



Application Notes for Configuring Trio Enterprise R4.1 from Enghouse Interactive AB with Avaya Communication Server 1000E R7.6 and Avaya Aura® Session Manager R6.3 using a SIP connection – Issue 1.0

Abstract

These Application Notes describe how to configure an Avaya Communication Server 1000E 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 1000E 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.

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 Aura® Session Manager R6.3 with Trio Enterprise R4.1 from Enghouse Interactive AB. Trio Enterprise is a client/server based application running on Microsoft Windows 2008 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 1000E 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 (CS1000E). The Trio Enterprise server uses a SIP connection to the CS1000E call server via Session Manager. See **Figure 1** for a network diagram. A basic Distance Steering Code configuration (DSC) was configured on the CS1000E to route all calls to the Trio attendant position. 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 three different simulated PSTN trunks were used which included a QSIG ISDN trunk, a H.323 trunk and a SIP trunk. 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. 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 1140E was used as the attendant's deskphone. When the attendant is called the Trio Enterprise server calls the 1140E 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 calls
- Supervised and unsupervised transfer with answer
- Directing calls to busy extensions
- Call queuing and retrieval
- Loop detection for busy and unanswered extensions

2.2. Test Results

Tests were performed to insure full interoperability between the Trio Enterprise and the CS1000E. The tests were all functional in nature and performance testing was not included. All the test cases passed successfully.

2.3. Support

For technical support on Trio products, please use the following web link.

<http://www.trio.com/web/Support.aspx>

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 during compliance testing. Trio Enterprise is connected to the CS1000E using a SIP connection via Session Manager. The Trio Enterprise Server is configured as a SIP Endpoint. System Manager is used to configure Session Manager.

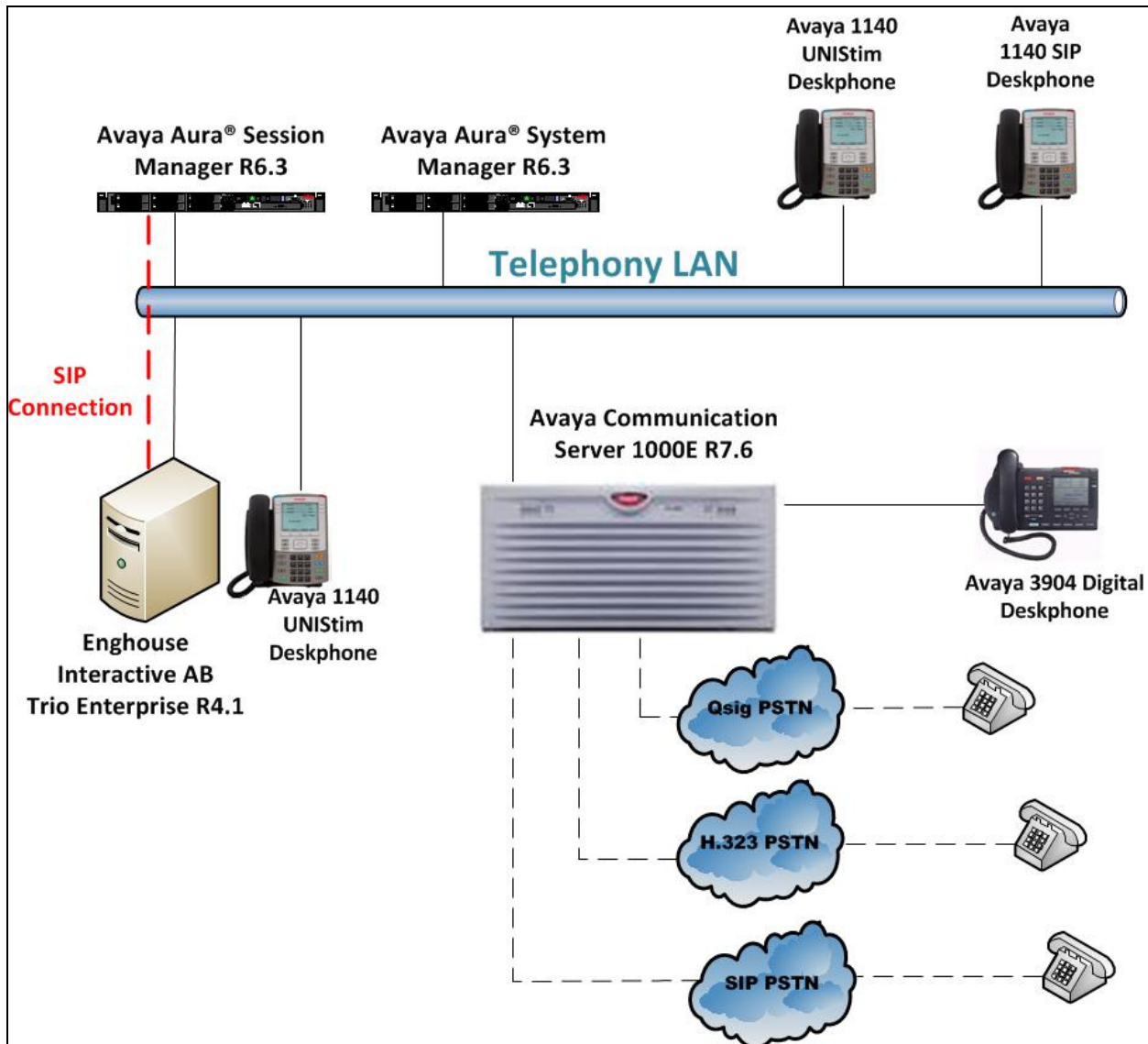


Figure 1: Configuration for Avaya Communication Server 1000E R7.6, Avaya Aura® Session Manager R6.3 and Trio Enterprise R4.1

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 SP2 (See Appendix A for list of patches) |
| Avaya Aura® System Manager | System Manager 6.3.0 - FP2 Build No. - 6.3.0.8.5682-6.3.8.1814 Software Update Revision No: 6.3.3.5.1719 |
| Avaya Aura® Session Manager | Session Manager R6.3 SM 6.3.3.0.633004 |
| Avaya 1140 UNISTim Deskphone | UNISTim V0625C8D |
| Avaya 1140 SIP Deskphone | SIP 04.03.12 |
| Avaya 3904 Digital Deskphone | Core V2.4 Flash V9.4 |
| Trio Enterprise Running on a Windows 2008 R2 64-bit server | Version 4.1 |

Note: The Avaya Aura® System Manager and Session Manager are virtual instances running on VMware ESXi 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 simulated PSTN connections are 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).

| Prompt | Response | Description |
|--|----------------------------|------------------|
| > | LD 22 | Enter Overlay 22 |
| 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 | 2000 LEFT 1992 USED | 8 |
| DECT USERS | 2000 LEFT 2000 USED | 0 |
| IP USERS | 4000 LEFT 3978 USED | 22 |
| BASIC IP USERS | 2000 LEFT 1998 USED | 2 |
| TEMPORARY IP USERS | 2000 LEFT 2000 USED | 0 |
| DECT VISITOR USER | 2000 LEFT 2000 USED | 0 |
| ACD AGENTS | 2000 LEFT 1995 USED | 5 |
| MOBILE EXTENSIONS | 2000 LEFT 2000 USED | 0 |
| TELEPHONY SERVICES | 2000 LEFT 2000 USED | 0 |
| CONVERGED MOBILE USERS | 2000 LEFT 2000 USED | 0 |
| AVAYA SIP LINES | 2000 LEFT 1997 USED | 3 |
| THIRD PARTY SIP LINES | 2000 LEFT 1998 USED | 2 |
| PCA | 2000 LEFT 2000 USED | 0 |
| ITG ISDN TRUNKS | 2000 LEFT 2000 USED | 0 |
| H.323 ACCESS PORTS | 2000 LEFT 1990 USED | 10 |
| AST | 2000 LEFT 1981 USED | 19 |
| SIP CONVERGED DESKTOPS | 2000 LEFT 2000 USED | 0 |
| SIP CTI TR87 | 2000 LEFT 1992 USED | 8 |
| SIP ACCESS PORTS | 2000 LEFT 1970 USED | 30 |
| RAN CON | 2000 LEFT 2000 USED | 0 |
| MUS CON | 2000 LEFT 2000 USED | 0 |
| IP RAN CON | 2000 LEFT 2000 USED | 0 |

| | | | | | |
|---------------------------------|-------|------|-------|------|-----|
| IP MUS CON | 2000 | LEFT | 2000 | USED | 0 |
| IP MEDIA SESSIONS | 2000 | LEFT | 1997 | USED | 3 |
| TNS | 10000 | LEFT | 9805 | USED | 195 |
| ACDN | 24000 | LEFT | 23979 | USED | 21 |
| AML | 16 | LEFT | 12 | USED | 4 |
| IDLE_SET_DISPLAY Cores3 Rls 7.5 | | | | | |
| LTID | 2000 | LEFT | 2000 | USED | 0 |
| RAN RTE | 512 | LEFT | 510 | USED | 2 |
| ATTENDANT CONSOLES | 100 | LEFT | 99 | USED | 1 |
| IP ATTENDANT CONSOLES | 2000 | LEFT | 1999 | USED | 1 |
| BRI DSL | 10000 | LEFT | 10000 | USED | 0 |
| MPH DSL | 100 | LEFT | 100 | USED | 0 |
| DATA PORTS | 2000 | LEFT | 2000 | USED | 0 |
| PHANTOM PORTS | 2000 | LEFT | 1995 | USED | 5 |
| TRADITIONAL TRUNKS | 2000 | LEFT | 1962 | USED | 38 |
| ELC ACCESS PORTS | 2000 | LEFT | 2000 | USED | 0 |
| DCH | 255 | LEFT | 252 | USED | 3 |

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 66 (i.e. **DCH 66**) was created. At the **CTYP** prompt, enter **DCIP**. This signifies the SIP D-Channel.

LD 17

| Prompt | Response | Description |
|-------------|---------------|-------------------------------------|
| > | LD 17 | Enter Overlay 17 |
| REQ | CHG | Change |
| TYPE | ADAN | Change the Action Device and Number |
| ADAN | NEW | Create New Action Device and Number |
| TYPE | DCH 66 | Create new D-Channel 66 |
| CTYP | DCIP | Card type is IP D-Channel |
| USR | ISDL | Integrated Services Digital Line |
| IFC | SL1 | D-Channel interface type |

5.2.2. 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** and **PCID** is **SIP**.

LD 16

| Prompt | Response | Description |
|-------------|--------------|-------------------------------------|
| > | LD 16 | Enter Overlay 16 |
| REQ | NEW | Create new |
| TYPE | RDB | Route Data block |
| CUST | 0 | Customer Number as defined in LD15 |
| ROUT | 20 | Route Number |
| TKTP | TIE | Route Type |
| VTRK | YES | Virtual Route |
| PCID | SIP | Protocol ID for route |
| DTRK | NO | Digital Trunk Route |
| ISDN | YES | Integrated Services Digital Network |
| MODE | ISDL | mode of operation |
| IFC | SL1 | Interface type |
| ACOD | 8020 | Access Code for trunk route |

5.2.3. Adding 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 | 20 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 | 36 | Route list Index number |
| ENTR | 0 | First entry for the RLI |
| ROUT | 20 | 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 **4000**, **FLEN** is **4** and the **RLI** is **36**.

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 | 4000 | Distant Steering code |
| FLEN | 4 | Flexible Length number of digits |
| RLI | 36 | 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 | 14 | Data number |
| PORT | 2 | 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 |

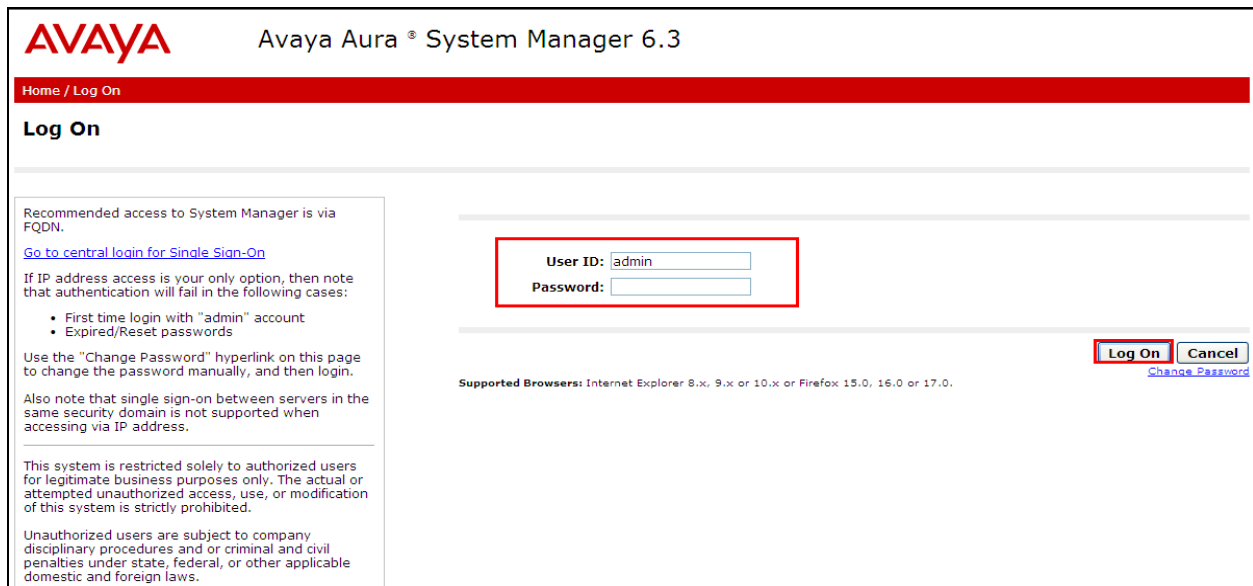
5.5.2. Configuration of ICP in the Customer Data Block

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 | 14 | Port number configured in Section 5.5.1 |
| NIPN | 9 | Number of Intercept positions |
| ICCR | NO | Intercept Position Cancelling Reply |
| ICDN | 4002 | Internal Call DN |
| ECDN | 4002 | External Call DN |
| ICDL | 4 | 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 Avaya Aura® System Manager 6.3

Home / Log On

Log On

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

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.

User ID:

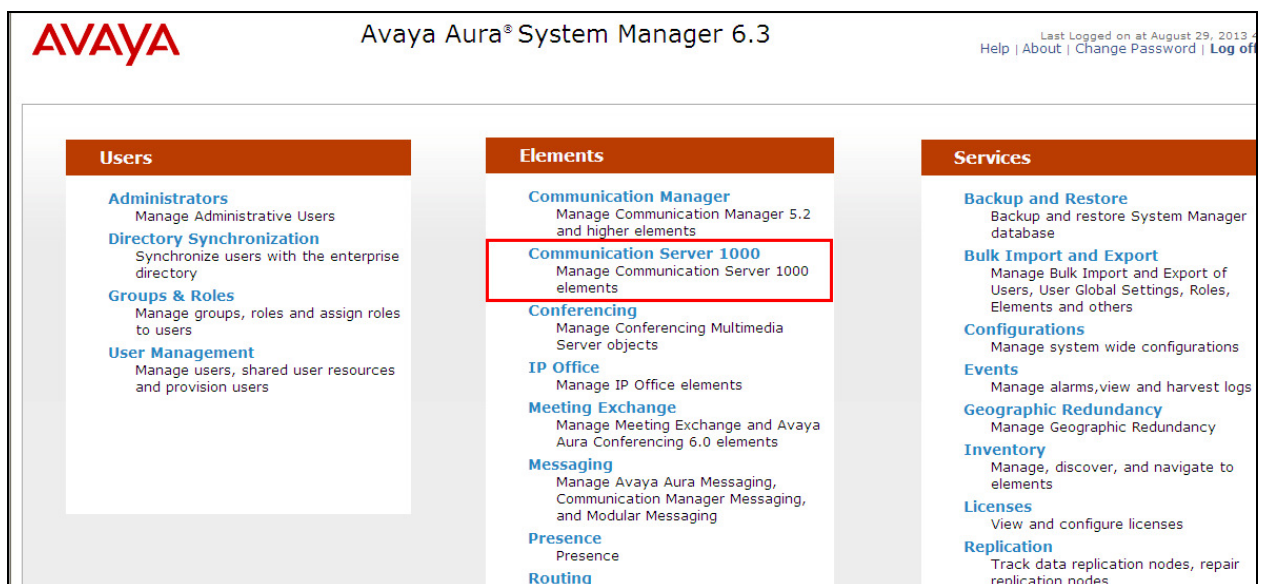
Password:

Log On Cancel

[Change Password](#)

Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 15.0, 16.0 or 17.0.

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



AVAYA Avaya Aura® System Manager 6.3

Help | About | Change Password | Log off

Users

- Administrators**
Manage Administrative Users
- Directory Synchronization**
Synchronize users with the enterprise directory
- Groups & Roles**
Manage groups, roles and assign roles to users
- User Management**
Manage users, shared user resources and provision users

Elements

- Communication Manager**
Manage Communication Manager 5.2 and higher elements
- Communication Server 1000**
Manage Communication Server 1000 elements
- Conferencing**
Manage Conferencing Multimedia Server objects
- IP Office**
Manage IP Office elements
- Meeting Exchange**
Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements
- Messaging**
Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging
- Presence**
Presence
- Routing**

Services

- Backup and Restore**
Backup and restore System Manager database
- Bulk Import and Export**
Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
- Configurations**
Manage system wide configurations
- Events**
Manage alarms, view and harvest logs
- Geographic Redundancy**
Manage Geographic Redundancy
- Inventory**
Manage, discover, and navigate to elements
- Licenses**
View and configure licenses
- Replication**
Track data replication nodes, repair replication nodes

Once **Communication Server 1000** is selected the following screen appears, click on the Element Manager link highlighted below.

The screenshot shows the Avaya Aura System Manager 6.3 interface. The left sidebar contains a navigation tree with categories like Network, CS 1000 Services, User Services, External Authentication, Password, Security, Roles, Policies, and Active Sessions. The main content area is titled 'Elements' and displays a table of registered elements. A red box highlights the element 'EM on cs1kpg1' in the table.

| Element Name | Element Type | Release | Address | Desc |
|---|--------------------------|---------|----------------|--------------|
| 1 smor63ympq.devconnect.local (primary) | Base OS | 7.6 | 10.10.40.32 | Base element |
| 2 EM on cs1kpg1 | CS1000 | 7.6 | 192.168.40.101 | New element |
| 3 cs1kpg1.devconnect.local (member) | Linux Base | 7.6 | 10.10.40.101 | Base element |
| 4 192.168.40.102 | Media Gateway Controller | 7.6 | 192.168.40.102 | New element |
| 5 NRSML on cs1kpg1 | Network Routing Service | 7.6 | 192.168.40.101 | New element |

Click on **IP Network → Nodes: Servers, Media Cards** in the left window. Click on the **Node ID** displayed in the right window. Note the IP address of this node as it will be required in **Section 7.7**.

The screenshot shows the CS1000 Element Manager interface. The left sidebar contains a navigation tree with categories like UCM Network Services, Home, Links, Virtual Terminals, System, Alarms, Maintenance, Core Equipment, Peripheral Equipment, IP Network, Nodes: Servers, Media Cards, Maintenance and Reports, Media Gateways, Zones, Host and Route Tables, Network Address Translation, QoS Thresholds, Personal Directories, Unicode Name Directory, Interfaces, Engineered Values, Emergency Services, and Software. The main content area is titled 'IP Telephony Nodes' and displays a table of nodes. A red box highlights the node '111' in the table.

| Node ID | Components | Enabled Applications | ELAN IP | Node/TLAN IPv4 | Node/TLAN IPv6 | Status |
|---------|------------|---|---------|----------------|----------------|--------------|
| 111 | 1 | SIP Line, LTPS, Gateway (SIPGw, H323Gw) | - | 10.10.40.111 | | Synchronized |

Select **Gateway (SIPGw&H323Gw)** highlighted below.

AVAYA CS1000 Element Manager

Managing: 192.168.40.101 Username: admin
System » IP Network » IP Telephony Nodes » Node Details

Node Details (ID: 111 - SIP Line, LTPS, Gateway (SIPGw, H323Gw))

Subnet mask: 255.255.255.0 * Subnet mask: 255.255.255.0 *
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 & H323Gw)**
- Personal Directories (PLD)
- 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

Managing: 192.168.40.101 Username: admin
System » IP Network » IP Telephony Nodes » Node Details » Virtual Trunk Gateway Configuration

Node ID: 111 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services | H.323 Gateway Settings

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

CTI settings

Customer number: [0] National: []
Maximum associations per DN: [3] International: []
International calls: ☐ Place as national Location code call: []
For calls within this country. Special number: []
Subscriber: []

CTI CLID presentation

Dialing plan: [CDP] Calling device URI format: [phone-context=dialstring]
Home location code: [] Country code (CCC): []

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

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

AVAYA CS1000 Element Manager

Managing: 192.168.0.10 Username: paul
System » IP Network » IP Telephony Nodes » Node Details

Node Details (ID: 100 - SIP Line, LTPS, 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:

* Required Value.

Save Cancel

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

AVAYA CS1000 Element Manager

Managing: 192.168.0.10 Username: paul
System » IP Network » IP Telephony Nodes » Node Saved

Node Saved

Node ID: 100 has been saved on the call server.

The new configuration must also be transferred to associated servers and media cards.

Transfer Now... You will be given an option to select individual servers, or transfer to all.

Show Nodes You may initiate a transfer manually at a later time.

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

AVAYA CS1000 Element Manager

Managing: 192.168.0.10 Username: paul
System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

Synchronize Configuration Files (Node ID <100>)

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"/> | cs1kpg | Signaling_Server | SIP Line, LTPS, Gateway, 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 synchronised the application will require a restart, select the **Hostname** and click on **Restart Applications** as highlighted below.

AVAYA CS1000 Element Manager

Managing: 192.168.0.10 Username: paul
System » IP Network » IP Telephony Nodes » Synchronize Configuration Files

Synchronize Configuration Files (Node ID <100>)

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"/> | cs1kpg | Signaling_Server | SIP Line, LTPS, Gateway, 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.

7. Creating SIP Entities on the Avaya Aura® Session Manager

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

- Create a SIP Domain
- Create a SIP location
- Create a SIP Entity
- Create an Entity Link
- Create a Routing Policy
- Create a Dial Pattern

Configuration of Session Manager is achieved by accessing the browser-based GUI of 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 Avaya Aura® System Manager 6.3

Home / Log On

Log On

Recommended access to System Manager is via FQDN.
[Go to central login for Single Sign-On](#)

If IP address access is your only option, then note that authentication will fail in the following cases:

- First time login with "admin" account
- Expired/Reset passwords

Use the "Change Password" hyperlink on this page to change the password manually, and then login.

Also note that single sign-on between servers in the same security domain is not supported when accessing via IP address.

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.

User ID:

Password:

Log On

[Change Password](#)

Supported Browsers: Internet Explorer 8.x, 9.x or 10.x or Firefox 15.0, 16.0 or 17.0.

Once logged in select the **Routing** link under the **Elements** column.

AVAYA

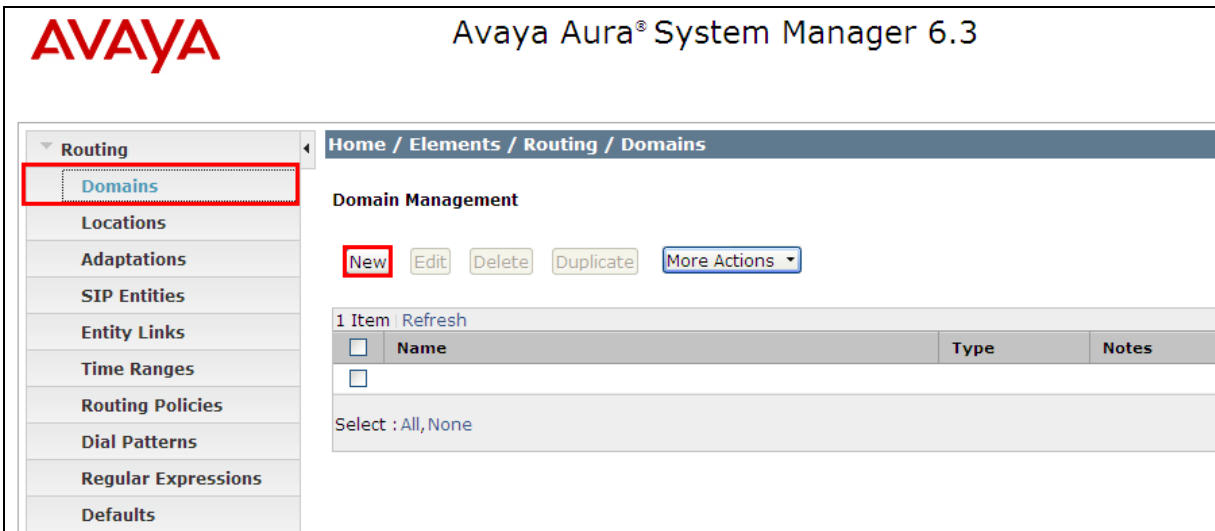
Avaya Aura® System Manager 6.3

Last Logged on at August 29, 2013
[Help](#) | [About](#) | [Change Password](#) | [Log out](#)

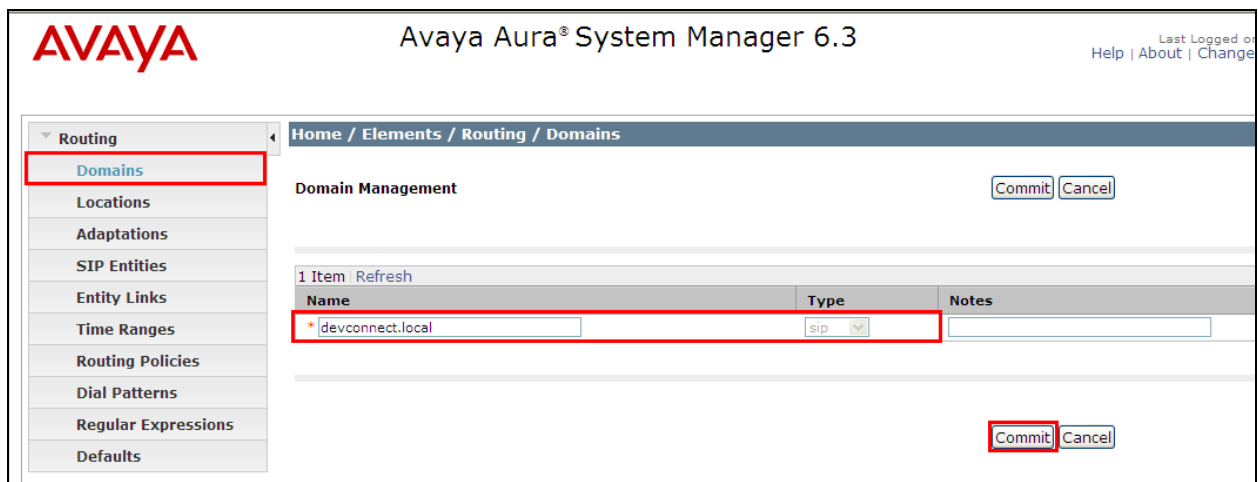
| Users | Elements | Services |
|---|---|---|
| Administrators Manage Administrative Users | Communication Manager Manage Communication Manager 5.2 and higher elements | Backup and Restore Backup and restore System Manager database |
| Directory Synchronization Synchronize users with the enterprise directory | Communication Server 1000 Manage Communication Server 1000 elements | Bulk Import and Export Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others |
| Groups & Roles Manage groups, roles and assign roles to users | Conferencing Manage Conferencing Multimedia Server objects | Configurations Manage system wide configurations |
| User Management Manage users, shared user resources and provision users | IP Office Manage IP Office elements | Events Manage alarms, view and harvest logs |
| | Meeting Exchange Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements | Geographic Redundancy Manage Geographic Redundancy |
| | Messaging Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging | Inventory Manage, discover, and navigate to elements |
| | Presence Presence | Licenses View and configure licenses |
| | Routing Session Manager Routing Administration | Replication Track data replication nodes, repair replication nodes |
| | Session Manager | Scheduler Schedule, track, cancel, update and delete jobs |

7.1. Administer SIP Domain

SIP domains are created as part of the Session Manager basic configuration. There will be at least one for which System Manager is the authoritative SIP controller. In these sample notes it is **devconnect.local**. In the left column click on **Routing** → **Domains**, click on **New** in the main window.

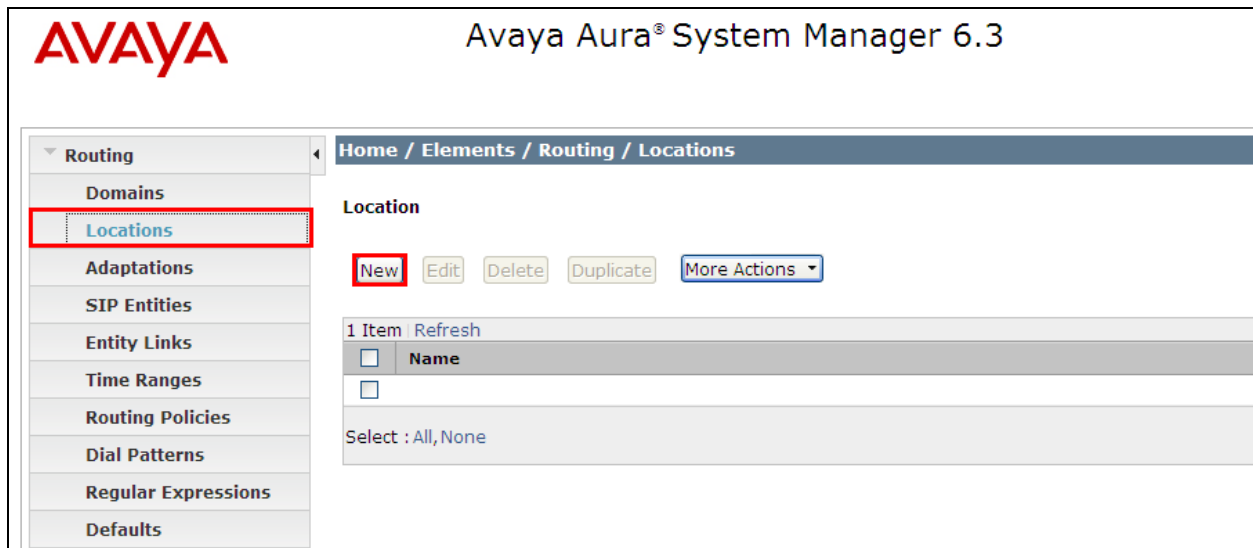


Enter the domain **Name**, set the **Type** as **sip** and click **Commit**.



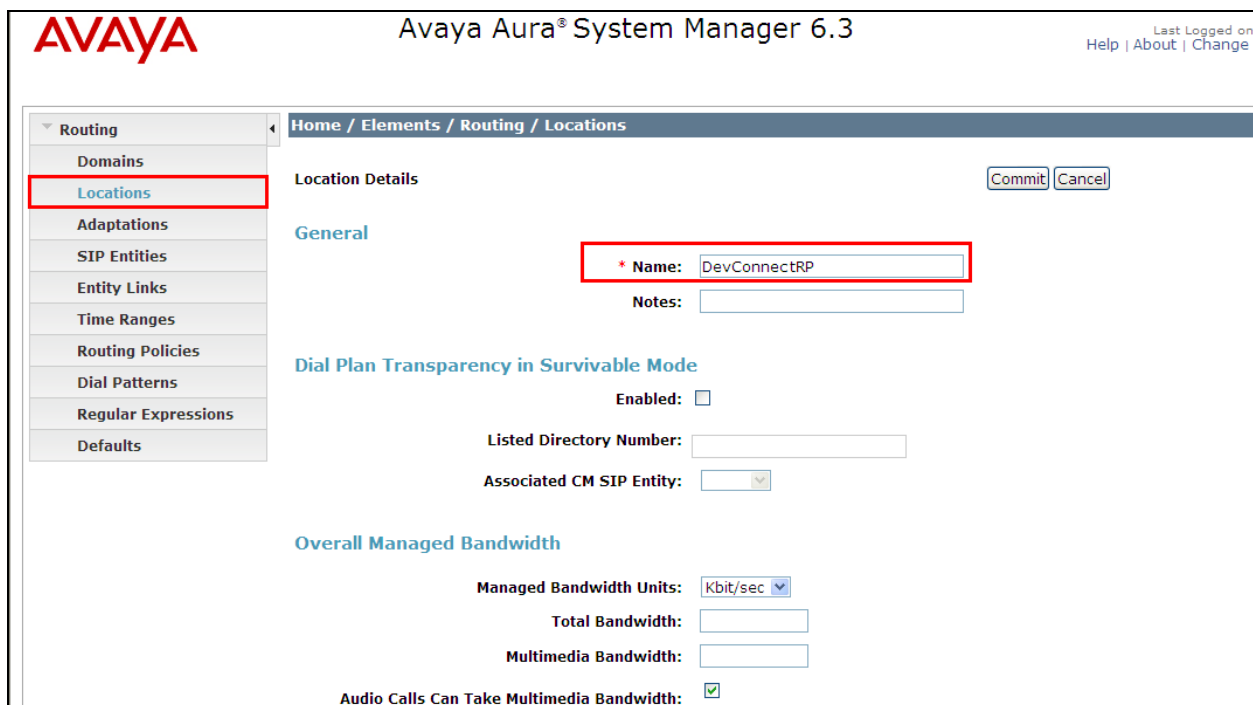
7.2. Administer Location

Session Manager uses the origination location to determine which dial patterns to look at when routing a call. In the left window under **Routing** select **Locations**, in the main window click on **New**.



The screenshot shows the Avaya Aura System Manager 6.3 interface. On the left, a navigation menu under 'Routing' has 'Locations' highlighted with a red box. The main area shows the 'Location' page with a breadcrumb 'Home / Elements / Routing / Locations'. Below the breadcrumb are buttons: 'New' (highlighted with a red box), 'Edit', 'Delete', 'Duplicate', and 'More Actions'. A table below shows '1 Item' with a 'Refresh' link. The table has a checkbox and a 'Name' column. Below the table is a 'Select : All, None' dropdown.

Enter an identifying **Name**, as shown below.



The screenshot shows the 'Location Details' page in Avaya Aura System Manager 6.3. The left navigation menu has 'Locations' highlighted with a red box. The main area shows the 'Location Details' page with a breadcrumb 'Home / Elements / Routing / Locations'. Below the breadcrumb are 'Commit' and 'Cancel' buttons. The 'General' section has a red box around the '* Name: DevConnectRP' field. Below it is a 'Notes' field. The 'Dial Plan Transparency in Survivable Mode' section has an 'Enabled' checkbox. Below it are fields for 'Listed Directory Number' and 'Associated CM SIP Entity'. The 'Overall Managed Bandwidth' section has a 'Managed Bandwidth Units' dropdown set to 'Kbit/sec', and fields for 'Total Bandwidth' and 'Multimedia Bandwidth'. At the bottom, the 'Audio Calls Can Take Multimedia Bandwidth' checkbox is checked.

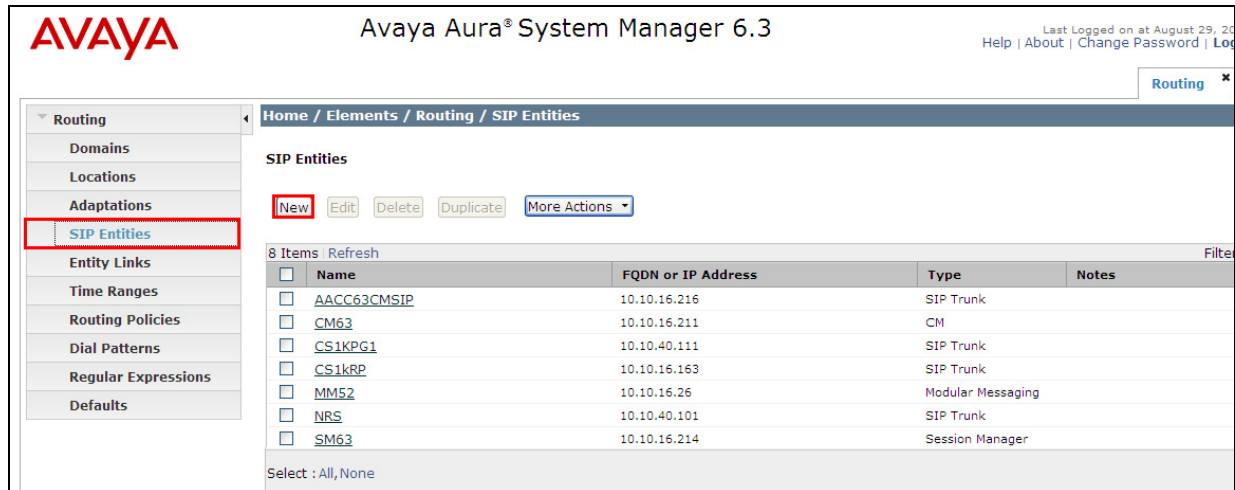
Scroll down to the bottom of the same page where the **Location Pattern** is defined. Click **Add** and enter the IP address range used to logically identify the location. In this case the **IP Address Pattern** is **10.10.16.*** as shown below. Click **Commit** when done.

The screenshot displays a configuration interface with the following sections and controls:

- Bandwidth Settings:**
 - Maximum Multimedia Bandwidth (Inter-Location): 1000 Kbit/Sec
 - * Minimum Multimedia Bandwidth: 64 Kbit/Sec
 - * Default Audio Bandwidth: 80 Kbit/sec (dropdown)
- Alarm Threshold:**
 - Overall Alarm Threshold: 80 %
 - Multimedia Alarm Threshold: 80 %
 - * Latency before Overall Alarm Trigger: 5 Minutes
 - * Latency before Multimedia Alarm Trigger: 5 Minutes
- Location Pattern:**
 - Buttons: Add, Remove
 - 1 Item Refresh
 - Table with columns: ☐ IP Address Pattern, Notes
 - Row 1: ☐ * 10.10.16.*
 - Select: All, None
 - Buttons: Commit, Cancel

7.3. Create Trio Enterprise SIP Entity

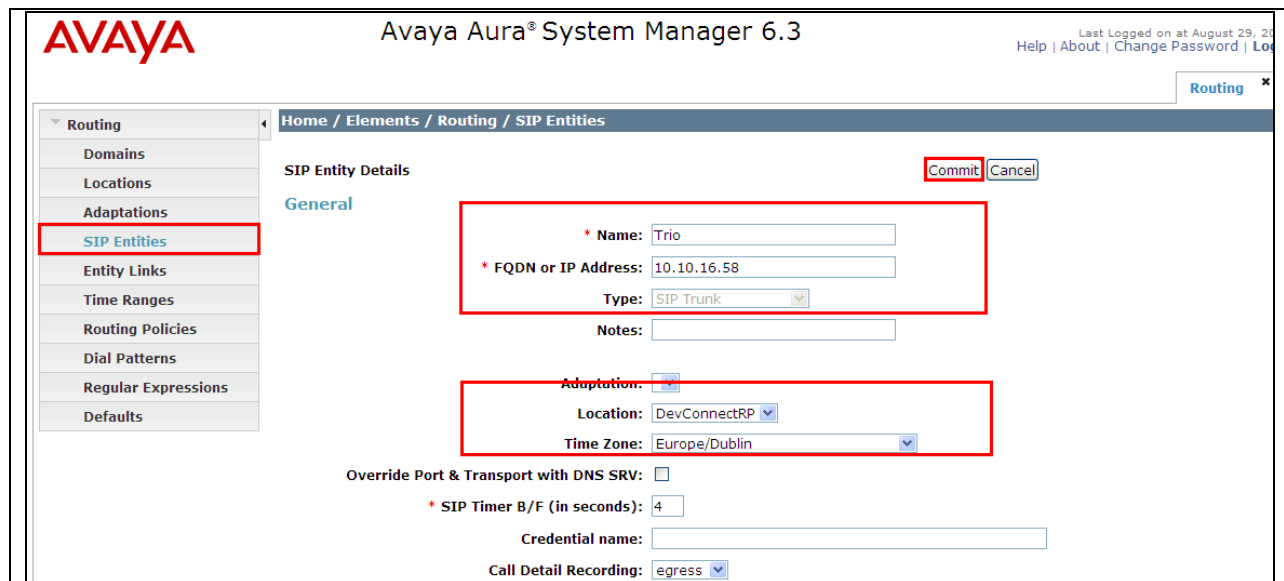
A SIP Entity must be added for Trio Enterprise Endpoint. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button in the main window.



Enter the following for the Trio SIP Entity.

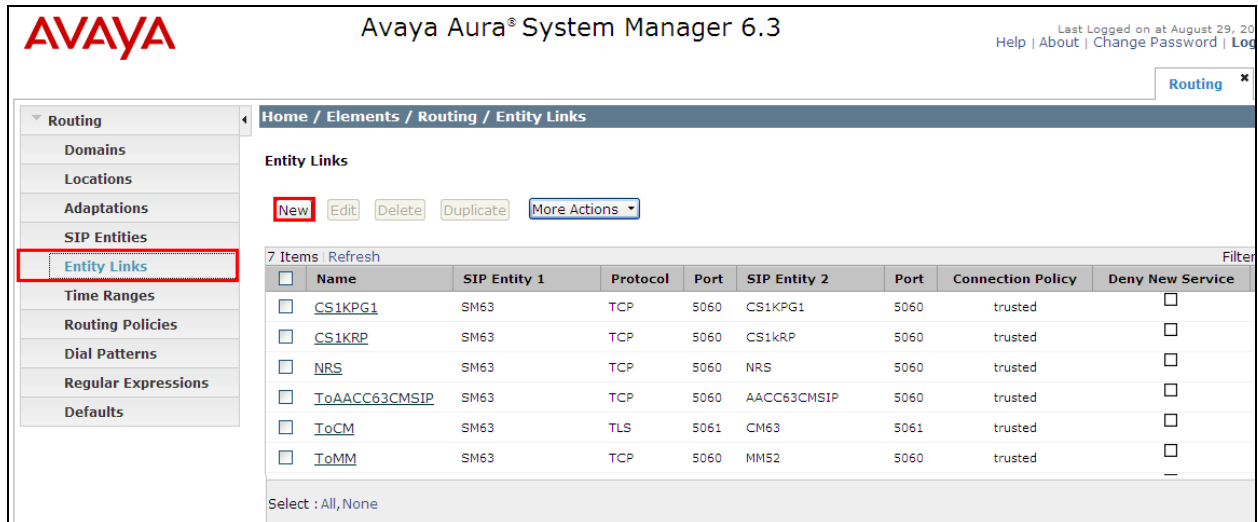
- **Name** An informative name (e.g., **Trio**)
- **FQDN or IP Address** IP address of the signalling interface on the Trio Enterprise
- **Type** **SIP Trunk** for Trio
- **Location** Location created in **Section 7.2**
- **Time Zone** Time zone for this location **Europe/Dublin**

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



7.4. Create Entity Link to Trio Enterprise

To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button in the main window.



Avaya Aura® System Manager 6.3

Home / Elements / Routing / Entity Links

Entity Links

New Edit Delete Duplicate More Actions

7 Items Refresh

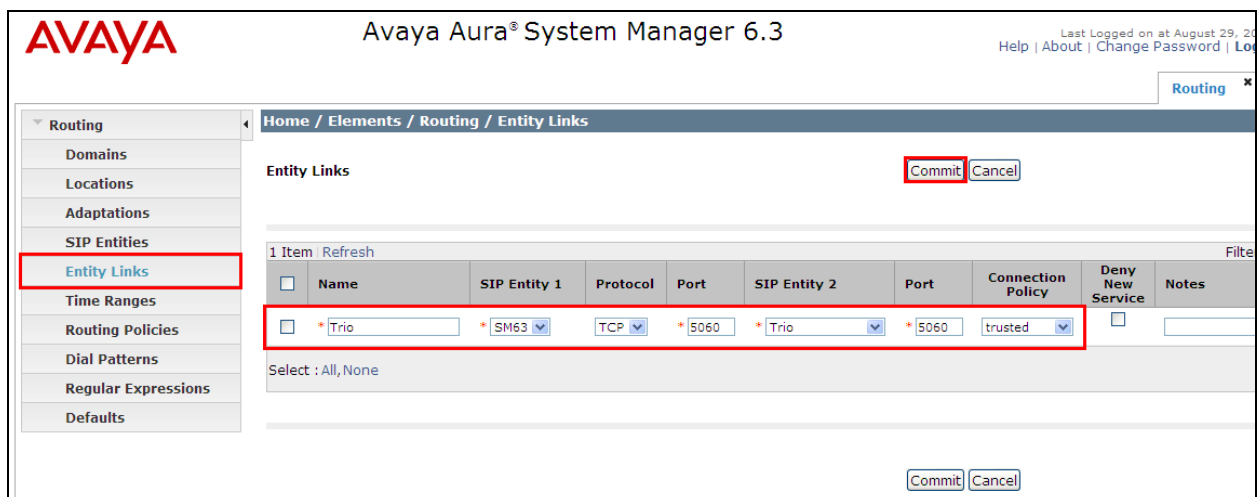
| <input type="checkbox"/> | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Connection Policy | Deny New Service |
|--------------------------|---------------|--------------|----------|------|--------------|------|-------------------|--------------------------|
| <input type="checkbox"/> | CS1KPG1 | SM63 | TCP | 5060 | CS1KPG1 | 5060 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | CS1KRP | SM63 | TCP | 5060 | CS1KRP | 5060 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | NRS | SM63 | TCP | 5060 | NRS | 5060 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | ToAACC63CMSIP | SM63 | TCP | 5060 | AACC63CMSIP | 5060 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | ToCM | SM63 | TLS | 5061 | CM63 | 5061 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | ToMM | SM63 | TCP | 5060 | MM52 | 5060 | trusted | <input type="checkbox"/> |

Select : All, None

Fill in the following fields in the new row that is displayed.

- **Name** An informative name, e.g. **Trio**
- **SIP Entity 1** Select **Session_Manager** from the **SIP Entity 1** dropdown box
- **Protocol** Select **TCP** from the Protocol
- **Port** Enter **5060** as the Port
- **SIP Entity 2** Select **Trio** from the **SIP Entity 2** dropdown box
- **Port** Enter **5060** as the Port
- **Trusted** Check the **Trusted** check box

Click **Commit** to save changes. The following screen shows the Entity Links used.



Avaya Aura® System Manager 6.3

Home / Elements / Routing / Entity Links

Entity Links

Commit Cancel

1 Item Refresh

| <input type="checkbox"/> | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Connection Policy | Deny New Service | Notes |
|--------------------------|-------|--------------|----------|-------|--------------|-------|-------------------|--------------------------|-------|
| <input type="checkbox"/> | *Trio | *SM63 | TCP | *5060 | *Trio | *5060 | trusted | <input type="checkbox"/> | |

Select : All, None

Commit Cancel

7.5. Create Routing Policy for Trio Enterprise

Create routing policies to direct calls to Trio Enterprise. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button in the main window.

The screenshot shows the Avaya Aura System Manager 6.3 interface. On the left, the 'Routing' menu is expanded, and 'Routing Policies' is highlighted. The main window displays the 'Routing Policies' section with a table of existing policies. The 'New' button is highlighted with a red box.

| Name | Disabled | Retries | Destination | Notes |
|-------------|--------------------------|---------|-------------|-------|
| CS1KPG1 | <input type="checkbox"/> | 0 | CS1KPG1 | |
| NRS | <input type="checkbox"/> | 0 | NRS | |
| TOAACC63SIP | <input type="checkbox"/> | 0 | AACC63CMSIP | |
| ToCM6.3 | <input type="checkbox"/> | 0 | CM63 | |
| ToCS1KRP | <input type="checkbox"/> | 0 | CS1kRP | |
| ToMM | <input type="checkbox"/> | 0 | MM52 | |

Enter an informative name, (e.g., **Trio**), under **SIP Entity as Destination**, click **Select**.

The screenshot shows the 'Routing Policy Details' form in Avaya Aura System Manager 6.3. The 'General' tab is active, and the 'Name' field is highlighted with a red box and contains the text 'Trio'. The 'SIP Entity as Destination' section has a 'Select' button highlighted with a red box. The 'Time of Day' section is also visible.

Routing Policy Details

General

* Name:

Disabled: ☐

* Retries:

Notes:

SIP Entity as Destination

| Name | FQDN or IP Address | Type | Notes |
|------|--------------------|------|-------|
|------|--------------------|------|-------|

Time of Day

Check the **Trio** radio button and click on the **Select** button to confirm the chosen options and then be returned to the **Routing Policies Details** screen.

Avaya Aura® System Manager 6.3

Home / Elements / Routing / Routing Policies

SIP Entities Select Cancel

8 Items Refresh

| Name | FQDN or IP Address | Type | Notes |
|---------------------------------------|--------------------|-------------------|-------|
| AACC63CMSIP | 10.10.16.216 | SIP Trunk | |
| CM63 | 10.10.16.211 | CM | |
| CS1KPG1 | 10.10.40.111 | SIP Trunk | |
| CS1kRP | 10.10.16.163 | SIP Trunk | |
| MMS2 | 10.10.16.26 | Modular Messaging | |
| NRS | 10.10.40.101 | SIP Trunk | |
| SM63 | 10.10.16.214 | Session Manager | |
| <input checked="" type="radio"/> Trio | 10.10.16.58 | SIP Trunk | |

Select : None

Click the **Commit** button to save. The following screen shows the **Routing Policy Details** for calls to Trio.

Avaya Aura® System Manager 6.3

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 |
|------|--------------------|-----------|-------|
| Trio | 10.10.16.58 | SIP Trunk | |

Time of Day

Add Remove View Gaps/Overlaps

7.6. Create Trio Enterprise Dial Pattern

A dial pattern must be defined that will direct calls to Trio Enterprise. During testing there were several numbers used beginning with 40xx in conjunction with the domain name created in **Section 7.1** called devconnect.local. To configure the Trio Enterprise dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button in the main window.

Avaya Aura® System Manager 6.3

Home / Elements / Routing / Dial Patterns

Dial Patterns

New Edit Delete Duplicate More Actions

| Pattern | Min | Max | Emergency Call | Emergency Type | Emergency Priority | SIP Domain | Notes |
|---------|-----|-----|--------------------------|----------------|--------------------|------------------|-----------|
| 1 | 4 | 4 | <input type="checkbox"/> | | | devconnect.local | MWI TO C |
| 1997 | 4 | 4 | <input type="checkbox"/> | | | -ALL- | |
| 30 | 4 | 4 | <input type="checkbox"/> | | | devconnect.local | To CS1000 |
| 32 | 4 | 4 | <input type="checkbox"/> | | | devconnect.local | |
| 35 | 4 | 4 | <input type="checkbox"/> | | | devconnect.local | To AACC |

Select : All, None

Enter the following information.

- **Pattern** Dialed number or prefix **40**
- **Min** Minimum length of dialed number **4**
- **Max** Maximum length of dialed number **4**
- **SIP Domain** Domain name of **devconnect.local** was used in the compliance testing (See **Section 7.1**)

Under **Originating Locations and Routing Policy List**, click on **Add**.

Avaya Aura® System Manager 6.3

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

Commit **Cancel**

General

* Pattern: 40

* Min: 4

* Max: 4

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: devconnect.local

Notes:

Originating Locations and Routing Policies

Add **Remove**

0 Items Refresh

| Originating Location Name | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled | Routing Policy Destination | Routing P Notes |
|---------------------------|----------------------------|---------------------|------|-------------------------|----------------------------|-----------------|
|---------------------------|----------------------------|---------------------|------|-------------------------|----------------------------|-----------------|

Check the **DevconRP** check box, and under **Routing Policies** check **Trio** check box. Click on the **Select** button to confirm the chosen options and then be returned to the Dial Pattern screen (shown previously).

The screenshot shows the 'Dial Patterns' configuration page in Avaya Aura System Manager 6.3. On the left, a sidebar contains navigation links: Time Ranges, Routing Policies, **Dial Patterns** (highlighted), Regular Expressions, and Defaults. The main area is divided into two sections. The top section, titled '1 Item Refresh', contains a table with columns 'Name' and 'Notes'. The row for 'DevConnectRP' has its checkbox checked. Below this is a 'Select : All, None' dropdown. The bottom section, titled 'Routing Policies', contains a table with columns 'Name', 'Disabled', 'Destination', and 'Notes'. The row for 'Trio' has its checkbox checked. At the bottom right, there are 'Select' and 'Cancel' buttons, with 'Select' highlighted.

Click the **Commit** button to save.

The screenshot shows the 'Dial Pattern Details' configuration page in Avaya Aura System Manager 6.3. The left sidebar is the same as the previous screenshot, with 'Dial Patterns' highlighted. The main area has a breadcrumb trail: Home / Elements / Routing / Dial Patterns. Below this is the 'Dial Pattern Details' section with a 'Commit' button highlighted. The 'General' tab is active, showing fields for 'Pattern' (40), 'Min' (4), 'Max' (4), 'Emergency Call' (unchecked), 'Emergency Priority' (1), 'Emergency Type' (empty), 'SIP Domain' (devconnect.local), and 'Notes' (empty). Below this is the 'Originating Locations and Routing Policies' section, which includes 'Add' and 'Remove' buttons. A table shows one item: 'DevConnectRP' with 'Trio' as the 'Routing Policy Name'. At the bottom, there are 'Add' and 'Remove' buttons and a 'Filter' dropdown.

7.7. Create CS1000E SIP Entity

A SIP Entity may already be in place for the CS1000E but in the event there is none present the procedure is the same as that above for the Trio SIP Entity. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button in the main window.

The screenshot shows the Avaya Aura System Manager 6.3 interface. On the left, the 'Routing' menu is expanded, and 'SIP Entities' is highlighted with a red box. The main window displays the 'SIP Entities' page with a 'New' button highlighted in red. Below the buttons is a table with 8 items. The table has columns for Name, FQDN or IP Address, Type, and Notes.

| Name | FQDN or IP Address | Type | Notes |
|-------------|--------------------|-------------------|-------|
| AACC63CMSIP | 10.10.16.216 | SIP Trunk | |
| CM63 | 10.10.16.211 | CM | |
| CS1KPG1 | 10.10.40.111 | SIP Trunk | |
| CS1kRP | 10.10.16.163 | SIP Trunk | |
| MM52 | 10.10.16.26 | Modular Messaging | |
| NRS | 10.10.40.101 | SIP Trunk | |
| SM63 | 10.10.16.214 | Session Manager | |

Enter the following for the CS1000E SIP Entity.

- **Name** An informative name (e.g., **CS1KPG1**)
- **FQDN or IP Address** CS1000E Node IP address as per **Section 6**
- **Type** **SIP Trunk** for the CS1000E
- **Location** Location created in **Section 7.2**
- **Time Zone** Time zone for this location **Europe/Dublin**

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

The screenshot shows the 'SIP Entity Details' form in Avaya Aura System Manager 6.3. The 'General' tab is selected. The 'Name' field is set to 'CS1KPG1' and the 'FQDN or IP Address' field is set to '10.10.40.111'. The 'Type' dropdown is set to 'SIP Trunk'. The 'Location' dropdown is set to 'DevConnectRP' and the 'Time Zone' dropdown is set to 'Europe/Dublin'. The 'Commit' button is highlighted in red.

SIP Entity Details

General

* Name: CS1KPG1

* FQDN or IP Address: 10.10.40.111

Type: SIP Trunk

Notes:

Adaptation: [v]

Location: DevConnectRP

Time Zone: Europe/Dublin

Override Port & Transport with DNS SRV: [x]

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

Credential name:

Call Detail Recording: egress

7.8. Create an Entity link for the CS1000E

To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button in the main window.

Avaya Aura® System Manager 6.3

Home / Elements / Routing / Entity Links

Entity Links

New Edit Delete Duplicate More Actions

7 Items Refresh

| | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Connection Policy | Deny New Service |
|--------------------------|---------------|--------------|----------|------|--------------|------|-------------------|--------------------------|
| <input type="checkbox"/> | CS1KPG1 | SM63 | TCP | 5060 | CS1KPG1 | 5060 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | CS1KRP | SM63 | TCP | 5060 | CS1kRP | 5060 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | NRS | SM63 | TCP | 5060 | NRS | 5060 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | ToAACC63CMSIP | SM63 | TCP | 5060 | AACC63CMSIP | 5060 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | ToCM | SM63 | TLS | 5061 | CM63 | 5061 | trusted | <input type="checkbox"/> |
| <input type="checkbox"/> | ToMM | SM63 | TCP | 5060 | MM52 | 5060 | trusted | <input type="checkbox"/> |

Select : All, None

Fill in the following fields in the new row that is displayed.

- **Name** An informative name, e.g. **CS1KPG1**
- **SIP Entity 1** Select **Session_Manager** from the **SIP Entity 1** dropdown box
- **Protocol** Select **TCP** from the Protocol
- **Port** Enter **5060** as the Port
- **SIP Entity 2** Select **Trio** from the **SIP Entity 2** dropdown box
- **Port** Enter **5060** as the Port
- **Trusted** Check the **Trusted** check box

Click **Commit** to save changes. The following screen shows the Entity Links used.

Avaya Aura® System Manager 6.3

Home / Elements / Routing / Entity Links

Entity Links

Commit Cancel

1 Item Refresh

| | Name | SIP Entity 1 | Protocol | Port | SIP Entity 2 | Port | Connection Policy | Deny New Service |
|--------------------------|-----------|--------------|----------|--------|--------------|--------|-------------------|--------------------------|
| <input type="checkbox"/> | * CS1KPG1 | * SM63 | TCP | * 5060 | * CS1KPG1 | * 5060 | trusted | <input type="checkbox"/> |

Select : All, None

Commit Cancel

7.9. Create Routing Policy for CS1000E

Create routing policies to direct calls to the CS1000E. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button in the main window.

The screenshot shows the Avaya Aura System Manager 6.3 interface. On the left, the 'Routing' menu is expanded, and 'Routing Policies' is selected. The main window displays the 'Routing Policies' list with columns: Name, Disabled, Retries, Destination, and Notes. The list contains seven items: CS1KPG1, NRS, TOAAC63SIP, ToCM6.3, ToCS1KRP, and ToMM. The 'New' button is highlighted in the top left of the main window.

| Name | Disabled | Retries | Destination | Notes |
|------------|--------------------------|---------|-------------|-------|
| CS1KPG1 | <input type="checkbox"/> | 0 | CS1KPG1 | |
| NRS | <input type="checkbox"/> | 0 | NRS | |
| TOAAC63SIP | <input type="checkbox"/> | 0 | AACC63CMSIP | |
| ToCM6.3 | <input type="checkbox"/> | 0 | CM63 | |
| ToCS1KRP | <input type="checkbox"/> | 0 | CS1kRP | |
| ToMM | <input type="checkbox"/> | 0 | MM52 | |

Enter an informative name, (e.g., **CS1KPG1**), under **SIP Entity as Destination**, click **Select**. Check the **CS1000E** radio button and click on the **Select** button to confirm the chosen options (not shown) and then be returned to the **Routing Policies Details** screen as shown below. Click the **Commit** button to save.

The screenshot shows the 'Routing Policy Details' screen for the policy 'CS1KPG1'. The 'General' section includes fields for Name (CS1KPG1), Disabled (checkbox), Retries (0), and Notes. The 'SIP Entity as Destination' section has a 'Select' button. Below this is a table with columns: Name, FQDN or IP Address, Type, and Notes. The table contains one entry: CS1KPG1, 10.10.40.111, SIP Trunk.

| Name | FQDN or IP Address | Type | Notes |
|---------|--------------------|-----------|-------|
| CS1KPG1 | 10.10.40.111 | SIP Trunk | |

7.10. Create CS1000E Dial Pattern

A dial pattern must be defined that will direct calls to the CS1000E. During testing there were several numbers used on the CS1000E beginning with 30xx. To configure the CS1000E dial pattern select **Dial Patterns** on the left panel menu and then click on the **New** button in the main window.

Avaya Aura® System Manager 6.3

Home / Elements / Routing / Dial Patterns

Dial Patterns

New Edit Delete Duplicate More Actions

8 Items Refresh

| Pattern | Min | Max | Emergency Call | Emergency Type | Emergency Priority | SIP Domain | Notes |
|---------|-----|-----|--------------------------|----------------|--------------------|------------------|-----------|
| 1 | 4 | 4 | <input type="checkbox"/> | | | devconnect.local | MWI TO C |
| 1997 | 4 | 4 | <input type="checkbox"/> | | | -ALL- | |
| 30 | 4 | 4 | <input type="checkbox"/> | | | devconnect.local | To CS1000 |
| 32 | 4 | 4 | <input type="checkbox"/> | | | devconnect.local | |
| 35 | 4 | 4 | <input type="checkbox"/> | | | devconnect.local | To AACC |

Select : All, None

Enter the following information.

- **Pattern** Dialed number or prefix **30**
- **Min** Minimum length of dialled number **4**
- **Max** Maximum length of dialled number **4**
- **SIP Domain** Domain name of **devconnect.local** was used in the compliance testing (See Section 7.1)

Under **Originating Locations and Routing Policy List**, click on **Add**.

Avaya Aura® System Manager 6.3

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

Commit Cancel

General

* Pattern: 30

* Min: 4

* Max: 4

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: devconnect.local

Notes: To CS1000E RP

Originating Locations and Routing Policies

Add Remove

0 Items Refresh

| Originating Location Name | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled | Routing Policy Destination | Routing P Notes |
|---------------------------|----------------------------|---------------------|------|-------------------------|----------------------------|-----------------|
|---------------------------|----------------------------|---------------------|------|-------------------------|----------------------------|-----------------|

Check the **DevconRP** check box (location setup in **Section 7.2**), and under **Routing Policies** check **CS1KPG1** check box. Click on the **Select** button to confirm the chosen options and then be returned to the **Dial Pattern** screen (shown previously).

Time Ranges
Routing Policies
Dial Patterns
Regular Expressions
Defaults

1 Item Refresh

| <input checked="" type="checkbox"/> | Name | Notes |
|-------------------------------------|--------------|-------|
| <input checked="" type="checkbox"/> | DevConnectRP | |

Select : All, None

Routing Policies

7 Items Refresh

| <input type="checkbox"/> | Name | Disabled | Destination | Notes |
|-------------------------------------|-------------|--------------------------|-------------|-------|
| <input checked="" type="checkbox"/> | CS1KPG1 | <input type="checkbox"/> | CS1KPG1 | |
| <input type="checkbox"/> | NRS | <input type="checkbox"/> | NRS | |
| <input type="checkbox"/> | TOAACC63SIP | <input type="checkbox"/> | AACC63CMSIP | |
| <input type="checkbox"/> | ToCM6.3 | <input type="checkbox"/> | CM63 | |
| <input type="checkbox"/> | ToCS1KRP | <input type="checkbox"/> | CS1kRP | |
| <input type="checkbox"/> | ToMM | <input type="checkbox"/> | MM52 | |
| <input type="checkbox"/> | Trio | <input type="checkbox"/> | Trio | |

Select : All, None

Select Cancel

Click the **Commit** button to save.

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

Commit Cancel

General

* Pattern: 30

* Min: 4

* Max: 4

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: devconnect.local

Notes: To CS1000E RP

Originating Locations and Routing Policies

Add Remove

1 Item Refresh

| <input type="checkbox"/> | Originating Location Name | Originating Location Notes | Routing Policy Name | Rank | Routing Policy Disabled | Routing Policy Destination |
|-------------------------------------|---------------------------|----------------------------|---------------------|------|--------------------------|----------------------------|
| <input checked="" type="checkbox"/> | DevConnectRP | | CS1KPG1 | | <input type="checkbox"/> | CS1KPG1 |

Select : All, None

8. Configure TRIO Enterprise

Trio Enterprise connects as a SIP endpoint to the CS1000E 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 5.3**. 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. The steps to configure SIP Trunks are as follows.

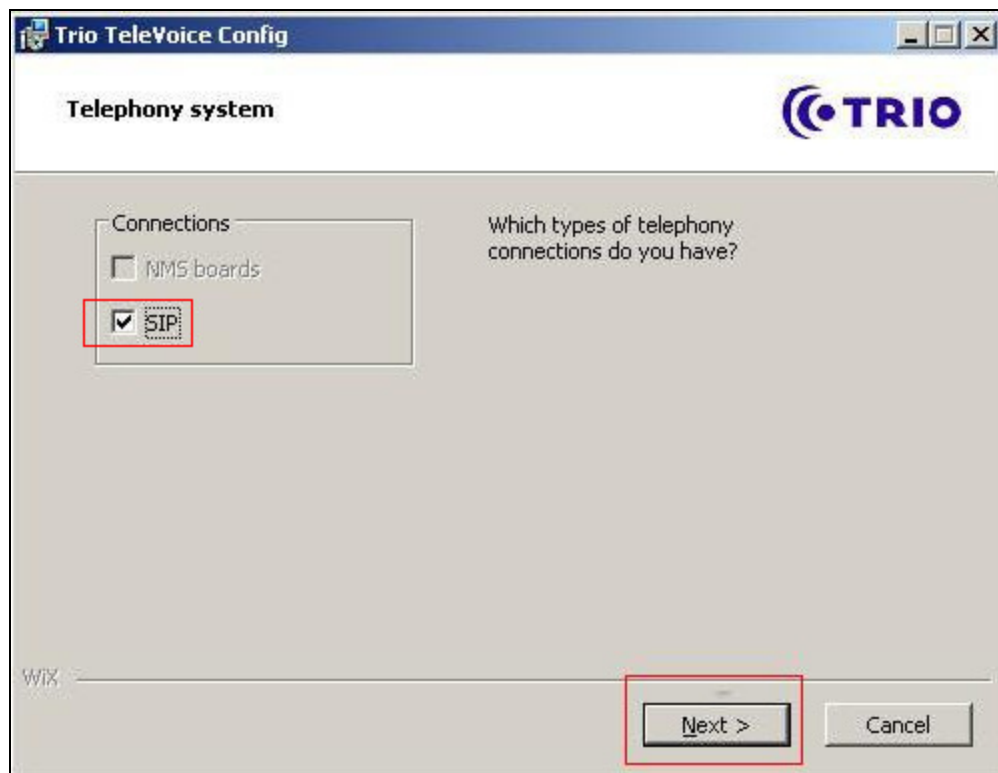
8.1. Configure Trio Enterprise to use SIP Trunks

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).

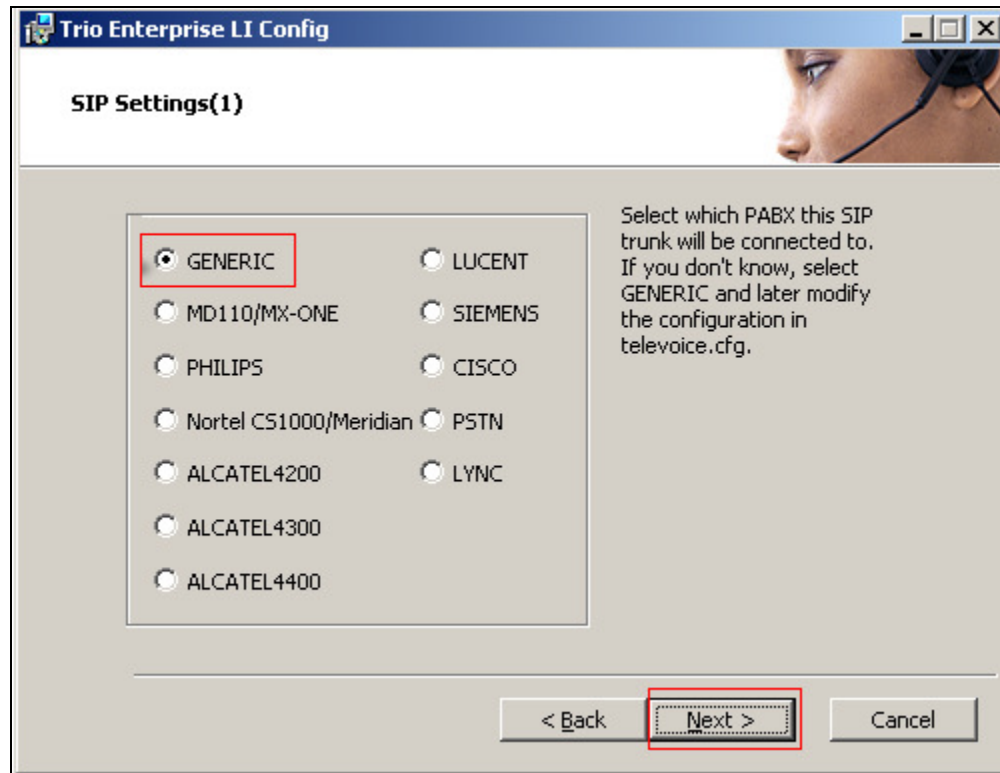
Launch the Trio configuration application. Select **Start → Programs → Trio Enterprise → Line Interface** and click on the **Config** entry (not shown). The configuration application starts up and presents the screenshot below.

- Ensure the **SIP** entry in the **Connections** area is checked.

Click **Next >** to continue.



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



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

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

Click **Next** to continue.

Trio Enterprise LI Config

SIP Settings(2)

SIP settings

Local IP: 10.10.16.58

Port: 5060

Target IP: 10.10.16.214

Port: 5060

Number of channels: 30

Codecs

☒ Enable G711 mu-law codec

TE 4.1.19

< Back Next > Cancel

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

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

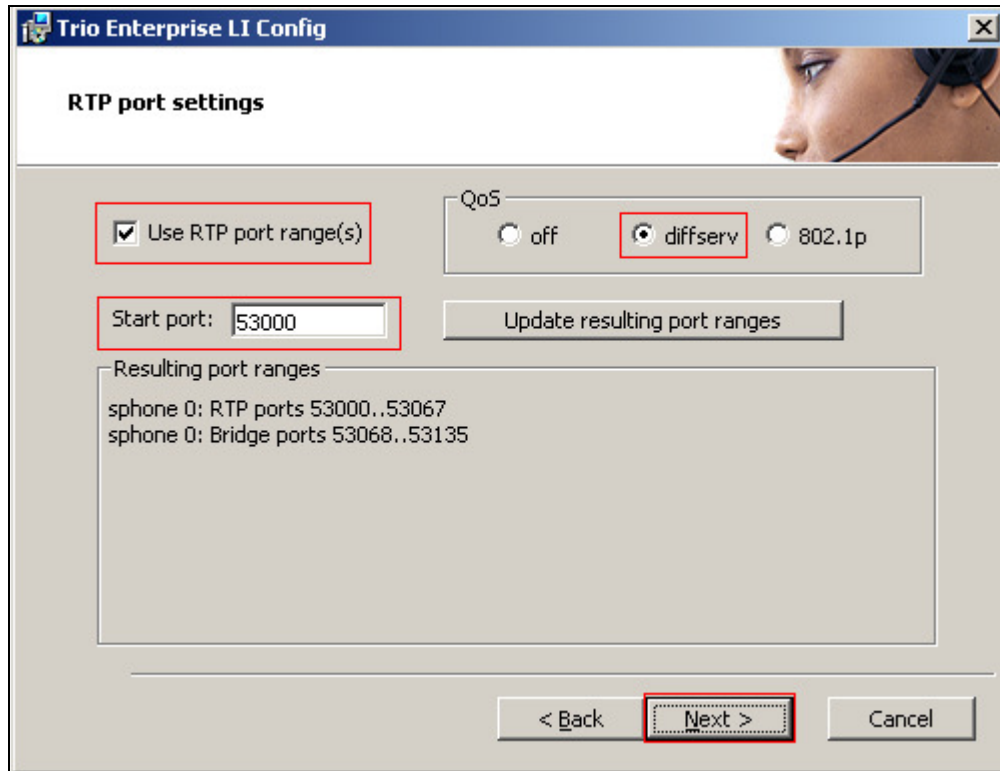
Click **Next** to continue.

The screenshot shows the 'Trio Enterprise LI Config' window with the 'SIP Settings(3)' tab active. The window has a blue title bar and a close button. The main content area is divided into two sections: 'Address Space (AS)' and 'Routing'. In the 'Address Space (AS)' section, there are three radio buttons: 'Use LI Address Space' (selected), 'AS Name:' (with an empty text box), and 'No Address Space'. In the 'Routing' section, there is a checked checkbox for 'Enable IP routing'. At the bottom of the window, there are three buttons: 'Additional SIP Trunk', '< Back', and 'Next >' (highlighted with a red box), and a 'Cancel' button.

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

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

Click **Next** to continue.



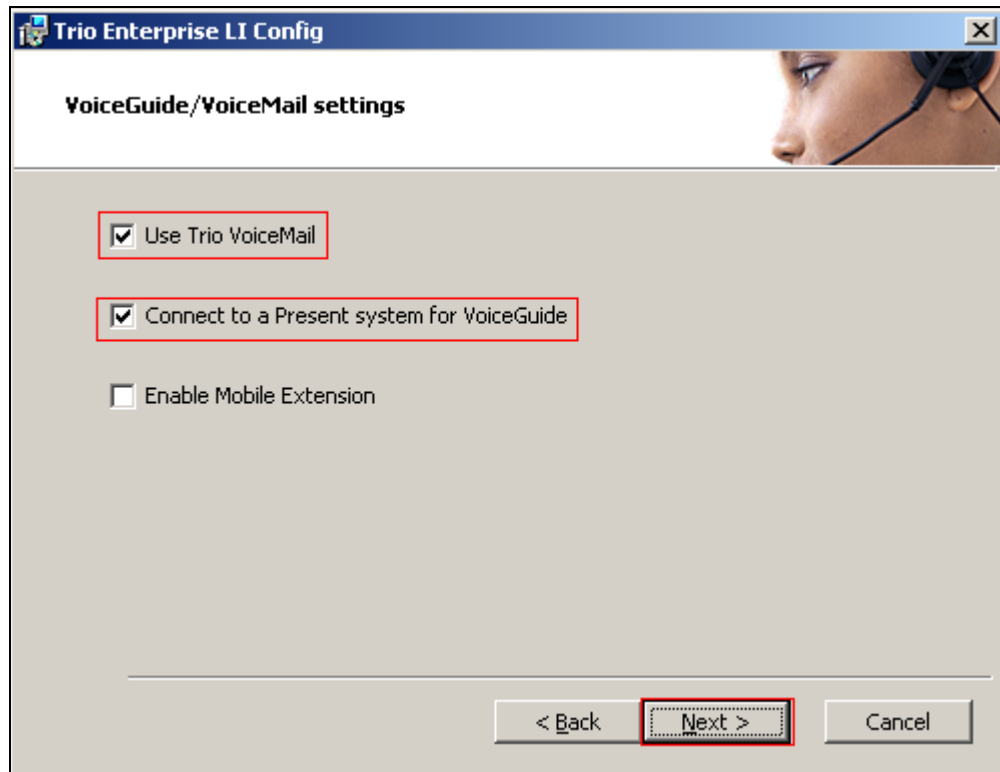
The screenshot shows the 'Trio Enterprise LI Config' window with the 'RTP port settings' tab selected. The window has a blue title bar and a header image of a person wearing a headset. The settings are as follows:

- Use RTP port range(s):** Checked (indicated by a red box).
- QoS:** Three radio buttons: 'off', 'diffserv' (selected, indicated by a red box), and '802.1p'.
- Start port:** A text box containing '53000' (indicated by a red box).
- Update resulting port ranges:** A button.
- Resulting port ranges:** A text area displaying:
 - sphone 0: RTP ports 53000..53067
 - sphone 0: Bridge ports 53068..53135
- Navigation:** At the bottom, there are three buttons: '< Back', 'Next >' (indicated by a red box), and 'Cancel'.

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

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

Click **Next** to continue.

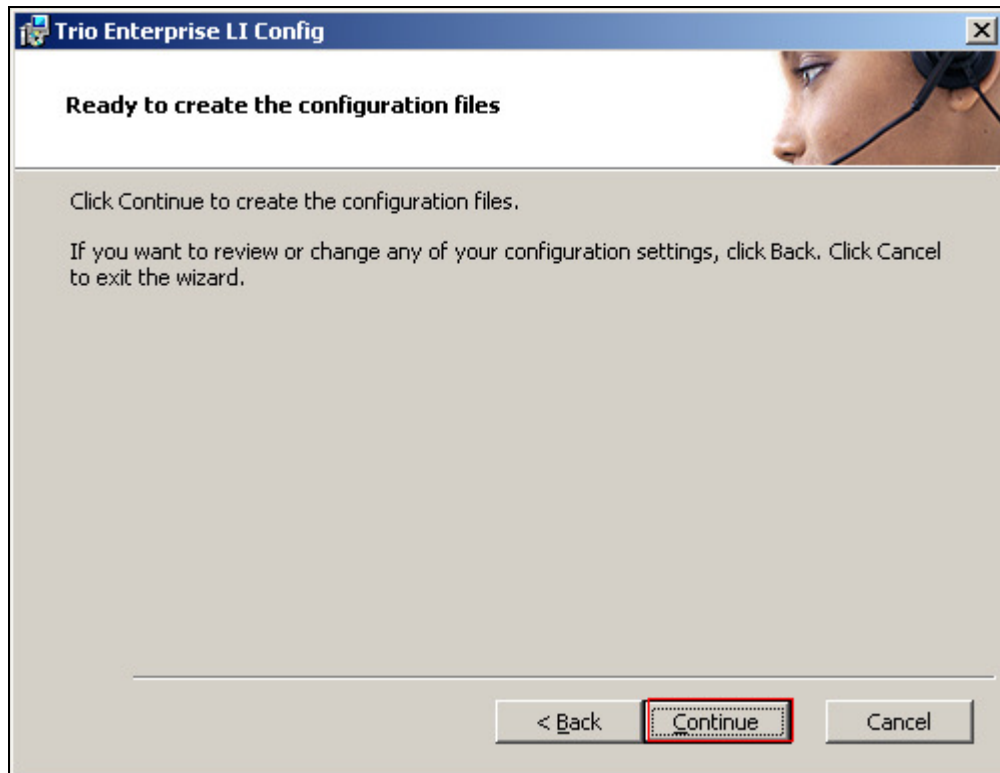


The screenshot shows a window titled "Trio Enterprise LI Config" with a close button in the top right corner. Below the title bar, there is a header area with the text "VoiceGuide/VoiceMail settings" and a small image of a person wearing a headset. The main content area contains three settings:

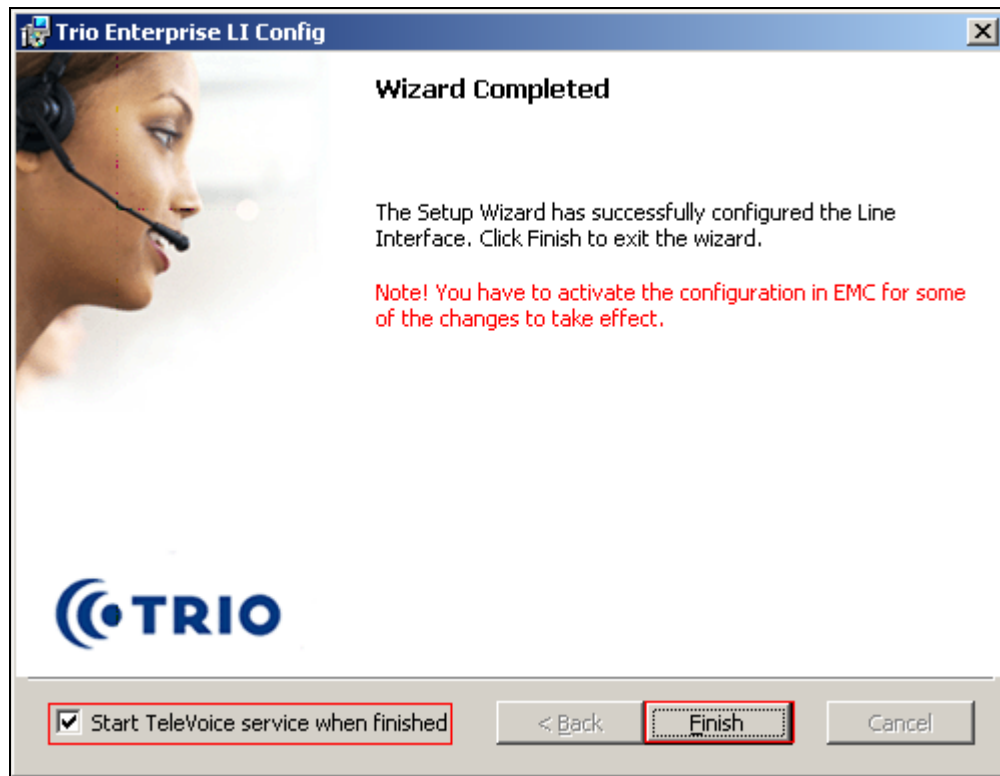
- ☒ Use Trio VoiceMail
- ☒ Connect to a Present system for VoiceGuide
- ☐ Enable Mobile Extension

At the bottom of the window, there are three buttons: "< Back", "Next >", and "Cancel". The "Next >" button is highlighted with a red border.

On the **Ready to create the configuration files** page click on **C**ontinue button.



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



8.2. InteractionStudio Configuration

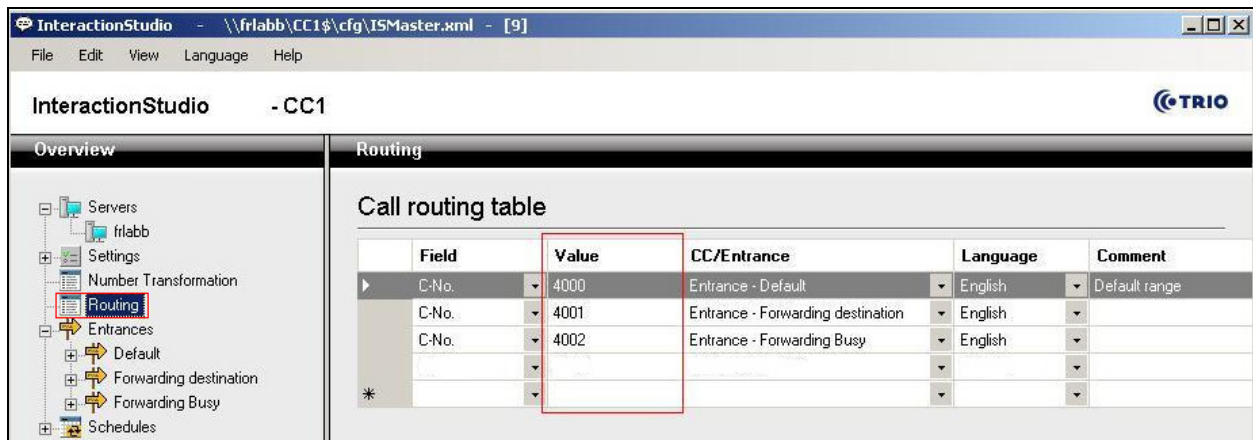
The InteractionStudio is used to configure many features for Trio Enterprise. For compliance testing, the following were configured.

- 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

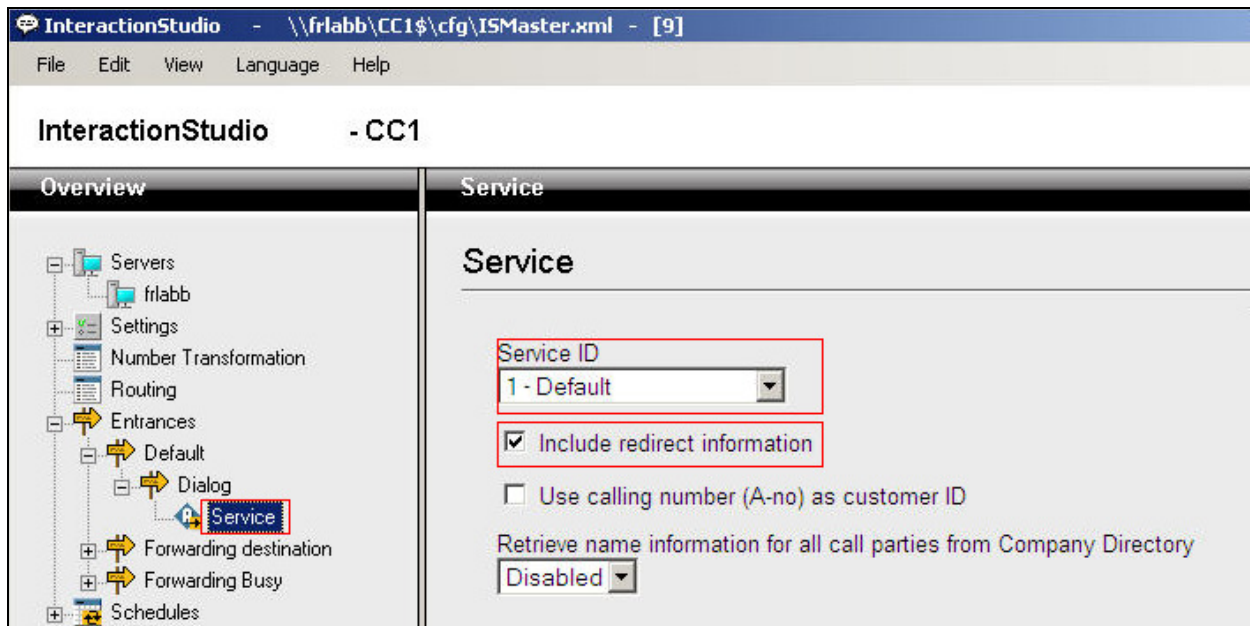
On the Trio Enterprise server, click the **Start** button → **Programs** → **Trio Enterprise** → **Contact Center** → **CC1** → **Interaction Studio** (not shown). When the InteractionStudio window opens, navigate to **Routing**. A **Call routing table** will open. In the example below:

- Extension **4000** is the main queue number
- Extension **4001** is the number that calls go to when Call forward No Answer is activated
- Extension **4002** is the number that calls go to when Call forward Busy is activated



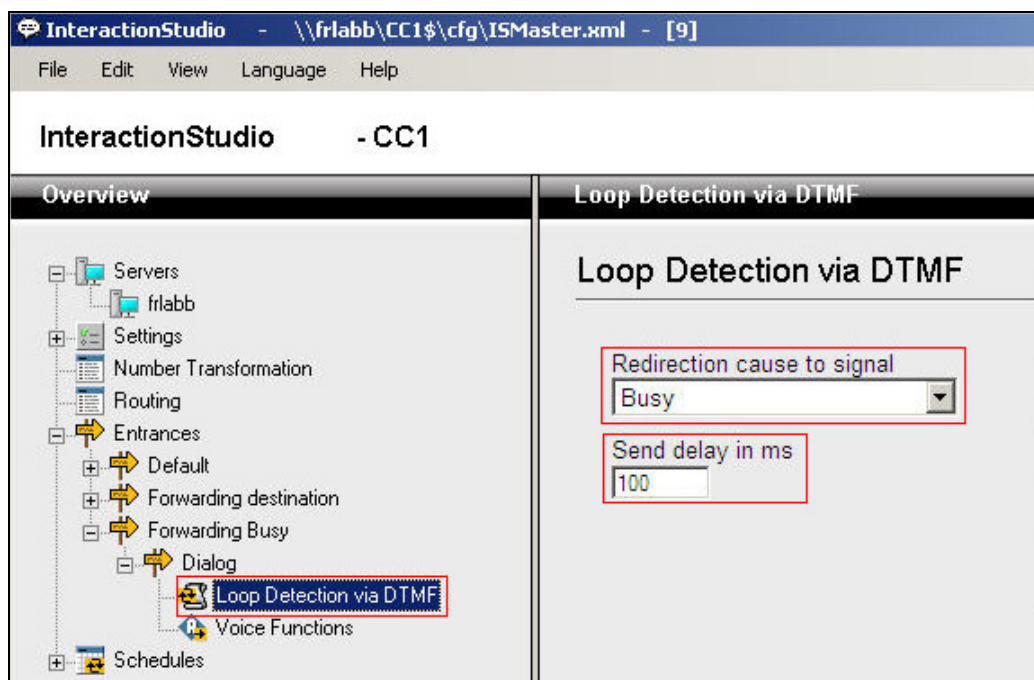
8.2.2. Configure Attendant Service

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



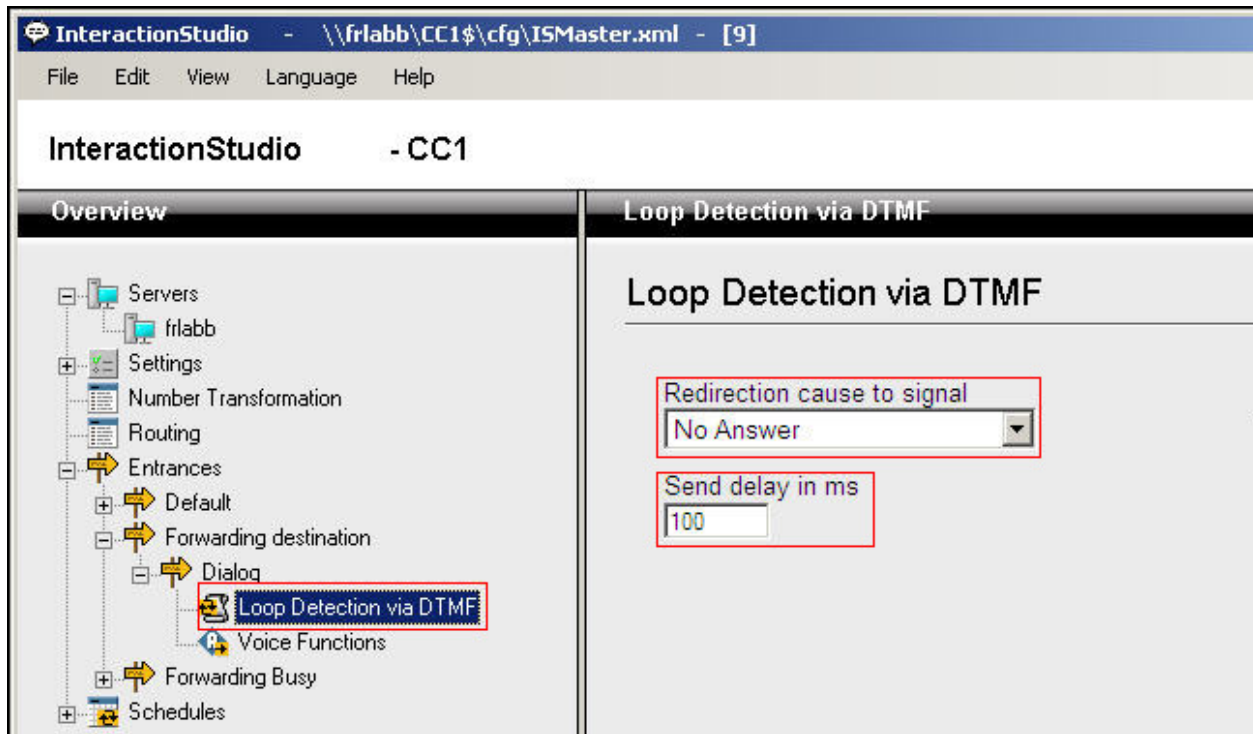
8.2.3. Configure Loop Detection via DTMF for Busy signal

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 to 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**, select the CS1000E telephone number that will be used as the agent's audio device (number **3032** in this example). Ensure the correct Trio Enterprise server is selected if there is more than one on the network (default is the current Trio server). Confirm **Phone type** is set to **Standard phone**. Click on the **OK** button when finished.

Trio Agent - Login

Trio Enterprise®

User ID: default

Password:

Extension: 3032

Server: trioserver.galctlab.com

Phone type: Standard phone

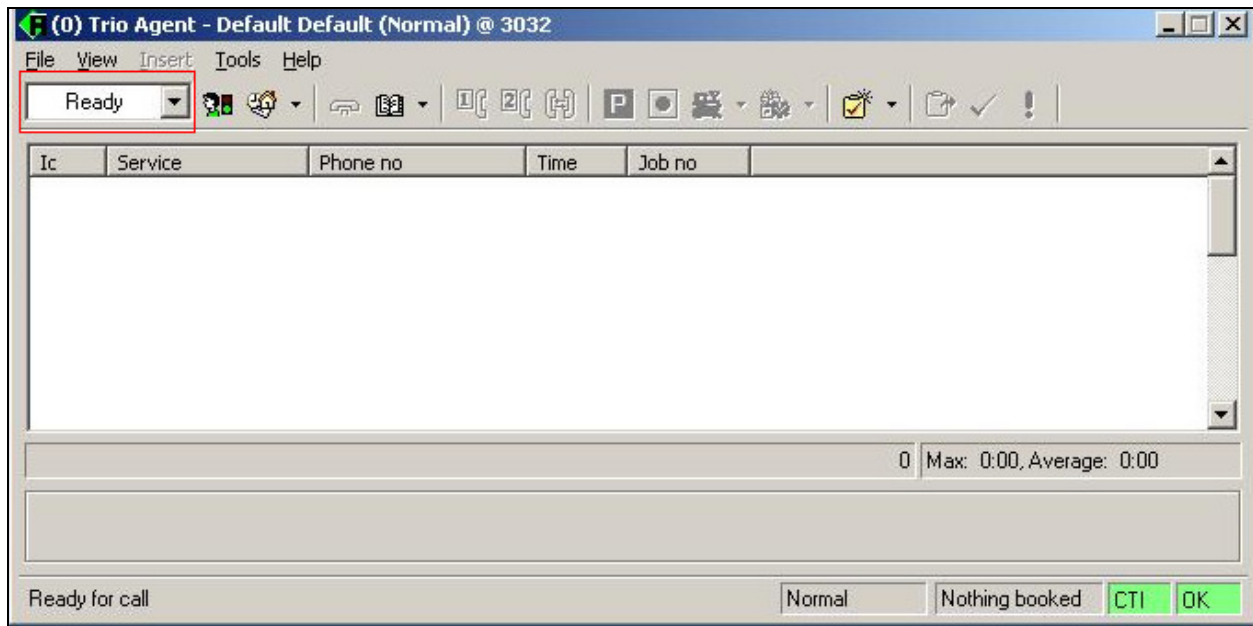
☐ Attach with Contact Center privileges

☐ Attach with Attendant privileges

OK Guest Cancel

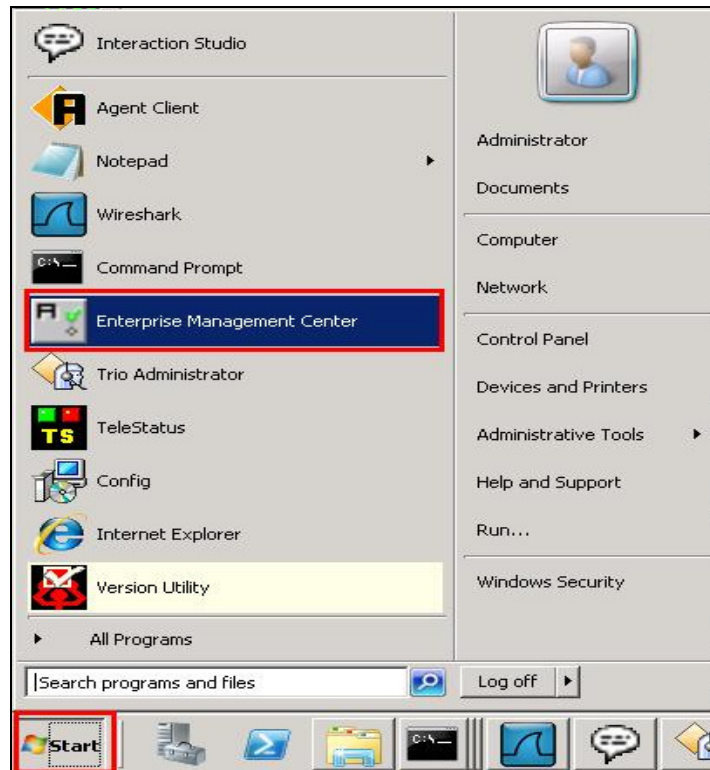
TRIO

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.

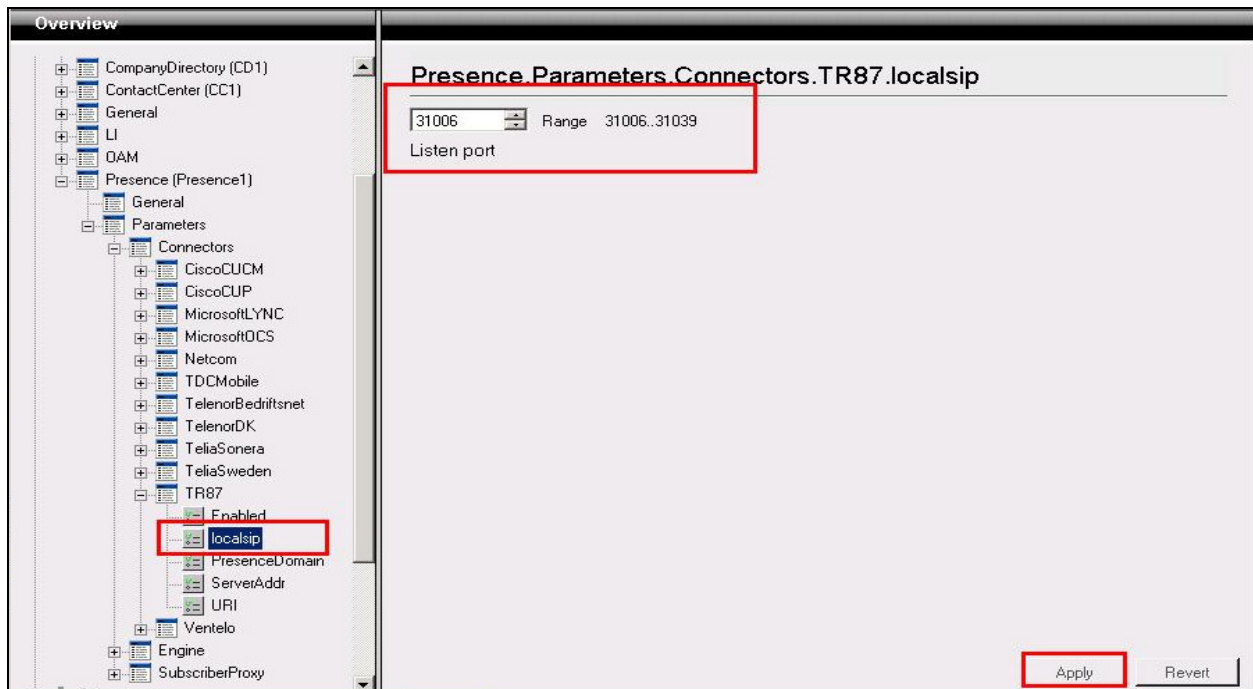
The screenshot shows the 'Overview' pane on the left with a tree view. The path 'Parameters' → 'Presence (Presence1)' → 'Parameters' → 'Connectors' → 'TR87' is highlighted with red boxes. The main pane on the right is titled 'Presence.Parameters.Connectors.TR87' and contains a table with the following data:

| Name | Value | Comment |
|----------------|------------------------|---------------------------------------|
| Enabled | true | Enable TR87 Presence Connector |
| localsip | 31006 | Listen port |
| PresenceDomain | 10.10.40.111 | Presence Domain name (domain...) |
| ServerAddr | 10.10.40.111 | TR87 server FQDN or IP Address |
| URI | sip.tetr87@10.10.16.58 | Uri of TE enterprise Server (defa...) |

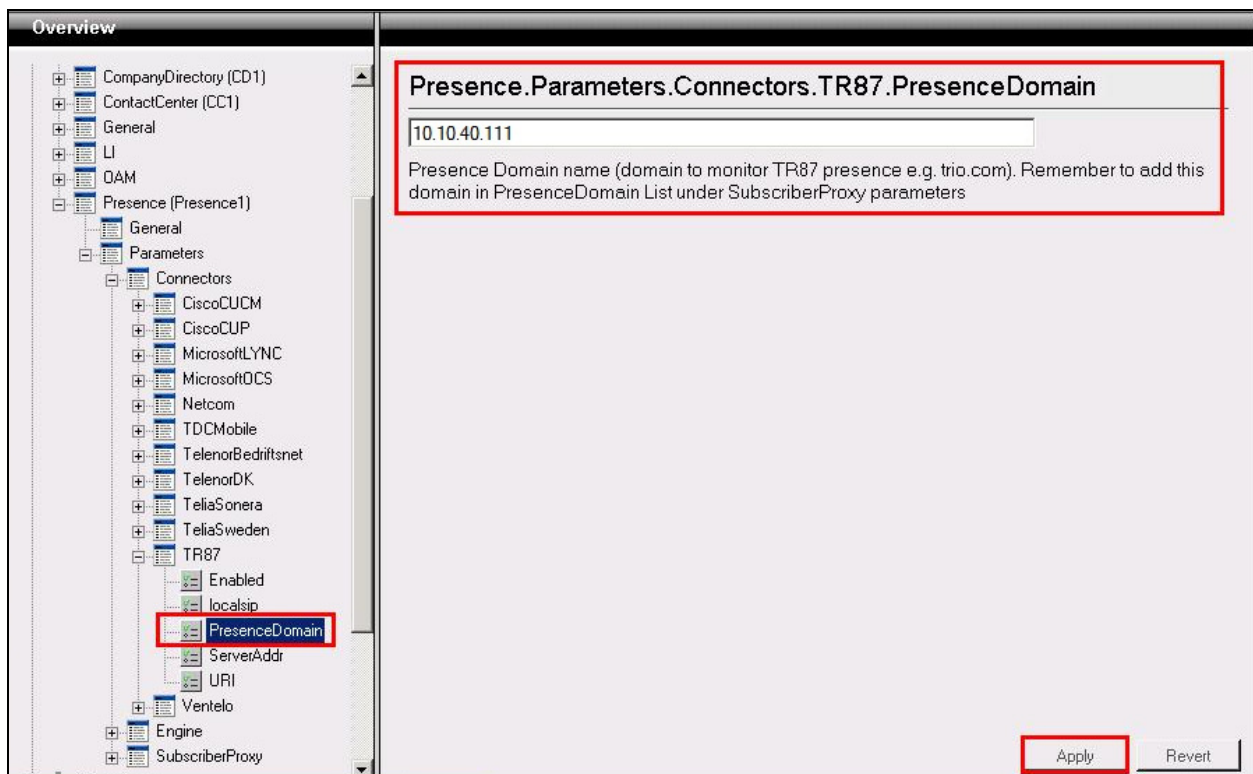
Under **TR87** select **Enabled** in the left window. Ensure that **Enable TR87 Presence Connector** is ticked as shown below. Click **Apply** to continue.

The screenshot shows the 'Overview' pane on the left with the tree view expanded to 'Parameters' → 'Connectors' → 'TR87'. The 'Enabled' checkbox is selected and highlighted with a red box. The main pane on the right is titled 'Presence.Parameters.Connectors.TR87.Enabled' and contains a single checkbox labeled 'Enable TR87 Presence Connector', which is also checked and highlighted with a red box. At the bottom right of the main pane, the 'Apply' button is highlighted with a red box.

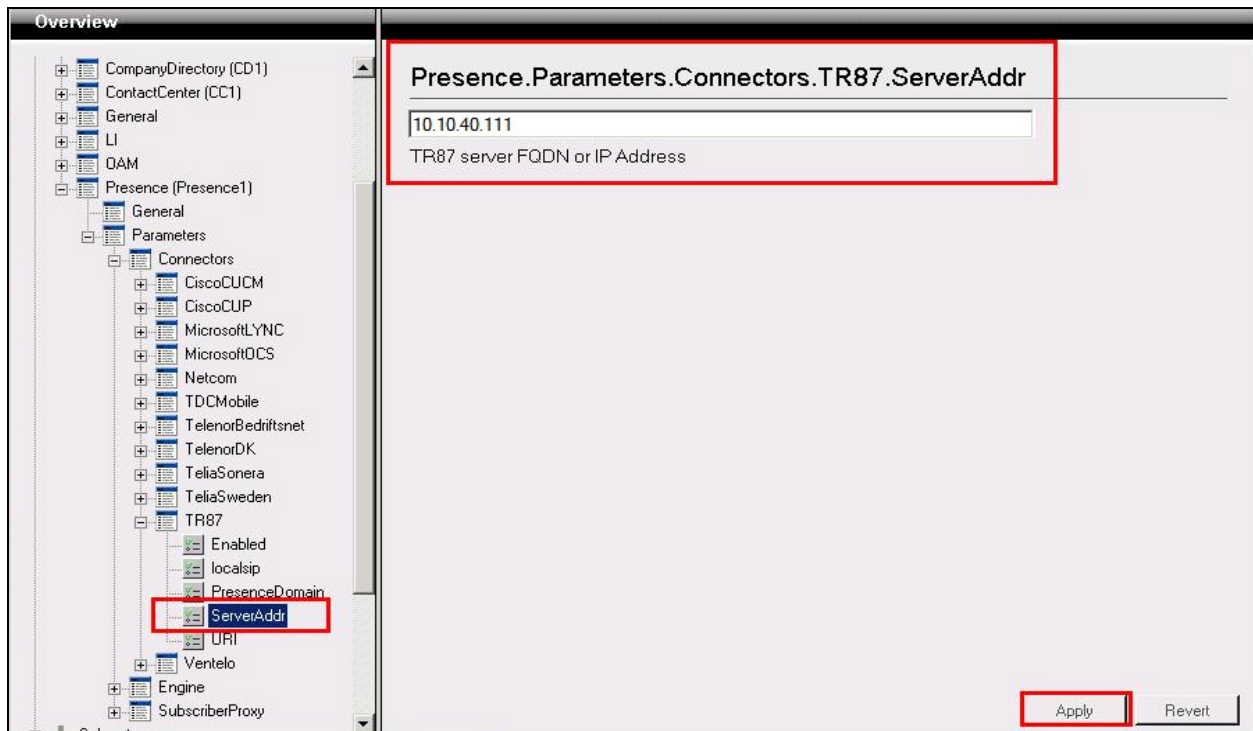
Select **localsip** under **TR87** in the left window and select the **Listen port** for TR87, for compliance testing this was left as default **31006** as shown below. Click **Apply** to continue.



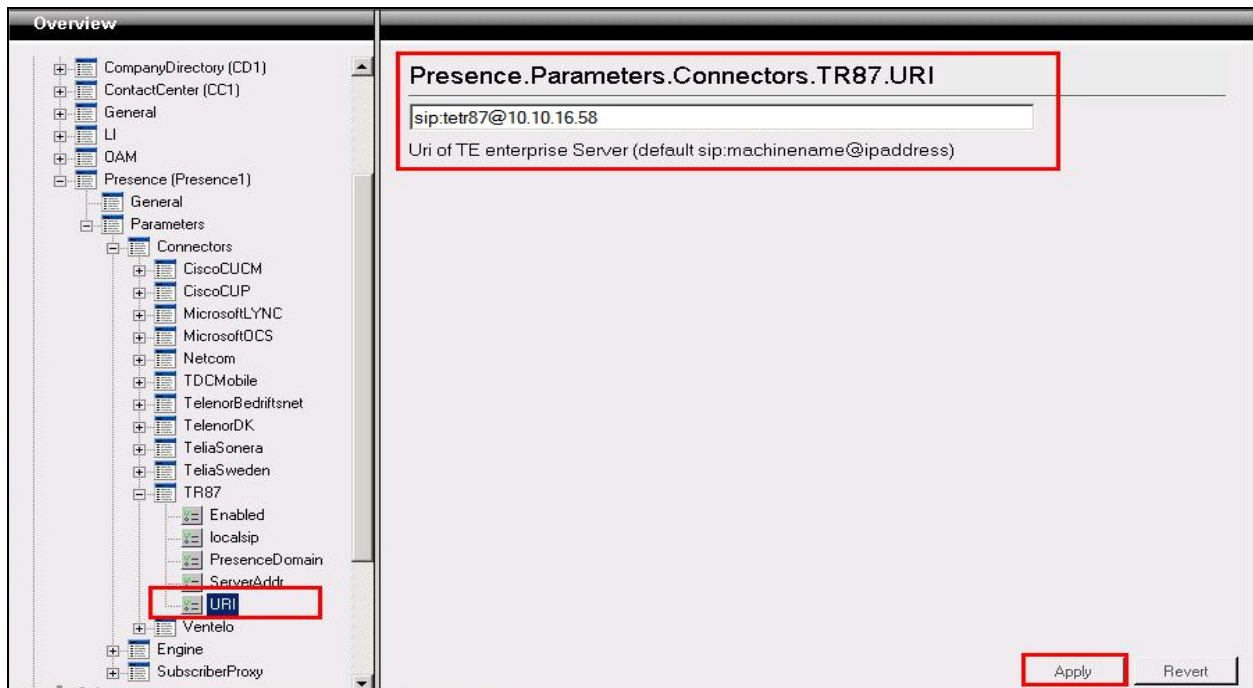
Select **PresenceDomain** under **TR87** in the left window. Enter the Node IP address of the CS1000E as per **Section6**. Click **Apply** to continue.



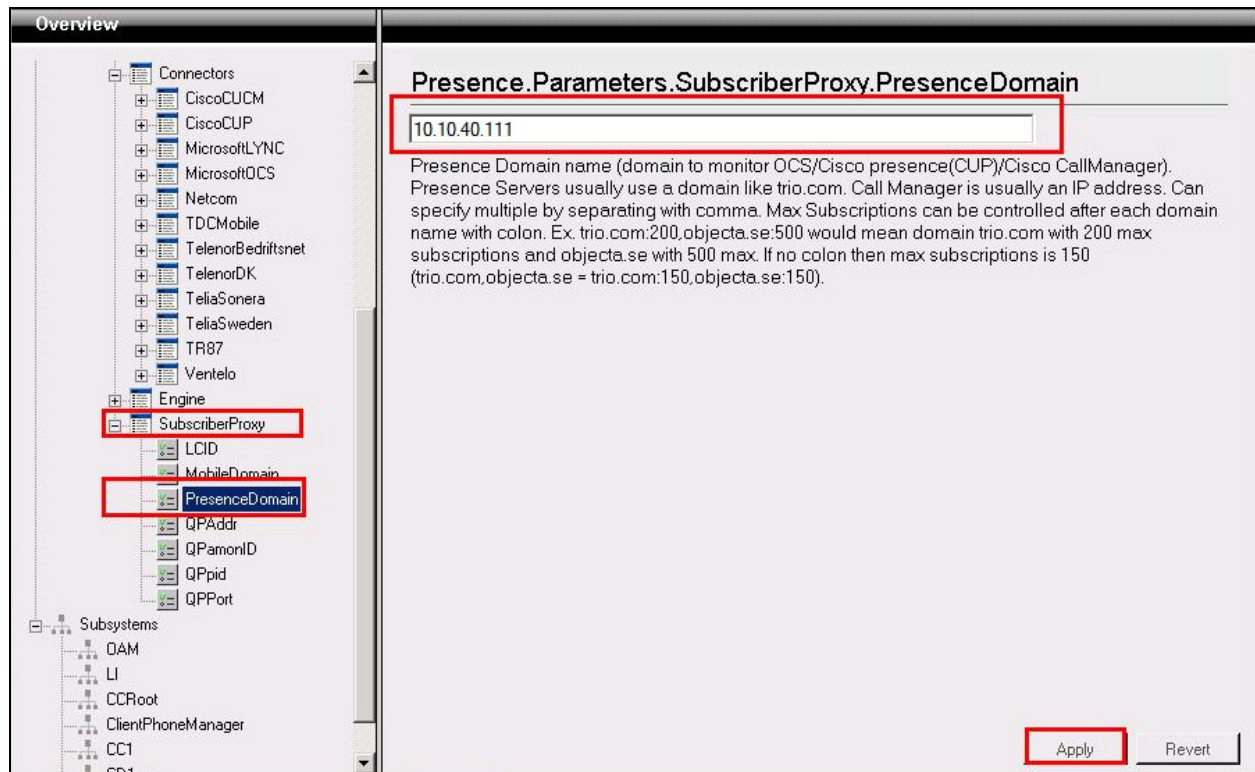
Select **ServerAddr** under **TR87** in the left window and again enter the Node IP address of the CS1000E. Click **Apply** to continue.



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

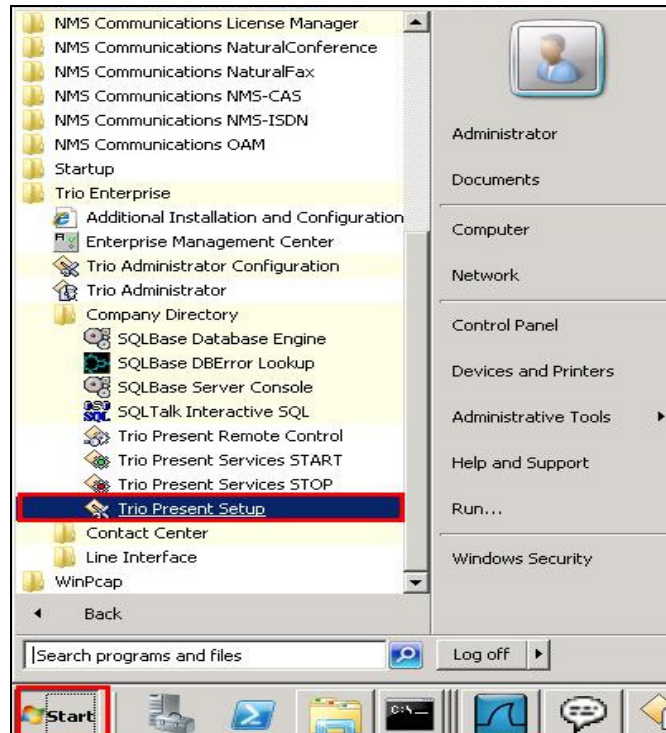


Select **PresenceDomain** under **SubscriberProxy** in the left window. Enter the Node IP address of the CS1000E in the right window. Click **Apply** to continue.

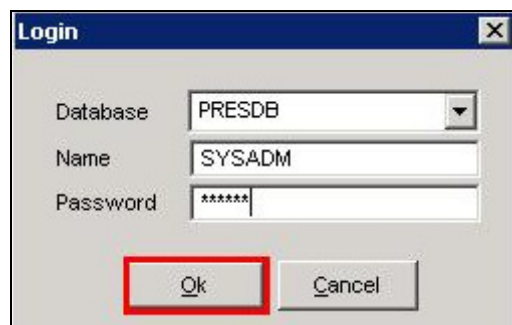


8.5. Configure ICP on TRIO

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



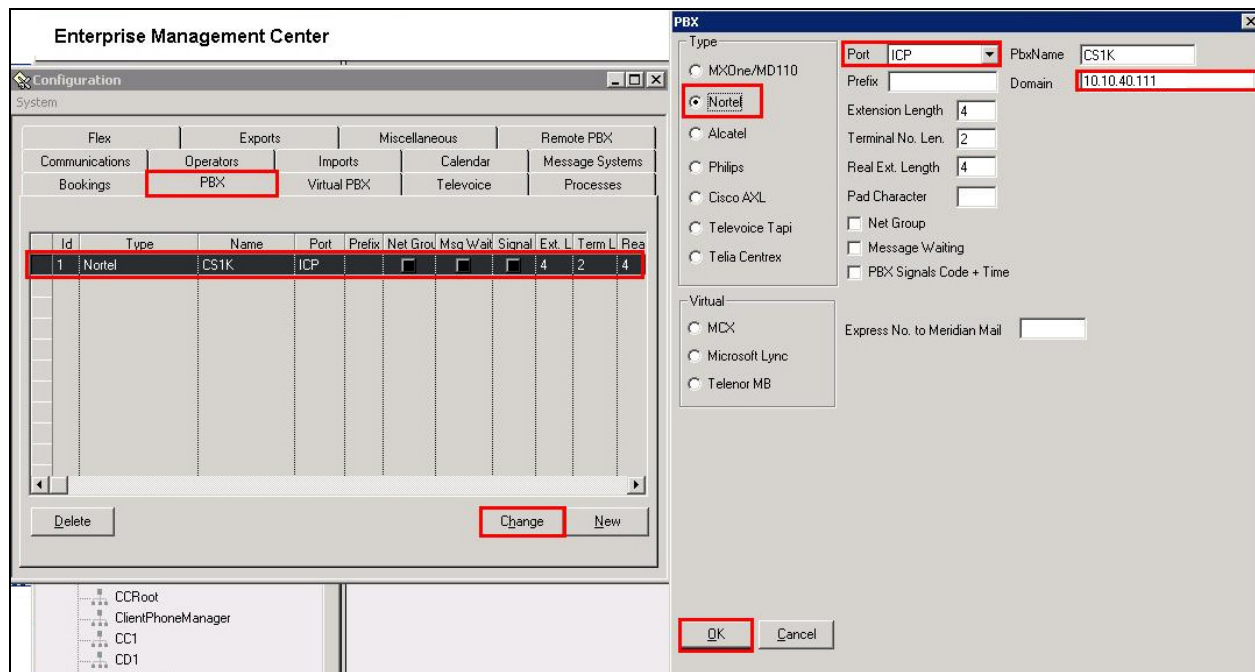
Enter the proper credentials and click **Ok**.



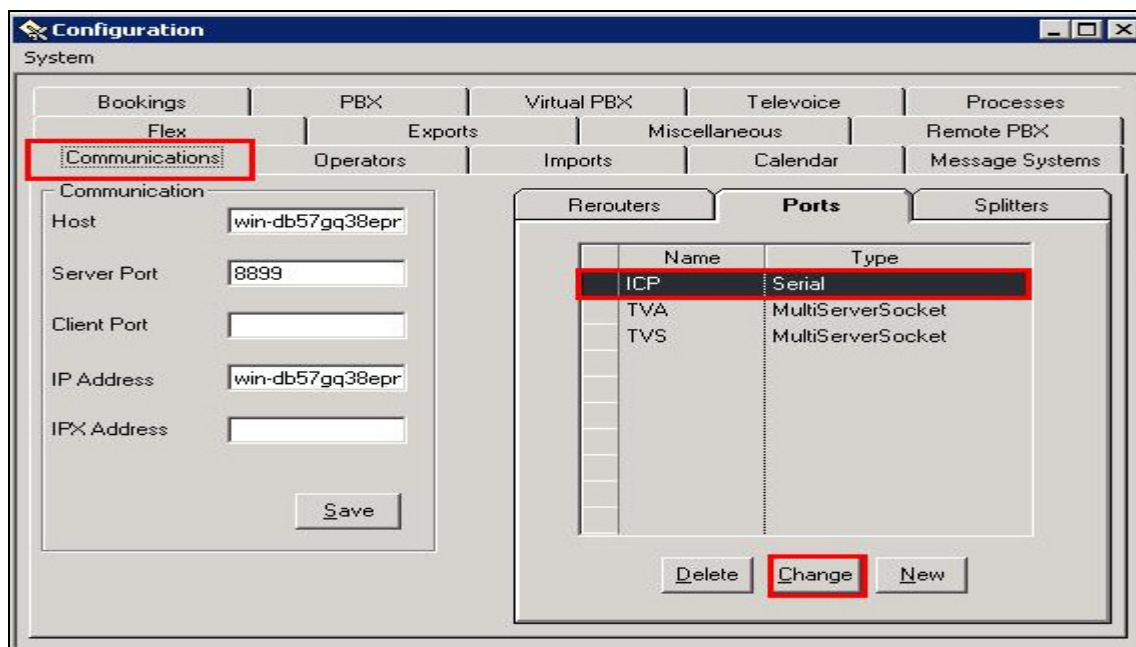
Highlight the selected PBX under the PBX tab and click on change. This opens the window displayed on the right. Ensure the following are selected.

- **Type** Nortel
- **Port** ICP
- **Domain** Node IP Address of CS1000E

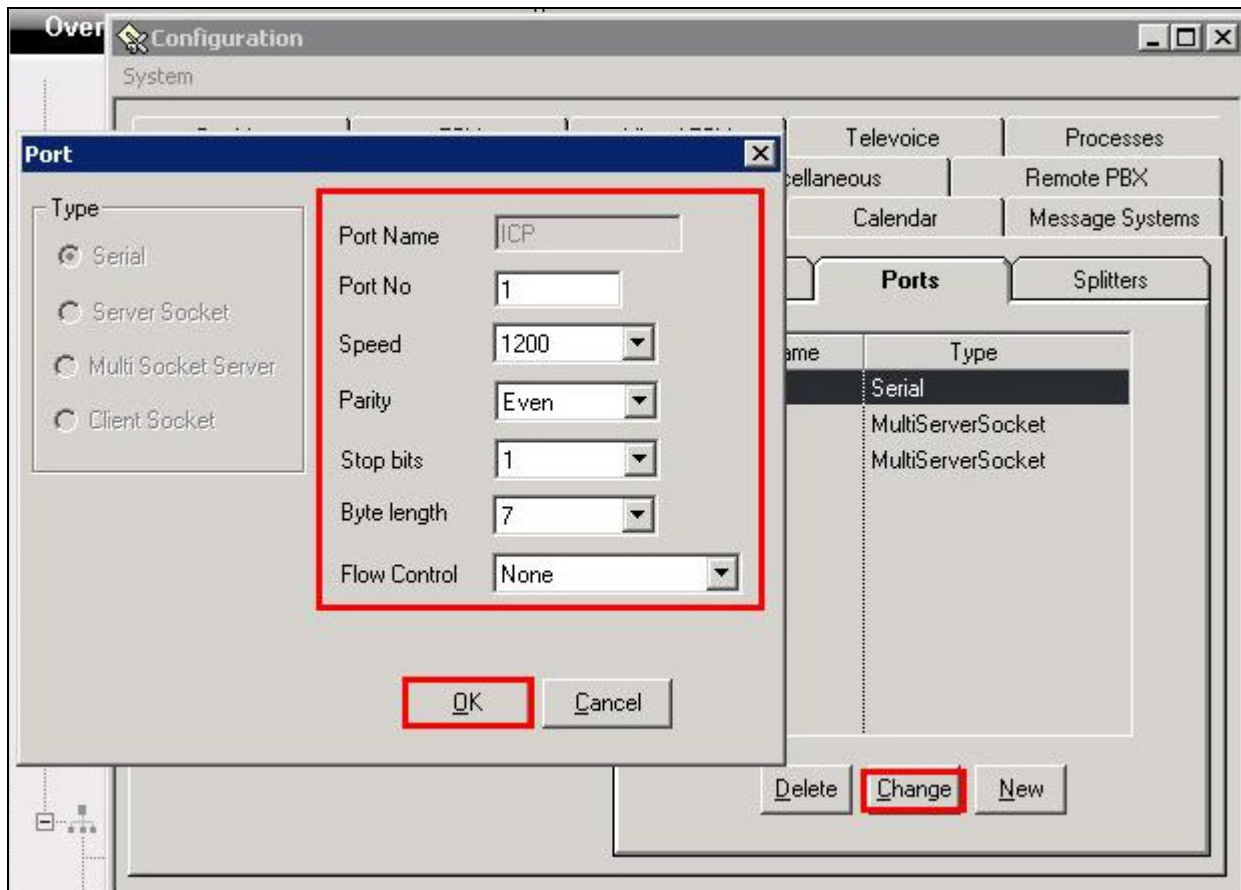
Select **OK** once the correct information is entered.



Under the **Communications** tab select **ICP** as highlighted below and click **Change**.



Enter the information that was entered in **Section 5.5.1** previous, click **OK** once all correct information is added.



9. Verification Steps

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

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

Check the status of the D-channel setup in **Section 5.2.1** by running the command **STAT DCH** in overlay 96 as shown below. The example below shows that D-Channel **66** is operational and established.

LD 96

| Prompt | Response | Description |
|---------|----------|--------------------------------|
| > | LD 96 | Enter Overlay 96 |
| . | STAT DCH | Check status of all D-Channels |
| DCH 066 | OPER EST | DES :to_Trio |

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 R4.1 from Enghouse Interactive AB to successfully interoperate with Avaya Communication Server 1000E R7.6 and Avaya Aura® Session Manager R6.3 using SIP trunks. Trio Enterprise passed all compliance testing successfully; please see **Section 2.2** of these Application Notes 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>

Appendix A

Installed CS1000E Dependency List

CS1000E Linux Service Pack 2

| In System service updates: 26 | | | | | |
|-------------------------------|------------|----------|---------|-----------|--|
| PATCH# | IN_SERVICE | DATE | SPECINS | REMOVABLE | NAME |
| 0 | Yes | 27/08/13 | NO | yes | cs1000-linuxbase-7.65.16.21-04.i386.000 |
| 1 | Yes | 27/08/13 | NO | YES | cs1000-patchWeb-7.65.16.21-04.i386.000 |
| 2 | Yes | 27/08/13 | NO | YES | cs1000-dmWeb-7.65.16.21-01.i386.000 |
| 3 | Yes | 28/08/13 | NO | yes | cs1000-snmp-7.65.16.00-01.i686.000 |
| 4 | Yes | 28/08/13 | NO | YES | cs1000-nrsm-7.65.16.00-03.i386.000 |
| 5 | Yes | 28/08/13 | NO | YES | cs1000-oam-logging-7.65.16.01-01.i386.000 |
| 6 | Yes | 28/08/13 | NO | yes | cs1000-cs1000WebService_6-0-7.65.16.21-00.i386.000 |
| 7 | Yes | 28/08/13 | NO | YES | cs1000-sps-7.65.16.21-01.i386.000 |
| 8 | Yes | 28/08/13 | NO | YES | cs1000-pd-7.65.16.21-00.i386.000 |
| 9 | Yes | 28/08/13 | NO | YES | cs1000-shared-carrdtct-7.65.16.21-01.i386.000 |
| 10 | Yes | 28/08/13 | NO | YES | cs1000-shared-tpselect-7.65.16.21-01.i386.000 |
| 11 | Yes | 28/08/13 | NO | YES | cs1000-emWebLocal_6-0-7.65.16.21-01.i386.000 |
| 12 | Yes | 28/08/13 | NO | yes | cs1000-dbcom-7.65.16.21-00.i386.000 |
| 13 | Yes | 28/08/13 | NO | YES | cs1000-csmWeb-7.65.16.21-05.i386.000 |
| 14 | Yes | 28/08/13 | NO | YES | cs1000-shared-xmsg-7.65.16.21-00.i386.000 |
| 15 | Yes | 28/08/13 | NO | YES | cs1000-vtrk-7.65.16.21-29.i386.000 |
| 16 | Yes | 28/08/13 | NO | YES | cs1000-tps-7.65.16.21-05.i386.000 |
| 17 | Yes | 28/08/13 | NO | YES | cs1000-mscAnnc-7.65.16.21-02.i386.001 |
| 18 | Yes | 28/08/13 | NO | YES | cs1000-mscAttn-7.65.16.21-04.i386.001 |
| 19 | Yes | 28/08/13 | NO | YES | cs1000-mscConf-7.65.16.21-02.i386.001 |
| 20 | Yes | 28/08/13 | NO | YES | cs1000-mscMusc-7.65.16.21-02.i386.001 |
| 21 | Yes | 28/08/13 | NO | YES | cs1000-mscTone-7.65.16.21-03.i386.001 |
| 22 | Yes | 28/08/13 | NO | YES | cs1000-bcc-7.65.16.21-21.i386.000 |
| 23 | Yes | 28/08/13 | NO | YES | cs1000-Jboss-Quantum-7.65.16.21-3.i386.000 |
| 24 | Yes | 28/08/13 | NO | YES | cs1000-emWeb_6-0-7.65.16.21-06.i386.000 |
| 25 | Yes | 28/08/13 | NO | yes | cs1000-cs-7.65.P.100-01.i386.001 |

CS1000E Call Server Patches

VERSION 4121
 RELEASE 7
 ISSUE 65 P +
 DepList 1: core Issue: 01 (created: 2013-06-14 03:54:33 (est))

IN-SERVICE PEPS

| PAT# | CR # | PATCH REF # | NAME | DATE | FILENAME | SPECINS |
|------|------------|-------------|----------|------------|--------------|---------|
| 000 | wi01052968 | ISS1:1OF1 | p32540_1 | 28/08/2013 | p32540_1.cpl | NO |
| 001 | wi01045058 | ISS1:1OF1 | p32214_1 | 28/08/2013 | p32214_1.cpl | NO |
| 002 | wi01085855 | ISS1:1OF1 | p32658_1 | 28/08/2013 | p32658_1.cpl | NO |
| 003 | wi01053314 | ISS1:1OF1 | p32555_1 | 28/08/2013 | p32555_1.cpl | NO |
| 004 | wi01060382 | iss1:1of1 | p32623_1 | 28/08/2013 | p32623_1.cpl | YES |
| 005 | wi01070580 | ISS1:1OF1 | p32380_1 | 28/08/2013 | p32380_1.cpl | NO |
| 006 | wi01067822 | ISS1:1OF1 | p32466_1 | 28/08/2013 | p32466_1.cpl | YES |
| 007 | wi01061481 | ISS1:1OF1 | p32382_1 | 28/08/2013 | p32382_1.cpl | NO |
| 008 | wi01072032 | ISS1:1OF1 | p32448_1 | 28/08/2013 | p32448_1.cpl | NO |
| 009 | wi01022599 | ISS1:1OF1 | p32080_1 | 28/08/2013 | p32080_1.cpl | NO |
| 010 | wi01035976 | ISS1:1OF1 | p32173_1 | 28/08/2013 | p32173_1.cpl | NO |
| 011 | wi01065922 | ISS1:1OF1 | p32516_1 | 28/08/2013 | p32516_1.cpl | NO |
| 012 | wi01055480 | ISS1:1OF1 | p32712_1 | 28/08/2013 | p32712_1.cpl | NO |
| 013 | wi01041453 | ISS1:1OF1 | p32587_1 | 28/08/2013 | p32587_1.cpl | NO |
| 014 | wi01078723 | ISS1:1OF1 | p32532_1 | 28/08/2013 | p32532_1.cpl | NO |
| 015 | WI0110261 | ISS1:1OF1 | p32758_1 | 28/08/2013 | p32758_1.cpl | NO |
| 016 | wi01064599 | iss1:1of1 | p32580_1 | 28/08/2013 | p32580_1.cpl | NO |
| 017 | wi01048457 | ISS1:1OF1 | p32581_1 | 28/08/2013 | p32581_1.cpl | NO |
| 018 | wi01072027 | ISS1:1OF1 | p32689_1 | 28/08/2013 | p32689_1.cpl | NO |
| 019 | wi01059388 | iss1:1of1 | p32628_1 | 28/08/2013 | p32628_1.cpl | NO |
| 020 | wi01074003 | ISS1:1OF1 | p32421_1 | 28/08/2013 | p32421_1.cpl | NO |
| 021 | wi00933195 | ISS1:1OF1 | p32491_1 | 28/08/2013 | p32491_1.cpl | NO |
| 022 | wi00996734 | ISS1:1OF1 | p32550_1 | 28/08/2013 | p32550_1.cpl | NO |
| 023 | wi01065118 | ISS1:1OF1 | p32397_1 | 28/08/2013 | p32397_1.cpl | NO |
| 024 | wi01063864 | ISS1:1OF1 | p32410_1 | 28/08/2013 | p32410_1.cpl | YES |
| 025 | wi01072023 | ISS1:1OF1 | p32130_1 | 28/08/2013 | p32130_1.cpl | YES |
| 026 | wi01075359 | ISS1:1OF1 | p32671_1 | 28/08/2013 | p32671_1.cpl | NO |
| 027 | wi01080753 | ISS1:1OF1 | p32518_1 | 28/08/2013 | p32518_1.cpl | NO |
| 028 | wi01070473 | ISS1:1OF1 | p32413_1 | 28/08/2013 | p32413_1.cpl | NO |
| 029 | wi01075355 | ISS1:1OF1 | p32594_1 | 28/08/2013 | p32594_1.cpl | NO |
| 030 | wi01071379 | ISS1:1OF1 | p32522_1 | 28/08/2013 | p32522_1.cpl | NO |
| 031 | wi01070756 | ISS1:1OF1 | p32444_1 | 28/08/2013 | p32444_1.cpl | NO |
| 032 | wi01075353 | ISS1:1OF1 | p32613_1 | 28/08/2013 | p32613_1.cpl | NO |
| 033 | wi01062607 | ISS1:1OF1 | p32503_1 | 28/08/2013 | p32503_1.cpl | NO |
| 034 | wi01068851 | ISS1:1OF1 | p32439_1 | 28/08/2013 | p32439_1.cpl | NO |
| 035 | wi01075352 | ISS1:1OF1 | p32603_1 | 28/08/2013 | p32603_1.cpl | NO |
| 036 | wi01092300 | ISS1:1OF1 | p32692_1 | 28/08/2013 | p32692_1.cpl | NO |
| 037 | wi01063263 | ISS1:1OF1 | p32573_1 | 28/08/2013 | p32573_1.cpl | NO |
| 038 | wi01087528 | ISS1:1OF1 | p32700_1 | 28/08/2013 | p32700_1.cpl | NO |
| 039 | wi01055300 | ISS1:1OF1 | p32543_1 | 28/08/2013 | p32543_1.cpl | NO |
| 040 | wi01039280 | ISS1:1OF1 | p32423_1 | 28/08/2013 | p32423_1.cpl | NO |
| 041 | wi01068669 | ISS1:1OF1 | p32333_1 | 28/08/2013 | p32333_1.cpl | NO |
| 042 | wi01069441 | ISS1:1OF1 | p32097_1 | 28/08/2013 | p32097_1.cpl | NO |
| 043 | wi01058621 | ISS1:1OF1 | p32339_1 | 28/08/2013 | p32339_1.cpl | NO |
| 044 | wi01032756 | ISS1:1OF1 | p32673_1 | 28/08/2013 | p32673_1.cpl | NO |
| 045 | wi01070465 | iss1:1of1 | p32562_1 | 28/08/2013 | p32562_1.cpl | NO |
| 046 | wi01053920 | ISS1:1OF1 | p32303_1 | 28/08/2013 | p32303_1.cpl | NO |
| 047 | wi00897254 | ISS1:1OF1 | p31127_1 | 28/08/2013 | p31127_1.cpl | NO |
| 048 | wi01057403 | ISS1:1OF1 | p32591_1 | 28/08/2013 | p32591_1.cpl | NO |
| 049 | wi01066991 | ISS1:1OF1 | p32449_1 | 28/08/2013 | p32449_1.cpl | NO |
| 050 | wi01094305 | ISS1:1OF1 | p32640_1 | 28/08/2013 | p32640_1.cpl | NO |
| 051 | wi01058359 | ISS1:1OF1 | p32331_1 | 28/08/2013 | p32331_1.cpl | NO |
| 052 | wi01047890 | ISS1:1OF1 | p32697_1 | 28/08/2013 | p32697_1.cpl | NO |

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| 053 | wi01060241 | ISS1:1OF1 | p32381_1 | 28/08/2013 | p32381_1.cpl | NO |
| 054 | wi01034307 | ISS1:1OF1 | p32615_1 | 28/08/2013 | p32615_1.cpl | NO |
| 055 | wi01052428 | ISS1:1OF1 | p32606_1 | 28/08/2013 | p32606_1.cpl | NO |
| 056 | wi00884716 | ISS1:1OF1 | p32517_1 | 28/08/2013 | p32517_1.cpl | NO |
| 057 | wi01070468 | iss1:1of1 | p32418_1 | 28/08/2013 | p32418_1.cpl | NO |
| 058 | wi01091447 | ISS1:1OF1 | p32675_1 | 28/08/2013 | p32675_1.cpl | NO |
| 059 | wi01068042 | ISS1:1OF1 | p32669_1 | 28/08/2013 | p32669_1.cpl | NO |
| 060 | wi01061483 | ISS1:1OF1 | p32359_1 | 28/08/2013 | p32359_1.cpl | NO |
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| 062 | wi01056633 | ISS1:1OF1 | p32322_1 | 28/08/2013 | p32322_1.cpl | NO |
| 063 | wi01070474 | iss1:1of1 | p32407_1 | 28/08/2013 | p32407_1.cpl | NO |
| 064 | wi01053597 | ISS1:1OF1 | p32304_1 | 28/08/2013 | p32304_1.cpl | NO |
| 065 | wi01070471 | ISS1:1OF1 | p32415_1 | 28/08/2013 | p32415_1.cpl | NO |
| 066 | wi01025156 | ISS1:1OF1 | p32136_1 | 28/08/2013 | p32136_1.cpl | NO |
| 067 | wi01088775 | ISS1:1OF1 | p32659_1 | 28/08/2013 | p32659_1.cpl | NO |
| 068 | wi01083584 | ISS1:1OF1 | p32619_1 | 28/08/2013 | p32619_1.cpl | NO |
| 069 | wi01075360 | iss1:1of1 | p32602_1 | 28/08/2013 | p32602_1.cpl | NO |
| 070 | wi01053195 | ISS1:1OF1 | p32297_1 | 28/08/2013 | p32297_1.cpl | NO |
| 071 | wi01043367 | ISS1:1OF1 | p32232_1 | 28/08/2013 | p32232_1.cpl | NO |
| 072 | wi01082456 | ISS1:1OF1 | p32596_1 | 28/08/2013 | p32596_1.cpl | NO |
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| 074 | wi01065842 | ISS1:1OF1 | p32478_1 | 28/08/2013 | p32478_1.cpl | NO |
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| 080 | wi01034961 | ISS1:1OF1 | p32144_1 | 28/08/2013 | p32144_1.cpl | NO |
| 081 | wi01056067 | ISS1:1OF1 | p32457_1 | 28/08/2013 | p32457_1.cpl | NO |
| 082 | WI01077073 | ISS1:1OF1 | p32534_1 | 28/08/2013 | p32534_1.cpl | NO |
| 083 | wi01073100 | ISS1:1OF1 | p32599_1 | 28/08/2013 | p32599_1.cpl | NO |
| 084 | wi01060341 | ISS1:1OF1 | p32578_1 | 28/08/2013 | p32578_1.cpl | NO |
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| MDP>USING DEPLIST ZIP FILE DOWNLOADED :2013-08-27 09:21:58(est) | | | | | | |

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