

* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNTP settings, SIP and H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding or removing application servers.

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

AVAYA CS1000 Element Manager

Managing: 10.97.73 Username: admin
System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

Synchronize Configuration Files (Node ID <557>)

Note: Select components to synchronize their configuration files with call server data. This process transfers server INI files to selected components, and requires a restart* of applications on affected server(s) when complete.

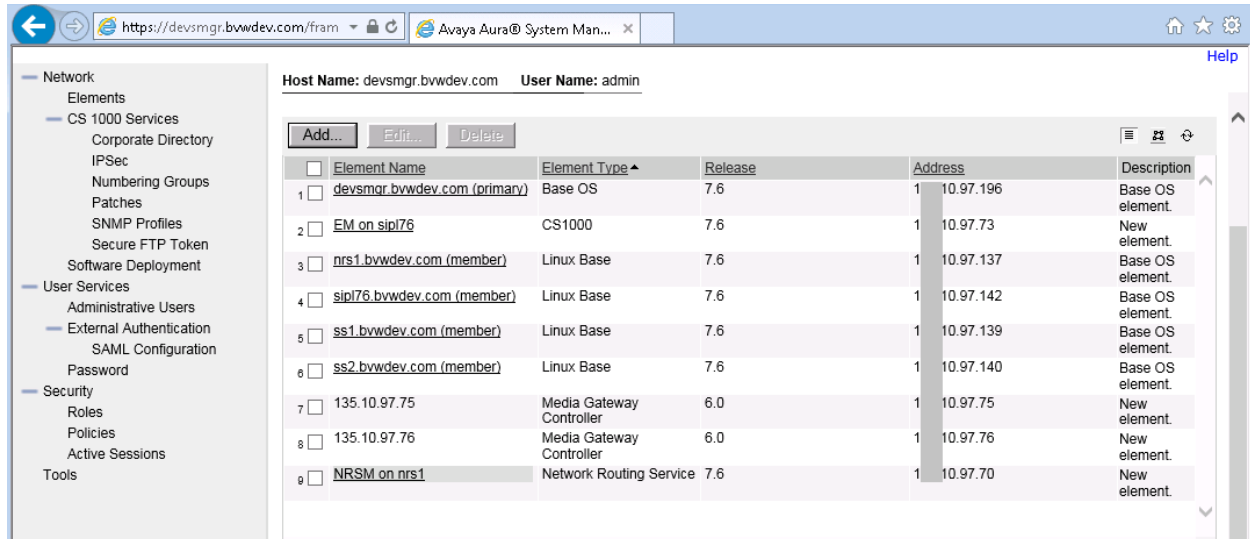
Start Sync Cancel **Restart Applications** [Print](#) [Refresh](#)

<input checked="" type="checkbox"/>	Hostname	Type	Applications	Synchronization Status
<input checked="" type="checkbox"/>	ss2	Signaling_Server	SIP Line, LTPS, Gateway (SIP/H323), PD, Presence Publisher, IP Media Services	Synchronized

* Application restart is only required for initial system configuration or if changes have been made to general LAN configurations, SNTP settings, SIP and H323 Gateway settings, network connectivity related parameters like ports and IP address, enabling or disabling services, or adding or removing application servers.

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.

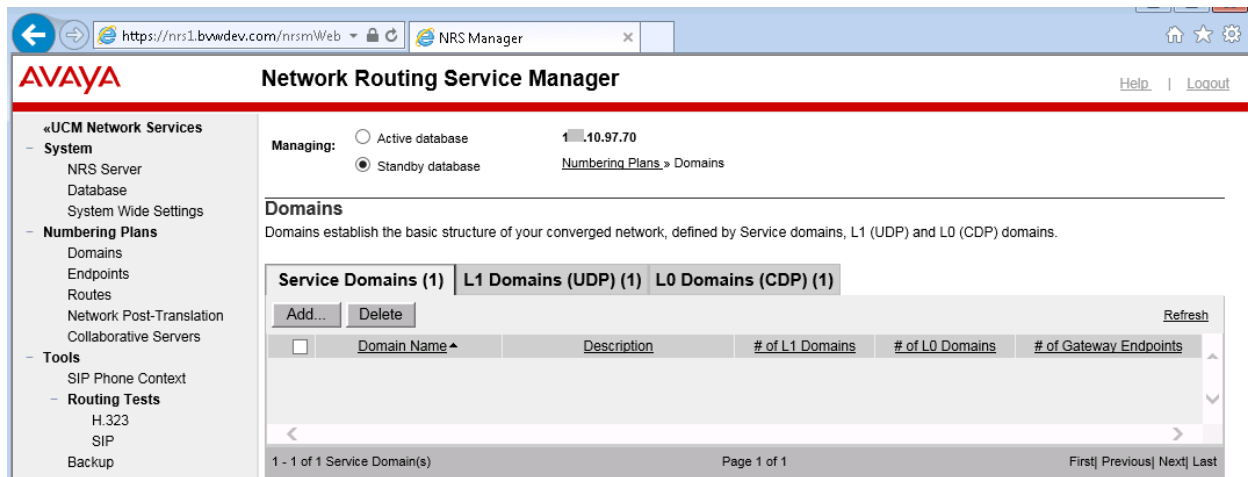


7.1. Add a New Domain

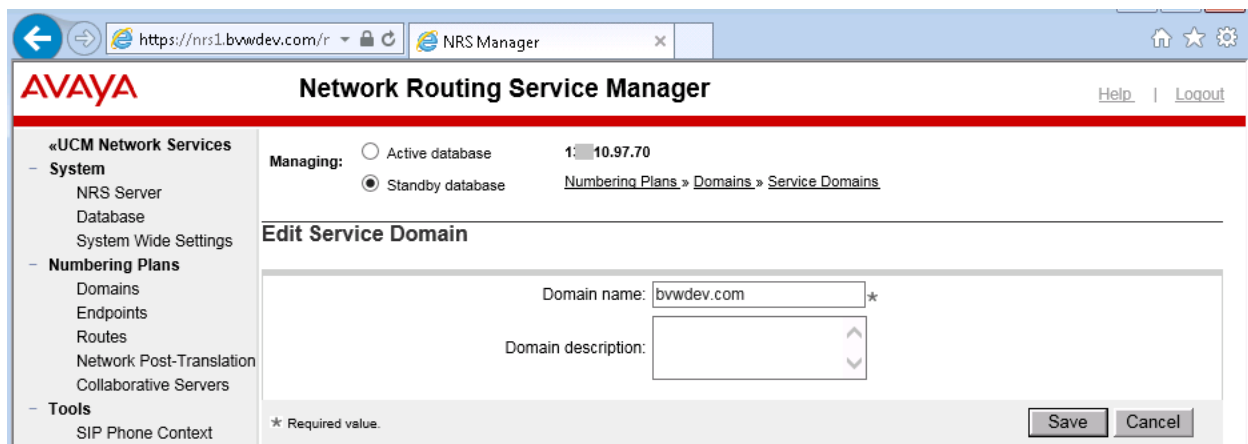
In the event there is no domain present a new one must be added as follows. Note that there are three separate entries for this domain and all three must be added.

- 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**.

AVAYA Network Routing Service Manager [Help](#) | [Logout](#)

«UCM Network Services

- System
 - NRS Server
 - Database
 - System Wide Settings
- Numbering Plans
 - Domains
 - Endpoints
 - Routes
 - Network Post-Translation
 - Collaborative Servers
- Tools
 - SIP Phone Context
- Routing Tests
 - H.323
 - SIP
 - Backup
 - Restore
 - GK/NRS Data upgrade

Managing: ☐ Active database **10.97.70**
☒ Standby database [Numbering Plans » Domains](#)

Domains

Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.

Service Domains (1) | L1 Domains (UDP) (1) | L0 Domains (CDP) (1)

Filter by Domain :

[Refresh](#)

<input type="checkbox"/>	ID	Description	# of L0 Domains	# of Gateway Endpoints	# of Routing Entries	Context

1 - 1 of 1 L1 Domain(s) Page 1 of 1 [First](#) | [Previous](#) | [Next](#) | [Last](#)

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

«UCM Network Services

- System
 - NRS Server
 - Database
 - System Wide Settings
- Numbering Plans
 - Domains
 - Endpoints
 - Routes
 - Network Post-Translation
 - Collaborative Servers
- Tools
 - SIP Phone Context
- Routing Tests
 - H.323
 - SIP
 - Backup
 - Restore
 - GK/NRS Data upgrade

Managing: ☐ Active database **10.97.70**
☒ Standby database [Numbering Plans » Domains » L1 Domain](#)

Edit L1 Domain (bvwddev.com)

Domain name: *

Domain description:

Endpoint authentication enabled:

Authentication password:

E.164 country code:

E.164 area code:

E.164 international dialing access code:

E.164 international dialing code length: (0-99)

E.164 national dialing access code:

E.164 national dialing code length: (0-99)

E.164 local (subscriber) dialing access code:

E.164 local (subscriber) dialing code length: (0-99)

Private L1 domain (UDP location) dialing access code:

Private L1 domain (UDP location) dialing code: (0-99)

* Required value

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

The screenshot shows the Avaya Network Routing Service Manager interface. The left sidebar contains a navigation menu with categories: «UCM Network Services», System (NRS Server, Database, System Wide Settings), Numbering Plans (Domains, Endpoints, Routes, Network Post-Translation, Collaborative Servers), and Tools (SIP Phone Context, Routing Tests, H.323, SIP, Backup, Restore, GK/NRS Data upgrade). The main content area is titled 'Network Routing Service Manager' and includes a 'Help' and 'Logout' link. Below the title, there's a 'Managing' section with radio buttons for 'Active database' and 'Standby database', and a status indicator '1: 10.97.70'. A breadcrumb trail reads 'Numbering Plans » Domains'. The 'Domains' section has a sub-header 'Domains' and a description: 'Domains establish the basic structure of your converged network, defined by Service domains, L1 (UDP) and L0 (CDP) domains.' Below this, there are three tabs: 'Service Domains (1)', 'L1 Domains (UDP) (1)', and 'L0 Domains (CDP) (1)'. The 'L0 Domains (CDP) (1)' tab is selected. A filter box shows 'Filter by Domain : bvwdev.com / udp'. There are 'Add...' and 'Delete' buttons, and a 'Refresh' link. A table with columns 'ID', 'Description', '# of Gateway Endpoints', '# of Routing Entries', and 'Context' is shown. The table has one entry with ID '1'. At the bottom, it says '1 - 1 of 1 L0 Domain(s)', 'Page 1 of 1', and navigation links 'First| Previous| Next| Last'.

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

The screenshot shows the 'Edit L0 Domain (bvwdev.com / udp)' form. The left sidebar is the same as the previous screenshot. The main content area has a 'Managing' section with 'Active database' and 'Standby database' radio buttons, and a status indicator '1: 10.97.70'. A breadcrumb trail reads 'Numbering Plans » Domains » L0 Domain'. The form title is 'Edit L0 Domain (bvwdev.com / udp)'. The form fields include: 'Domain name' (cdp), 'Domain description' (empty), 'Endpoint authentication enabled' (Not configured), 'Authentication password' (empty), 'E.164 country code' (empty), 'E.164 area code' (empty), 'Private unqualified number label' (PrivateUnknown), 'E.164 international dialing access code' (empty), 'E.164 international dialing code length' (empty, range 0-99), 'E.164 national dialing access code' (empty), 'E.164 national dialing code length' (empty, range 0-99), 'E.164 local (subscriber) dialing access code' (empty), 'E.164 local (subscriber) dialing code length' (empty, range 0-99), and 'Private L1 domain (UDP location) dialing access code' (empty). At the bottom, there's a note '* Required value.' and 'Save' and 'Cancel' buttons.

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**.

The screenshot shows the Avaya Network Routing Service Manager interface. On the left is a navigation tree with 'UCM Network Services' expanded, showing 'System', 'Numbering Plans', and 'Tools'. Under 'Numbering Plans', 'Endpoints' is selected. The main area shows 'Managing: Standby database' and '10.97.70'. Below this is the 'Search for Endpoints' section with a text input for 'Endpoint ID' (containing '*') and three dropdown menus for 'Limit results to Domain' (bvwddev.com), 'udp', and 'cdp'. A 'Search' button is at the bottom right of the search area. Below the search area are two tabs: 'Gateway Endpoints (4)' and 'User Endpoints (0)'. The 'Gateway Endpoints' tab is active, showing a table with columns: ID, Supported Protocols, SIP mode, Call Signaling IP, Description, # of Routing Entries, and Context. The table is currently empty.

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

The screenshot shows the 'Edit Gateway Endpoint' form for the domain 'bvwddev.com / udp / cdp'. The form has several fields: 'End point name' (Trio), 'Description' (empty), 'Trust Node' (checked), 'Tandem gateway endpoint name' (Not Applicable), 'Endpoint authentication enabled' (Authentication off), 'Authentication password' (empty), 'E.164 country code' (empty), 'E.164 area code' (empty), 'E.164 international dialing access code' (empty), 'E.164 international dialing code length' (empty, range 0-99), 'E.164 national dialing access code' (empty), 'E.164 national dialing code length' (empty, range 0-99), 'E.164 local (subscriber) dialing access code' (empty), and 'E.164 local (subscriber) dialing code length' (empty, range 0-99). A 'Save' button and a 'Cancel' button are at the bottom right. A note at the bottom left says '* Required value'.

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.

AVAYA

Network Routing Service Manager

Help

Logout

«UCM Network Services

- System

NRS Server

Database

System Wide Settings

- Numbering Plans

Domains

Endpoints

Routes

Network Post-Translation

Collaborative Servers

- Tools

SIP Phone Context

- Routing Tests

H.323

SIP

Backup

Restore

GK/NRS Data upgrade

Search for Endpoints

Hide

Enter an endpoint ID (use * for all) and click Search. You may narrow the search by specifying a particular domain.

Endpoint ID: *

Limit results to Domain: bwvdev.com / udp / cdp

Results per page: 50

Search

Gateway Endpoints (4)

User Endpoints (0)

SIP phone context...

Refresh

<input type="checkbox"/>	ID	Supported Protocols	SIP mode	Call Signaling IP	Description	# of Routing Entries	Context
1 <input type="checkbox"/>	car2-core	Dynamic SIP endpoint	Proxy Mode	Not registered		1	bwvdev.com / udp / cdp
2 <input type="checkbox"/>	Trio	Dynamic SIP endpoint	Proxy Mode	Not registered		1	bwvdev.com / udp / cdp
3 <input type="checkbox"/>	SS1	Dynamic SIP endpoint	Proxy Mode	10.97.189	SIP trunk to the Middle CS1K	5	bwvdev.com / udp / cdp

Copyright 2002-2015 Avaya Inc. All rights reserved.

https://nrs1.bwvdev.com/nrsWeb_6_0/SECURE_OBJECT_ID/com.nortel.ems.NRS/9d3b7020-6371-11e4-8cca-fbb0e29d817d/pages/endpoints.faces?si

Internet | Protected Mode: Off

100%

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**.

The screenshot shows the 'Network Routing Service Manager' interface. On the left is a navigation tree with categories: «UCM Network Services, System (NRS Server, Database, System Wide Settings), Numbering Plans (Domains, Endpoints, Routes, Network Post-Translation, Collaborative Servers), Tools (SIP Phone Context, Routing Tests, H.323, SIP, Backup, Restore, GK/NRS Data upgrade). The main area is titled 'Managing: 1 10.97.70' and 'Numbering Plans » Routes'. Below this is a 'Search for Routing Entries' section with a text input for 'Enter a DnPrefix and Dn Type (use * for all) and click Search. You may narrow the search by specifying a particular domain.' The search criteria are: DN Prefix: *, DN Type: All DN Types, Limit results to Domain: bvwdev.com, udp, cdp, and Endpoint Name: Trio. A 'Search' button is present. Below the search section is a table titled 'Routing Entries (1) | Default Routes (0) | Emergency Fallback Routes (0)'. The table has columns: DN Prefix, DN Type, Route Cost, SIP URI Phone Context, and Context. The first row shows a single entry. At the bottom, it says '1 - 1 of 1 Routing Entry(ies)', 'Page 1 of 1', and navigation links 'First| Previous| Next| Last'.

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.

The screenshot shows the 'Network Routing Service Manager' interface with the 'Edit Routing Entry' form. The navigation tree on the left is the same as in the previous screenshot. The main area is titled 'Managing: 1 10.97.70' and 'Numbering Plans » Routes » Routing Entry'. Below this is a section titled 'Edit Routing Entry (bvwdev.com / udp / cdp / Trio)'. The form contains three fields: 'DN type' with a dropdown menu set to 'Private level 0 regional (CDP steering code)', 'DN prefix' with a text input containing '710' and an asterisk, and 'Route cost' with a text input containing '1' and an asterisk. A 'Save' button and a 'Cancel' button are at the bottom right. A note at the bottom left says '* Required value.'

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

AVAYA **Network Routing Service Manager** [Help](#) | [Logout](#)

Results per page: [50] [Clear](#)

Routing Entries (7) **Default Routes (0)** **Emergency Fallback Routes (0)**

[Routing test...](#) [Export...](#) [Refresh](#)

<input type="checkbox"/>	DN Prefix	DN Type	Route Cost	SIP URI Phone Context	Context
1 <input type="checkbox"/>	28	Private level 0 regional (CDP steering code)	1	cdp.udp	bwwdev.com / udp / cdp / SS1
2 <input type="checkbox"/>	46	Private level 0 regional (CDP steering code)	1	cdp.udp	bwwdev.com / udp / cdp / car2-cores
3 <input type="checkbox"/>	52	Private level 0 regional (CDP steering code)	1	cdp.udp	bwwdev.com / udp / cdp / SS1
4 <input type="checkbox"/>	53	Private level 0 regional (CDP steering code)	1	cdp.udp	bwwdev.com / udp / cdp / SS1
5 <input type="checkbox"/>	54	Private level 0 regional (CDP steering code)	1	cdp.udp	bwwdev.com / udp / cdp / SS1
6 <input type="checkbox"/>	55	Private level 0 regional (CDP steering code)	1	cdp.udp	bwwdev.com / udp / cdp / SS1
7 <input type="checkbox"/>	710	Private level 0 regional (CDP steering code)	1	cdp.udp	bwwdev.com / udp / cdp / Trio

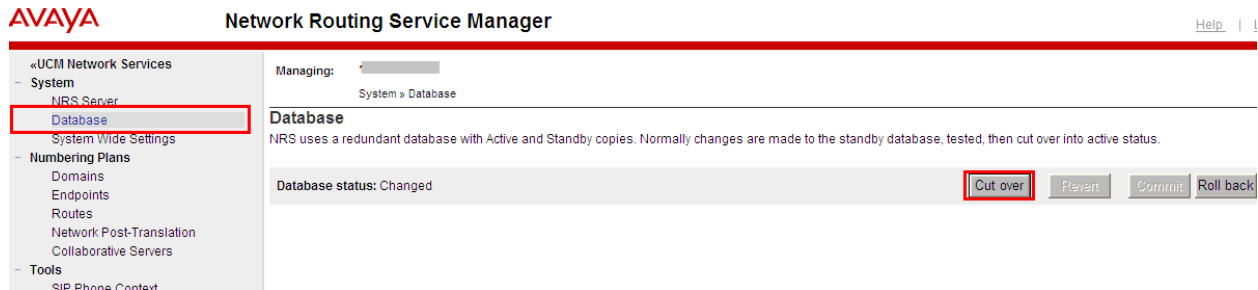
1 - 7 of 7 Routing Entry(ies) Page 1 of 1 [First](#) [Previous](#) [Next](#) [Last](#)

Copyright © 2002-2015 Avaya Inc. All rights reserved.

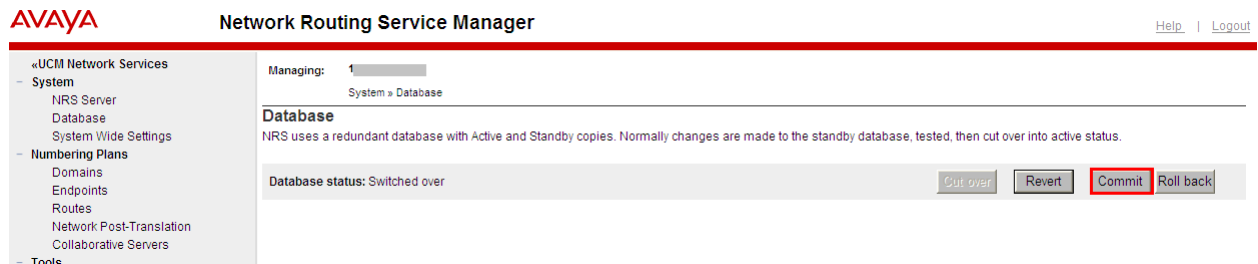
Internet | Protected Mode: Off 100%

7.4. Save New Changes on NRS

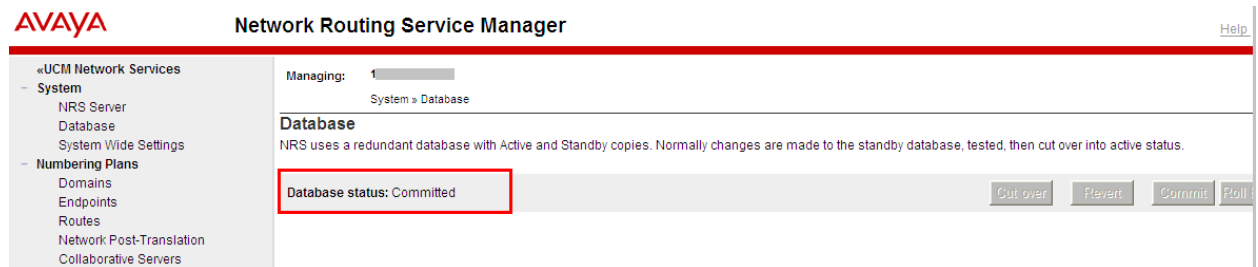
To save the new entries to the database on the NRS the database must be cut over and committed. Select **Database** under **System** in the left window and click on **Cut over** in the right window.



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

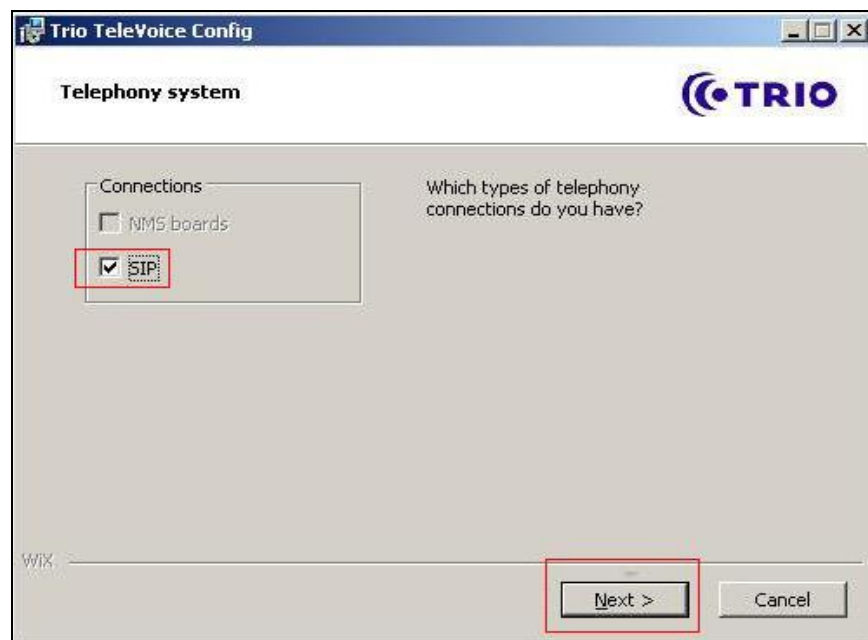
This section describes how to integrate Trio Enterprise with the CS1000E using dynamic SIP. Trio Enterprise is added to the NRS as a Dynamic SIP endpoint and calls are routed to the Trio Enterprise server according to the dialing plan setup in **Section 5.3**. This section shows how to configure Trio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

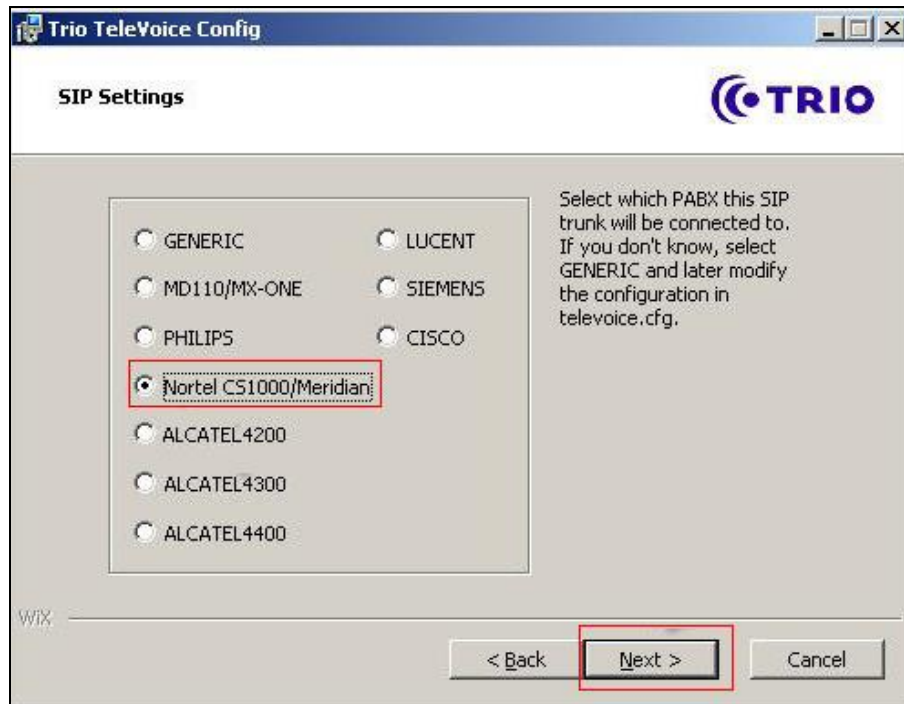
8.1. Configure Trio Enterprise to use SIP Trunks

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.

Trio Enterprise LI Config

SIP Settings(2)

SIP settings

Local IP: 10.10.97.236

Port: 5060

Target IP: 10.10.97.137

Port: 5060

Number of channels: 30

Nortel settings

Service Domain: bvwdev.com

L0 Domain: cdp

L1 Domain: udp

Endpoint name: Trio

Codecs

☒ Enable G711 mu-law codec

TE 4.1.19

< Back **Next >** Cancel

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.

The screenshot shows the 'TeleVoice Product Configuration' dialog box with the 'General' tab selected. The 'PBX' section has 'Ext. length' set to 5. The 'Default language' section has 'Default language' set to 'en'. The 'Customer group data' section has 'Group' set to 0, 'Number to operator' set to 71000, and empty fields for 'Phrase for operator-option', 'Number to AutoAttendant', 'Number to AutoAttendant, English option', and 'Beginning digits in extensions'. The 'Outgoing calls' section has an empty field for 'Prefix for outgoing calls'. The 'Attendant extensions' section has 'Attendant' set to 0 and an empty field for 'Extension'. At the bottom are 'OK', 'Cancel', and 'Apply' buttons.

Section	Field	Value
PBX	Ext. length	5
	Default language	en
Customer group data	Group	0
	Number to operator	71000
	Phrase for operator-option	
	Number to AutoAttendant	
	Number to AutoAttendant, English option	
	Beginning digits in extensions	
Outgoing calls	Prefix for outgoing calls	
Attendant extensions	Attendant	0
	Extension	

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.

The screenshot shows the InteractionStudio CC1 (Administrator) - [14] window. The left pane displays a tree view with the following structure:

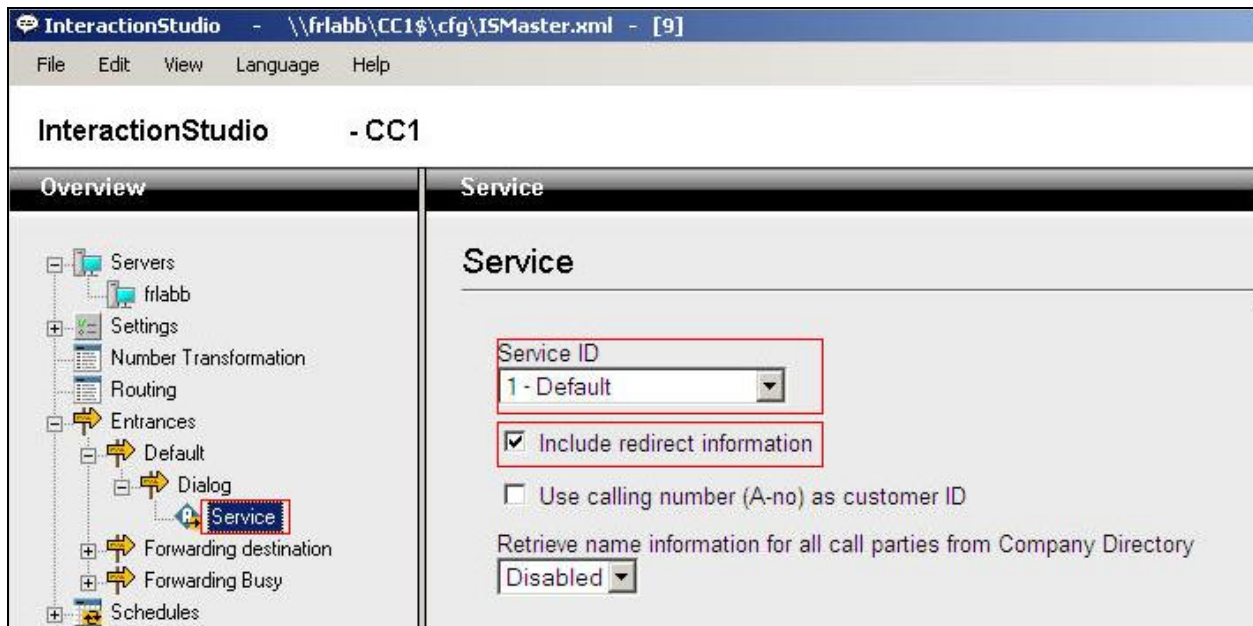
- Servers
- Settings
- Number Transformation
- Routing
 - Entrances
 - Default
 - PLAY
 - Busy
 - Dialog
 - Loop Detection via DTMF
 - Voice Functions
 - No Answer
 - Dialog
 - Loop Detection via DTMF
 - Voice Functions
 - Absent
- Schedules

The right pane shows the **Call routing table** configuration. The table has the following columns: **Field**, **Value**, **CC/Entrance**, **Language**, and **Comment**.

Field	Value	CC/Entrance	Language	Comment
C.No.	71000	Entrance - Default	English	Default range
C.No.	71001	Entrance - Busy	English	Busy
C.No.	71002	Entrance - No Answer	English	No Answer
C.No.	71003	Entrance - Absent	English	
*				

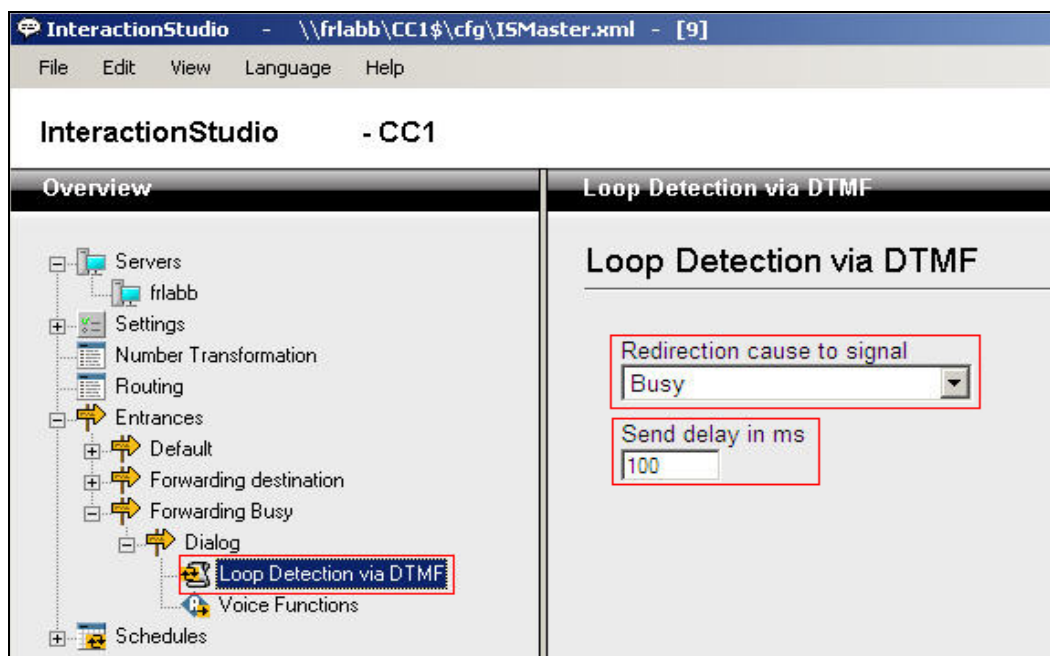
8.2.2. Configure Attendant Service

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



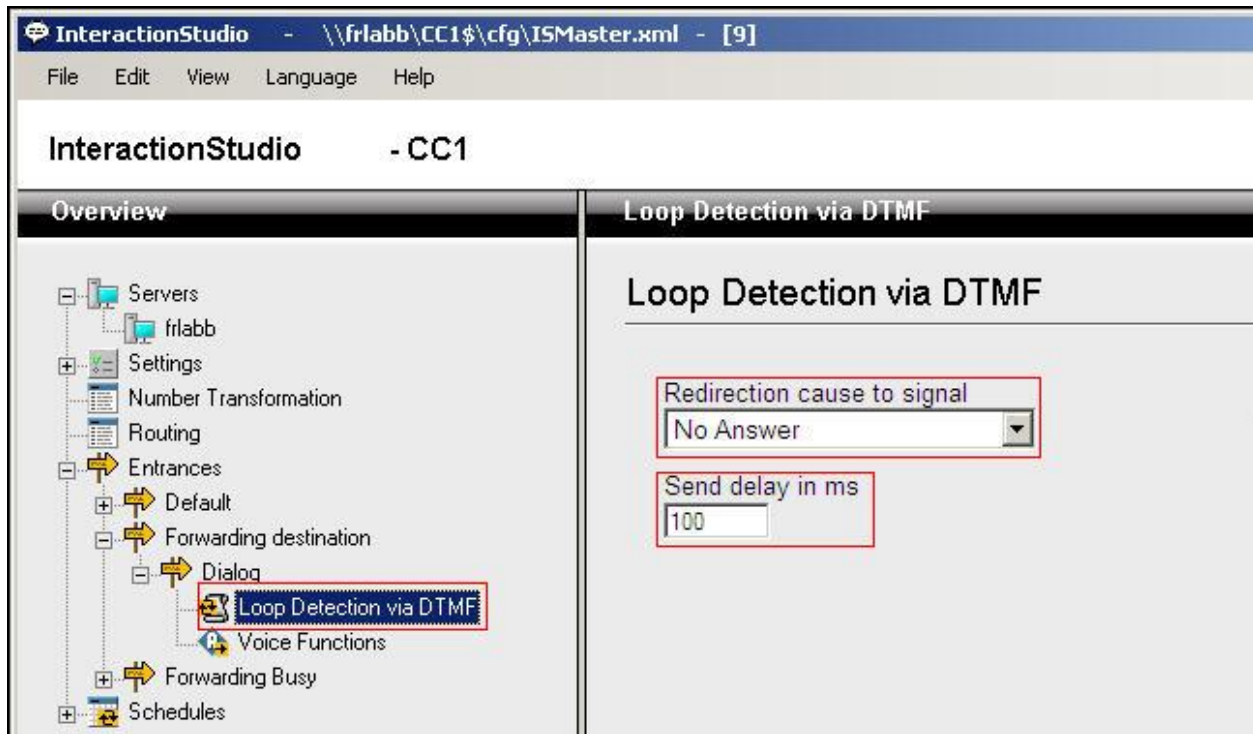
8.2.3. Configure Loop Detection via DTMF for Busy Signal

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.



Trio Agent - Login

Trio Enterprise®

User ID: default

Password:

Extension: 3032

Server: trioserver.galctlab.com

Phone type: Standard phone

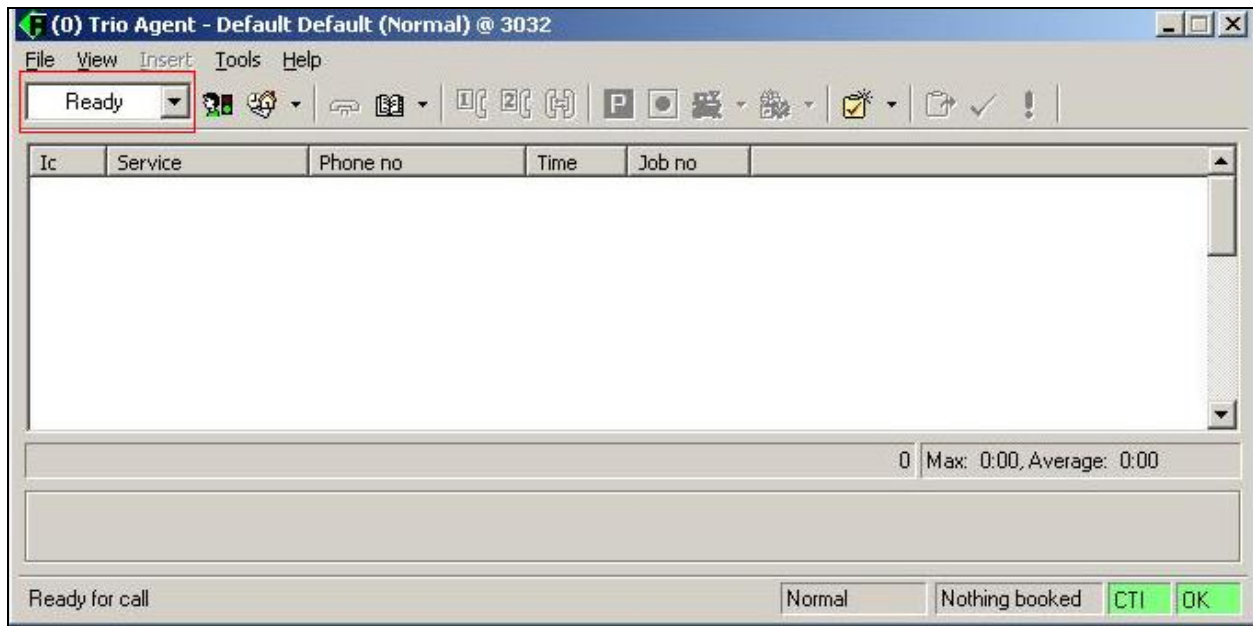
☐ Attach with Contact Center privileges

☐ Attach with Attendant privileges

OK Guest Cancel

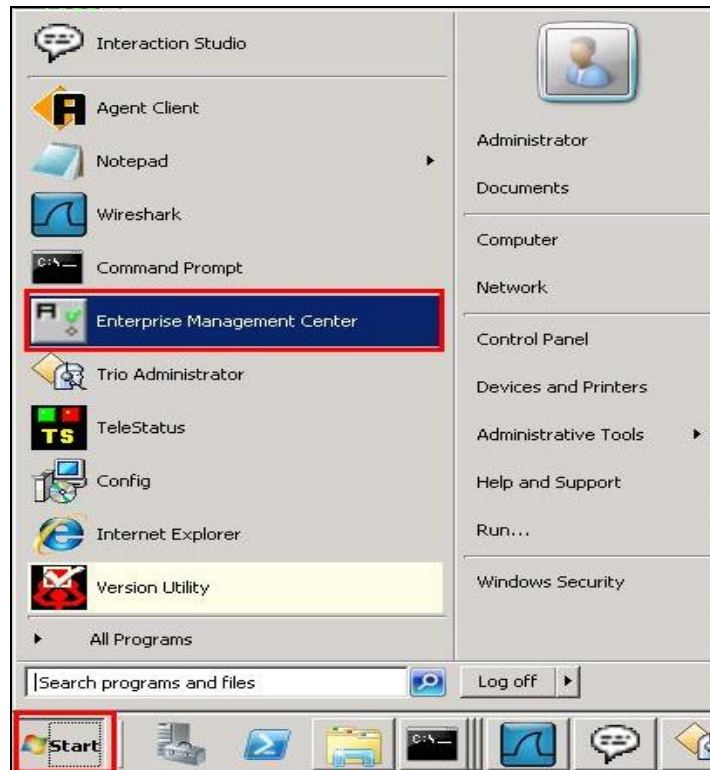
TRIO

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.

Name	Value	Comment
Enabled	true	Enable TR87 Presence Connector
localsip	31006	Listen port
PresenceDomain	10.97.138	Presence Domain name (domain...)
ServerAddr	10.97.138	TR87 server FQDN or IP Address
URI	sip:server236@10.97.138	Uri of TE enterprise Server (defa...)

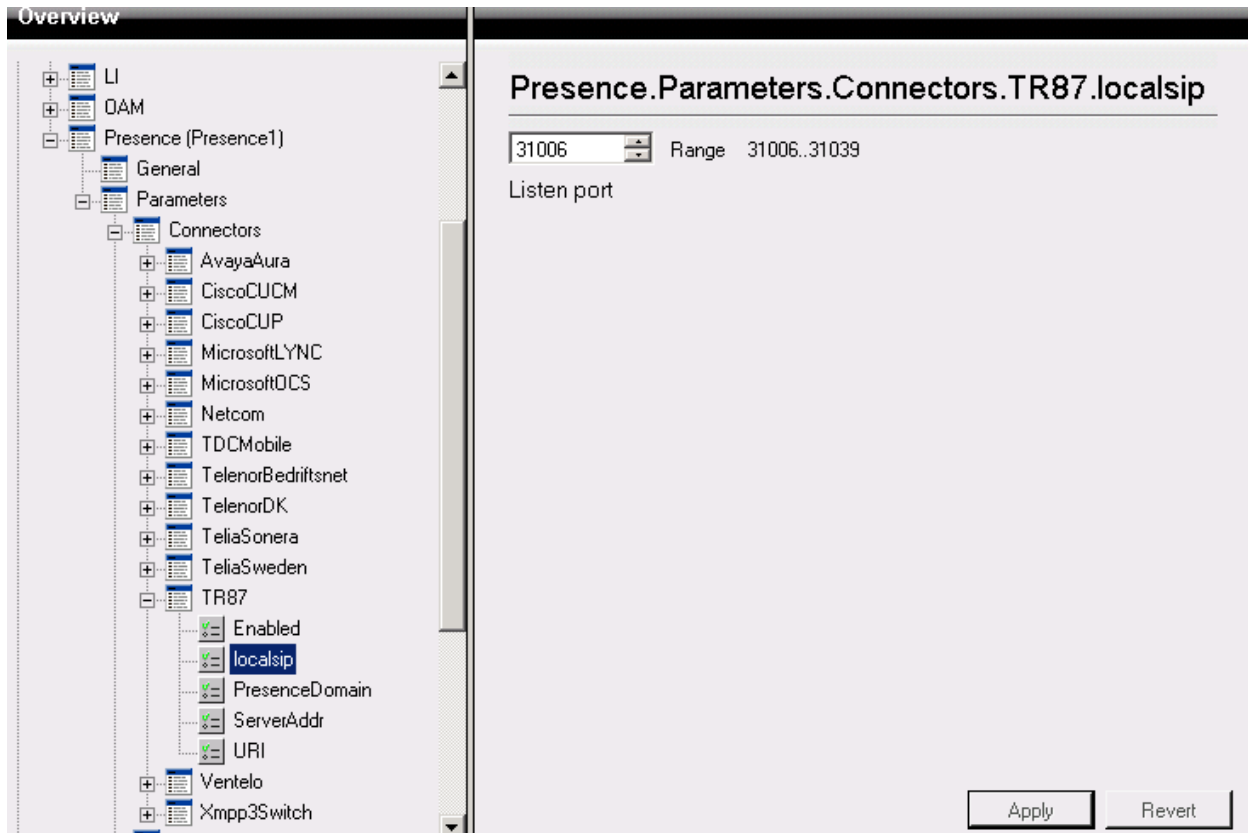
Under **TR87** select **Enabled** in the left window. Ensure that **Enable TR87 Presence Connector** is ticked as shown below. Click **Apply** to continue.

Presence.Parameters.Connectors.TR87.Enabled

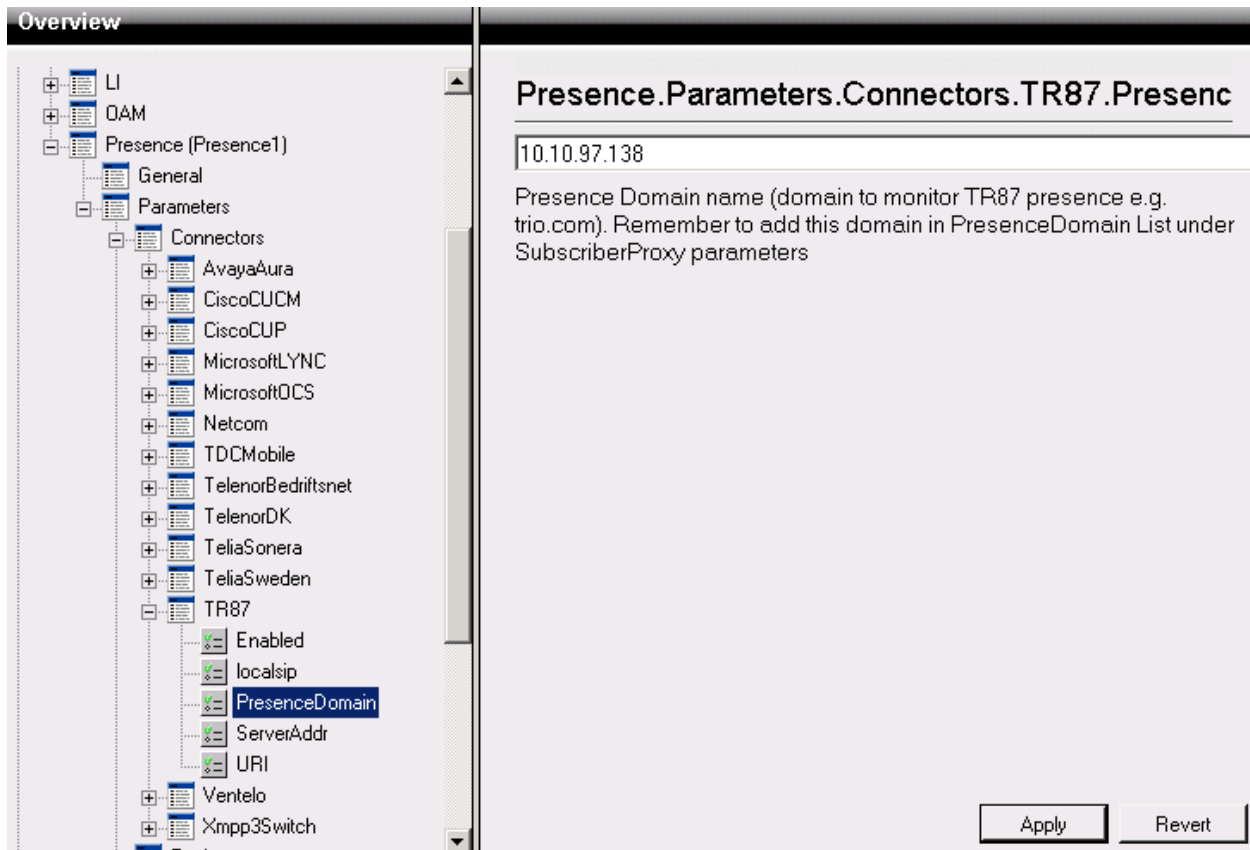
☒ Enable TR87 Presence Connector

Apply Revert

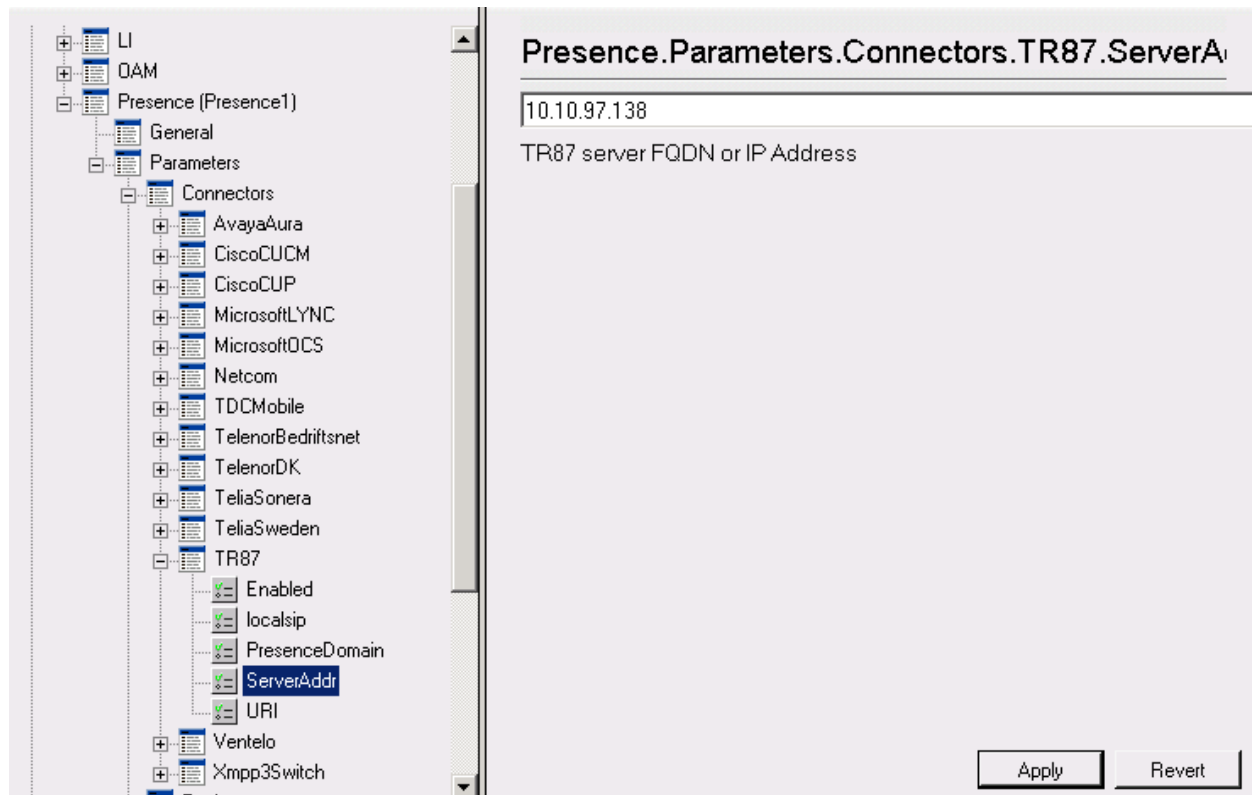
Select **localsip** under **TR87** in the left window and select the **Listen port** for TR87, for compliance testing this was left as default **31006** as shown below. Click **Apply** to continue.



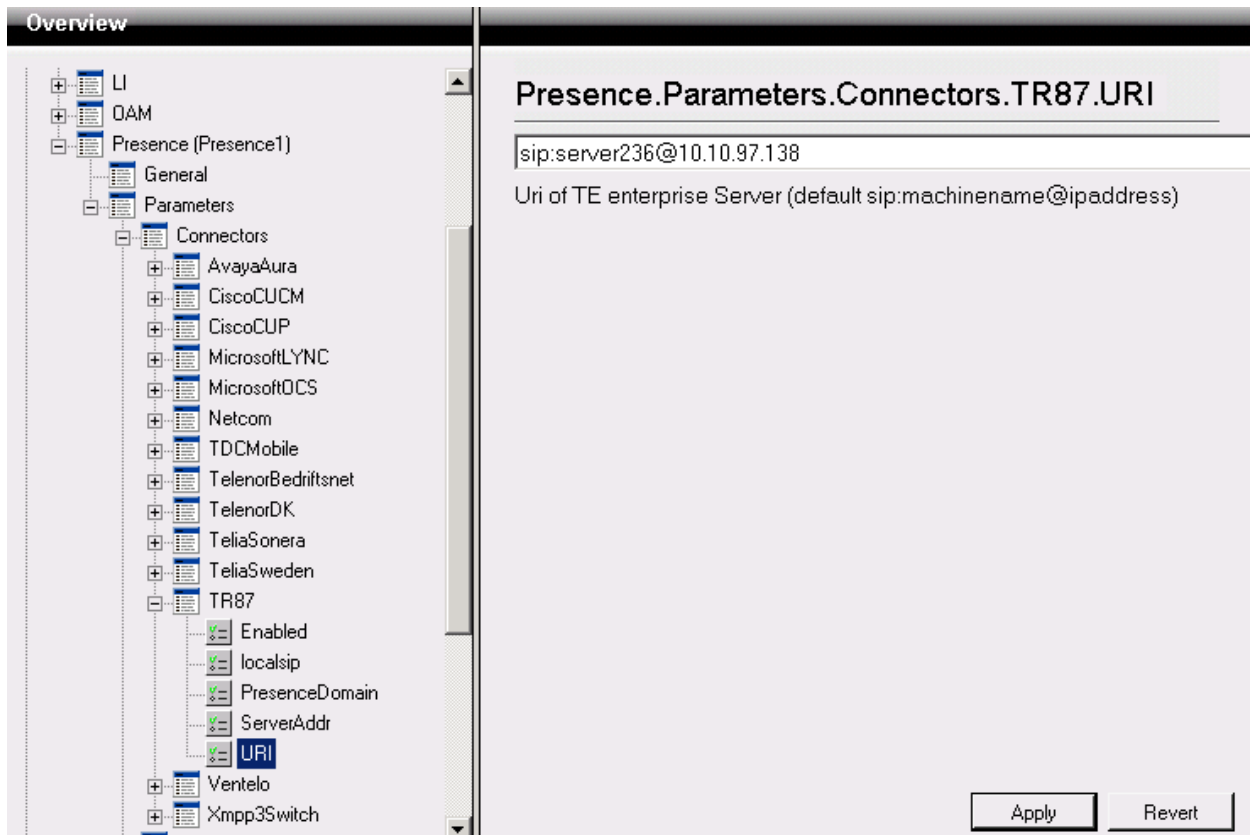
Select **PresenceDomain** under **TR87** in the left window. Enter the Node IP address of the CS1000E as per **Section 6**. Click **Apply** to continue.



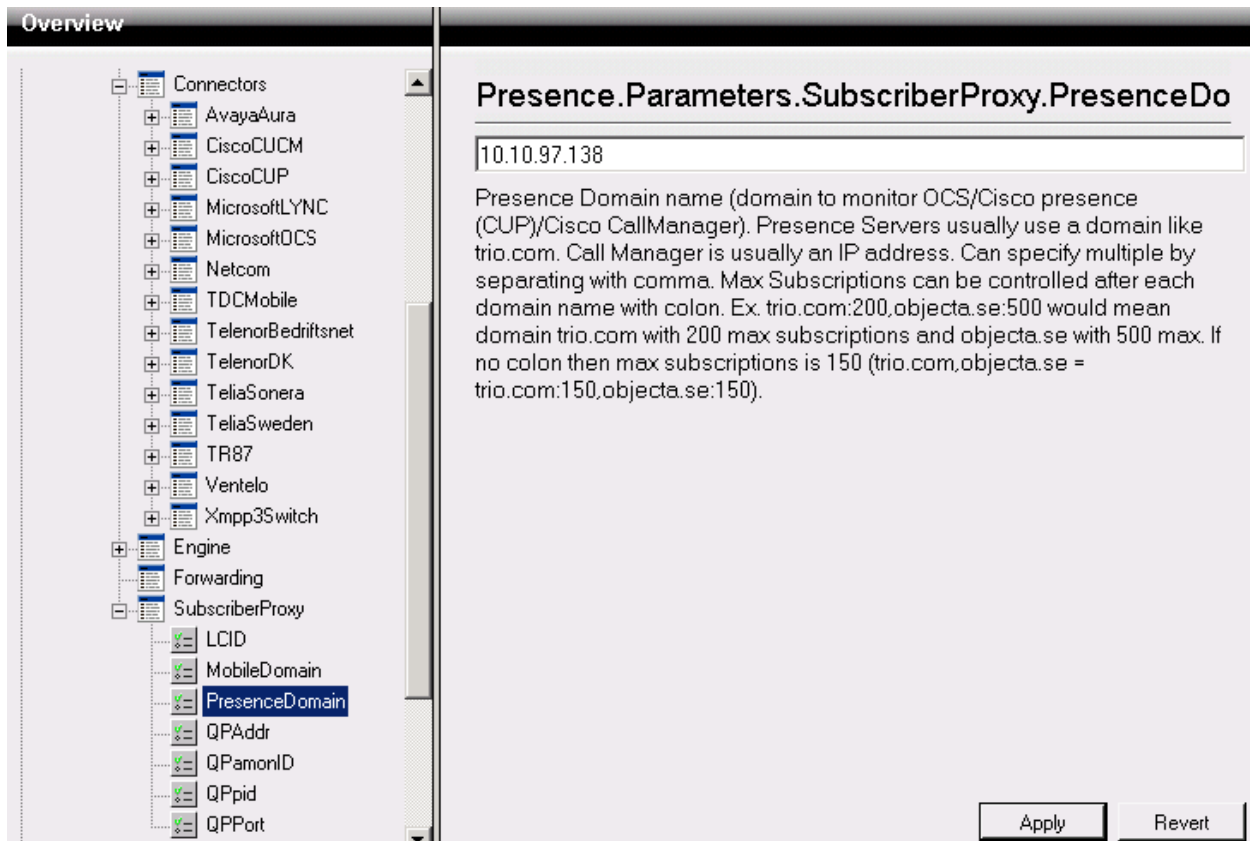
Select **ServerAddr** under **TR87** in the left window and again enter the Node IP address of the CS1000E. Click **Apply** to continue.



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

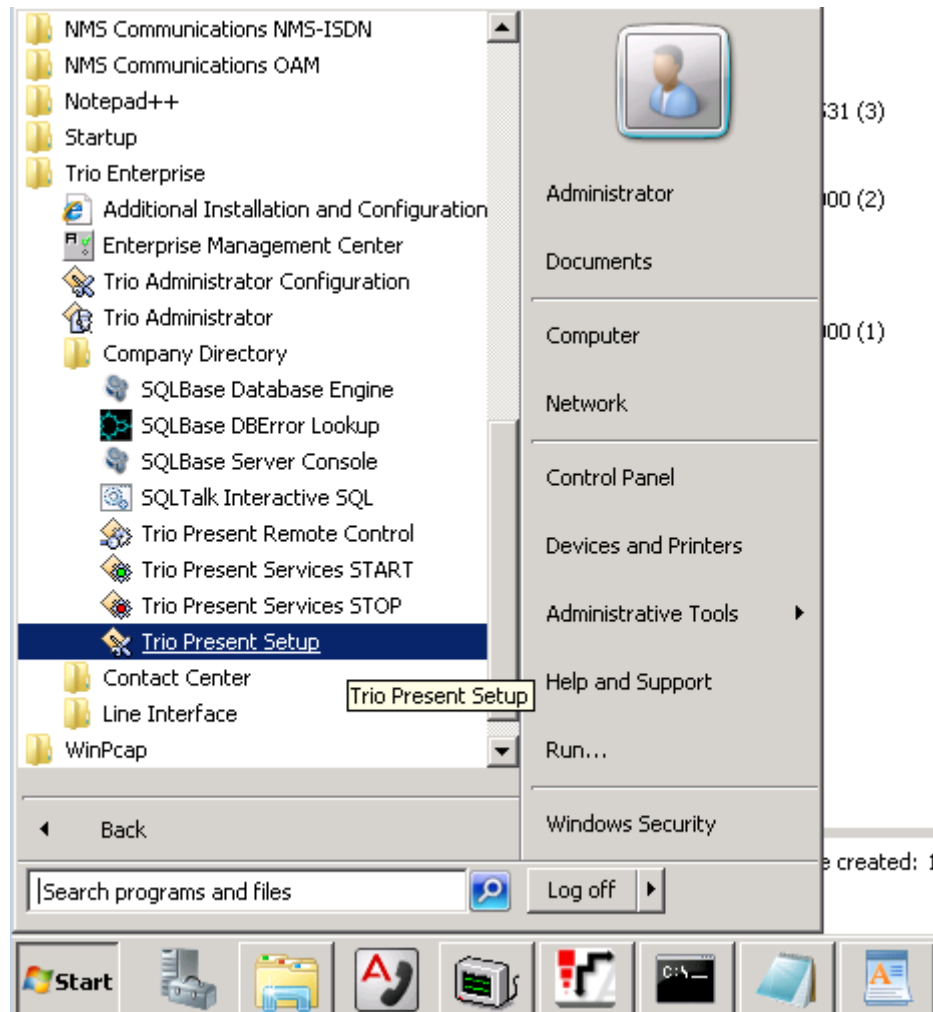


Select **PresenceDomain** under **SubscriberProxy** in the left window. Enter the Node IP address of the CS1000E in the right window. Click **Apply** to continue.

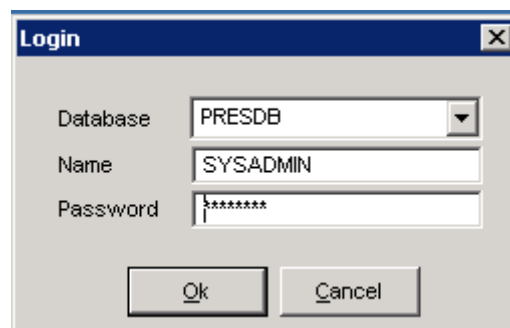


8.5. Configure ICP on Trio

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



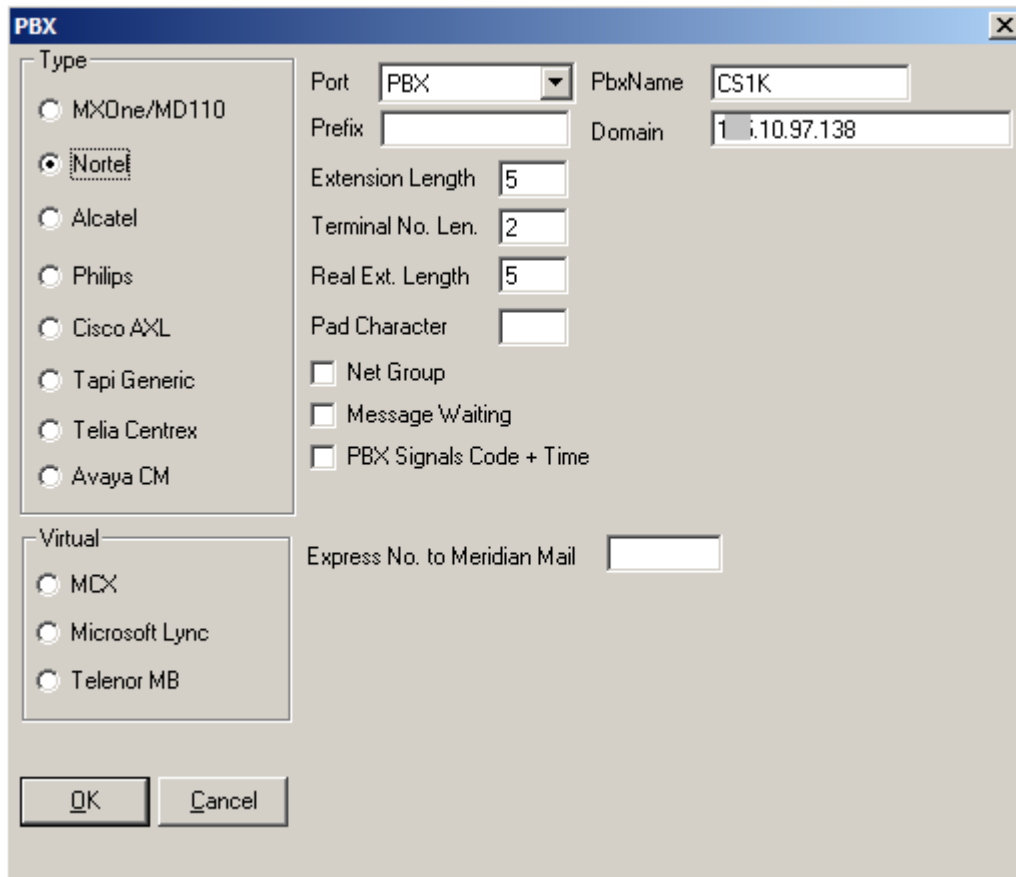
Enter the proper credentials and click **Ok**.



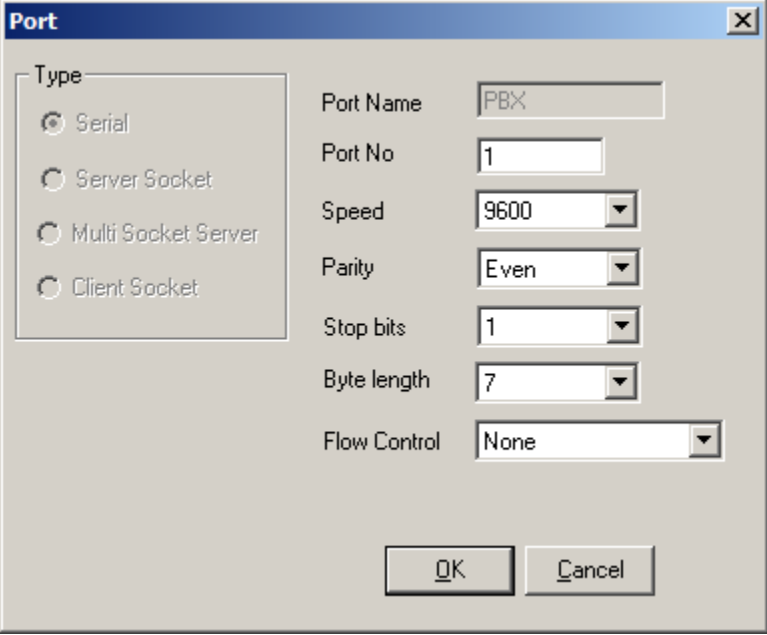
Highlight the selected PBX under the PBX tab and click on **Change** (not shown). This opens the window displayed on the right. Ensure the following are selected.

- **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.

A screenshot of a 'PBX' configuration window. The window has a title bar with 'PBX' and a close button. It is divided into several sections. On the left, there are two groups of radio buttons: 'Type' and 'Virtual'. The 'Type' group includes options like 'MXOne/MD110', 'Nortel' (which is selected), 'Alcatel', 'Philips', 'Cisco AXL', 'Tapi Generic', 'Telia Centrex', and 'Avaya CM'. The 'Virtual' group includes 'MCX', 'Microsoft Lync', and 'Telenor MB'. To the right of these groups, there are several input fields and checkboxes. The 'Port' field is a dropdown menu showing 'PBX'. The 'PbxName' field is a text box containing 'CS1K'. The 'Prefix' field is empty. The 'Domain' field is a text box containing '10.97.138'. Below these are three more text boxes: 'Extension Length' with '5', 'Terminal No. Len.' with '2', and 'Real Ext. Length' with '5'. There is also a 'Pad Character' text box. Below these are three checkboxes: 'Net Group', 'Message Waiting', and 'PBX Signals Code + Time', all of which are unchecked. At the bottom right, there is an 'Express No. to Meridian Mail' text box. At the bottom left, there are 'OK' and 'Cancel' buttons.

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.



The image shows a 'Port' configuration dialog box. On the left, under the 'Type' section, there are four radio buttons: 'Serial' (selected), 'Server Socket', 'Multi Socket Server', and 'Client Socket'. On the right, there are several input fields: 'Port Name' with the text 'PBX', 'Port No' with the value '1', 'Speed' with a dropdown menu showing '9600', 'Parity' with a dropdown menu showing 'Even', 'Stop bits' with a dropdown menu showing '1', 'Byte length' with a dropdown menu showing '7', and 'Flow Control' with a dropdown menu showing 'None'. At the bottom right, there are two buttons: 'OK' and 'Cancel'.

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>

©2015 Avaya Inc. All Rights Reserved.

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

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.