Avaya Solution & Interoperability Test Lab

Application Notes for Enghouse Interactive Attendant Console 6.0 to interoperate with Avaya Communication Server1000E R7.6 and Avaya Aura® Session Manager R6.3 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required Enghouse Interactive Attendant Console 6.0 to interoperate with Avaya Communication Server1000E R7.6 and Avaya Aura® Session Manager R6.3

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 outline the steps necessary to configure Enghouse Interactive Attendant Console to interoperate with Avaya Communication Server R6.3 via Avaya Aura® Session Manager R6.3. Enghouse Interactive Attendant Console provides a Computer Telephony Interface (CTI) to PBX environments that do not support a traditional Third-Party Call Control API or Protocol. Instead, the CTI Connect Media Gateway implements a Back-to-Back User Agent (B2BUA) model where customer and agent call legs are directed from the PBX to the Attendant Console, where they are controlled via a conference bridge. In addition, the Attendant Console enables a set of Media Control APIs to be supported by Attendant Console for these PBXs. These PBXs can be used to support applications such as Self-service IVR.
Attendant Console 6.0 introduces support for an equivalent SIP Trunk interface to Avaya Communication Server1000E via Avaya Aura® Session Manager.
The Attendant Console supports two types of CTI Connect channel:

- **Route Channel** – a CTC application can use this channel to play media to the caller, receive notification of DTMF key presses, receive audio input and route the call to a new destination on the PBX.
- **Device channel** – The Attendant Console joins the inbound call in a conference with the PBX Extension that was associated with the "To:" header. A Device Channel may be used to initiate an outbound call on behalf of the PBX extension. In this case, a call leg will be established from the Media Gateway to the extension and then conferenced with the outbound leg to the IP-PBX which will route it to the appropriate destination.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using Communication Server 1000E (CS1000E). The Attendant Console Communicates with CS1000E using a SIP trunk via session Manager. See Figure 1 for a network diagram. A Dial plan was configured on CS1000E to route calls to queues on the Attendant Console. Calls placed to these queues are automatically bridged to the telephone the Attendant is using for answering purposes.

Note: During compliance testing an Avaya 1140 IP deskphone was used as the attendant's telephone.

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 interoperability compliance testing included feature and serviceability testing. The serviceability testing introduced failure scenarios to see if the Attendant Console could resume after a link failure with CS1000E/Session Manager. The testing included:

- 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

2.2. Test Results

Tests were performed to verify interoperability between the Attendant Console and CS1000E. The tests were all functional in nature and performance testing was not included. All the test cases passed successfully with one observation. The Attendant Console can only initiate a 3 way conference.

2.3. Support

For technical support for Enghouse Interactive products, please use the following web link.
https://mysupport.enghouseinteractive.com

Enghouse Interactive can also be contacted as follows.
Phone: +44 (0)870 220 2205 (EMEA)
+1 800.657.1530 (Americas)
E-mail: Support@Datapulse.com

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of a CS1000E and a Session Manager which has a SIP Trunk connection to the CTI Connect Server. An Avaya 1140 deskphone telephone was used as the Attendant telephone during compliance testing. System Manager was used to configure Session Manager. Digital and Unistim and SIP telephones were configured on the CS1000E to generate outbound/inbound calls to/from the PSTN. A QSIG trunk was configured to connect to the PSTN.

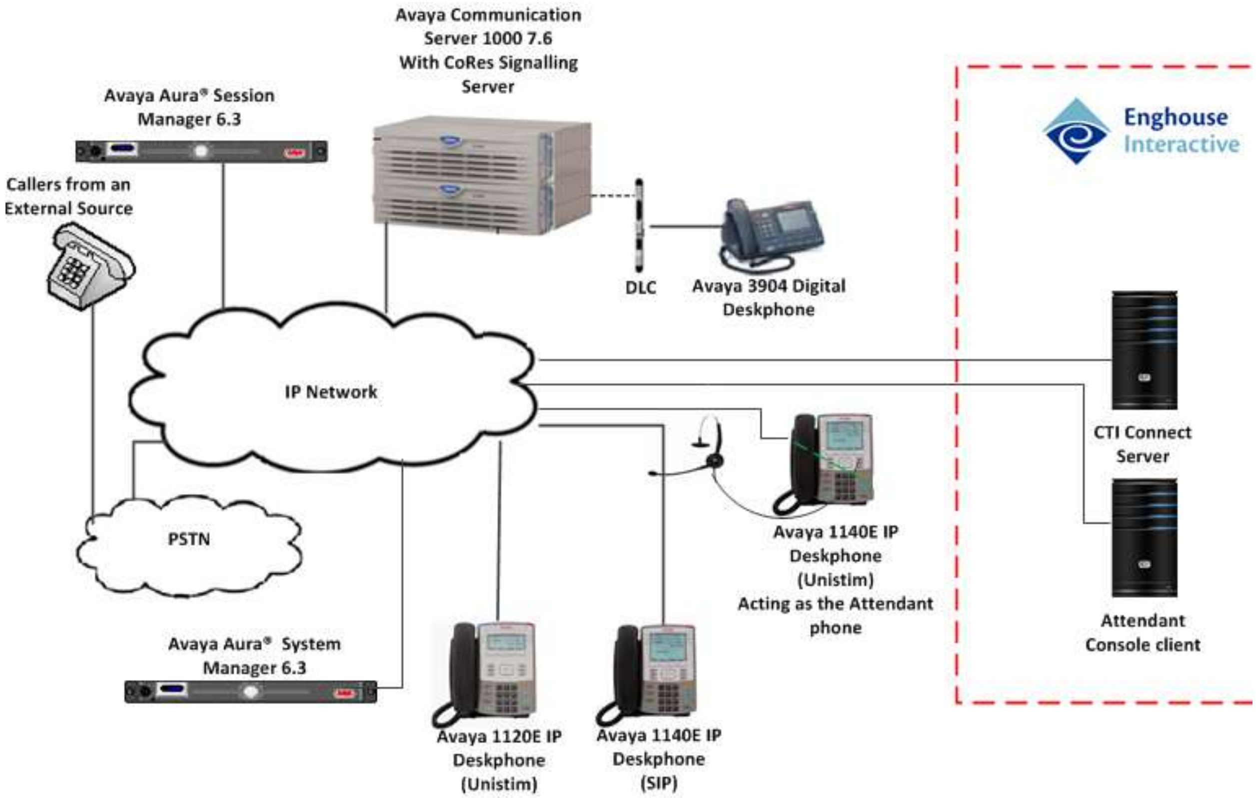


Figure 1: Avaya and Enghouse Interactive Reference Configuration

4. Equipment and Software Validated

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

Avaya Equipment	Software / Firmware Version
Call Processor Pentium Mobile (CPPM) NTDW61BA Avaya Media Gateway NTDW60BA	Avaya Communication Server 1000E R7.6 FPGA AA18
Avaya Aura® Session Manager	R6.3.7 Software Update 6.3.7.0.637008
Avaya Aura® System Manager	R6.3.7 Build 6.3.0.8.5682-6.3.8.2826 Update 6.3.5.52017
Avaya 1100 series IP Telephones • 1120E (IP) • 1140E (IP) • 1140E (SIP) • 3904 (Digital)	 0624C8Q 0624C8Q 04.03.12.00 N/A
Enghouse Interactive Equipment	**Software / Firmware Version**
CTI Connect Media Gateway Attendant Console server both running on Microsoft Windows 2008 R2 SP1 Server	Version 8.1 Version 6.0.0.6
Attendant Console client running on Microsoft Windows 7 Enterprise SP1	Version 6.0.0.6

5. Configure Avaya CS1000E

The configuration operations illustrated in this section were performed using terminal access to the CS1000E over a telnet session. It is implied a working system is already in place, including a Route and D-Channel, however the basic steps to create these are outlined in the sections below. For all other provisioning information such as Installation and Configuration, please refer to the product documentation in **Section 12**. The configuration operations described in this section can be summarized as follows: (Note: during compliance testing all inputs not highlighted in Bold were left as Default).

- Verify Licences
- Configuring a SIP Connection on CS1000E
- Configure Dialling Plan

5.1. Verify Licences

Check that **SIP ACCESS PORTS** are installed (in bold below).

LD 22

```
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
```

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 Session Manager. 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	**21**	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 16**	Enter Overlay 16
REQ	**NEW**	Create new
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	21 1	Route number and Member number

5.3. Configure Dialling Plan

To route calls to CTI Connect a dialing plan is required. The numbers configured are routed to the Session Manager, where a Dialing Pattern (see **Section 7.5**) is configured to route the calls to CTI Connect. There are a number of ways to setup a dialing plan. For compliance testing a Coordinated Dialing Plan (CDP) was used. During compliance testing CTI Connect was configured with two queue numbers 4700 was used for the Internal queue and 4701 was used for the External queue. All 4 digit numbers beginning with 47 were routed to CTI Connect via a Dial Pattern on Session Manager.

5.3.1. Create a Route List Index

In order to create a CDP a Route List Index (RLI) in overlay 86 is required. Use the **NEW** command in **LD 86** to create a **RLI**. In the example below **FEAT** is **RLB**, and **TYPE** is **DSC**. **Note:** Rout 21 was used.

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	**22**	Route list Index number
ENTR	0	First entry for the RLI
ROUT	**21**	Enter the route number

5.3.2. Create a Coordinated Dialling Plan

Use the **NEW** command in **LD 87** to create a CDP entry to route calls to the Session Manager. In the example below, **FEAT** is **CDP**, **DSC** is **47, FLEN** is **4** and the **RLI** is **22** (RLI created in **Section 5.3.1**).

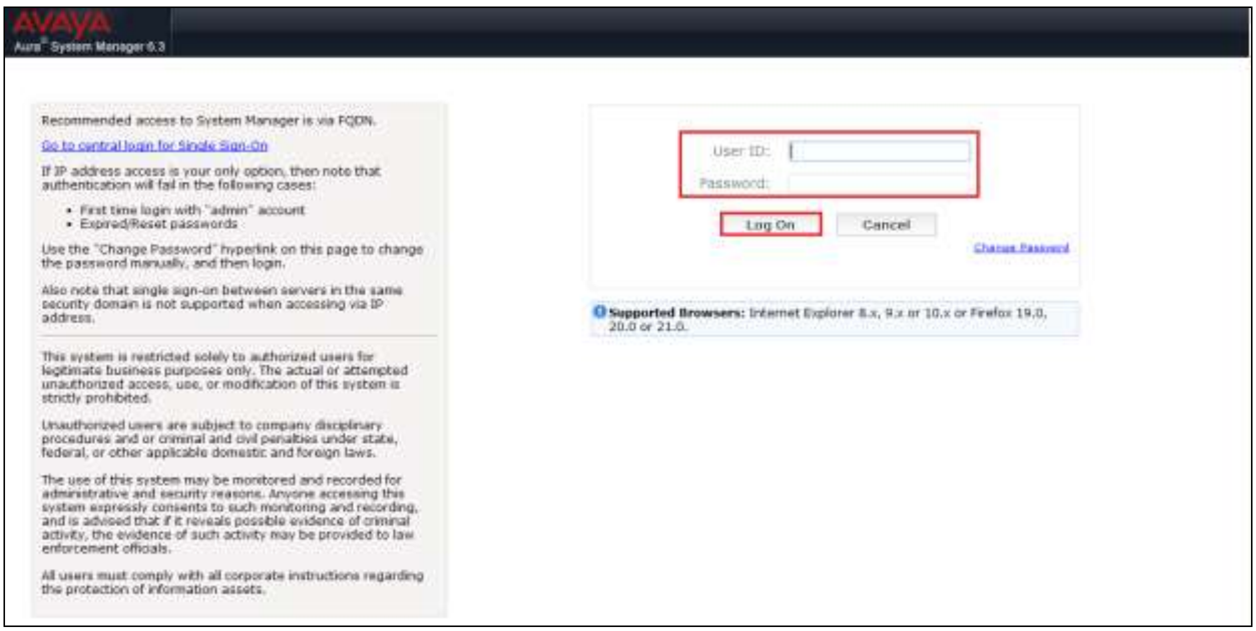
LD 87

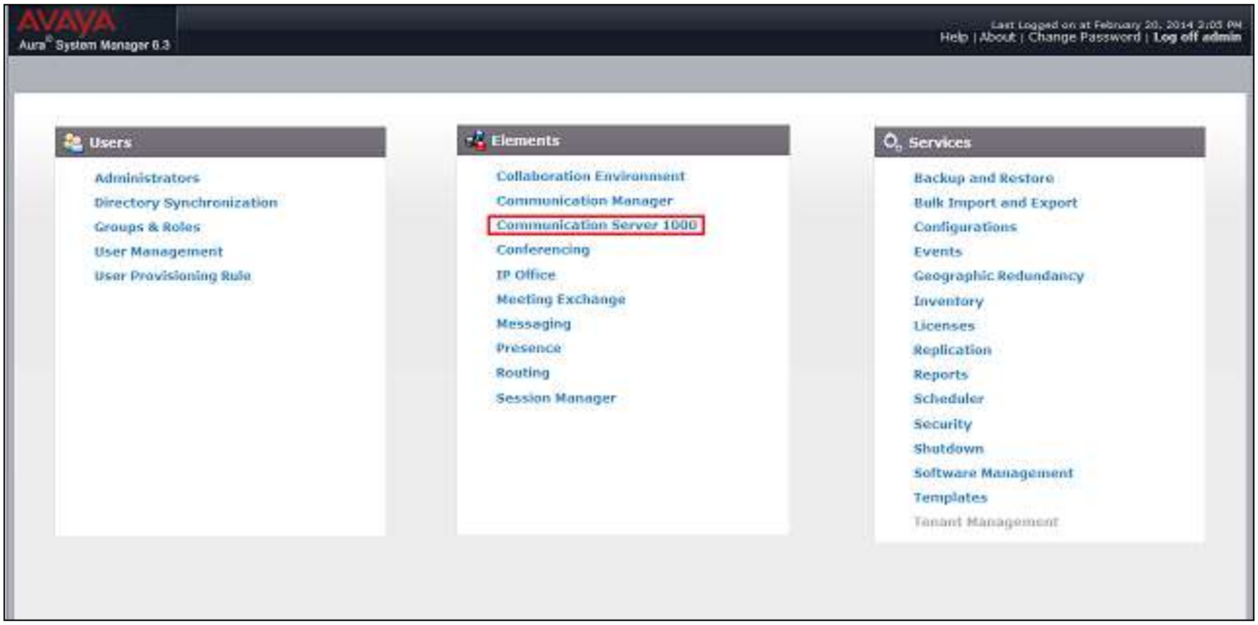
Prompt	Response	Description
>	LD 87	Enter Overlay 87
REQ	**NEW**	Create new
CUST	0	Customer Number as defined in LD15
FEAT	**CDP**	Coordinated dialing plan
TYPE	DSC	Distance Steering code
DSC	**47**	Distant Steering code
FLEN	**4**	Flexible Length number of digits
RLI	**22**	Route list index Number

6. Configure Trunk Group for Session Manager

The Virtual Trunk Gateway on the Signalling Server needs to be configured to route calls to the Session Manager. It is implied that the Signalling Server is already in place, and a Node is configured and is part of the security framework. The following configuration was used during compliance testing. The Virtual Trunk Gateway is configured using the CS1000 Element Manager WEB interface, accessed via a link from System Manager. After logging into the System Manger using the appropriate credentials (not Shown), select **Communication Server 1000** from the **Elements** column.

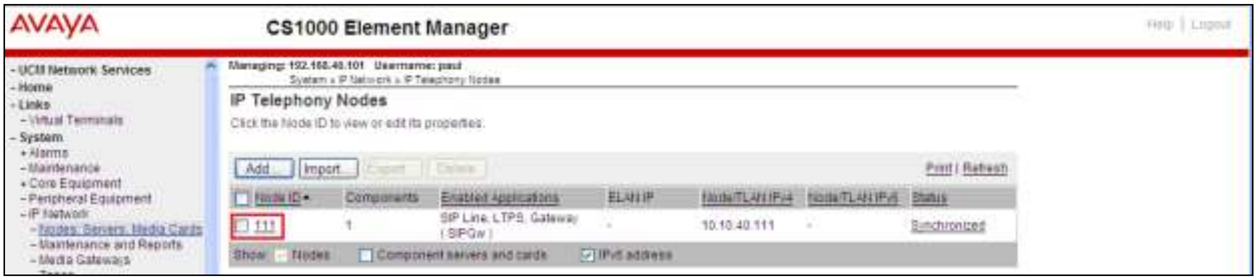
Log on by accessing the browser-based GUI of System Manager, using the URL "http://<fqdn>/SMGR" or "http://<ip-address>/SMGR", where:
"<fqdn> is the fully qualified domain name of the System Manager or the"<ipaddress>" is the IP address of System Manager.
Once the System Manager web page opens, log in with the appropriate credentials.





Once the Elements page opens select the Element Manager of the CS1000E to administer and login using the appropriate credentials (not shown). Once the CS1000 Element Manager page opens, navigate to **IP Network → Nodes: Services, Media Cards**.

Once the **IP Telephony Nodes page** opens click on the appropriate node (During compliance testing node **111** was used.

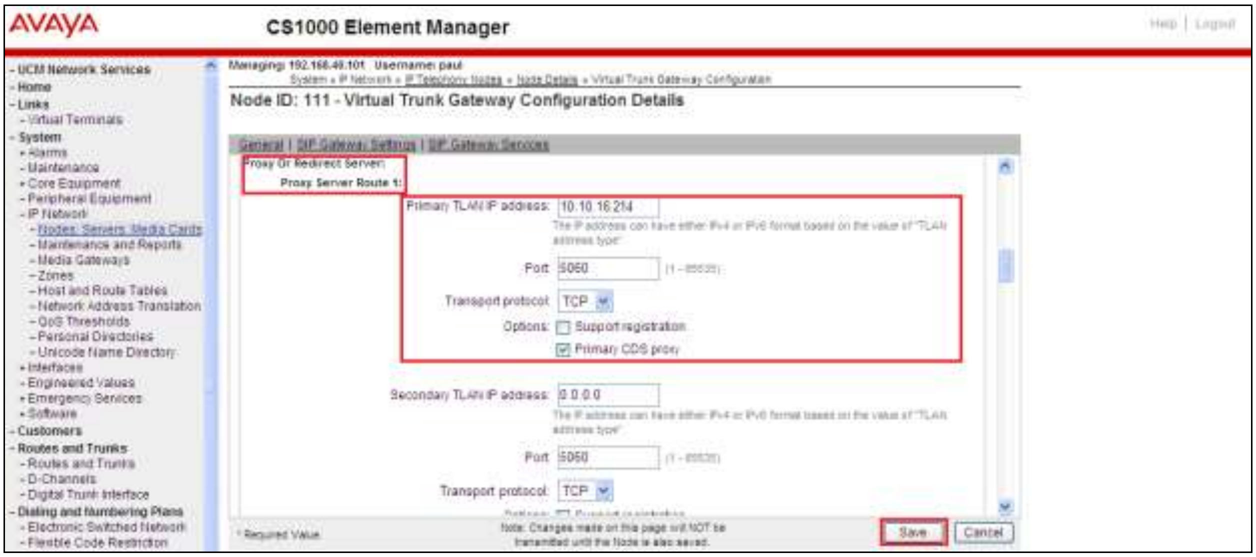


Once the **Node Details** page opens scroll down using the vertical scroll bar on the right side of the page and click on **Gateway (SIPGw)**.

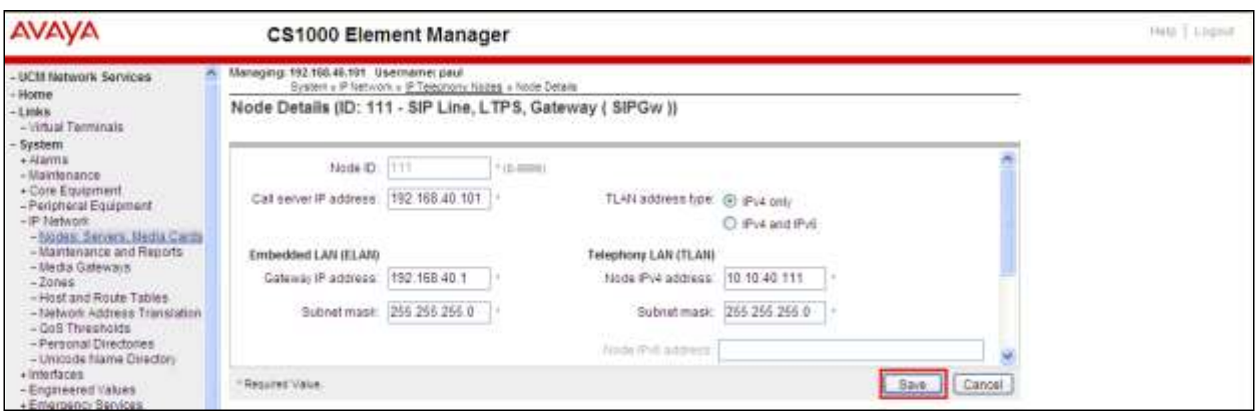
Once the **Virtual Trunk Gateway Configuration Details** page opens, scroll down using the vertical scroll bar on the right side of the page to **Proxy Or Redirect Server** (**Proxy Server route 1**) and enter the following:

- **Primary TLAN IP address** Enter the IP address of the Session Manager (10.10.16.214)
- **Port** Enter **5060**
- **Transport protocol** Select **TCP** from the dropdown box
- **Options** Click the **Primary CDS proxy** check box

Click on the **Save** button to save the configuration.



Once the Virtual Trunk Gateway Configuration is saved the Node must also be saved. On the **Node Details** page, click on the **Save** button.



On the **Node Saved** page click on the **Transfer Now** button.

On the **Synchronize Configuration Files** page select the appropriate Signalling Server and click on the **Start Sync** button.

Once the synchronization is complete the applications must be restarted. Click on the **Restart Applications** button.

7. Configuring Avaya Aura® Session Manager

A number of configurations are required to enable CS1000E to route call to CTI connect and vice versa. All configurations of Session Manager are performed using System Manager. The configuration operations described in this section can be summarized as follows:

- Create CS1000E Adaptation
- Create CTI Adaptation
- Create CTI Connect as a SIP Entity
- Create an Entity Link for CTI Connect
- Create a Routing Policy for CTI Connect
- Create a Dial Pattern for CTI Connect

Note: It is implied a working system is already in place including a Domain (**devconnect.local**) and a Location (**DevConnectRP**). During Compliance testing a SIP Entity and an Entity Link for the CS1000E were created. Also a Routing Policy and a Dial Pattern to route calls to the CS1000E were created and are outside the scope of these Application Notes.

7.1. Create CS1000E Adaptation

A CS1000E Adaptation is required for CTI Connect to interoperate with CS1000E. This Adaptation is to the SIP Entity of the CS1000E (see **Appendix A**). After logging on to System Manager, select **Routing** from the elements column and select **Adaptations.** When the Adaptations window opens, click on the **New** button (not shown). When the **Adaptation Details** window opens enter the following:

- **Adaptation Name:** Enter an informative name (i.e. CS1000 Adapter)
- **Module Name:** Select **CS1000Adapter** from the dropdown box
- **Module Parameter Type:** Select **Name-Value Parameter** from the dropdown box

Click on the **Add** button, and in the **Name** field enter **fromto**, in the **Value** field enter **true**.

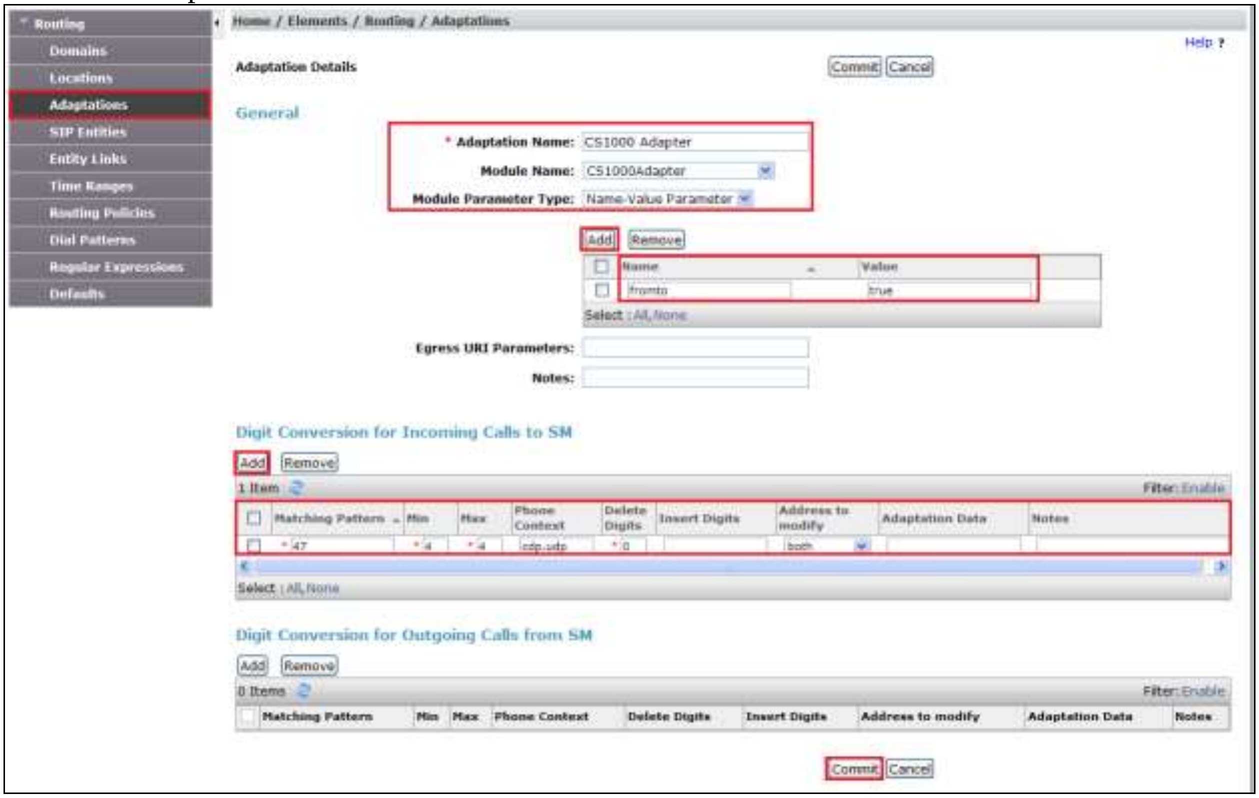
In the **Digit Conversion for Incoming Calls to SM** section click on the **Add** button, and enter the following:

- **Matching Pattern** Enter the **Dial Pattern** number as was configured to route calls to the CTI Connect (during testing all 4 digit numbers beginning with **47** were sent to the Attendant Console. (See **Section 5.3**)
- **Min** Enter **4**
- **Max** Enter **4**
- **Phone Context** Enter **cdp.udp** (this entry may differ for other CS1000E systems. CTI Connect requires all phone context be removed. To find out what phone context that is being sent to CTI Connect, verify what settings are used form the Element Manager on the CS1000E or by placing a call to CTI Connect, and using WireShark or traceSM to see what phone context is being sent).
- **Delete Digits** Enter the number of digits to be removed (i.e. 0)
- **Insert Digits** Leave blank
- **Address to modify** select **Both** from the dropdown box
- **Adaptation Data** Leave blank

Click on the **Commit** button to save the configuration.

Note: See screen shot on next page.

CS1000E Adaptation



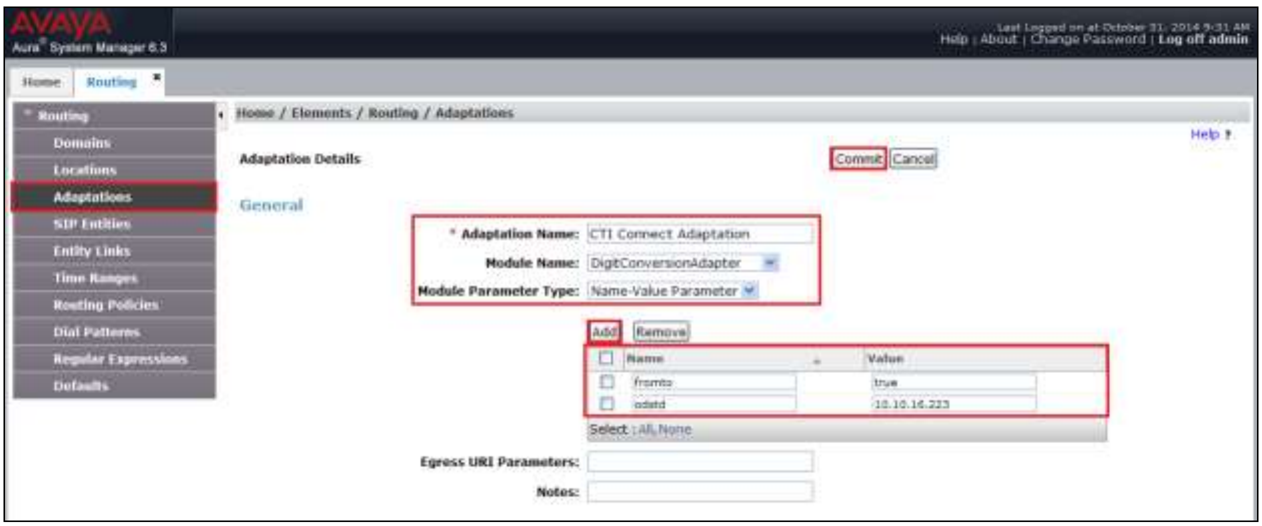
7.2. Create CTI Adaptation

A CTI Adaptation is required for CTI Connect to interoperate with CS1000E. Select **Routing** from the elements column and select **Adaptations.** When the Adaptations window opens, click on the **New** button (not shown). When the **Adaptation Details** window opens enter the following:

- **Adaptation Name:** Enter an informative name (i.e. CTI Connect Adaptation)
- **Module Name:** Select **CTI Connect Adaptation** from the dropdown box
- **Module Parameter Type:** Select **Name-Value Parameter** from the dropdown box

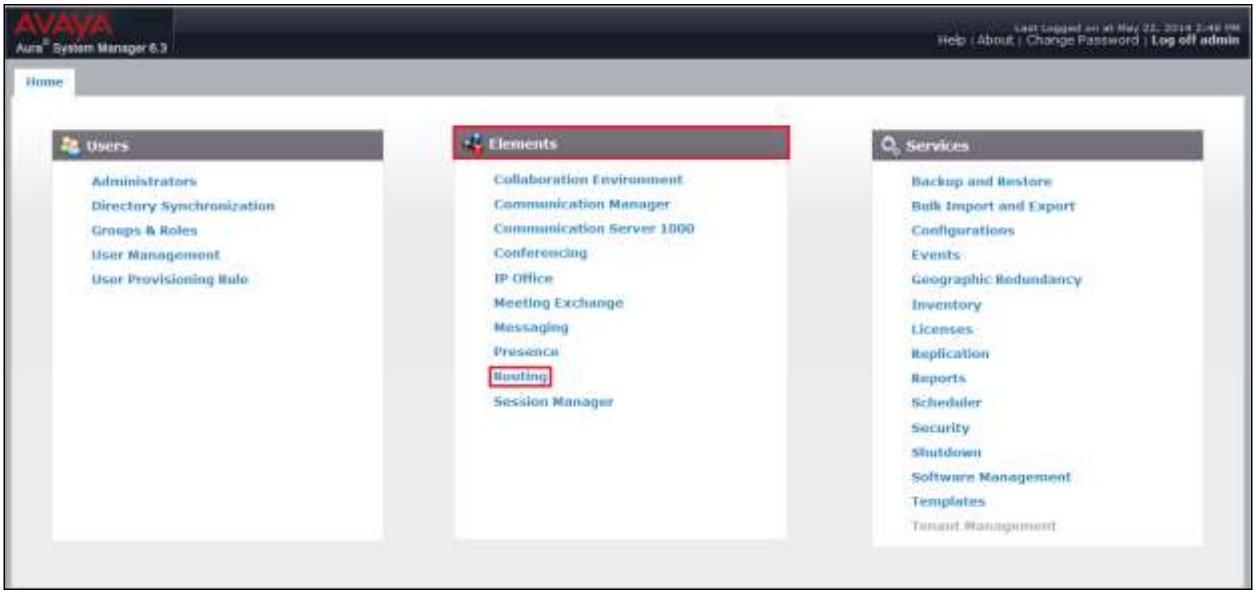
Click on the **Add** button, and in the **Name** field enter **fromto**, in the **Value** field enter **true**. Click on the **Add** button a second time, and in the **Name** field enter **odstd**, in the **Value** field enter the IP address of the CTI Connect Server.

Click on the **Commit** button to save the configuration.



7.3. Create CTI Connect as a SIP Entity

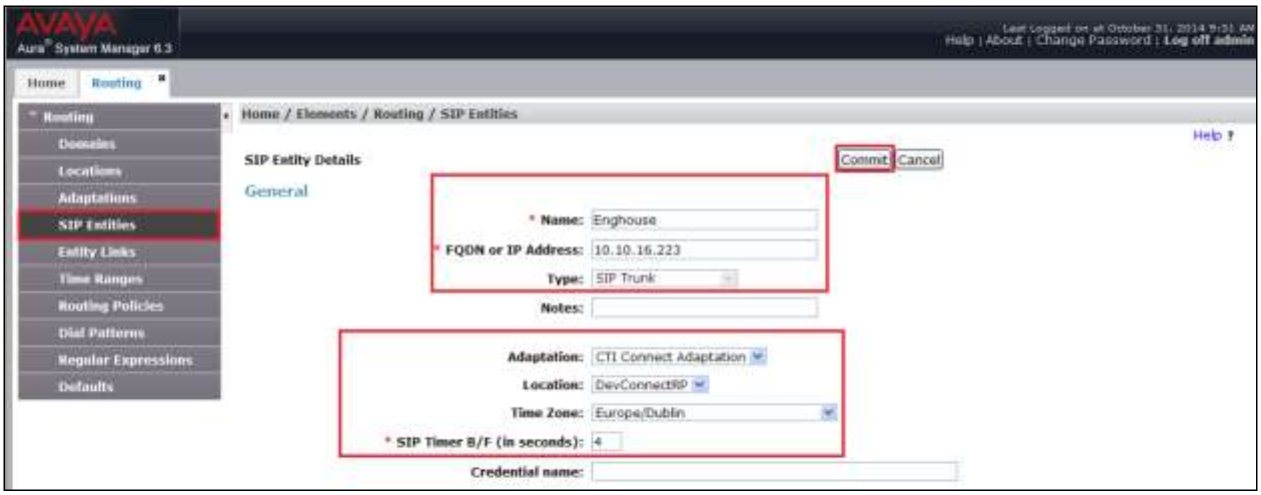
Once logged in select the **Routing** Link under the **Elements** column.



A SIP Entity must be added for CTI Connect. To add a SIP Entity, select **SIP Entities** on the left panel menu and then click on the **New** button (not shown).
Note: A SIP Entity was already configured for the CS1000E and was called **cores3**.
Enter the following for ACS SIP Entity:
Under **General:**

- **Name** Enter an informative name (e.g., **Enghouse**)
- **FQDN or IP Address** Enter the IP address of the signalling interface on CTI connect
- **Type** Select **SIP Trunk** from the dropdown box
- **Adaptation** Select the Adaptation as created in **Section 7.3**
- **Location** Select the required Location (i.e. **DevConnectRP**) from the dropdown box
- **Time Zone** Select time zone for this location from the dropdown box
- **SIP Timer** Enter **4**

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

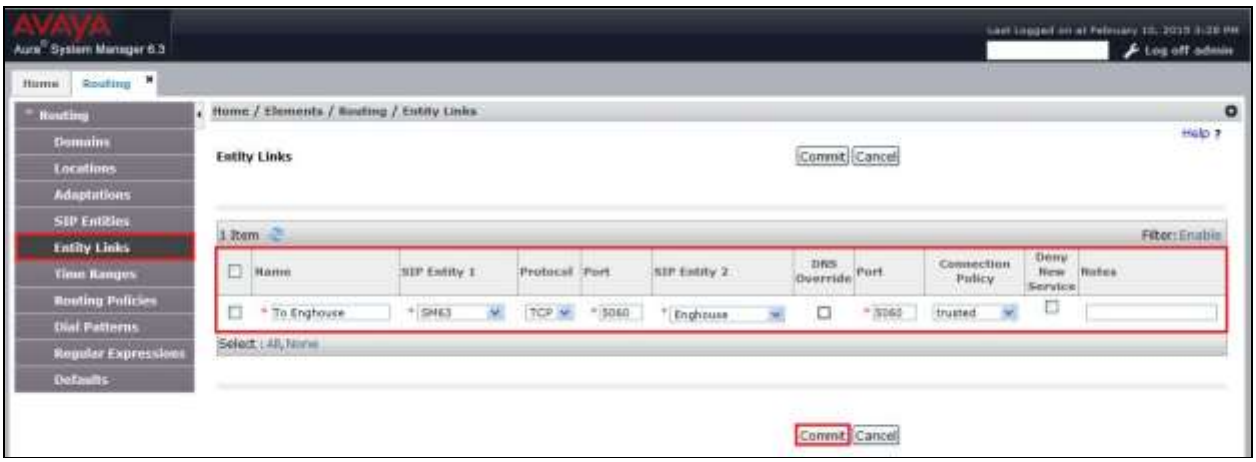


7.4. Create an Entity Link for CTI Connect

The SIP trunk between Session Manager and CTI Connect Server requires an Entity Link. To add an Entity Link, select **Entity Links** on the left panel menu and click on the **New** button (not shown), and enter the following:

- **Name** An informative name, (e.g. **To Enghouse**)
- **SIP Entity 1** Select **SM63** from the **SIP Entity 1** dropdown box
- **Protocol** Select **TCP** from the Protocol drop down box
- **Port** Enter **5060**
- **SIP Entity 2** Select **Enghouse** from the **SIP Entity 2** dropdown box (configured in **Section 7.2**)
- **Port** Enter **5060** as the Port
- **Connection Policy** Select **Trusted** from the **Connection Policy** dropdown

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



7.5. Create a Routing Policy for CTI Connect

Create routing policies to direct calls to CTI Connect. To add a routing policy, select **Routing Policies** on the left panel menu and then click on the **New** button (not shown). In **Routing Policy Details** enter an informative name in the **Name** field (example, **To Enghouse**) and enter **0** in the **Retries** field**.** In **SIP Entity as Destination,** click **Select**.

Once the SIP Entity List screen opens, check the **Enghouse** radio button. Click on the **Select** button to confirm the chosen options and then return to the **Routing Policies Details** screen and select **Commit** button (not shown) to save.

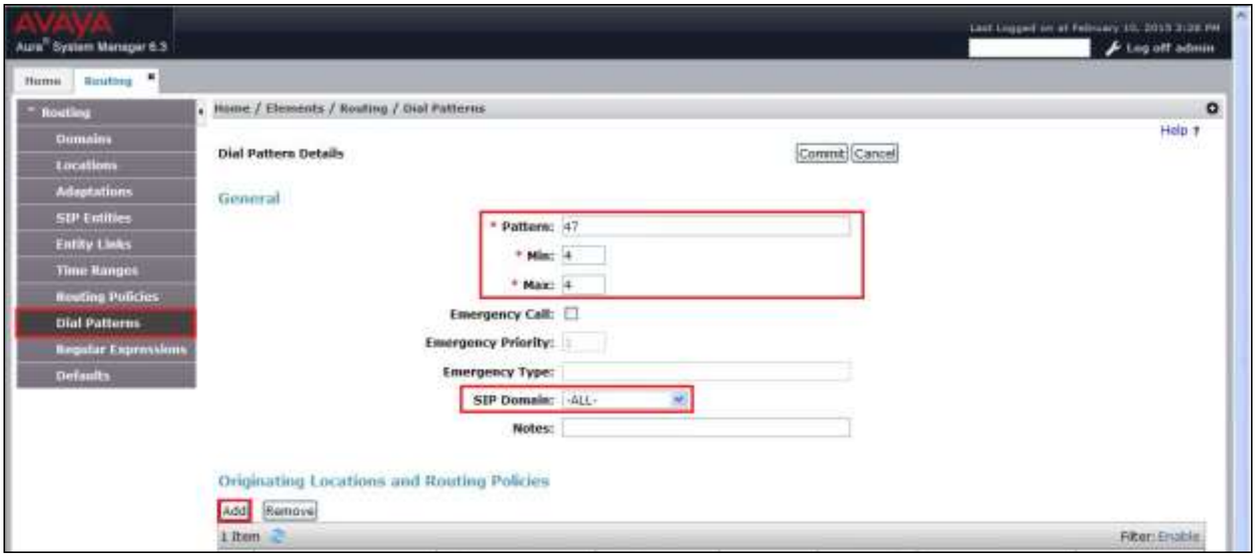
7.6. Create a Dial Pattern for CTI Connect

A dial pattern must be created on Session Manager to route calls to and from CTI Connect. During testing numbers beginning with **47** were sent to CTI Connect. To configure the CTI Connect Dial Pattern select **Dial Patterns** on the left panel menu and then click on the **New** button (not shown).

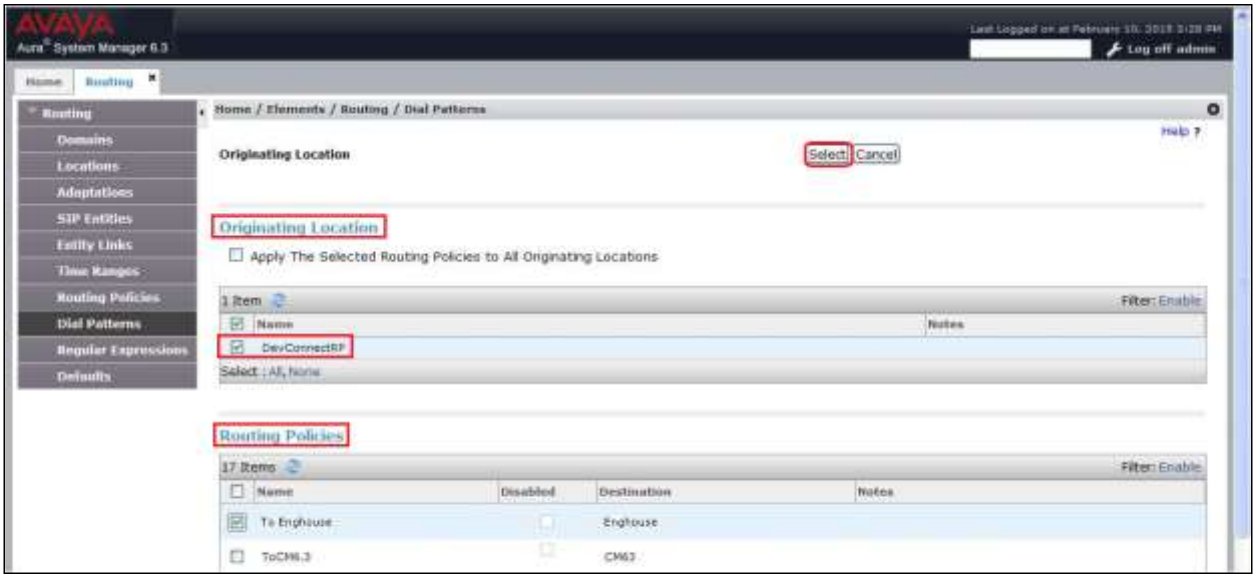
Under **General** carry out the following for each number:

- **Pattern** Enter **47**
- **Min** Enter **4** as the minimum length of dialed number
- **Max** Enter **4** as the maximum length of dialed number
- **SIP Domain** Select **All** from the drop down box

Click the **Add** button in **Originating Locations and Routing Policies**.



In **Originating Location** check the **DevConnectRP** check box. Under **Routing Policies** check the **To Enghouse** check box. Click on the **Select** button to confirm the chosen options and then be returned to the Dial Pattern screen (shown previously), select **Commit** button (not shown) to save.



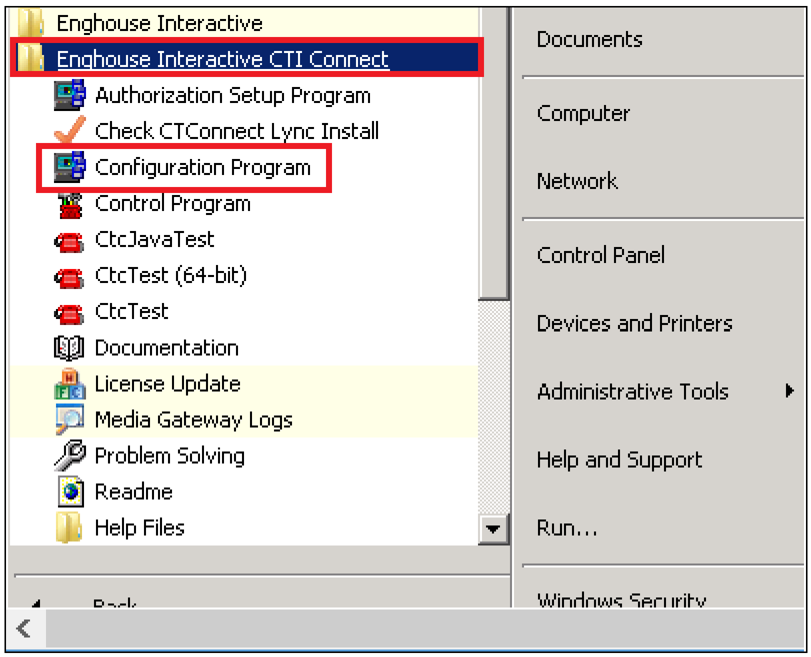
8. Configure CTI Connect server

This section shows how to configure CTI Connect server to successfully connect to CS1000E via Session Manager. The installation of the CTI Connect server software is assumed to be completed and the CTI Connect services are up and running. The steps to configure the CTI Connect server are as follows:

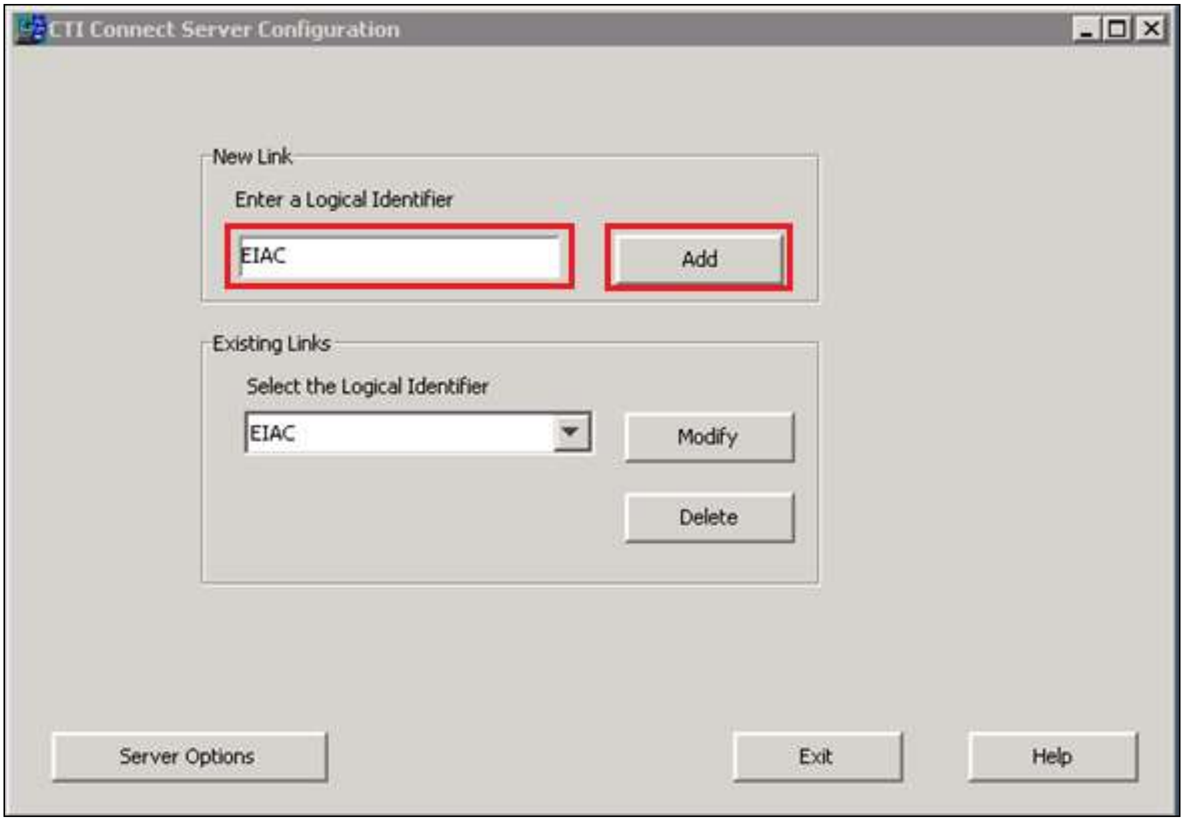
- Configure CTI Connect Configuration Program
- Turn on Link State

8.1. Configure CTI Connect Configuration Program

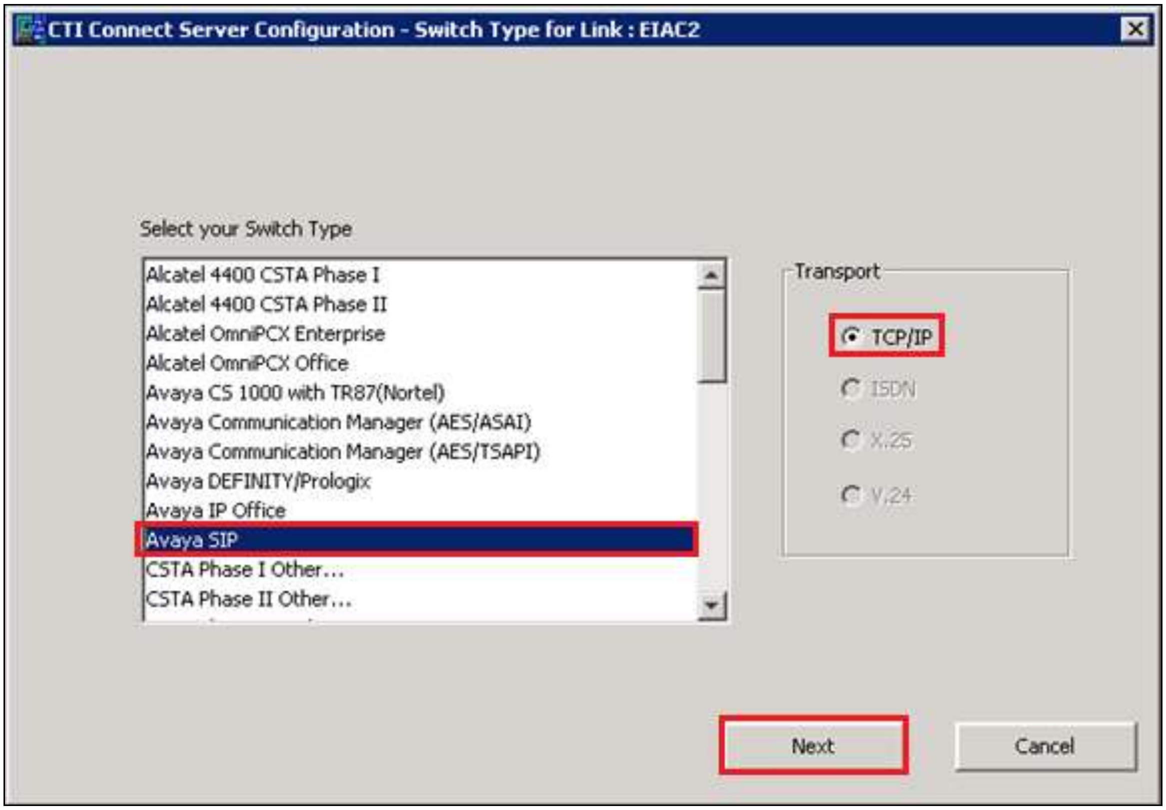
On the CTI Connect server launch the **Configuration Program** select **Start → All Programs** → (not shown) **Enghouse Interactive CTI Connect →Configuration Program**.



In the subsequent window, enter the **Logical Identifier** (i.e. EIAC, which is mandatory and cannot be anything different)and click on the **Add** button.



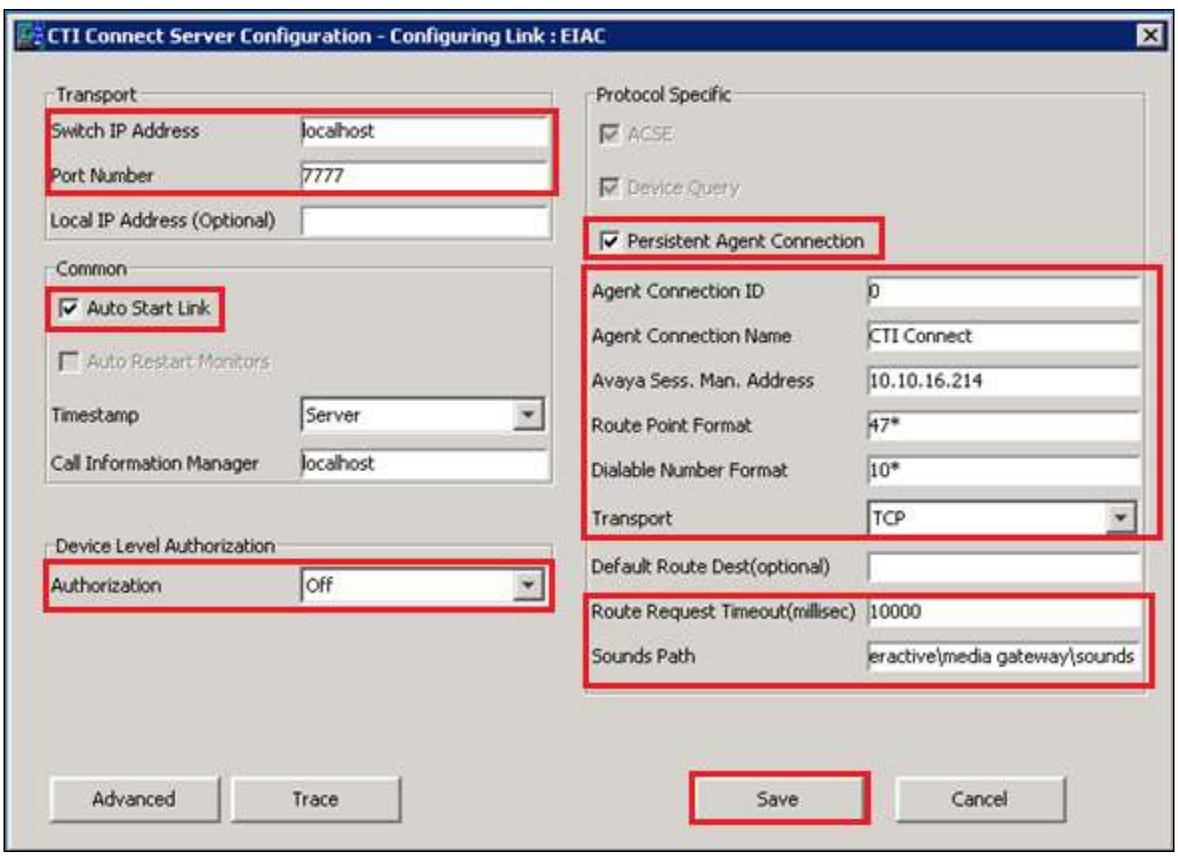
In the subsequent window click on the **TCP/IP** radio button and select **Avaya** SIP from the **Select your Switch Type** pane followed by the **Next** button to continue.



In the subsequent window enter the following settings:

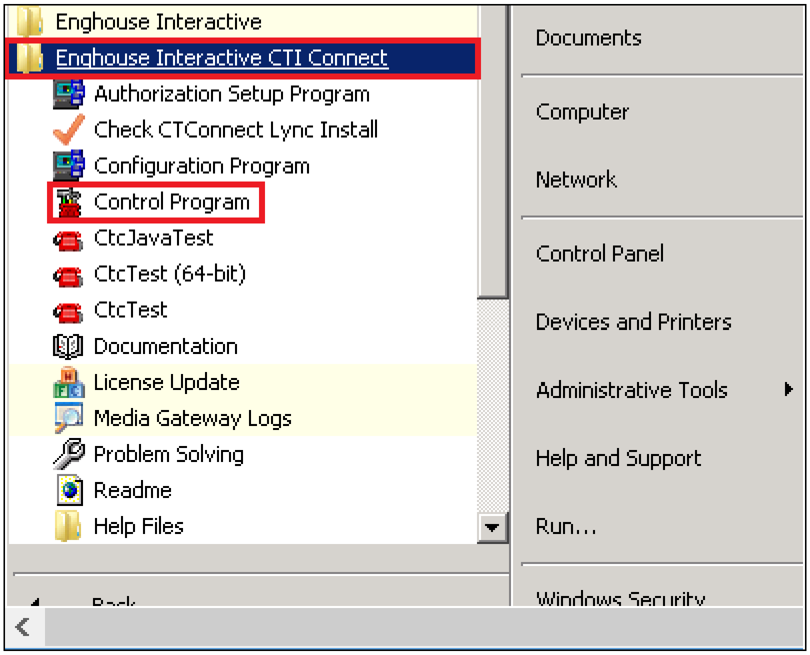
- **Switch IP Address** Enter **localhost**
- **Port Number** Enter **7777**
- **Auto start Link** Tick the check box
- **Authorization** Select **Off** from the dropdown list
- **Persistent Agent connection** Tick the check box
- **Agent Connection ID** Enter **0**
- **Agent connection Name** Enter **CTI Connect**
- **Avaya Sess. Man. Address** Enter the IP address of the Session Manager as seen in **Section 5.1**
- **Route Point Format** Enter **47*** (see **Section 6.5**)
- **Dialable Number Format** Enter the number range of CS1000E (i.e. **10***, any number beginning with 10 will be sent to CS1000E via Session Manager)
- **Transport** Select **TCP** from the dropdown list
- **Route Request Timeout** Enter **10000**
- **Sound Path** Enter the path where the voice prompts are stored

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

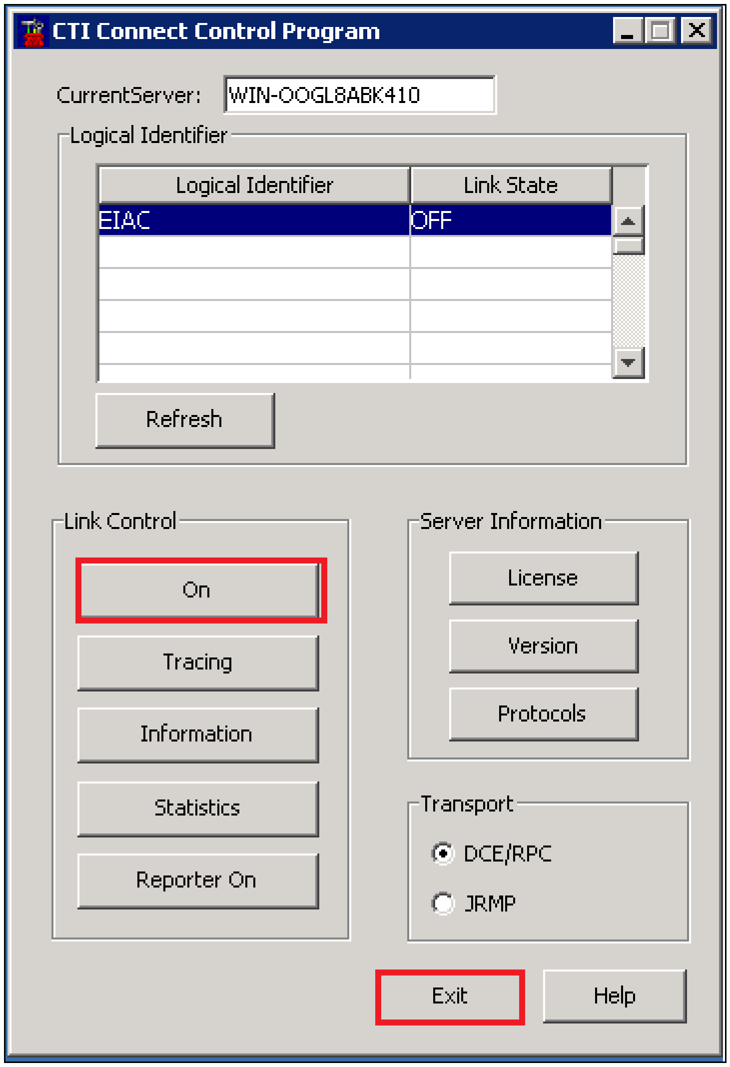


8.2. Turn on Link State

To turn on the Link State go to **Control Program** select **Start → All Programs →** (not shown) **Enghouse Interactive CTI Connect →Control Program**.

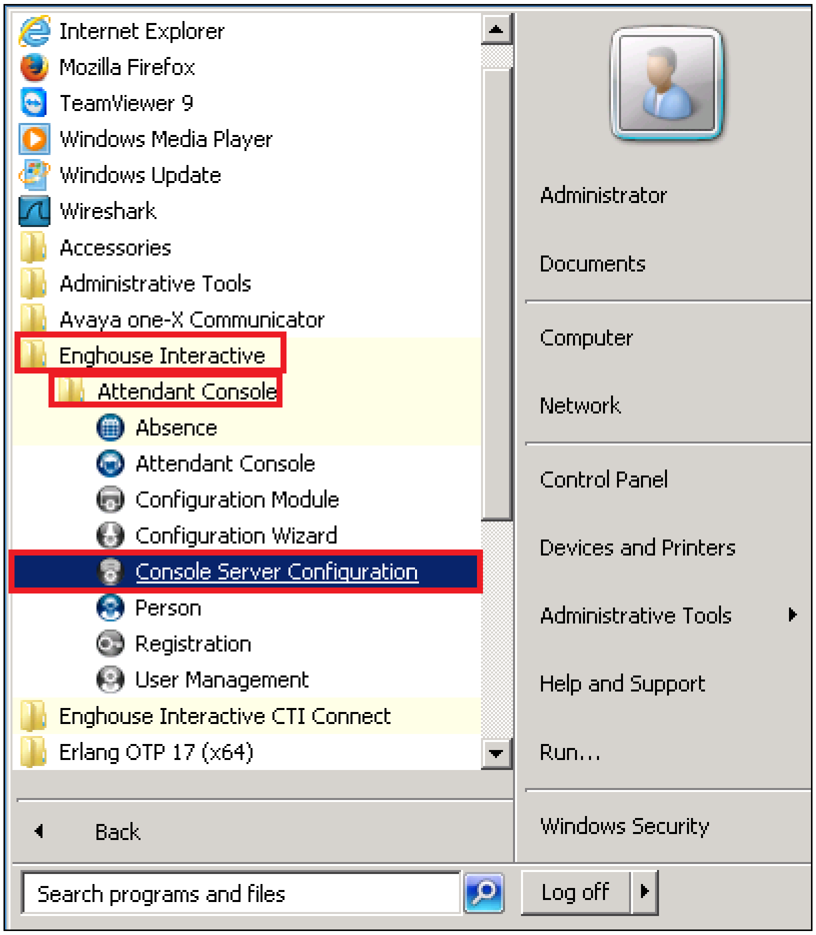


In the subsequent window click the **On** button, wait for the state to change and then **Exit**.



9. Configure Console server

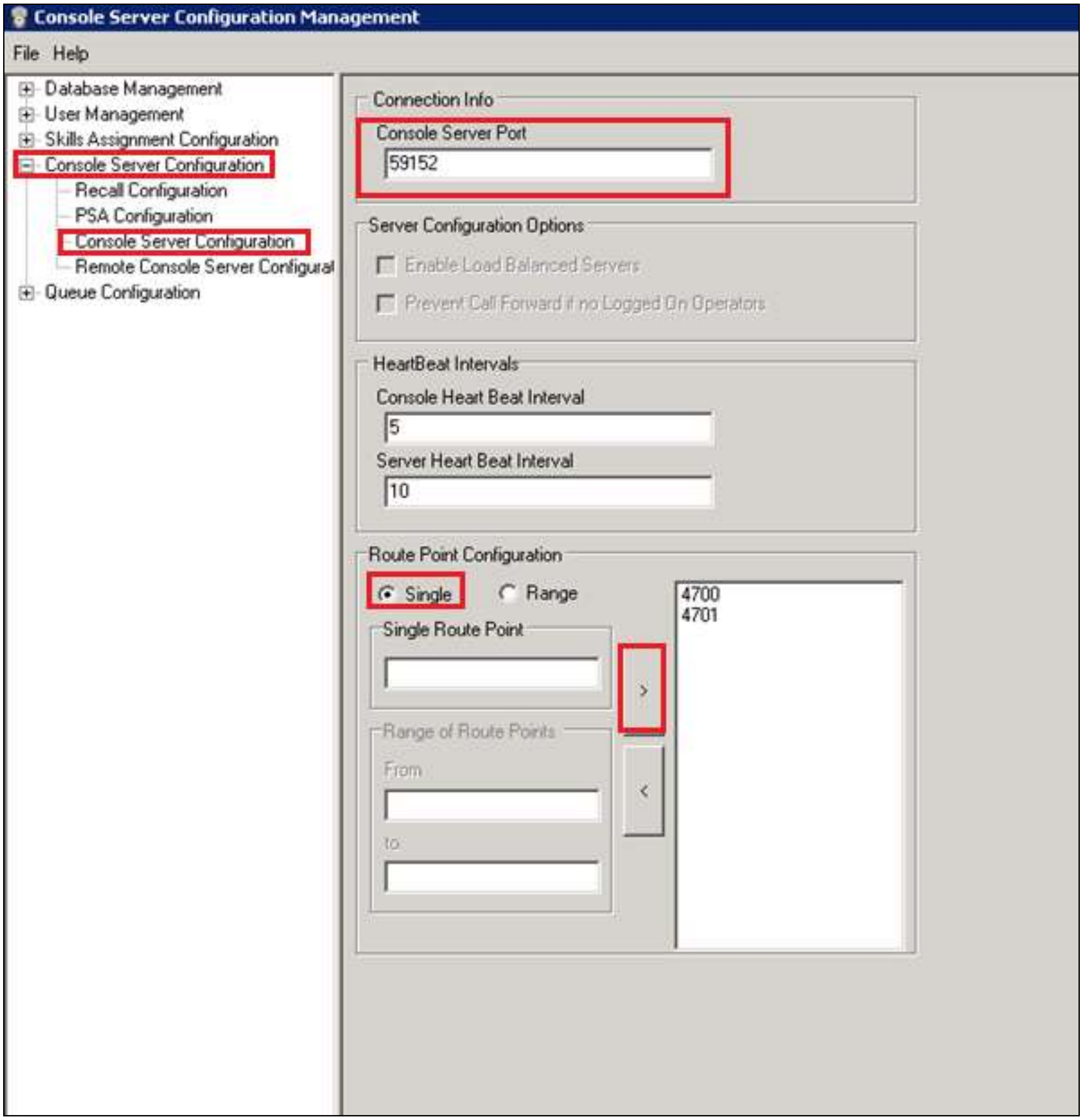
This section shows how to configure Enghouse Interactive Console server. To launch the **Console Server Configuration** select **Start** → **All Programs** → (not shown) **Enghouse Interactive** → **Attendant Console** → **Console Server Configuration**.



In the subsequent window enter **Intuition** in the **Database** field, **admin** in the **Login Name** field and click on the **OK** button.
Note: No Password is required.

9.1. Add Route Points for Queue

In the subsequent window navigate to **Console Server Configuration → Console Server Configuration**. In the **console Server Port** enter **59152**. Click the **Single** radio button. In the **Single Route Point** field enter the required Queue numbers. (In the example below **4700** and **4701** were added.

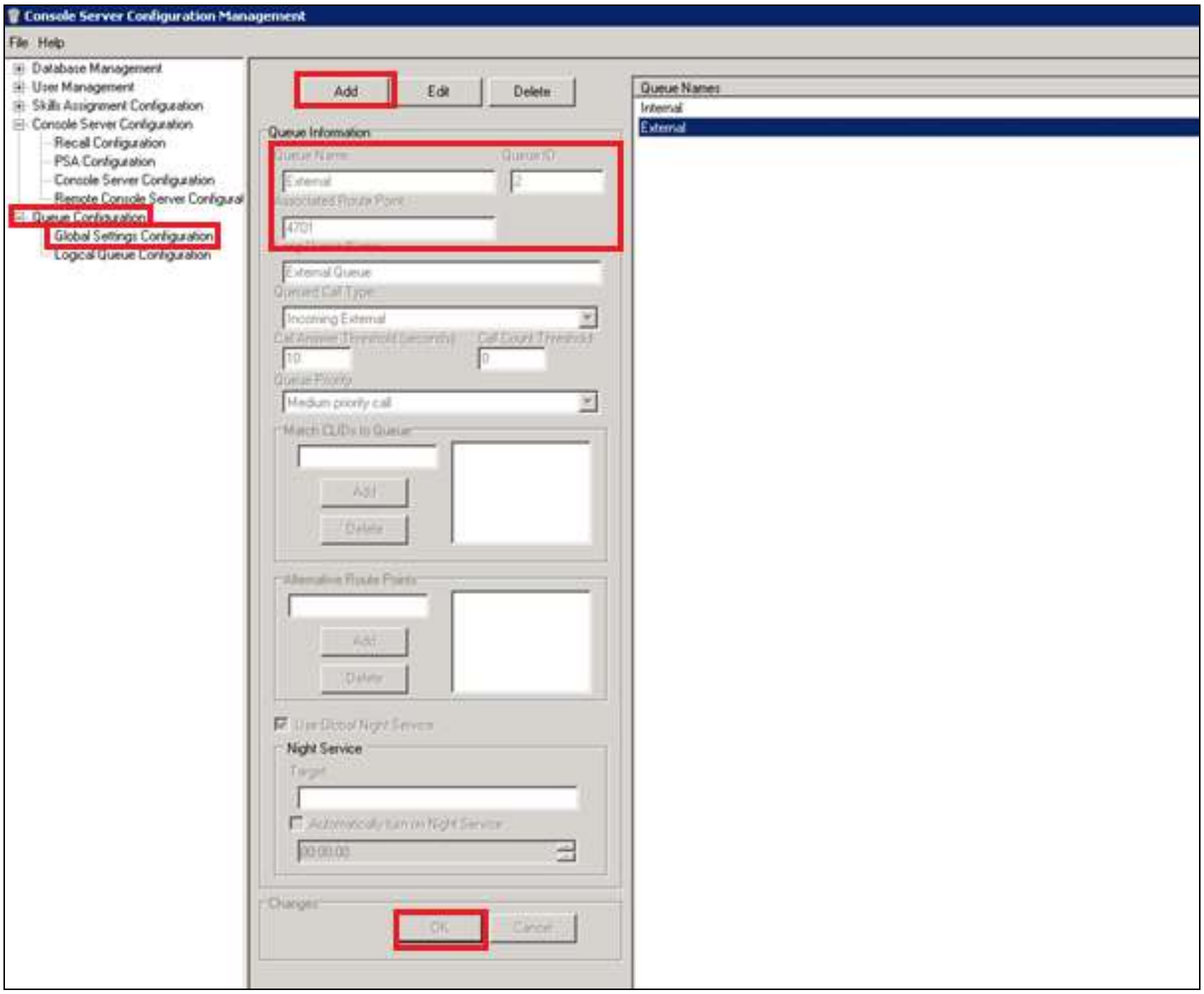


9.2. Add Queues

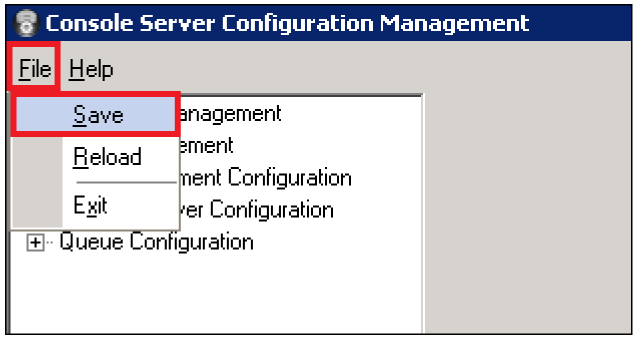
In the example below the **External** queue is configured. To add the Queues navigate to **Queue Configuration → Global Settings Configuration** and enter the following:

- **Queue Name** Enter **External**
- **Queue ID** Enter **2**
- **Associated Route Point** Enter **4701**

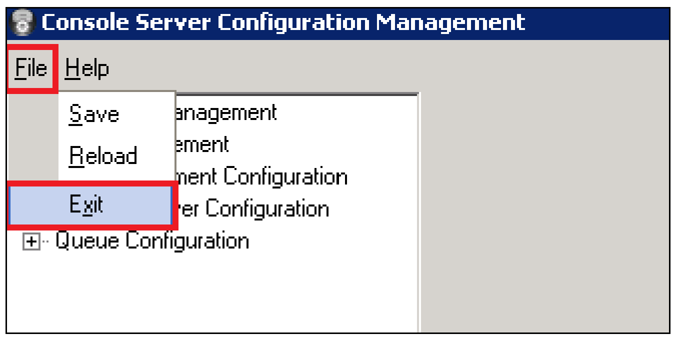
Click on the **Add** button. Repeat for the internal queue, where 4700 is the **Associated Route Point** and the **Queue ID** is **1**. When complete and click on the **OK** button.



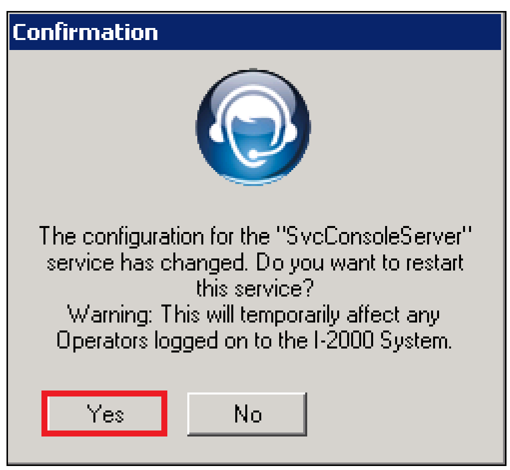
To save the configuration click on **File** followed by **Save**.



To exit **Console Server Configuration** click on **File** followed by **Exit**.

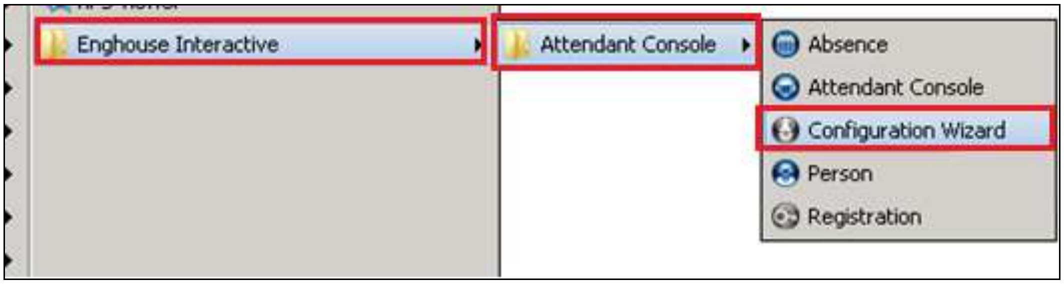


After exiting click on the **Yes** button in the subsequent **Confirmation** window to restart the Console server.



10. Configure Attendant Console Client

This section shows how to configure Enghouse Interactive CTI Connect Attendant Console Client. The installation of the Attendant Console Client software is assumed to be completed To configure Attendant Console client click on **Start →Programs →** (not shown) **Enghouse Interactive → Attendant Console → Configuration Wizard**.



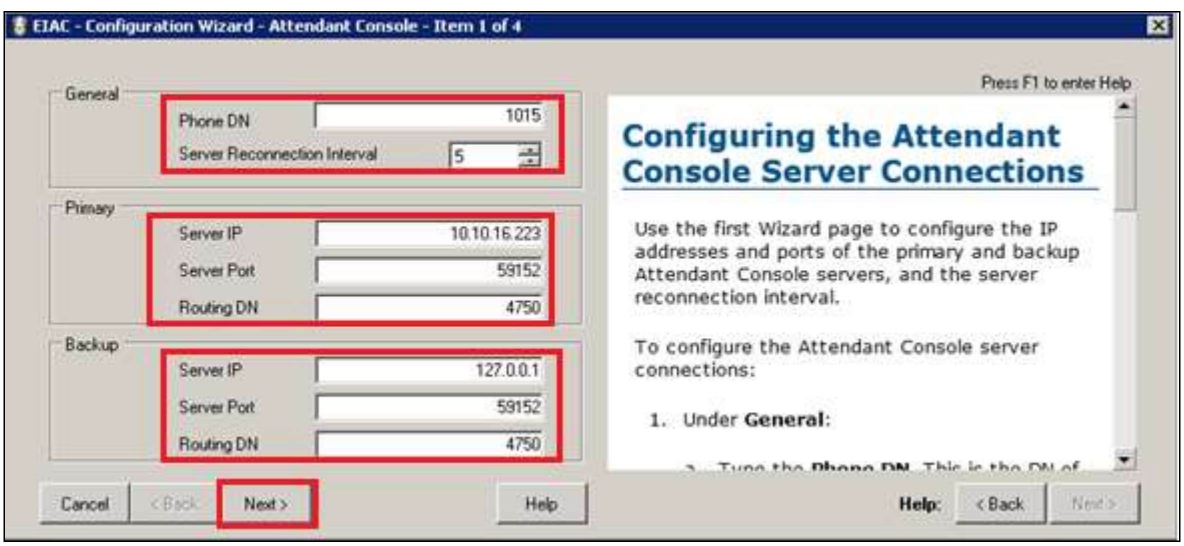
When the **Configuration Wizard** opens enter the following:

• **Phone DN**	Enter the extension of the telephone configured on the CS1000E to act as the Attendant telephone. (i.e. **1015**)
• **Server Reconnection Interval**	Enter **5**
• **Server IP**	Enter the IP address of the CTI Connect Server
• **Server Port**	Enter **59152**
• **Routing DN**	Enter **4750** (this is an internal number used for the agent using the Attendant client)

Backup (if no Backup server is available configure the following:

• **Server IP**	Enter **127.0.0.1**
• **Server Port**	Enter **59152**
• **Routing DN**	Enter **4750**

Click on the **Next** button (3 times, not shown) button to continue followed by the **Finish** button.



11. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the Avaya and Enghouse Interactive solution.

11.1. 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.

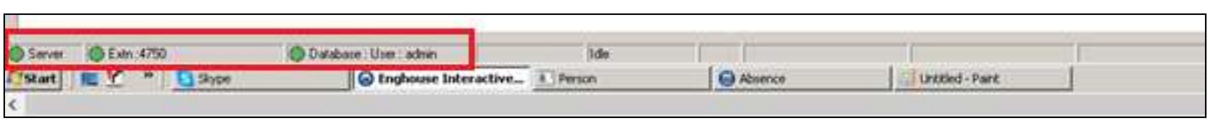
Prompt	Response	Description
>	**LD 96**	Enter Overlay 96
.	**STAT DCH**	Check status of all D-Channels
DCH 066	**OPER EST**	**DES : SM**

11.2. Verify the SIP Entity Link status for Attendant Console

From System Manager select **Session Manager** from under the **Elements** column, not shown. When the **Session Manager** tab opens select **System Status** followed by **SIP Entity Monitoring**, then click on **Enghouse** SIP Entity (not shown) created in **Section 7.4**, ensure that the **Conn. Status** is **Up**, the **Reason Code** is **200OK** and the **Link Status** is **Up**.

11.3. Verify Attendant Console

Login to a Attendant Console client using the appropriate credentials (not shown) and verify the **Server**, **Extn** and **Database** icons are green as per the screenshot below.



12. Conclusion

A full and comprehensive set of feature and functional test cases were performed during Compliance testing. Enghouse Interactive Attendant Console 8.1 is considered compliant with Avaya Communication Server 1000E 7.6 and Avaya Aura® Session Manager 6.3. All test cases have passed and met the all objectives with one observation as outlined in **Section 2.2**.

13. Additional References

These documents form part of the Avaya official technical reference documentation suite. Further information may be had from *http://support.avaya.com* or from an Avaya representative.

[1] Software Input Output Reference — Administration Avaya Communication Server 1000 7.6, NN43001-611, 06.01. March 2013

[2] Software Input Output Reference — Maintenance Avaya Communication Server 1000 7.6, NN43001-711, 06.01. March 2013

[3] Administering Avaya Aura® Session Manager, Release 6.3

[4] Administering Avaya Aura® System Manager, Release 6.3

Product Documentation for Enghouse Interactive AB can be obtained in the installed software or at: https://mysupport.enghouseinteractive.com

Appendix A

To add the CS1000E Adaptation to the CS1000E SIP Entity, in the navigation pane select **SIP Entities** (the complete configuration of the SIP Entity is not shown in this section) and select the SIP Entity configured for the CS1000E (i.e. CS1kRP).When the **SIP Entity Details** window opens select **CS1000 Adapter** (created in **Section 7.2**) from the **Adaptation** dropdown box.

Click on the **Commit** button to save the configuration.

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