



## **Application Notes for Configuring Trio Enterprise from Enghouse Interactive AB with Avaya Communication Server 1000 and Avaya Aura® Session Manager using a SIP Trunk connection – Issue 1.0**

### **Abstract**

These Application Notes describe how to configure an Avaya Communication Server 1000 and an Avaya Aura® Session Manager to interface with Trio Enterprise, which is operating as an attendant answering position. Trio Enterprise is a software application from Enghouse Interactive AB installed on a Windows server that interfaces with Avaya Communication Server 1000 using a SIP connection via Avaya Aura® Session Manager and provides users with the call functions of an attendant console without having to install a hardware attendant position.

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 1000 R7.6 and Avaya Aura® Session Manager R7.0 with Trio Enterprise R6.2 from Enghouse Interactive AB. Trio Enterprise is a client/server based application running on Microsoft Windows 2012 Server operating systems. Trio Enterprise provides users with an attendant answering position for Avaya Communication Server 1000E that does not require attendant telephony hardware e.g., Avaya 2250 attendant console. Trio Enterprise connects to the Avaya Communication Server 1000 using a SIP connection via Avaya Aura® Session Manager.

## 2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using an Avaya Communication Server 1000E (Communication Server 1000). The Trio Enterprise server uses a SIP connection to the Communication Server 1000 call server via Session Manager. See **Figure 1** for a network diagram. A basic Distance Steering Code configuration (DSC) was configured on the Communication Server 1000 to route all calls to the Trio attendant position. If a call is made from the Trio Enterprise attendant console to the PSTN the call will route from the Trio console via a SIP trunk to Session Manager then to the PSTN. During compliance testing simulated PSTN SIP trunks were used. Trio Enterprise can perform the usual range of attendant call functions, i.e., centralized answering position; extend PSTN calls to users, place PSTN calls on behalf of internal users, perform internal telephone directory lookups.

During tests, calls are placed to a number associated with the Trio attendant position. Session Manager 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 REFER to connect caller and called user directly. It is possible to have multiple Trio attendant positions on a Communication Server 1000 system.

A variety of Avaya telephones were installed and configured on the Communication Server 1000. 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 IP phone was used as the attendant's deskphone. When the attendant is called the Trio Enterprise server calls the Avaya IP 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.

- Incoming internal and external calls
- Outgoing internal and external calls
- Supervised and unsupervised transfer with answer
- Directing calls to busy extensions
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions
- Absence detection
- Status of the phones

## 2.2. Test Results

Tests were performed to insure full interoperability between the Trio Enterprise and the Communication Server 1000. 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 for Enghouse Interactive AB 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

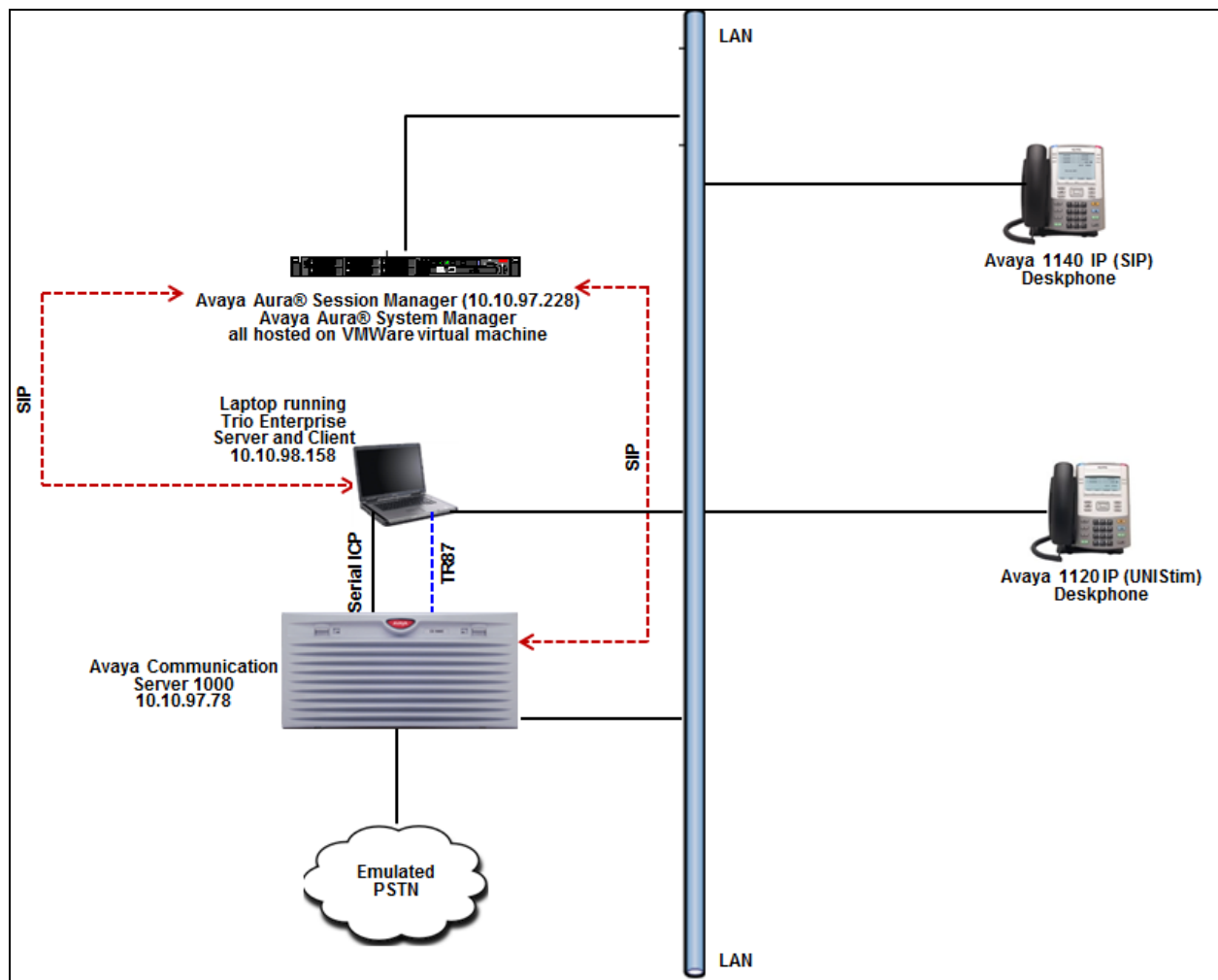
Fax: +46 (0)8 31 87 00

E-mail: [triosupport@enghouse.com](mailto:triosupport@enghouse.com)

### 3. Reference Configuration

**Figure 1** shows the network topology during compliance testing. Trio Enterprise is connected to the Communication Server 1000 using a SIP connection via Session Manager. System Manager is used to configure Session Manager. TR87 is accomplished using the SIP CTI Service and ICP is accomplished using a serial connection.

**Note:** The Trio Enterprise Attendant (client) was installed on the same server as the Trio Enterprise Server, but can be installed on a separate platform if required.



**Figure 1: Configuration for Avaya Communication Server 1000, Avaya Aura® Session Manager and Trio Enterprise**

## 4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Communication Server 1000	7.65 SP8
Avaya Aura® Session Manager running on virtualized environment	7.0.1.1.701114
Avaya Aura® System Manager running on virtualized environment	7.0.1.2 SP2
Avaya 11xx Series IP Telephone <ul style="list-style-type: none"><li>• 1120 (UNISim)</li><li>• 1140 (SIP)</li></ul>	0624C9 4.03.09
Trio Enterprise Server and Client running on Microsoft Windows 2012 R2 Server	6.2

## 5. Configure Avaya Communication Server 1000E

The configuration operations illustrated in this section were performed using terminal access to the Communication Server 1000 over an “SSH” session using “PuTTY”. The information provided in this section describes the configuration of the Communication Server 1000 for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 11**.

### Note:

- It is assumed that the SIP connection from Communication Server 1000 to Session Manager is in place and operational and will not be discussed in detail in this application notes. During compliance test, route number (**ROUT**) and route list index (**RLI**) is **6** to Session Manager, this information is needed in Section **02** to configure route to Trio number 71xxx.
- The configuration of the simulated PSTN connections is outside the scope of these Application Notes.
- Not all prompts need a response. 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 Communication Server 1000 is licensed for SIP CTI use **LD 22** and type **SLT** at the **REQ** prompt. Check for **SIP CTI TR87** and **AST** (in bold and red below).

```
>ld 22
PT2000

REQ  slt

System type is - Communication Server 1000E/CPPM Linux
CPPM - Pentium M 1.4 GHz

IPMGs Registered:          1
IPMGs Unregistered:       0
IPMGs Configured/unregistered: 0

TRADITIONAL TELEPHONES 32767    LEFT 32767    USED    0
DECT USERS              32767    LEFT 32767    USED    0
IP USERS                32767    LEFT 32682    USED   85
BASIC IP USERS          32767    LEFT 32764    USED    3
TEMPORARY IP USERS      32767    LEFT 32765    USED    2
DECT VISITOR USER      10000    LEFT 10000    USED    0
ACD AGENTS              32767    LEFT 32739    USED   28
MOBILE EXTENSIONS       32767    LEFT 32761    USED    6
TELEPHONY SERVICES     32767    LEFT 32767    USED    0
CONVERGED MOBILE USERS  32767    LEFT 32767    USED    0
AVAYA SIP LINES         32767    LEFT 32755    USED   12
THIRD PARTY SIP LINES   32767    LEFT 32740    USED   27

PCA                     32767    LEFT 32764    USED    3
ITG ISDN TRUNKS         32767    LEFT 32767    USED    0
H.323 ACCESS PORTS     32767    LEFT 32767    USED    0
AST                   32767    LEFT 32716    USED   51
SIP CONVERGED DESKTOPS  32767    LEFT 32767    USED    0
SIP CTI TR87         32767    LEFT 32734    USED   33
SIP ACCESS PORTS       32767    LEFT 32703    USED   64
```

## 5.2. Configure Coordinated Dialing Plan

This section show steps on how to create CDP to route the call from CS1000 to Trio Enterprise via Session Manager.

Use the **NEW** command in **LD 87** to create a **CDP** entry for the Trio Enterprise. In the example below, the **DSC** is **71**, **FLEN** is **5** and the **RLI** is **6**.

```
REQ  new
CUST 0
FEAT cdp
TYPE dsc
DSC 71
FLEN 5
DSP  LSC
RRPA NO
RLI 6
CCBA NO
NPA
NXX
```



### 5.3. Configure TR87 , ELAN and Value Added Server on Communication Server 1000

This section show steps on configure the Class of Service for TR87 events to be allowed from a phone and also to configure the ELAN and Value Added Server (VAS), so that these events can be passed on to Trio Enterprise.

#### 5.3.1. Configure TR87 in Class of Service

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 **T87A** 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.

```
CLS  CTD FBD WTA LPR PUA MTD FND HTD TDD HFA CRPD
      MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
      POD SLKD CCSD SWD LND CNDA
      CFTD SFD MRD DDV CNID CDCA MSID DAPA BFED RCBF
      ICDD CDMD LLCN MCTD CLBD AUTU
      GPUD DPUD DNDA CFXD ARHD CLTD ASCD
      CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
      UDI RCC HBTB AHD IPND  DDGA NAMA MIND PRSD NRWD NRCD NROD
      DRDD EXR0
      USMD USRD ULAD CCBF RTDD RBDD RBHD PGND OCBF FLXD FTTC DNDY DNO3 MCBN
      FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87A SBMD
      KEM3 MSNV FRA  PKCH MUTA MWTD DVLD CROD ELCD VMSA
CPND _LANG ENG
RCO  0
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST  00
IAPG 1
```

### 5.3.2. Configure ELAN for Trio Enterprise Application

Log in to the command line interface (CLI) of the Communication Server 1000 using the proper credentials (not shown) and issue overlay **LD 17** to create a new ELAN for the Contact Center application. Screen below shows an already configured **ELAN 34**.

```
ADAN      ELAN 34
CTYP ELAN
DES  ELAN34
N1  512
```

### 5.3.3. Configure VAS for the ELAN of Trio Enterprise Application

Using the CLI, issue overlay **LD 17** to create a value added server (VAS) for the ELAN 34 that was configured above for the Contact Center application. Screen below shows an already configured **VSID 34**.

```
VSID 034
ELAN 034
SECU YES
INTL 0001
MCNT 9999
```

## 5.4. Configure Intercept Computer Update on Communication Server 1000

Trio uses Intercept Computer Update (ICP) on the Communication Server 1000 to change the presence state of the phoneset. A physical port on the Communication Server 1000 must be configured for ICP along with the ICP configuration in the Customer Data Block.

### 5.4.1. 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 0 connected to MGC card 4 0. Type **LD 17** at the > prompt to enter overlay 17. During compliance testing **TTY 13** was configured for ICP.

```
ADAN      TTY 13
CTYP MGC
IPMG 4 0
DNUM 13
PORT 0
DES  icp2
BPS 9600
BITL 7
STOP 1
PARY EVEN
FLOW NO
USER ICP
```

### 5.4.2. 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. **APL** is the auxiliary processor link used and in this case it is **13** as configured in **Section 5.4.1**. **IDCN** and **ECDN** are the numbers used for diversion when attendant is absent. During compliance testing **71003** was used. **ICDL** is the DN length and during compliance testing **5** was used.

```
TYPE: icp
TYPE ICP_DATA
CUST 0

TYPE ICP_DATA
CUST 00
ICP YES
  APL 13
  NIPN 9
  ICCR NO
  ICMM 0
  IDCN 71003
  ECDN 71003
  ICWN 0
  ICPS CIR
  ICDL 5
  ICPD 0
  ICTD YES
```

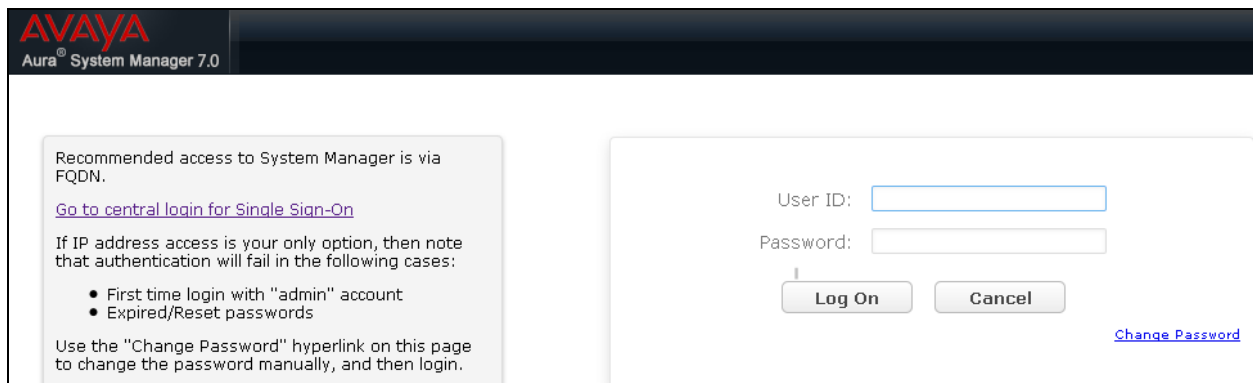
### 5.4.3. Configuration of Flexible Feature Code for ICP Activate and Deactivate

Enter Overlay 57 to add a code to Intercept Computer Interface Activate (ICPA) and Intercept Computer Interface Deactivate (ICPD) by typing **LD 57** at the > prompt and follow the instructions as shown below to configure them. These codes are required when setting the attendant as absent using the phones. During compliance testing **\*23** and **\*24** were used for activate and deactivate respectively.

```
REQ prt
TYPE ffc
CUST 0
CODE icpa
ICPA *23
CODE icpd
ICPD *24
CODE
```

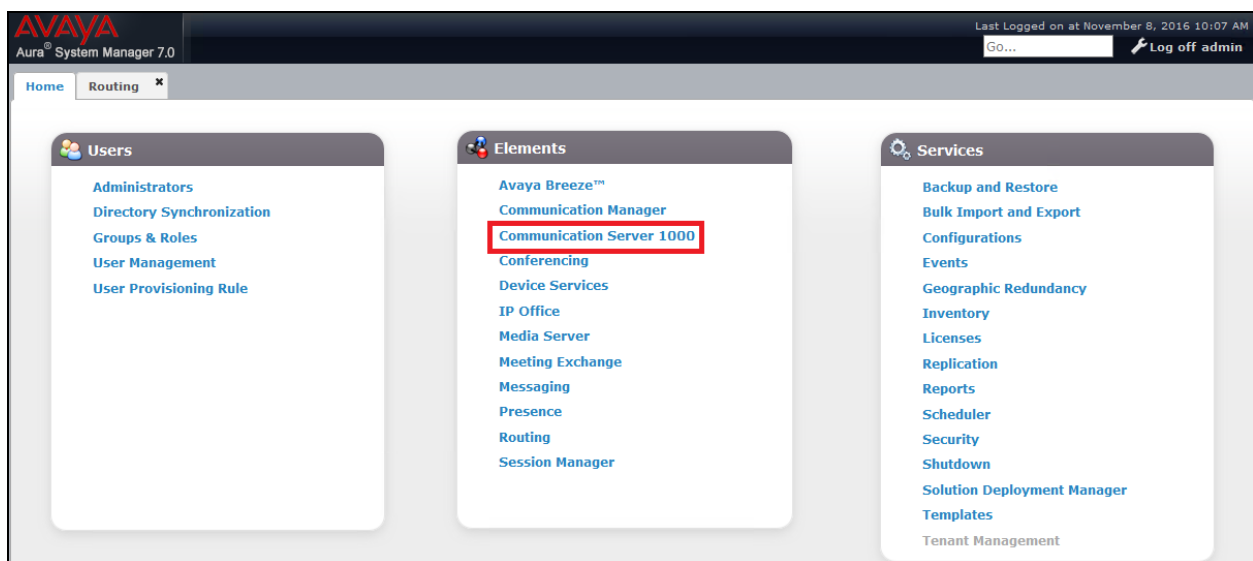
## 6. Configure Avaya Communication Server 1000 Signalling Server for TR87 events

SIP CTI (TR87) services must be enabled and configured on the Communication Server 1000 IP Telephony Node to allow applications obtain presence information or invoke a make-call operation. Changes on the Communication Server 1000 Node are performed using Element Manager which is accessible through the System Manager. To make changes in Element Manager, access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



The image shows the login page of the Avaya Aura System Manager 7.0. The header includes the Avaya logo and the text "Aura® System Manager 7.0". The main content area is divided into two sections. The left section contains a message: "Recommended access to System Manager is via FQDN. Go to central login for Single Sign-On". Below this, it states: "If IP address access is your only option, then note that authentication will fail in the following cases:" followed by a bulleted list: "• First time login with 'admin' account" and "• Expired/Reset passwords". It also mentions: "Use the 'Change Password' hyperlink on this page to change the password manually, and then login." The right section contains a login form with fields for "User ID:" and "Password:", and buttons for "Log On" and "Cancel". A "Change Password" link is located at the bottom right of the login form.

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 cppm3** link.

AVAYA  
Aura System Manager 7.0

Last Logged on at November 8, 2016 10:07 AM  
Go... Log off admin

Home Routing **Communication Server 1000**

Network  
Elements  
CS 1000 Services  
Corporate Directory  
IPSec  
Numbering Groups  
Patches  
SNMP Profiles  
Secure FTP Token  
Software Deployment  
User Services  
Administrative Users  
External Authentication  
SAML Configuration  
Password  
Security  
Roles  
Policies  
Active Sessions  
Tools

Host Name: dewmsmgr.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

<input type="checkbox"/>	Element Name	Element Type	Release	Address	Description
<input type="checkbox"/>	dewmsmgr.bvwdev.com (primary)	Base OS	7.6		Base OS element
<input checked="" type="checkbox"/>	EM on cppm3	CS1000	7.6		New element
<input type="checkbox"/>	cppm3.bvwdev.com (member)	Linux Base	7.6		Base OS element
<input type="checkbox"/>		Media Gateway Controller	7.6		New element

Click on **IP Network** → **Nodes: Servers, Media Cards** in the left window. Click on the **Node ID** displayed in the right window, during compliance test Node **510** is configured to connect to Session Manager. Note the IP address of this node as it used while configuring Communication Server 1000 as SIP Entity endpoint on Session Manager in **Section 7.5.2**. Trio Enterprise also gets TR87 events via Node **510** and hence this IP address will also be used in **Section Error! Reference source not found.4** and in **Section 8.5** to configure ICP

AVAYA  
CS1000 Element Manager

Managing: 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

<input type="checkbox"/>	Node ID	Components	Enabled Applications	ELAN IP	Node/TLAN IPv4	Node/TLAN IPv6	Status
<input checked="" type="checkbox"/>	510	1	SIP Line, LTPS, PD, Gateway (SIPGw)		10.10.97.149	-	Synchronized

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

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

**AVAYA CS1000 Element Manager**

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

**Node Details (ID: 510 - SIP Line, LTPS, PD, Gateway ( SIPGw ))**

Node ID:  \* (0-9999)

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

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

Gateway IP address:  \* Node IPv4 address:  \*

Subnet mask:  \* Subnet mask:  \*

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 for the field **SIP CTI Service**, the **Enable CTI service** box is checked as shown below and uncheck the **TLS endpoints only** (if this is ticked) box; retain default values for all other fields. Click on **Save** once finished.

**AVAYA CS1000 Element Manager**

Managing:  Username: admin  
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

**Node ID: 510 - Virtual Trunk Gateway Configuration Details**

General | SIP Gateway Settings | SIP Gateway Services

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

**CTI settings**

Customer number:

Maximum associations per DN:

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

**Dial plan prefixes**

National:

International:

Location code call:

Special number:

Subscriber:

**CTI CLID presentation**

Dialing plan:

Calling device URI format:

Home location code:

Country code (CCC):

Area code:  NPA in North America

\* Required Value. Note: Changes made on this page will NOT be transmitted until the Node is also saved. Save Cancel

Save and transmit (not shown) these Node properties to complete the SIPGw configuration. Once the components are synchronized the Signalling Gateway will require a restart.

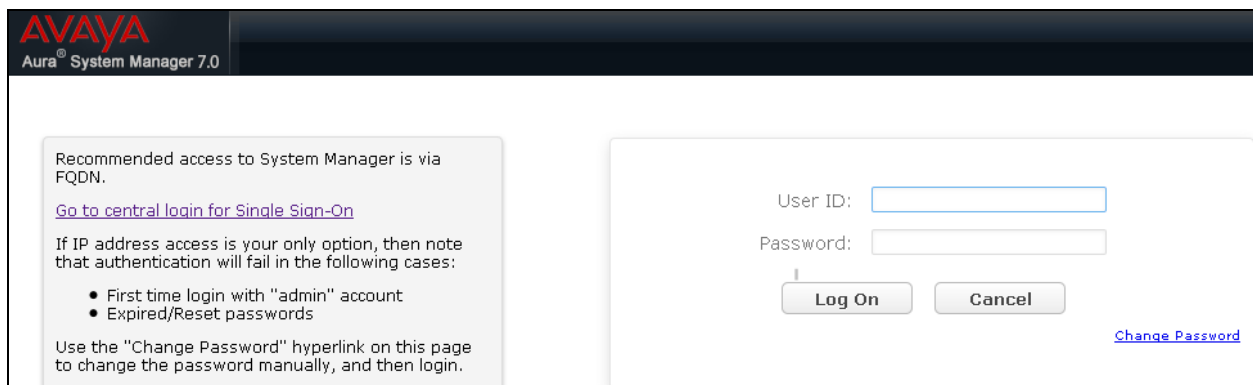
## 7. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include the following areas:

- Launch System Manager
- Administer Domain
- Administer locations
- Administer Adaptation
- Administer SIP entities
- Administer routing policies
- Administer dial patterns

### 7.1. Launch System Manager

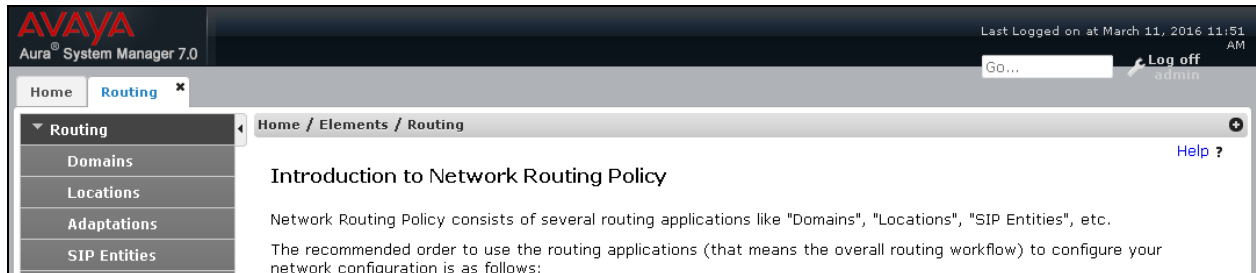
Access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of System Manager. Log in using the appropriate credentials.



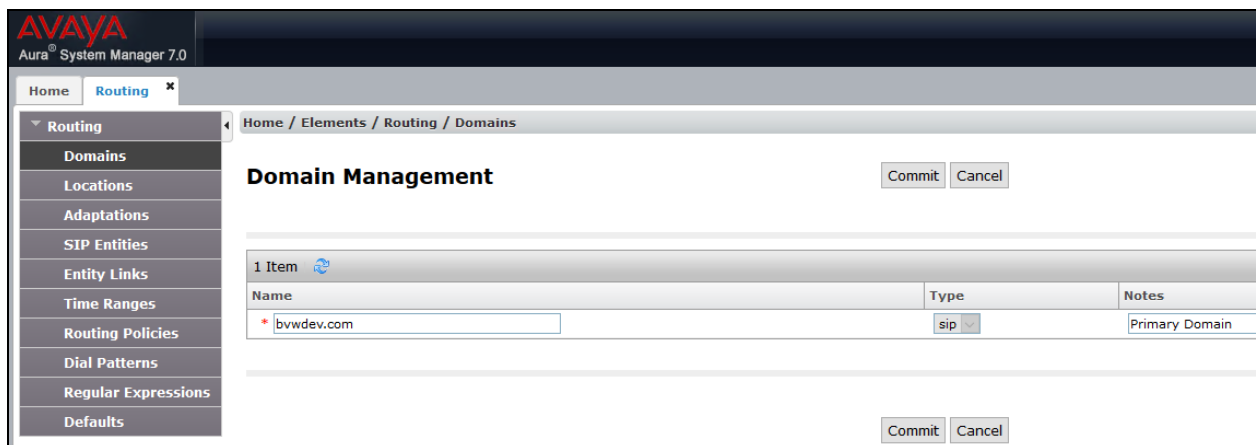
The screenshot shows the Avaya Aura System Manager 7.0 login interface. The header features the Avaya logo and the text "Aura® System Manager 7.0". The main content area is divided into two sections. The left section contains instructions: "Recommended access to System Manager is via FQDN." followed by a link "Go to central login for Single Sign-On". Below this, it states "If IP address access is your only option, then note that authentication will fail in the following cases:" and lists two bullet points: "• First time login with 'admin' account" and "• Expired/Reset passwords". It also mentions "Use the 'Change Password' hyperlink on this page to change the password manually, and then login." The right section contains the login form with fields for "User ID:" and "Password:", a "Log On" button, a "Cancel" button, and a "Change Password" link.

## 7.2. Administer Domain

In the subsequent screen (not shown), select **Elements** → **Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing** → **Domains** from the left pane, and click **New** in the subsequent screen (not shown) to add a new domain



The **Domain Management** screen is displayed. In the **Name** field enter the domain name, select *sip* from the **Type** drop down menu and provide any optional **Notes**.





### 7.3. Administer Locations

Select **Routing** → **Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for Trio Enterprise.

The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. Retain the default values in the remaining fields.

AVAYA  
Aura® System Manager 7.0

Home Routing

Home / Elements / Routing / Locations

### Location Details

Commit Cancel

**General**

\* **Name:** Belleville

**Notes:** Belleville DevConnect Lab

**Dial Plan Transparency in Survivable Mode**

Enabled: ☐

Scroll down to the **Location Pattern** sub-section, click **Add** and enter the IP address of all devices involved in the compliance testing in **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

Location Pattern

Add Remove

4 Items Filter: Enable

<input type="checkbox"/>	IP Address Pattern	Notes
<input type="checkbox"/>	* 10.10.5.*	
<input type="checkbox"/>	* 10.10.97.*	
<input type="checkbox"/>	* 10.10.98.*	
<input type="checkbox"/>	*	

Select : All, None

Commit Cancel

## 7.4. Administer Adaptation

During compliance test the simulated PSTN using SIP trunk was between Communication Server 1000 and Avaya Aura® Communication Manager via Session Manager. In order to make the call from and to Communication Manager via Session Manager, Adaptation to translate IP address into domain name is used for Trio SIP entity. Another adaptation was used for Communication Server 1000 to change the phone context information. Here is step on how to create Adaptation. Select **Adaptations** on the left panel menu and then click on the **New** button in the main window (not shown).

### 7.4.1. Adaptation for Trio SIP Entity

Enter the following for the Trio Adaptation.

- **Adaptation Name:** An informative name (e.g., **change IP to Domain Trio**)
- **Module Name:** Select “DigitConversionAdapter”
- **Module Parameter Type:** Select “Name-Value Parameter”

Click **Add** to add a new row for the following values as shown below table:

Name	Value
fromto	true
iodstd	Enter the domain name of system, ex: <b>bvwwdev.com</b>
iosrcd	Enter the domain name of system, ex: <b>bvwwdev.com</b>
odstd	Enter IP address of Trio, ex: <b>10.10.98.158</b>
osrcd	Enter IP Address of Session Manager, ex: <b>10.10.97.228</b>

Once the correct information is entered click the **Commit** button. Here the screenshot shows Adaptation created for Trio.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options: Routing, Domains, Locations, Adaptations (selected), SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Adaptation Details' and includes a 'General' tab. The 'Adaptation Name' is 'For\_Trio', the 'Module Name' is 'DigitConversionAdapter', and the 'Module Parameter Type' is 'Name-Value Parameter'. Below this is a table with columns 'Name' and 'Value'. The table contains three rows: 'fromto' with value 'true', 'iodstd' with value 'bvwddev.com', and 'iosrcd' with value 'bvwddev.com'. At the bottom of the table, it says 'Select : All, None' and 'Page 1 of 2'.

Name	Value
fromto	true
iodstd	bvwddev.com
iosrcd	bvwddev.com

(Continue) the screenshot show Adaptation created for Trio:

The screenshot shows the same Avaya Aura System Manager 7.0 interface as the previous one, but on page 2 of the table. The 'Adaptation Name' is 'For\_Trio', the 'Module Name' is 'DigitConversionAdapter', and the 'Module Parameter Type' is 'Name-Value Parameter'. The table contains two rows: 'iodstd' with value '10.10.98.158' and 'iosrcd' with value '10.10.97.228'. At the bottom of the table, it says 'Select : All, None' and 'Page 2 of 2'.

Name	Value
iodstd	10.10.98.158
iosrcd	10.10.97.228

## 7.4.2. Adaptation for Communication Server 1000 SIP Entity

Enter the following for the Communication Server 1000 Adaptation.

- **Adaptation Name:** An informative name (e.g., **CS1000Adapter**)
- **Module Name:** Select “CS1000Adapter”
- **Module Parameter Type:** Select “Name-Value Parameter”

Click **Add** to add a new row for the following values as shown below table:

Name	Value
fromto	true

Once the correct information is entered click the **Commit** button. Here the screenshot shows Adaptation created for Communication Server 1000.

The screenshot displays the Avaya Aura System Manager 7.0 interface. The left sidebar shows a navigation menu with 'Routing' selected. The main content area is titled 'Adaptation Details' and includes a 'General' tab. The 'Adaptation Name' is set to 'CS1000Adapter', the 'Module Name' is 'CS1000Adapter', and the 'Module Parameter Type' is 'Name-Value Parameter'. A table below these fields shows a single parameter: 'fromto' with a value of 'true'. The 'Commit' and 'Cancel' buttons are visible at the top right. The 'Notes' field at the bottom contains the text 'CS1000 adapter for Phone Context'.

Name	Value
fromto	true

## 7.5. Administer SIP Entities

Add two new SIP entities, one for Trio Enterprise and one for the new SIP trunks with Communication Server 1000.

### 7.5.1. SIP Entity for Trio Enterprise

Select **Routing** → **SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Trio Enterprise.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of the Trio Enterprise server.
- **Type:** “Other”
- **Notes:** Any desired notes.
- **Adaptation:** Select the adaptation configured in **Section 7.4.1**
- **Location:** Select the Trio Enterprise location name from **Section 7.3**.
- **Time Zone:** Select the applicable time zone.

The screenshot shows the AVAYA Aura System Manager 7.0 interface. The left navigation pane has 'SIP Entities' selected under the 'Routing' category. The main content area is titled 'SIP Entity Details' and contains the following fields:

- Name:** TrioATT
- FQDN or IP Address:** 10.10.98.158
- Type:** Other
- Notes:** SIP Entity for Trio by Enghouse
- Adaptation:** For\_Trio
- Location:** Belleville
- Time Zone:** America/Fortaleza
- SIP Timer B/F (in seconds):** 4
- Credential name:** (empty field)
- Securable:** ☐
- Call Detail Recording:** none
- CommProfile Type Preference:** (empty dropdown)
- Loop Detection Mode:** On
- Loop Count Threshold:** 5
- Loop Detection Interval (in msec):** 200
- SIP Link Monitoring:** Use Session Manager Configuration

Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DevvmSM”.
- **Protocol:** “TCP”
- **Port:** “5060”
- **SIP Entity 2:** The Trio Enterprise entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

Note that only TCP protocol was tested.

**Entity Links**

Override Port & Transport with DNS SRV: ☐

Add
Remove

1 Item

Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* DevvmSM_TrioATT_5060	DevvmSM	TCP	* 5060	TrioATT	* 5060	trusted	<input type="checkbox"/>

Select : All, None

## 7.5.2. SIP Entity for Communication Server 1000

Select **Routing** → **SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Communication Server 1000. Note that this SIP entity is used for integration with Trio Enterprise.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of an existing Communication Server 1000 node IP.
- **Type:** “Other”
- **Notes:** Any desired notes.
- **Adaptation:** Select the adaptation configured in **Section 7.4.2**
- **Location:** Select the applicable location for Communication Server 1000.
- **Time Zone:** Select the applicable time zone.

**AVAYA**  
Aura® System Manager 7.0

Home Routing x

Home / Elements / Routing / SIP Entities

### SIP Entity Details

Commit Cancel

**General**

\* Name: CS1K\_Bottom

\* FQDN or IP Address: 10.10.97.149

Type: Other

Notes: SIP connection to CS1K

Adaptation: CS1000Adapter

Location: Belleville

Time Zone: America/Toronto

\* SIP Timer B/F (in seconds): 4

Credential name:

Securable: ☐

Call Detail Recording: none

CommProfile Type Preference:

**Loop Detection**

Loop Detection Mode: On

Loop Count Threshold: 5

Loop Detection Interval (in msec): 200

**SIP Link Monitoring**

SIP Link Monitoring: Use Session Manager Configuration

Scroll down to the **Entity Links** sub-section, and click **Add** to add an entity link. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case “DevvmSM”.
- **Protocol:** “UDP”
- **Port:** “5060”
- **SIP Entity 2:** The Communication Server 1000 entity name from this section.
- **Port:** “5060”
- **Connection Policy:** “trusted”

**Entity Links**

Override Port & Transport with DNS SRV: ☐

Add
Remove

1 Item
Filter: Enable

<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Deny New Service
<input type="checkbox"/>	* LinktoCS1K_Bottom	DevvmSM	UDP	* 5060	CS1K_Bottom	* 5060	trusted	<input type="checkbox"/>

Select : All, None



## 7.6. Administer Routing Policies

Add two new routing policies, one for Trio Enterprise and one for the new SIP trunks with Communication Server 1000.

### 7.6.1. Routing Policy for Trio Enterprise

Select **Routing** → **Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Trio Enterprise.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Trio Enterprise entity name from **Section 7.5.1**. The screen below shows the result of the selection.

**Routing Policy Details** [Commit] [Cancel]

**General**

\* **Name:**

**Disabled:** ☐

\* **Retries:**

**Notes:**

**SIP Entity as Destination**

Name	FQDN or IP Address	Type	Notes
TrioATT	10.10.98.158	Other	SIP Entity for Trio by Enghouse

## 7.6.2. Routing Policy for Communication Server 1000

Select **Routing** → **Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Communication Server 1000.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**, and retain the default values in the remaining fields.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Communication Server 1000 entity name from **Section 7.5.2**. The screen below shows the result of the selection.

AVAYA  
Aura® System Manager 7.0

Last Logged on at November 8, 2016 10:07 AM

Go... Log off admin

Home Routing

Home / Elements / Routing / Routing Policies

### Routing Policy Details

Commit Cancel

#### General

\* Name:

Disabled: ☐

\* Retries:

Notes:

#### SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
CS1K_Bottom	10.10.97.149	Other	SIP connection to CS1K

## 7.7. Administer Dial Patterns

Add a new dial pattern for Trio Enterprise, and update existing dial patterns for Communication Server 1000.

### 7.7.1. Dial Pattern for Trio Enterprise

Select **Routing** → **Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Trio Enterprise. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case “71”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The domain name from **Section 7.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Trio Enterprise. In the compliance testing, the entry allowed for call originations from all Communication Server 1000 endpoints in locations “Belleville”. The Trio Enterprise routing policy from **Section 7.6.1** was selected as shown below.

**AVAYA**  
Aura® System Manager 7.0

Last Logged on at October 7, 2016 11:29  
GO... Log off adm

Home Routing

Home / Elements / Routing / Dial Patterns

### Dial Pattern Details

Commit Cancel

**General**

\* Pattern: 71

\* Min: 5

\* Max: 36

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: bvwddev.com

Notes: Dialing pattern to reach Trio Server

**Originating Locations and Routing Policies**

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name ^	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville	Belleville DevConnect Lab	Route_To_Trio	0	<input type="checkbox"/>	TrioATT	Routing to Trio Server

Select : All, None

## 7.7.2. Dial Pattern for Communication Server 1000

Select **Routing** → **Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Communication Server 1000. The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match, in this case “54”.
- **Min:** The minimum number of digits to match.
- **Max:** The maximum number of digits to match.
- **SIP Domain:** The domain name from **Section 7.2**.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create an entry for reaching Communication Server 1000. In the compliance testing, the entry allowed for call originations from all Trio Enterprise endpoints in locations “Belleville”. The Communication Server routing policy from **Section 7.6.2** was selected as shown below.

Follow the procedures in this section to make similar changes to the applicable Communication Server 1000 dial pattern to reach the PSTN (not shown).

**AVAYA**  
Aura® System Manager 7.0

Last Logged on at November 8, 2016 10:07 AM

Home / Elements / Routing / Dial Patterns

### Dial Pattern Details

Commit Cancel Help ?

**General**

\* Pattern: 54

\* Min: 5

\* Max: 36

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: bvwddev.com

Notes: Dial pattern to CS1K

**Originating Locations and Routing Policies**

Add Remove

1 Item Filter: Enable

<input type="checkbox"/>	Originating Location Name ^	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville	Belleville DevConnect Lab	Route_to_CS1K_Bottom	0	<input type="checkbox"/>	CS1K_Bottom	

Select : All, None

## 8. Configure TRIO Enterprise

Trio Enterprise connects as a SIP endpoint to the Communication Server 1000 through Session Manager. Trio Enterprise is added to Session Manager as a SIP Entity and calls are routed to the Trio Enterprise server according to the Coordinated Dial Plan setup in **Section 5.2**. This section shows how to configure Trio Enterprise to successfully connect to the Communication Server 1000 using SIP trunks. The installation of the Trio Enterprise software is assumed to be completed and the Trio services are up and running. The steps to configure SIP Trunks are as follows.

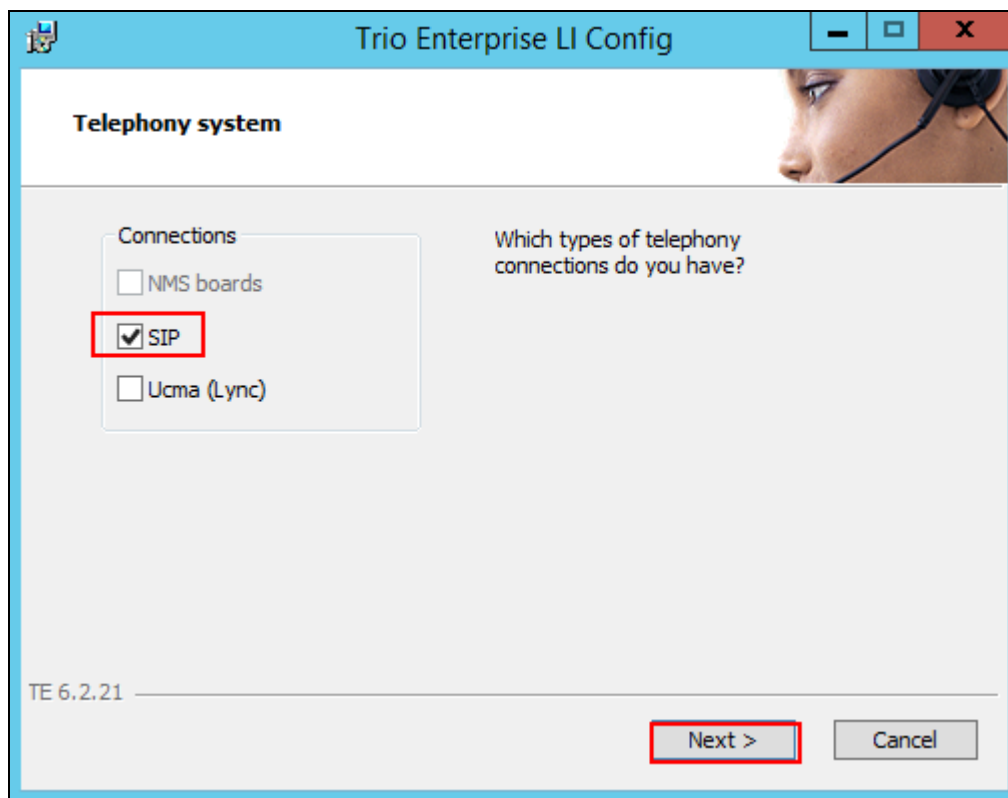
### 8.1. Configure Trio Enterprise to use SIP Trunks

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

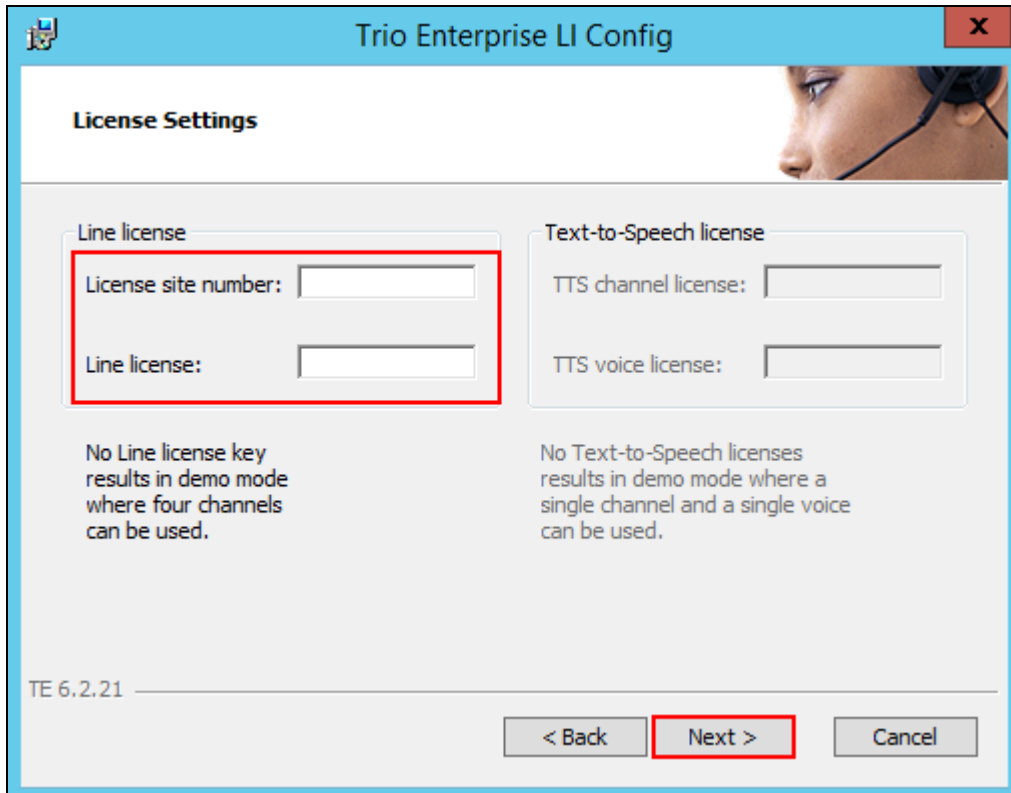


Launch the 'TeleVoice Config' shortcut

The configuration of the application starts, and when the new window opens, check the **SIP** check box followed by the **Next** button.



In the subsequent window, enter the **License site number:** and **Line licence:** as supplied directly by Enghouse Interactive AB or the Trio Enterprise reseller. Click on the **Next** button to continue.



The screenshot shows a software configuration window titled "Trio Enterprise LI Config". The window has a blue header bar with a close button (X) on the right. Below the header, the title "License Settings" is displayed. The main content area is divided into two columns. The left column is titled "Line license" and contains two input fields: "License site number:" and "Line license:". These two fields are enclosed in a red rectangular box. Below these fields, a note states: "No Line license key results in demo mode where four channels can be used." The right column is titled "Text-to-Speech license" and contains two input fields: "TTS channel license:" and "TTS voice license:". Below these fields, a note states: "No Text-to-Speech licenses results in demo mode where a single channel and a single voice can be used." At the bottom left of the window, the version "TE 6.2.21" is displayed. At the bottom right, there are three buttons: "< Back", "Next >", and "Cancel". The "Next >" button is highlighted with a red rectangular box.

**Trio Enterprise LI Config**

**License Settings**

**Line license**

License site number:

Line license:

No Line license key results in demo mode where four channels can be used.

**Text-to-Speech license**

TTS channel license:

TTS voice license:

No Text-to-Speech licenses results in demo mode where a single channel and a single voice can be used.

TE 6.2.21

< Back   **Next >**   Cancel

In the subsequent window, click on the **GENERIC** radio button followed by the **Next** button to continue.

**Trio Enterprise LI Config**

**SIP Settings(1)**

Select which PABX this SIP trunk will be connected to. If you don't know, select GENERIC and later modify the configuration in televoice.cfg.

☒ **GENERIC** ☐ LUCENT

☐ MD 110/MX-ONE ☐ SIEMENS

☐ PHILIPS ☐ CISCO

☐ Nortel CS1000/Meridian ☐ PSTN

☐ ALCATEL 4200

☐ ALCATEL 4300

☐ ALCATEL 4400

TE 6.2.21

< Back **Next >** Cancel

In the subsequent window enter the following settings:

- **Local IP:** Enter the local IP address of the Trio Enterprise server
- **Port:** Enter the SIP Port “5060”
- **Target IP:** Enter the IP address of the Session Manager
- **Port:** Enter the SIP Port 5060
- **Number of channels:** Enter **30** as the number of channels

Click on the **Next** button to continue.

**Trio Enterprise LI Config**

**SIP Settings(2)**

**SIP settings**

Local IP: 10.10.98.158

Port: 5060

Target IP: 10.10.97.228

Port: 5060

Number of channels: 30

**Codecs**

☐ Enable G711 mu-law codec

TE 6.2.21

< Back   **Next >**   Cancel



In the subsequent window enter the following settings:

- **Use LI Address Space:** Click on the radio button
- **Enable IP routing:** Check the box
- **UPDATE support:** Check the box

Click on the **Next** button to continue.

**Trio Enterprise LI Config**

**SIP Settings(3)**

Address Space (AS)

☒ Use LI Address Space

☐ AS Name:

☐ No Address Space

Sip Options

☒ UPDATE support

Routing

☒ Enable IP routing

TE 6.2.21

Additional SIP Trunk

< Back

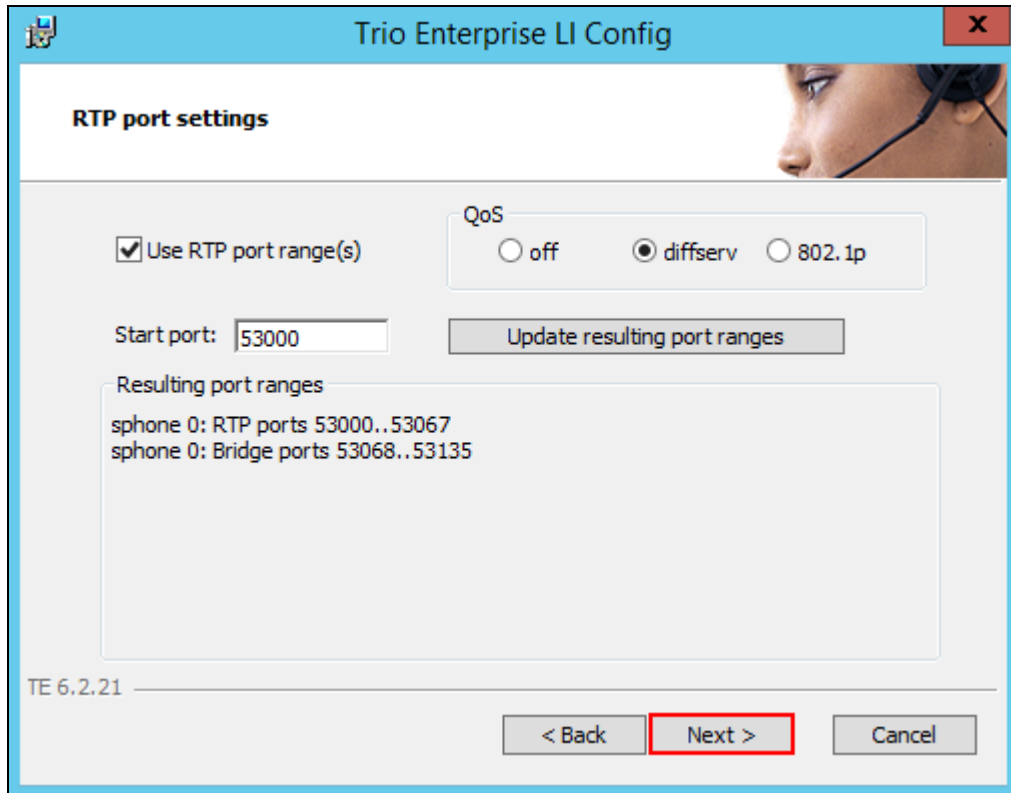
**Next >**

Cancel

In the subsequent window enter the following settings:

- **Use RPT port range(s):** Check the box
- **diffserv:** Click on the radio button
- **Start port:** Enter **53000**

Click on the **Next** button to continue.



The screenshot shows the 'Trio Enterprise LI Config' window with the 'RTP port settings' tab selected. The 'Use RTP port range(s)' checkbox is checked. The 'QoS' section has three radio buttons: 'off', 'diffserv' (which is selected), and '802.1p'. The 'Start port' field is set to '53000'. An 'Update resulting port ranges' button is visible. Below this, a text box displays the 'Resulting port ranges' for 'sphone 0': RTP ports 53000..53067 and Bridge ports 53068..53135. At the bottom, there are three buttons: '< Back', 'Next >' (highlighted with a red border), and 'Cancel'. The version 'TE 6.2.21' is shown in the bottom left corner.

**Trio Enterprise LI Config**

**RTP port settings**

☒ Use RTP port range(s)

QoS

☐ off ☒ diffserv ☐ 802.1p

Start port:

Resulting port ranges

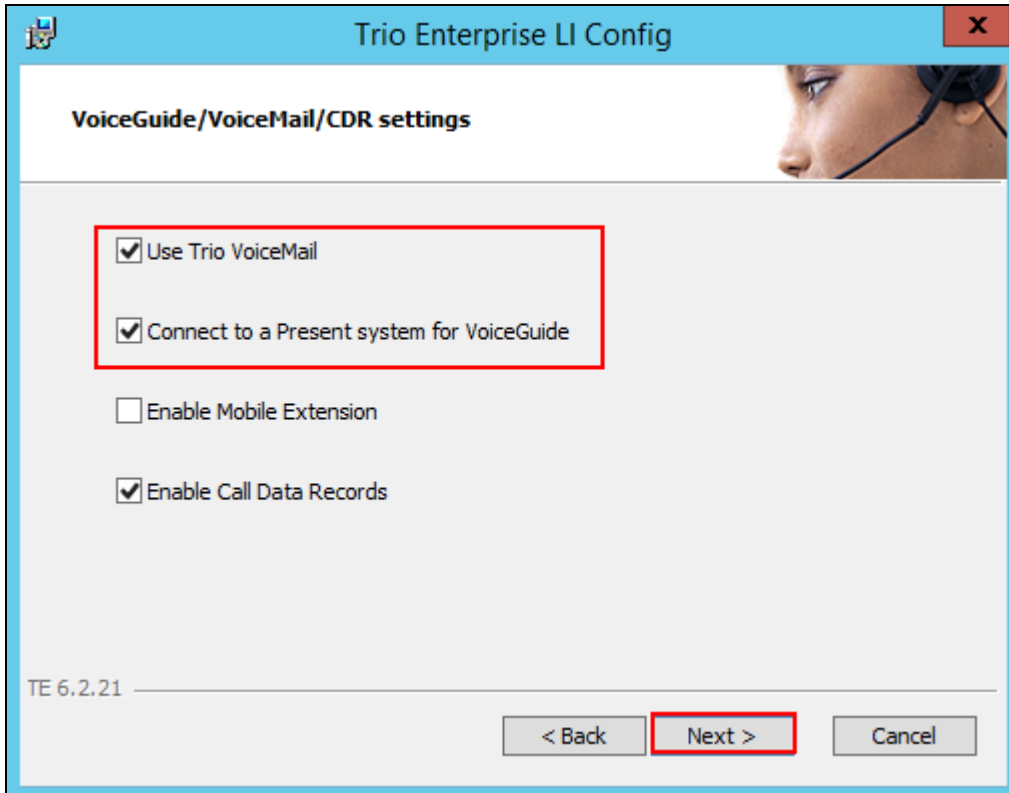
sphone 0: RTP ports 53000..53067  
sphone 0: Bridge ports 53068..53135

TE 6.2.21

In the subsequent window enter the following settings:

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

Click on the **Next** button to continue.

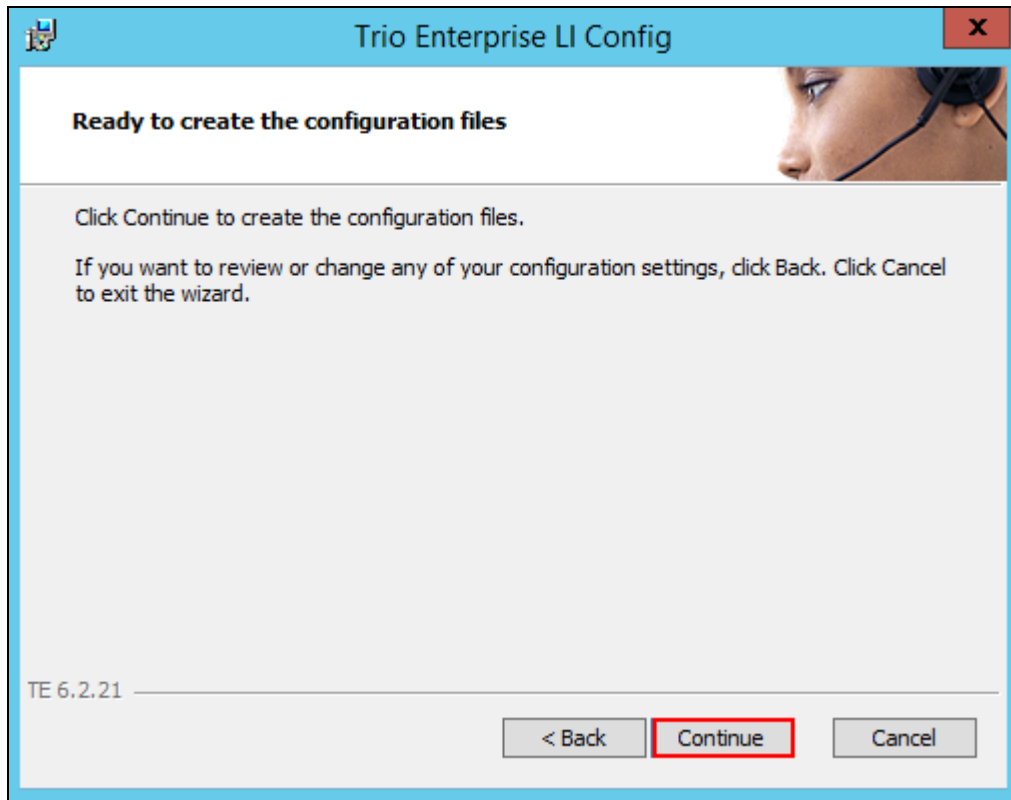


The screenshot shows a window titled "Trio Enterprise LI Config" with a close button (X) in the top right corner. The window contains a section titled "VoiceGuide/VoiceMail/CDR settings" with a background image of a person wearing a headset. Below the title, there are four checkboxes:

- ☒ Use Trio VoiceMail
- ☒ Connect to a Present system for VoiceGuide
- ☐ Enable Mobile Extension
- ☒ Enable Call Data Records

At the bottom left, the text "TE 6.2.21" is displayed. At the bottom right, there are three buttons: "< Back", "Next >", and "Cancel". The "Next >" button is highlighted with a red border.

In the subsequent window shown below, click on **Continue** button.



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



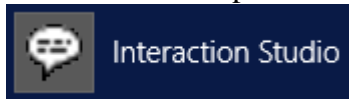
## 8.2. InteractionStudio Configuration

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

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

### 8.2.1. Configure Call Routing table

On the Trio Enterprise server, launch the 'Interaction Studio' shortcut



When the Interaction Studio window opens, navigate to **Routing**. A **Call routing table** will open. In the example below:

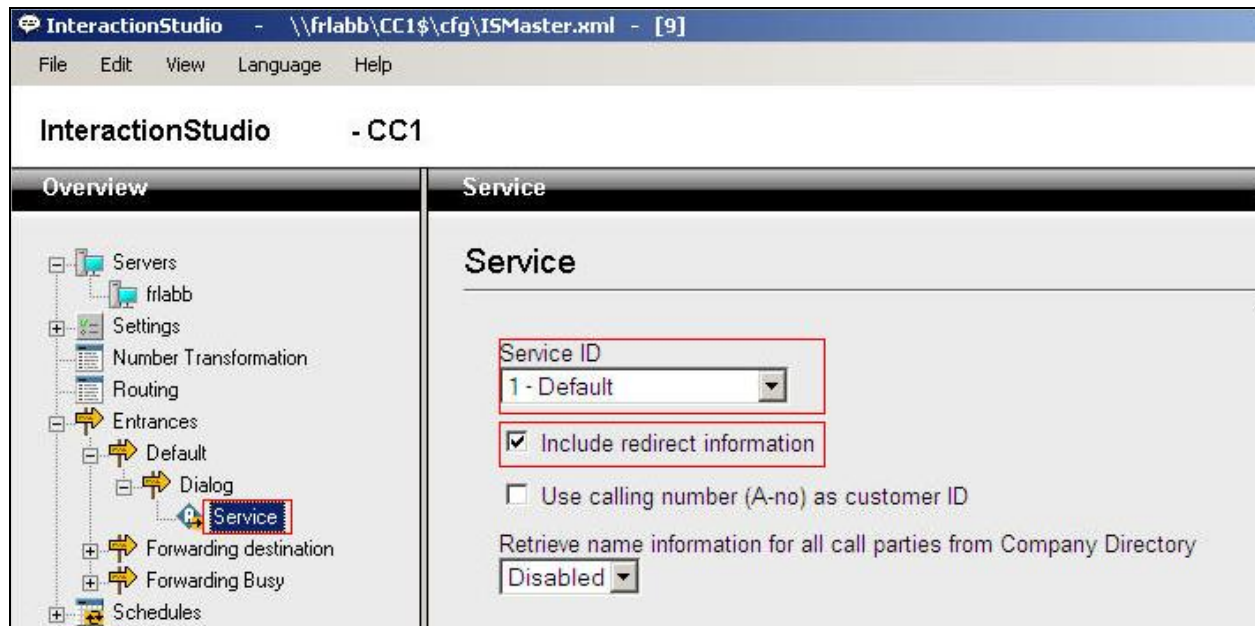
- Extension **71000** is the main queue number.
- Extension **71001** is the number that calls go to when Call forward Busy is activated.
- Extension **71002** is the number that calls go to when Call forward No Answer is activated.
- Extension **71003** is the number that calls go to when user absent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

The screenshot shows the InteractionStudio CC1 (Administrator) window. The left sidebar contains a tree view with categories: Servers, Settings, Number Transformation, Routing, Entrances, Default, PLAY, Busy, Dialog, Loop Detection via DTMF, Voice Functions, No Answer, Absent, and Schedules. The main area is titled "Routing" and displays a "Call routing table".

	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	
*					

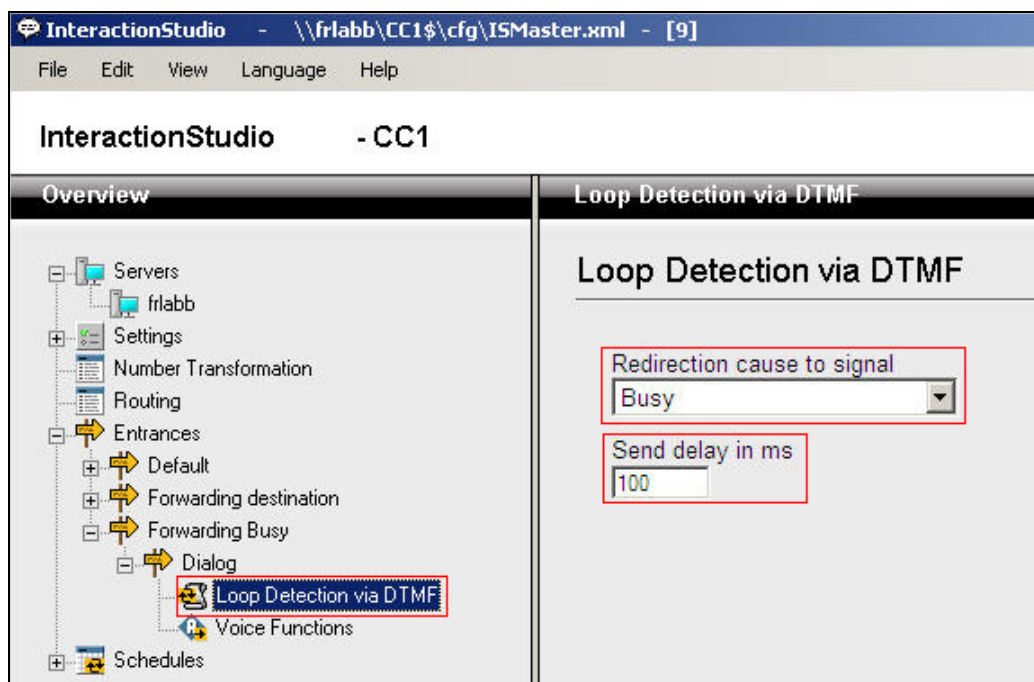
### 8.2.2. Configure Attendant Service

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



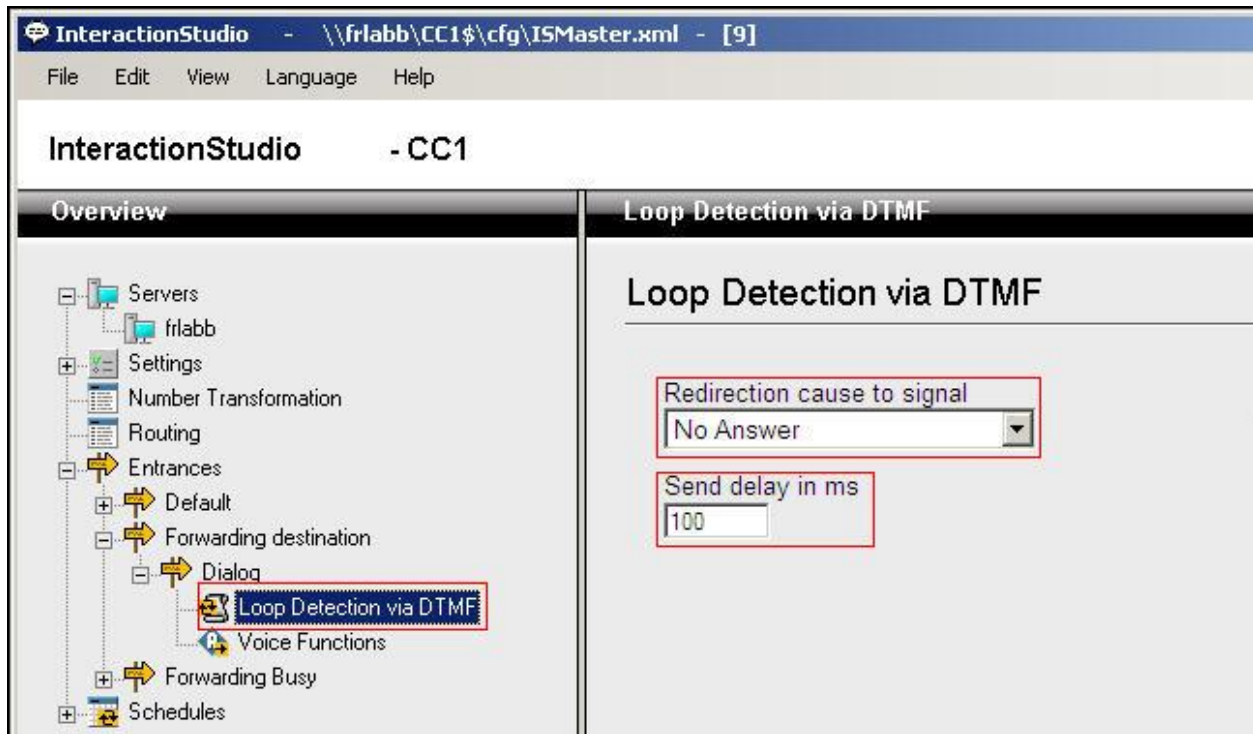
### 8.2.3. Configure Loop Detection via DTMF for Busy signal

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 Communication Server 1000 telephone to make and receive calls, which are directed to the phone by Trio Enterprise server. The steps to configure Trio Attendant are to launch the 'Agent Client' shortcut.



The window below opens. Enter a valid **User ID** and **Password**. Note this user ID and password is created during the installation of TRIO Enterprise Server. For **Extension**, select the Communication Server 1000 telephone number that will be used as the agent's audio device (number **54336** 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.

A screenshot of the 'Trio Agent - Login' window. The window has a blue title bar with the text 'Trio Agent - Login' and a close button. The main area has a light blue background with the 'Trio Enterprise®' logo at the top. Below the logo is a key icon. The login form includes fields for 'User ID' (containing 'op'), 'Password' (empty), 'Phone number' (containing '54336'), 'Phone type' (a dropdown menu showing 'Standard phone'), 'Location' (a dropdown menu showing 'Location 1'), 'Work mode' (a dropdown menu showing 'Switchboard operator'), and 'Server' (a dropdown menu showing 'te62cs1k76'). There are two checkboxes: 'Log in with Contact Center license (e-mail, fax, voice mail and tasks)' which is unchecked, and 'Log in with Enterprise Attendant license (extended switchboard features)' which is checked. At the bottom are three buttons: 'OK', 'Guest', and 'Cancel'. The footer shows 'Version 6.2.21.559', '© Enghouse Interactive AB', and the 'TRIO' logo.

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).

Trio Enterprise Attendant - Operator (Normal) @ 54336

File View Insert Tools Help

Ready  
Normal

Phone no Time Job no

Company Directory <All> <Current service> Dial

Availability	Icon	Returns	Extension	Last name	First name	State	T	Q	Extension infor...
<     >									

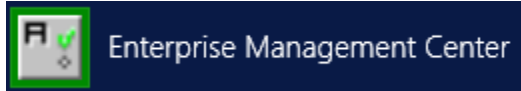
Extension information	E-mail	Subject
<     >		

Reason	From	To	Forward	Alternate answering
<     >				

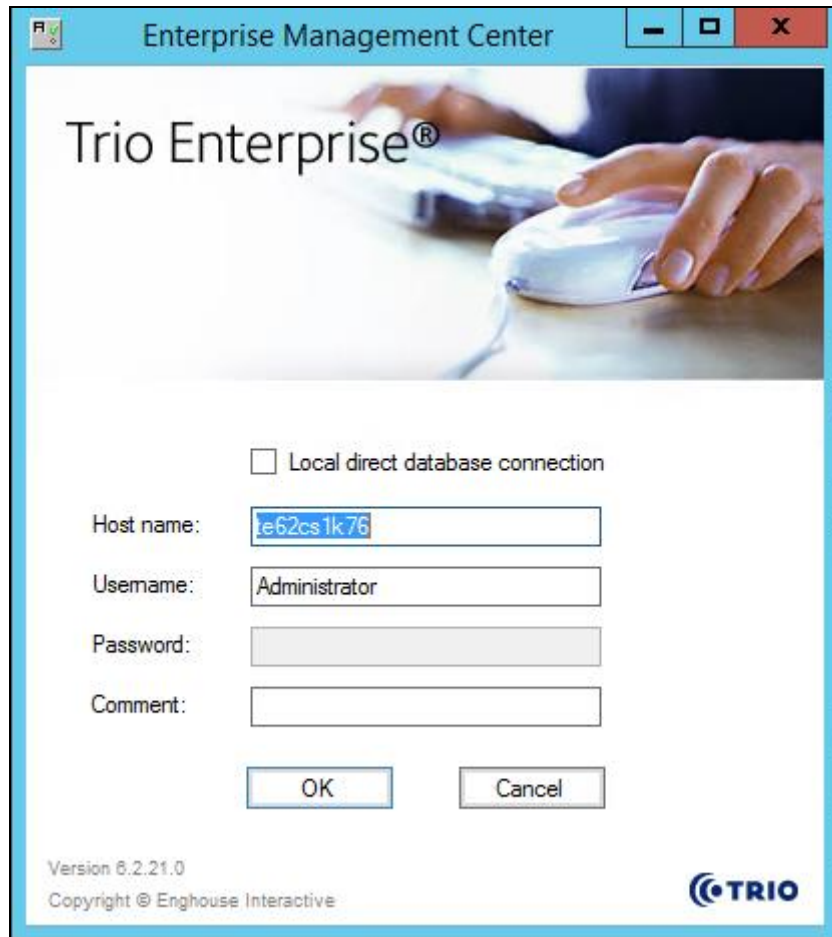
Ready for call Normal Nothing booked CTI 1: OK

## 8.4. Configure Presence (TR87) on Trio

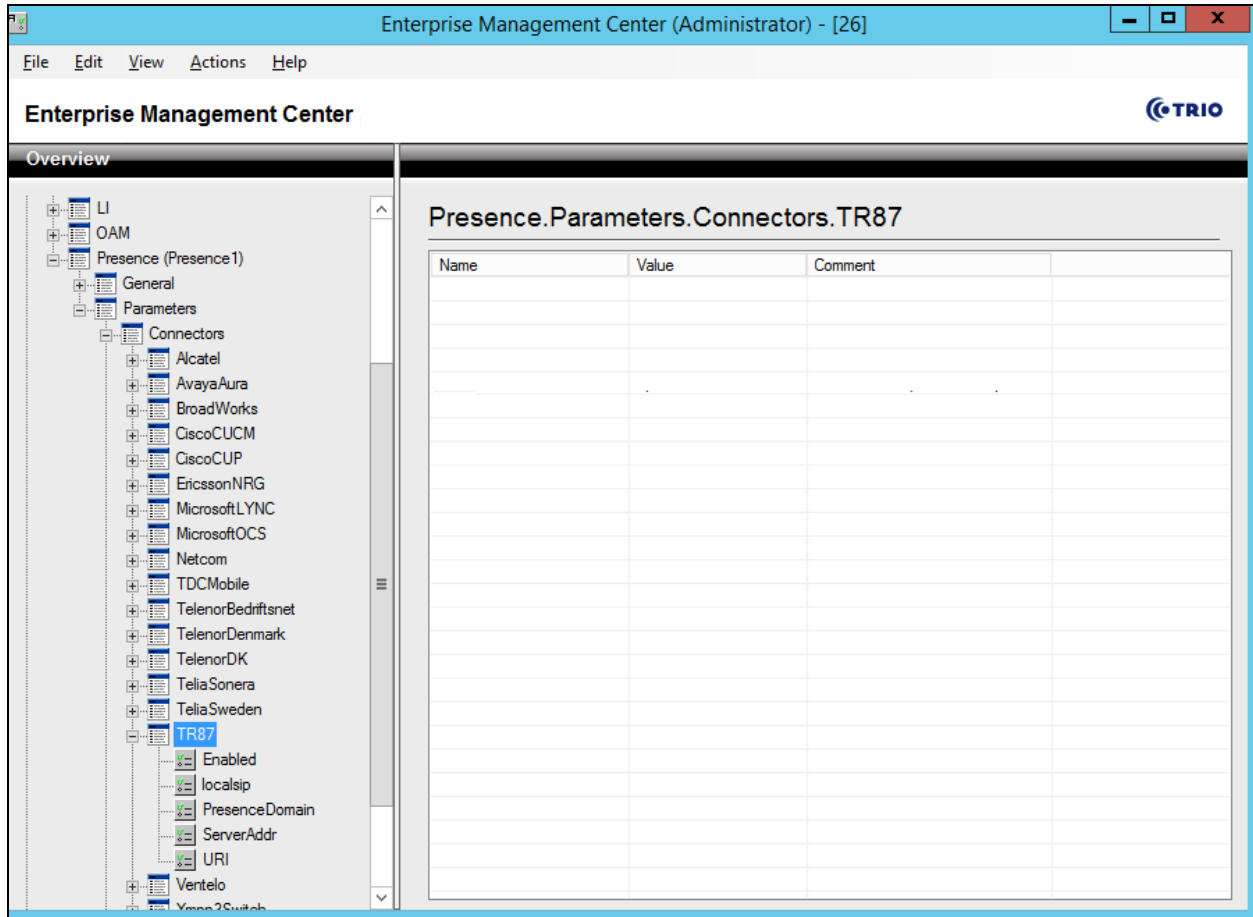
Launch the 'Enterprise Management Center' shortcut.



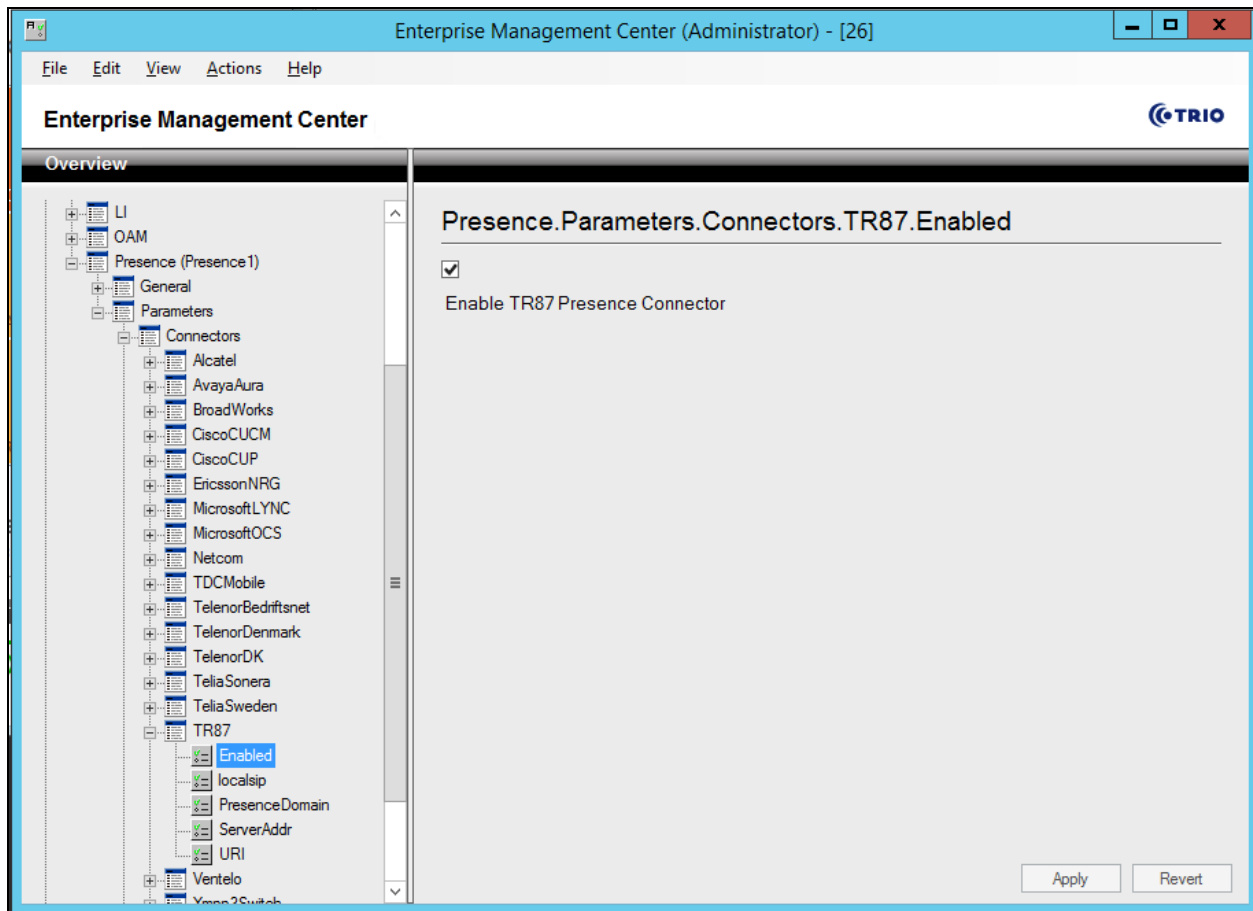
Enter the proper credentials and click on **OK**.

A screenshot of the 'Enterprise Management Center' login window. The window has a blue title bar with the text 'Enterprise Management Center' and standard Windows window controls. The main content area features a background image of hands using a computer mouse. The text 'Trio Enterprise®' is prominently displayed. Below this, there is a checkbox labeled 'Local direct database connection'. Further down are four input fields: 'Host name:' with the value 'e62cs1k76', 'Username:' with the value 'Administrator', 'Password:', and 'Comment:'. At the bottom are 'OK' and 'Cancel' buttons. The footer contains the text 'Version 6.2.21.0', 'Copyright © Enghouse Interactive', and the 'TRIO' logo.

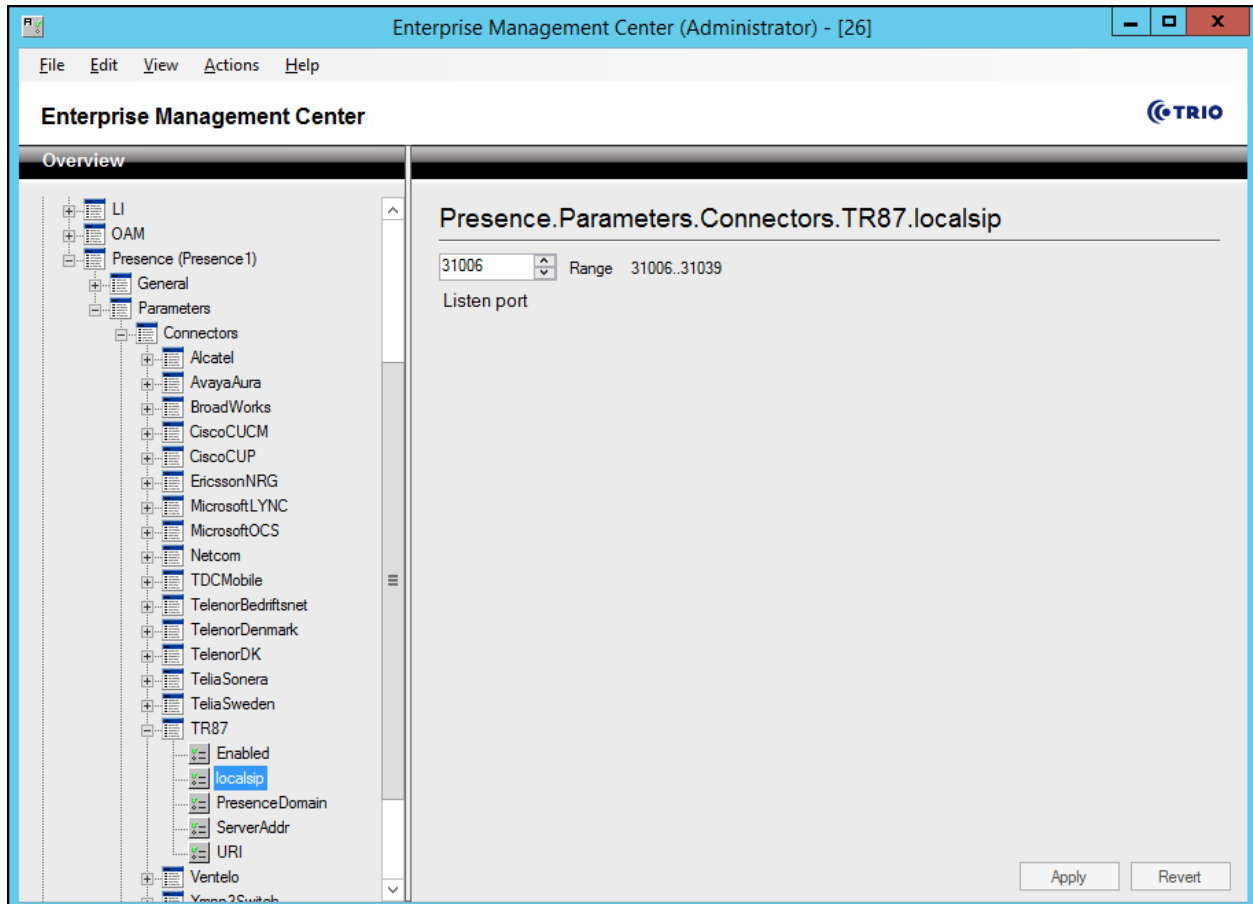
Click on **Parameters** → **Presence** → **Parameters** → **Connectors** → **TR87** in the left window.



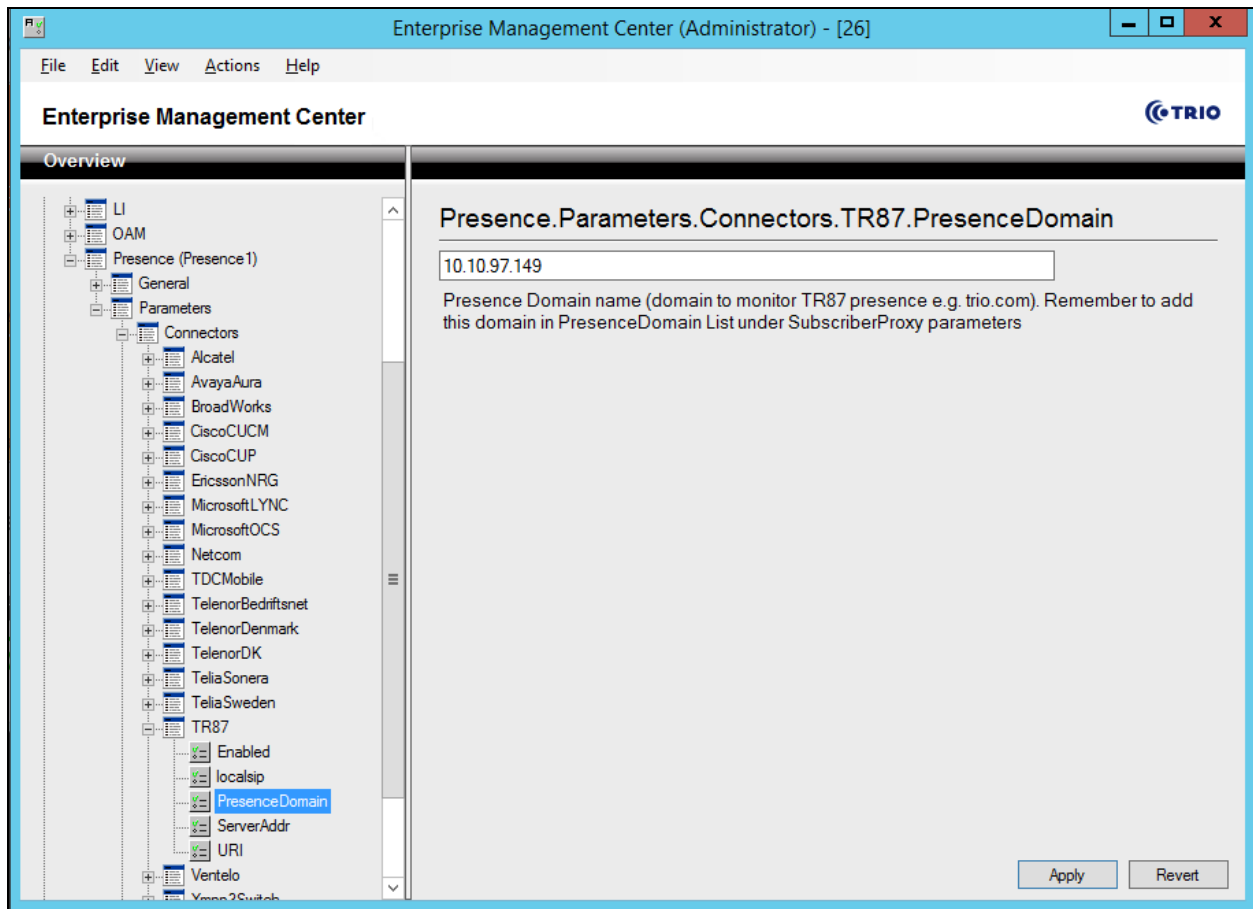
Under **TR87** select **Enabled** in the left window. Ensure that **Enable TR87 Presence Connector** is ticked as shown below. Click **Apply** to continue.



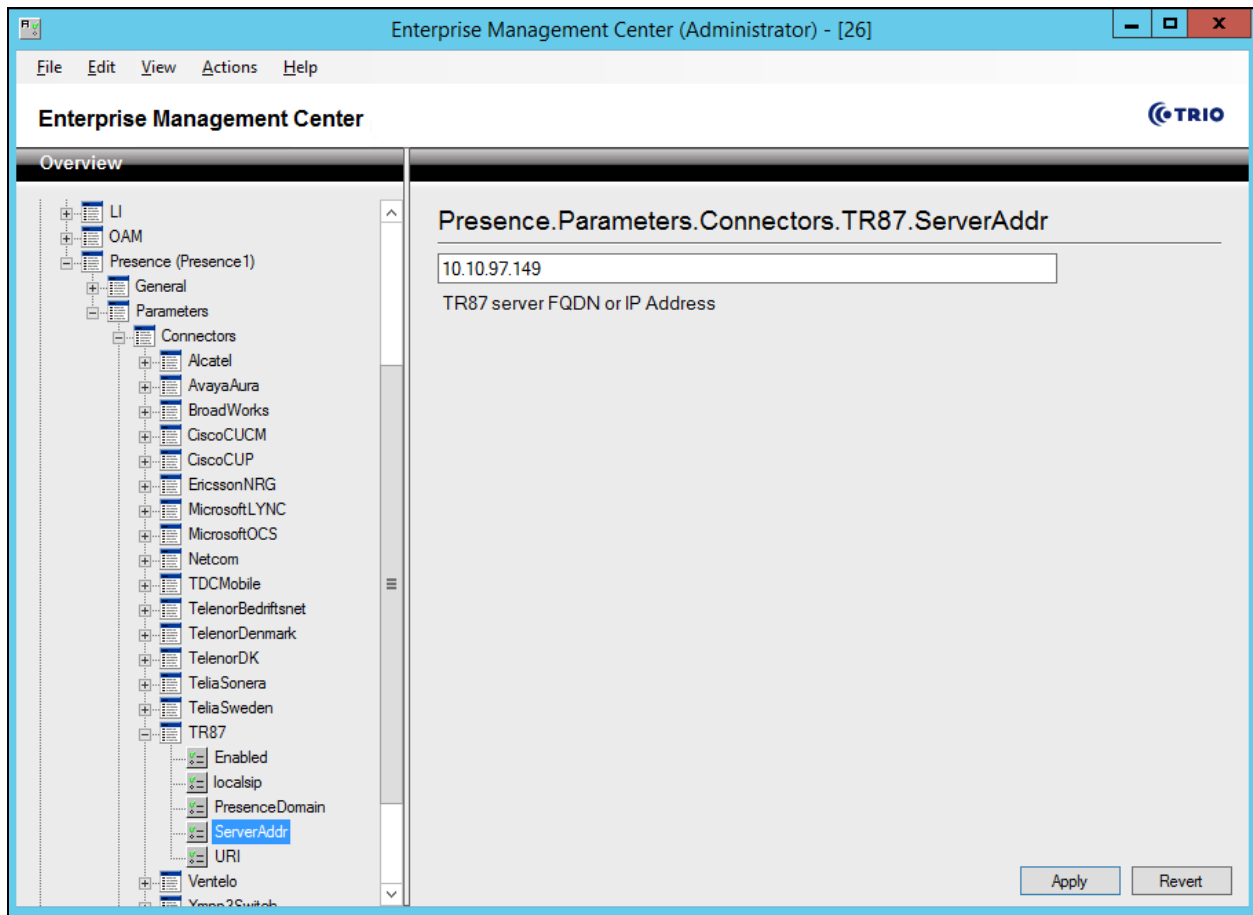
Select **localsip** under **TR87** in the left window and select the **Listen port** for TR87, for compliance testing this was left as default **31006** as shown below. Click **Apply** to continue.



Select **PresenceDomain** under **TR87** in the left window. Enter the Node IP address of the Communication Server 1000 as noted in **Section 6**. Click **Apply** to continue.

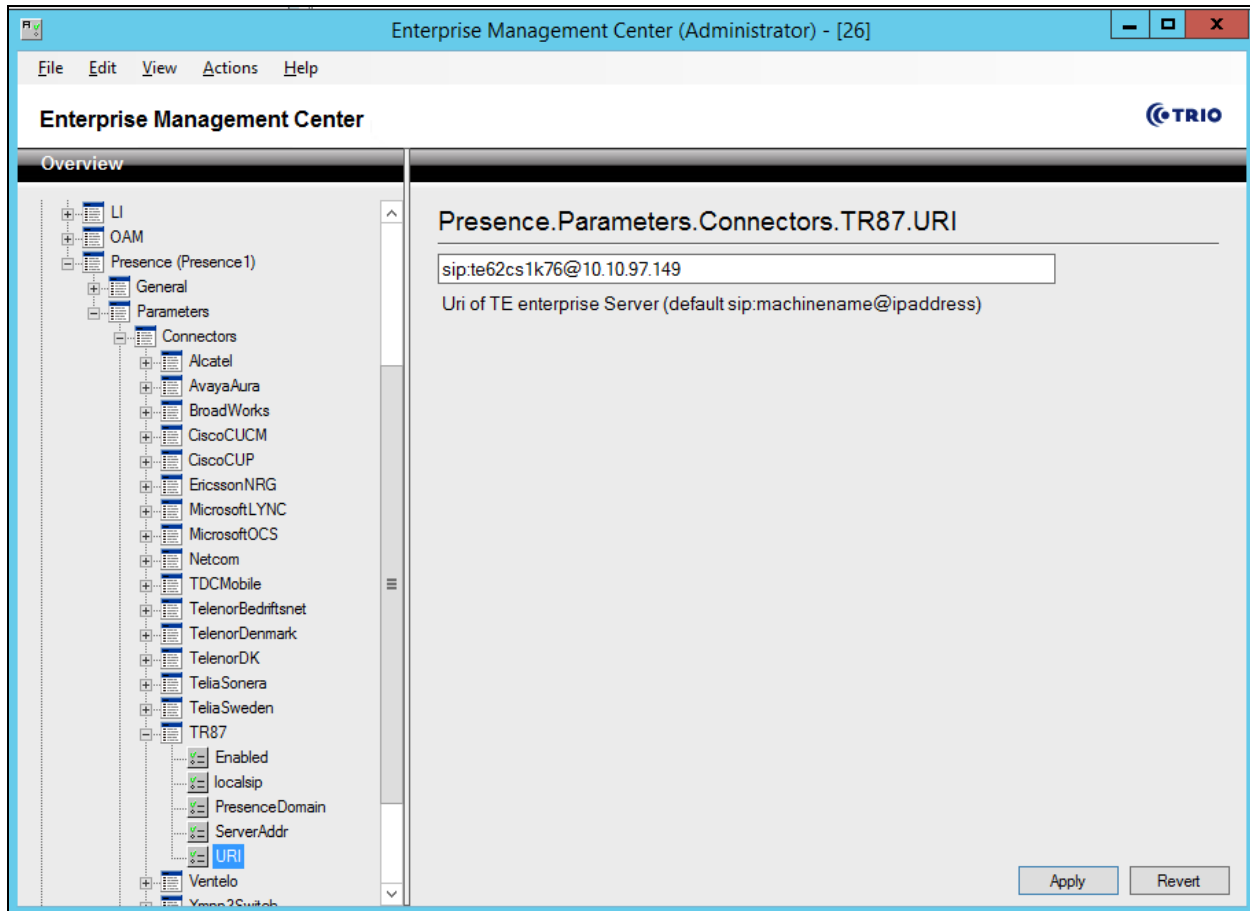


Select **ServerAddr** under **TR87** in the left window and again enter the Node IP address of the Communication Server 1000 as noted in **Section 6**. Click **Apply** to continue.

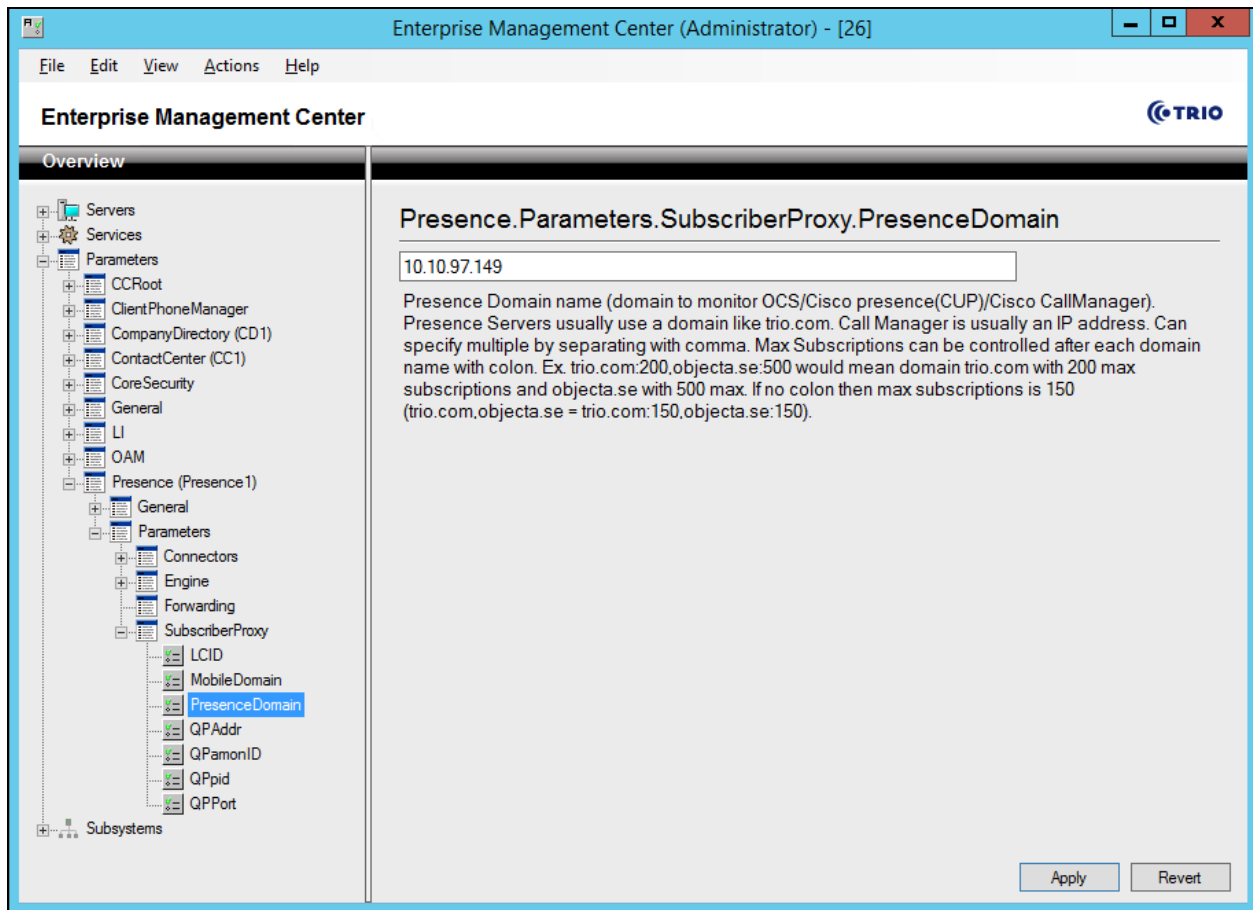




Select **URI** under **TR87** in the left window and enter the **machinename@ipaddress** preceded with **sip:** as shown below. Click Apply to continue.



Select **PresenceDomain** under **SubscriberProxy** in the left window. Enter the Node IP address of the Communication Server 1000 in the right window as noted in **Section 6**. Click **Apply** to continue.

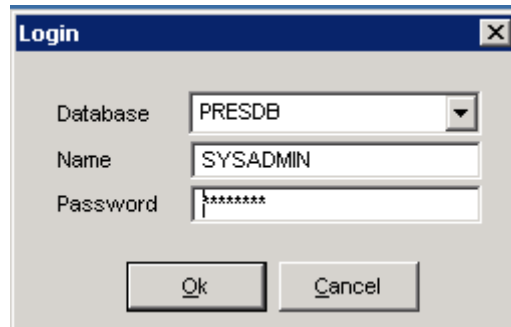


## 8.5. Configure ICP on Trio

Launch the 'Trio Present Setup' shortcut



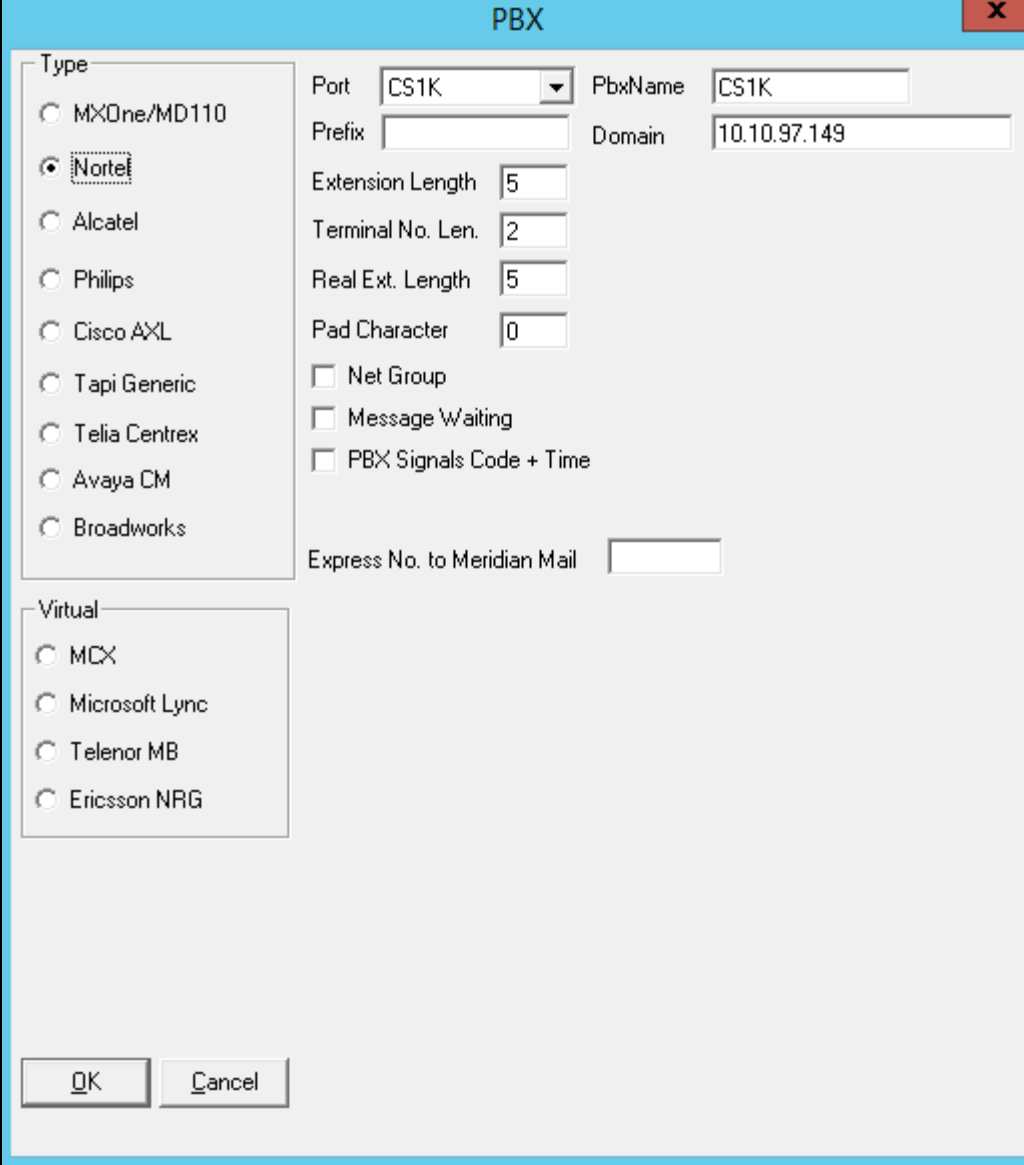
Enter the proper credentials and click **Ok**.

A 'Login' dialog box with a title bar containing the word 'Login' and a close button. The dialog has three input fields: 'Database' with a dropdown menu showing 'PRESDB', 'Name' with the text 'SYSADMIN', and 'Password' with masked characters '\*\*\*\*\*'. At the bottom are 'Ok' and 'Cancel' buttons.

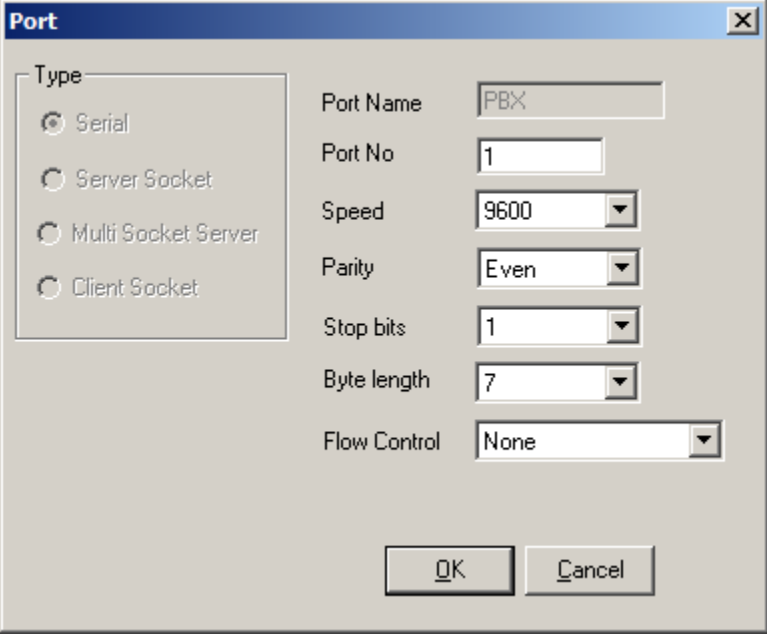
Highlight the selected PBX under the PBX tab and click on **Change** (not shown). This opens the window displayed on the right. Ensure the following are selected.

- **Type:** **Nortel**
- **Port:** Select the appropriate PBX from the drop down. In this case **CS1K**.
- **Domain:** Node IP Address of Communication Server 1000, in this case it is IP address of Node **510** as noted in **Section 6**.

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 two main sections: 'Type' and 'Virtual'. The 'Type' section contains a list of radio buttons: MXOne/MD110, Nortel (selected), Alcatel, Philips, Cisco AXL, Tapi Generic, Telia Centrex, Avaya CM, and Broadworks. The 'Virtual' section contains a list of radio buttons: MCX, Microsoft Lync, Telenor MB, and Ericsson NRG. To the right of these sections are several input fields: 'Port' (a dropdown menu showing 'CS1K'), 'PbxName' (a text box with 'CS1K'), 'Prefix' (an empty text box), 'Domain' (a text box with '10.10.97.149'), 'Extension Length' (a text box with '5'), 'Terminal No. Len.' (a text box with '2'), 'Real Ext. Length' (a text box with '5'), 'Pad Character' (a text box with '0'), and three checkboxes: 'Net Group', 'Message Waiting', and 'PBX Signals Code + Time', all of which are unchecked. At the bottom of the window is an 'Express No. to Meridian Mail' text box. At the very bottom are 'OK' and 'Cancel' buttons.

Under the **Communications** tab select **ICP** and click **Change** (not shown). Enter the information that was entered in **Section 5.4.1**. Click **OK** once all correct information is added.



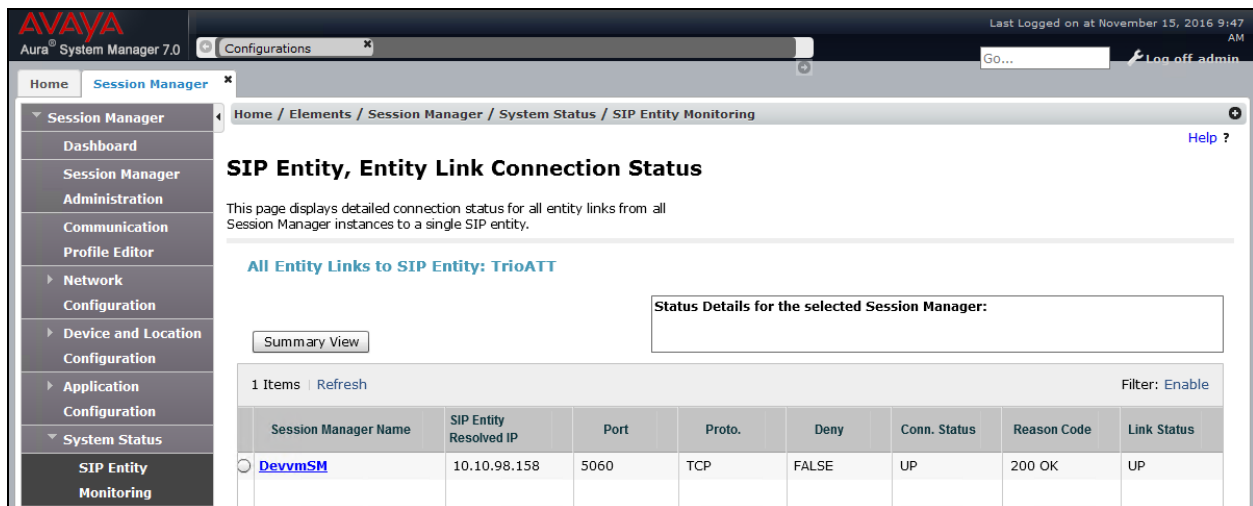
The image shows a 'Port' configuration dialog box. On the left, under the 'Type' section, there are four radio buttons: 'Serial' (selected), 'Server Socket', 'Multi Socket Server', and 'Client Socket'. On the right, there are several input fields: 'Port Name' with the text 'PBX', 'Port No' with the value '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 'OK' and 'Cancel' buttons.

## 9. Verification Steps

This section provides the tests that can be performed to verify correct configuration of Communication Server 1000 and Session Manager with TRIO Enterprise.

### 9.1. Verify status of Trio SIP Entity

In System manager web page, to confirm a successful Trio SIP entity connection to Session Manager, click on **Element** → **Session Manager** and then select **System Status** → **SIP Entity Monitoring**, click on **TrioATT** entity to verify its status. The detail page shows the link from **Trio** to **Session Manager** via **TCP** is **UP**.



The screenshot shows the Avaya Aura System Manager 7.0 web interface. The left sidebar contains a navigation menu with options like Session Manager, Dashboard, Session Manager Administration, Communication, Profile Editor, Network, Configuration, Device and Location Configuration, Application Configuration, and System Status. The main content area is titled "SIP Entity, Entity Link Connection Status" and displays a table of entity links for the selected Session Manager (DevvmSM). The table has columns for Session Manager Name, SIP Entity Resolved IP, Port, Proto., Deny, Conn. Status, Reason Code, and Link Status. The status of the link is UP.

Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
DevvmSM	10.10.98.158	5060	TCP	FALSE	UP	200 OK	UP

For feature testing the following were verified,

- Incoming internal and external calls
- Outgoing internal and external calls
- Supervised and unsupervised transfer with answer
- Directing calls to busy extensions
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions
- Absence detection
- Status of the phones

## 9.2. SIP Channels on Trio Enterprise

To confirm a successful Trio Enterprise connection with the Session Manager, launch the

‘Telestatus’ shortcut.



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 from Enghouse Interactive AB to successfully interoperate with Avaya Communication Server 1000 and Avaya Aura® Session Manager using SIP trunks. Trio Enterprise passed all compliance testing successfully; please see **Section 2.2** of these Application Notes for results and observations if any.

## 11. Additional References

This section references the product documentation relevant to these Application Notes. Product documentation for Avaya products may be found at <http://support.avaya.com>.

### Avaya:

1. *Communication Server 1000E Installation and Commissioning*, Release 7.6, NN43041-310
2. *Element Manager System Reference – Administration - Avaya Communication Server 1000*, Release 7.6, NN43001-632.
3. *Avaya Communication Server 1000 Co-resident Call Server and Signaling Server Fundamentals* Release 7.6, NN43001-509.
4. *Avaya Communication Server 1000 Unified Communications Management Common Services Fundamentals -*, Release 7.6, NN43001-116.
5. *Avaya Communication Server 1000 - Software Input Output Reference — Administration* Release 7.6, NN43001-611.
6. *Avaya Communication Server 1000 - ISDN Primary Rate Interface Installation and Commissioning*, Release 7.6, NN43001-301.
7. *Implementing Avaya Aura® Session Manager* Document ID 03-603473.
8. *Administering Avaya Aura® Session Manager*, Doc ID 03-603324.
9. *Deploying Avaya Aura® System Manager*, Release 7.0.
10. *Administering Avaya Aura® System Manager for Release 7.0*, Release 7.0.

All information on the product installation and configuration TRIO Enterprise Server can be found at <http://www.trio.com>



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