Avaya Solution & Interoperability Test Lab

Application Notes for Vision 8020HA to interoperate 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 Vision 8020HA, which is operating as an attendant answering position from Enghouse Interactive AB.

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 Vision 8020HA from Enghouse Interactive AB to interoperate with Avaya Communication Server 1000E R7.6 (CS1000E) and Avaya Aura® Session Manager R6.3 (Session Manager). Vision 8020HA is a client/server based application (Server running on CentOS and Client running on Microsoft Windows 2008 Server operating systems). Vision 8020HA provides users with an attendant answering position for CS1000E. The Vision 8020HA Attendant client provides a view of contacts, schedules, and communication tasks. Vision 8020HA also includes its own voicemail solution. Vision 8020HA connects to the CS1000E using a SIP trunk via Session Manager. Vision 8020HA is supplied with all prerequisite software.

2. General Test Approach and Test Results

The general test approach was to configure a simulated enterprise voice network using CS1000E. The Vision 8020HA server Communicates with the CS1000E using a SIP trunk 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 Vision 8020HA attendant position. If a call is made from the Vision 8020HA attendant to the PSTN the call will route from the Vision 8020HA attendant via a SIP trunk to the CS1000E, and then to the PSTN using the CS1000E PSTN connection. Vision 8020HA attendant 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 Vision 8020HA attendant. The CS1000E routes all calls destined for the Vision 8020HA attendant over the SIP connection. The Vision 8020HA server then automatically places a call to the telephone the attendant is using for answering purposes. When the attendant answers the call, the Vision 8020HA server bridges the two calls. It is possible to have multiple Vision 8020HA attendant positions on a CS1000E system.

Note: During compliance testing an Avaya 1140E was used as the attendant's deskphone. When the attendant is called the Vision 8020HA 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 interoperability compliance testing included feature and serviceability testing. The serviceability testing introduced failure scenarios to see if Vision 8020HA 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 insure full interoperability between Vision 8020HA and CS1000E. The tests were all functional in nature and performance testing was not included. All the test cases passed successfully.

Note: When a Vision 8020HA attendant transfers a call to another Telephone on the CS1000E or on the PSTN there is no route optimization. Route optimization/Path replacement is not supported on Vision 8020HA

2.3. Support

For technical support for Enghouse Interactive AB products, please use the following web link.
http://www.trio.com/web/Support.aspx

Enghouse Interactive AB can also be contacted as follows.
Phone: +46 (0)8 457 30 00
Fax: +46 (0)8 31 87 00
E-mail: triosupport@enghouse.com

3. Reference Configuration

Figure 1 illustrates the network topology used during compliance testing. The Avaya solution consists of a CS1000E, System Manager and Session Manager. The Vision 8020HA server connects the CS1000E using a SIP Trunk via Session Manager. The Vision 8020HA can control a telephone on the CS1000E to act as the Attendant telephone using the SIP CTI feature of the CS1000E. An Avaya 1140 IP deskphone was used as the Vision 8020HA Attendant telephone during compliance testing. Other Avaya deskphones were configured on the CS1000E to generate outbound/inbound calls to/from the PSTN.

Note: The Vision 8020HA Attendant (client) was installed on the same server as the Vision 8020HA Server, but can be installed on a separate platform if required.

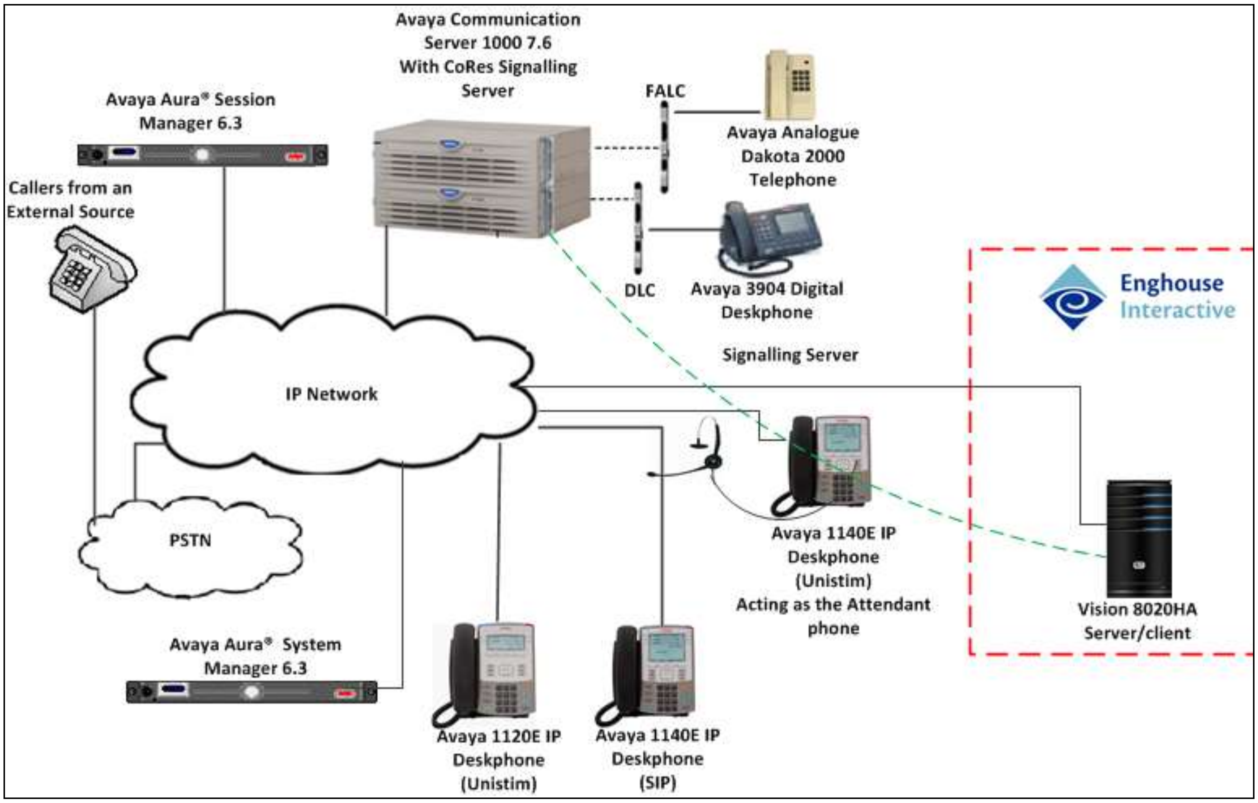


Figure 1: Avaya and Vision 8020HA Reference Configuration

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) • 500 (Analogue)	 0624C8Q 0624C8Q 04.03.12.00 N/A N/A
Enghouse Interactive Equipment	**Software / Firmware Version**
Vision 8020HA client running on Microsoft Windows 2008 R2 Server Vision 8020HA server running on Centos 6.0	Version 3.0 Version 3.0

5. Configure Avaya Aura® 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 (Rout 21) and D-Channel (DCH 66). For all other provisioning information such as Installation and Configuration, please refer to the product documentation in **Section 11**. 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

Both SIP CTI Licences and AST licenses are required to allow Trio observe TR87 events. To ensure the CS1000E is licensed for SIP CTI use **LD 22** and type **SLT** at the **REQ** prompt. Check for **SIP CTI TR87** and **AST** (in bold below).

LD 22

```
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 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	**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 the Vision 8020 attendant a dialling plan is required. The numbers configured are routed to the Session Manager, where a Dialling Pattern (see **Section 7.5**) is configured to route the calls to the Vision 8020 server. There are a number of ways to setup a dialling plan. For compliance testing a Coordinated Dialing Plan (CDP) was used. During compliance testing all 4 digit numbers beginning with 45 were routed to the Vision 8020HA server 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

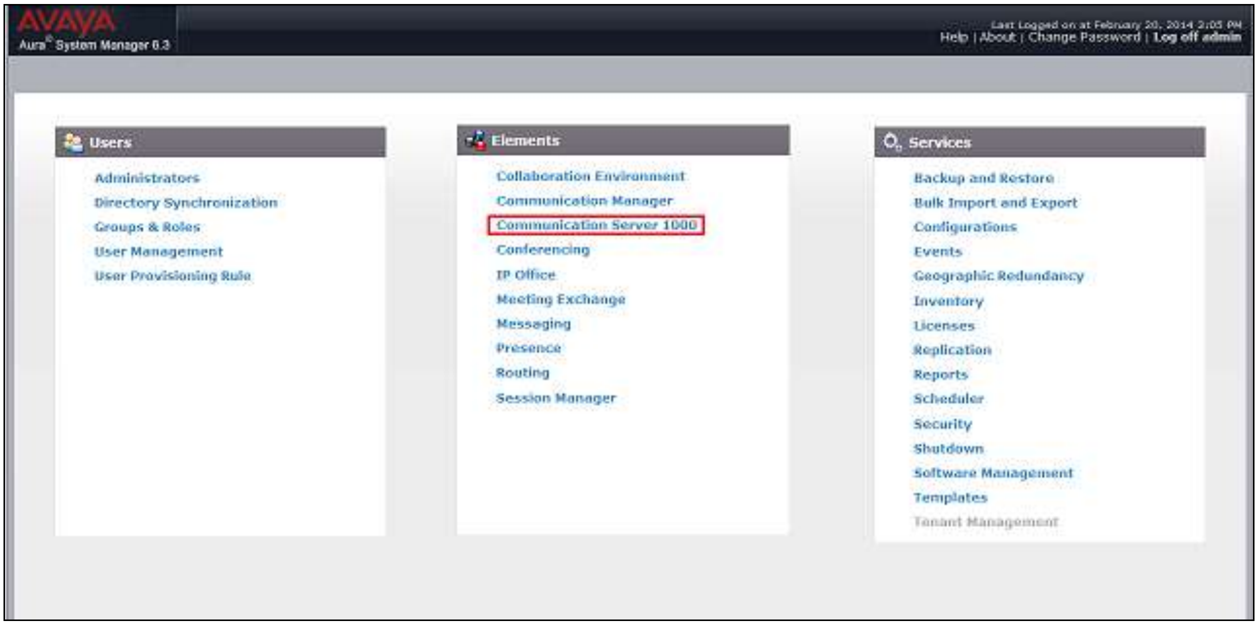
Use the **NEW** command in **LD 87** to create a CDP entry to route calls to the Session Manager. In the example below, **FEAT** is **CDP**, **DSC** is **45, FLEN** is **4** and the **RLI** is **22** (RLI created in **Section 5.4.1**).

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	**45**	Distant Steering code
FLEN	**4**	Flexible Length number of digits
RLI	**22**	Route list index Number

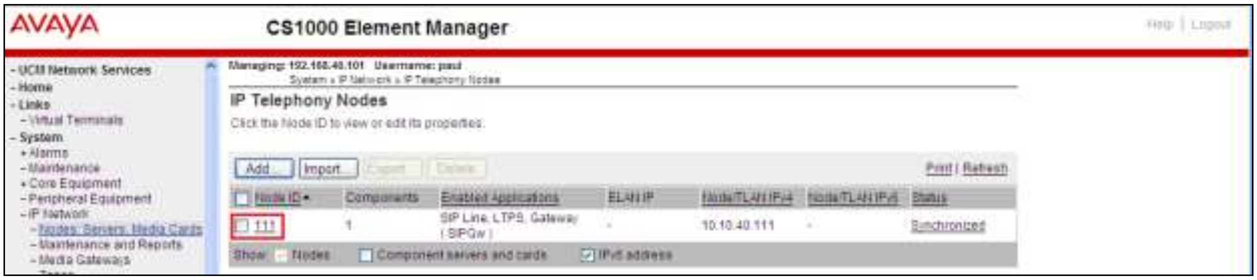
6. Configure Trunk Group for Vision 8020HA

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.



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.

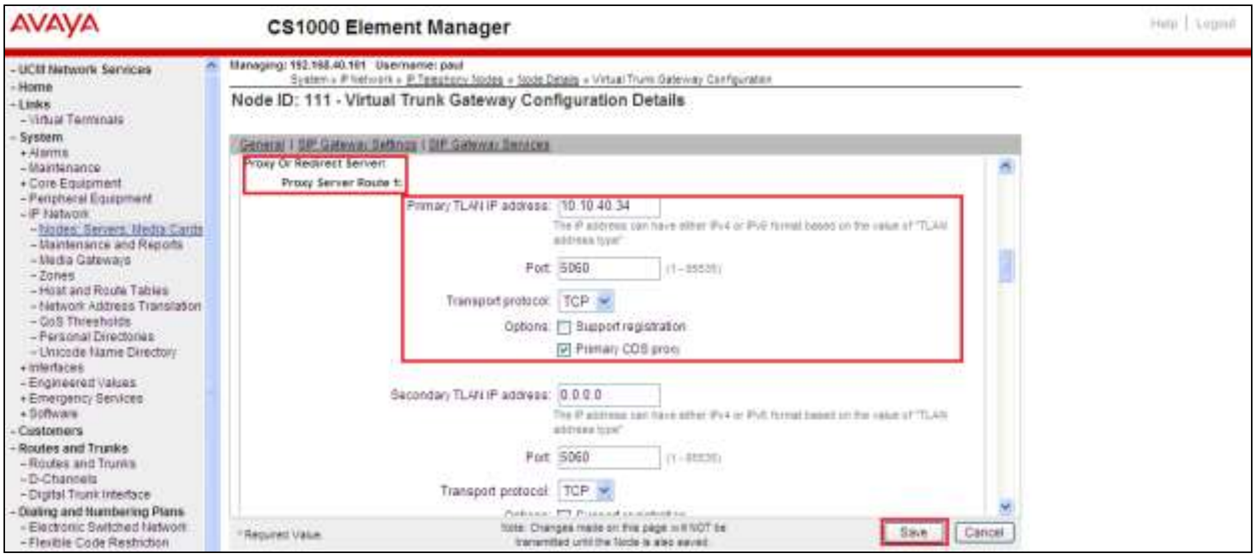


Once the **Node Details** page opens scroll down using the vertical scroll bar on the right side of the page and click on **Gateway (SIPGw)**.

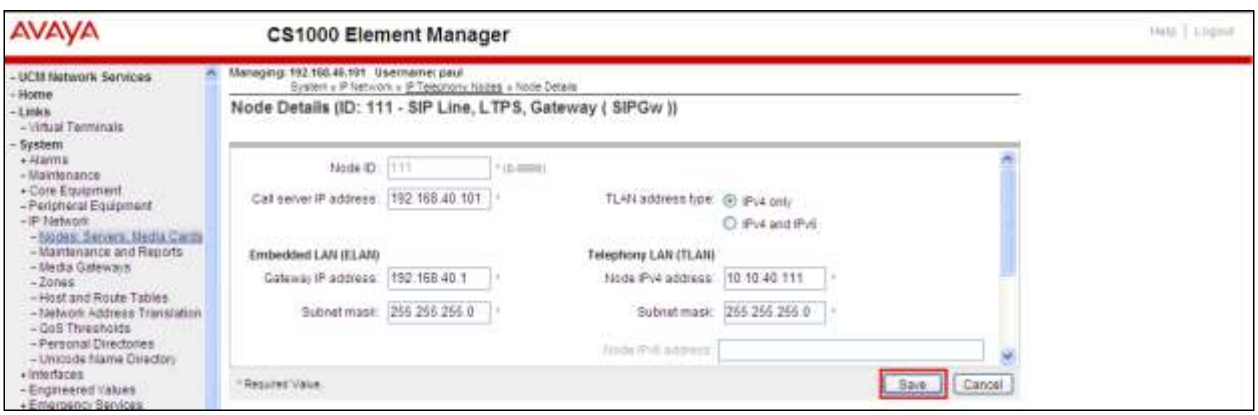
Once the **Virtual Trunk Gateway Configuration Details** page opens, scroll down using the vertical scroll bar on the right side of the page to **Proxy Or Redirect Server** (**Proxy Server route 1**) and enter the following:

- **Primary TLAN IP address** Enter the IP address of the Session Manager (10.10.40.34)
- **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

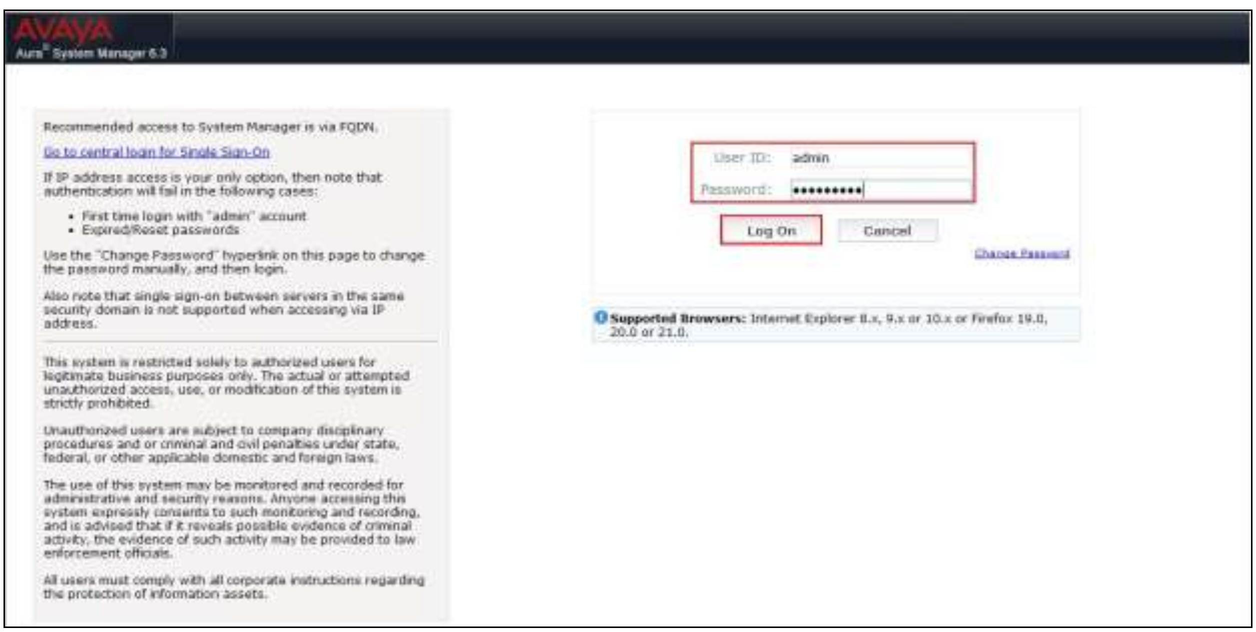
A number of configurations are required to enable the CS1000E to route calls to/from Vision8020HA. All configurations of the Session Manager are performed using System Manager. The configuration operations described in this section can be summarized as follows: (Note: during Compliance Testing all inputs not highlighted with a red box were left as Default)

- Logging on to System Manager
- Create Vision8020HA server as a SIP Entity
- Create an Entity Link for Vision 8020HA Server
- Create a Routing Policy for Vision 8020HA Server
- Create a Dial Pattern for Vision 8020HA Server

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 this Application Note.

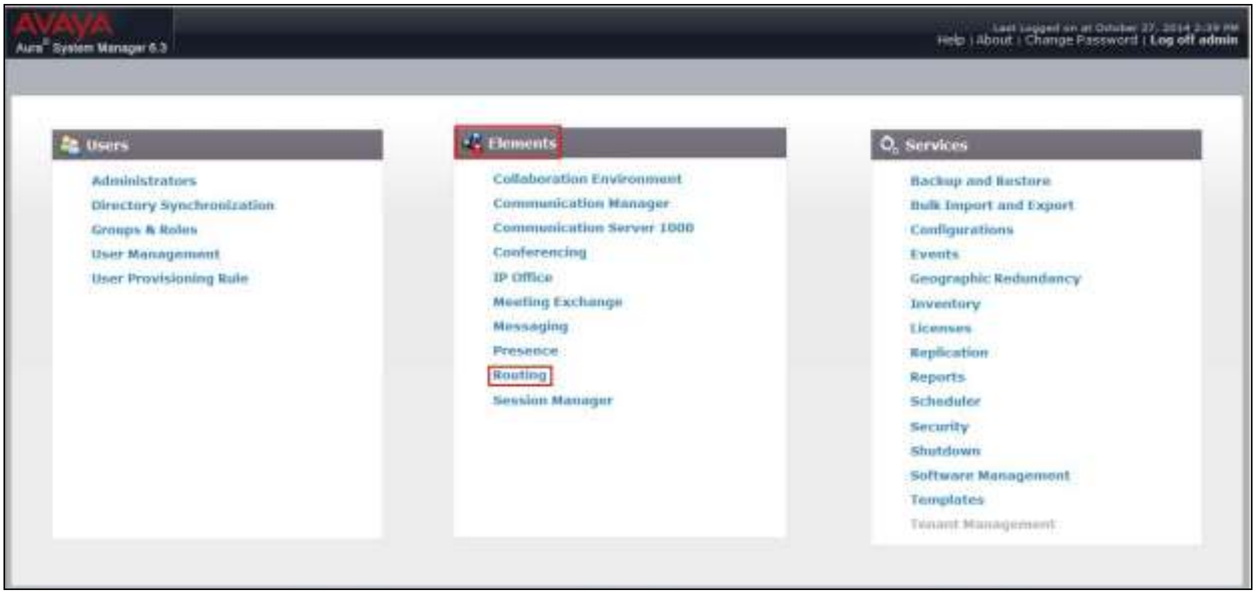
7.1. Logging on to System Manager

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 Avaya Aura® System Manager or the"<ipaddress>" is the IP address of Avaya Aura® System Manager. Once the System Manager Web page opens Log in with the appropriate credentials.



7.2. Create Vision 8020HA Server as a SIP Entity

A SIP Entity must be added for the BTS Call Director server.
Note: A SIP Entity was already configured for the CS1000E and was called **CS1KPG1**. Once logged into the System Manager, select the **Routing** Link under the **Elements** column.



Click on **SIP Entities** link followed by **New** (not shown), and once the SIP Entity Details page opens, enter the following:

- **Name** Enter an informative name (e.g. **Trio**)
- **FQDN or IP Address** enter the IP address of the signalling interface of the Vision 8020HA server
- **Type** Select **SIP Trunk** from the dropdown box
- **Location** Select an appropriate **location** from the dropdown box, (**DevConnectRP** was used during compliance testing)
- **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.

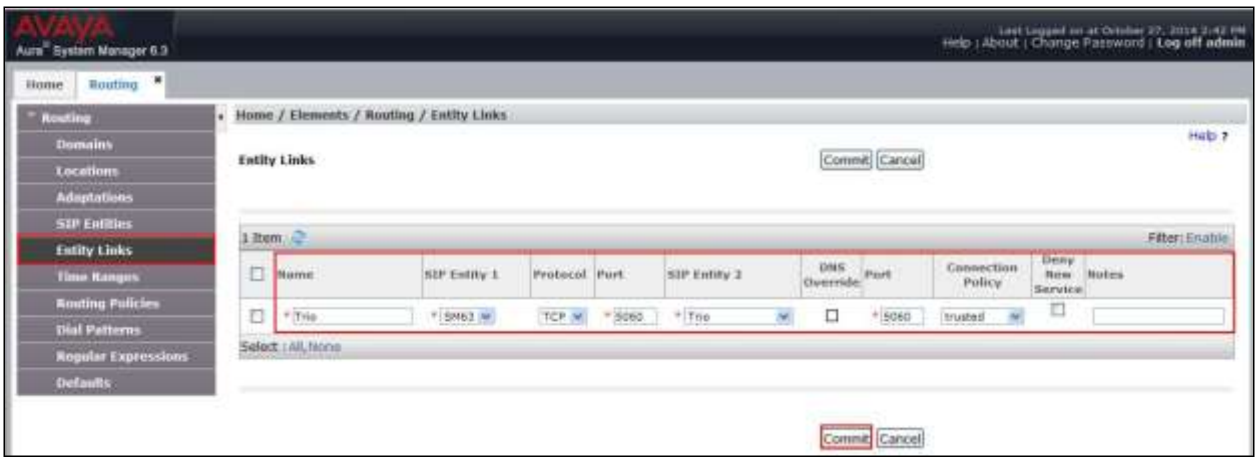
Note: During compliance testing **Adaptation** was left blank.

7.3. Create an Entity Link for Vision 8020HA Server

The SIP trunk between the Session Manager and the 8020HA 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. **Trio**)
- **SIP Entity 1** Select **Session Manager (SM63)** from the **SIP Entity 1** dropdown box
- **Protocol** Select **TCP** from the Protocol drop down box
- **Port** Enter **5060**
- **SIP Entity 2** Select **Trio** from the **SIP Entity 2** dropdown box (configured in **Section 7.2**)
- **Port** Enter **5060** as the Port
- **Connection Policy** Select **Trusted** from the dropdown box

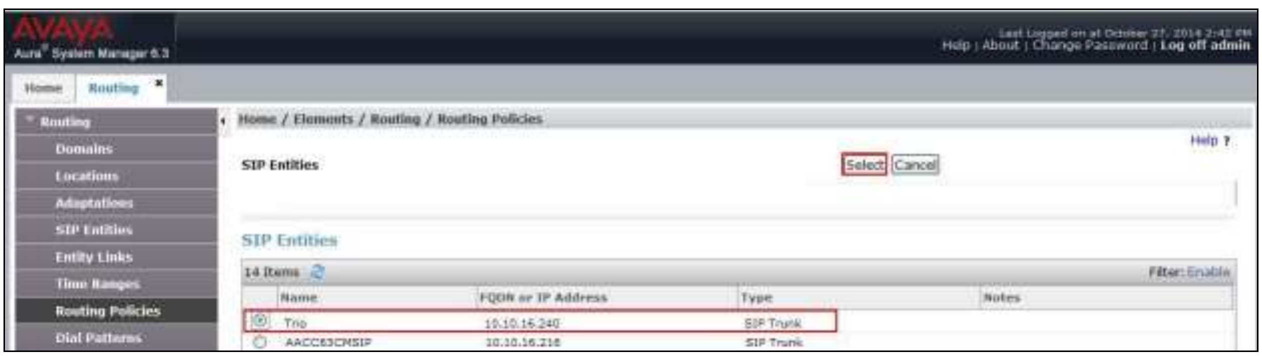
Click **Commit** to save changes. The following screen shows the Entity Links used.



7.4. Create a Routing Policy for Vision 8020HA Server

Create routing policies to direct calls to 8020HA Server. 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, **Trio**) and enter **0** in the **Retries** field**.** In **SIP Entity as Destination**, Click **Select**.

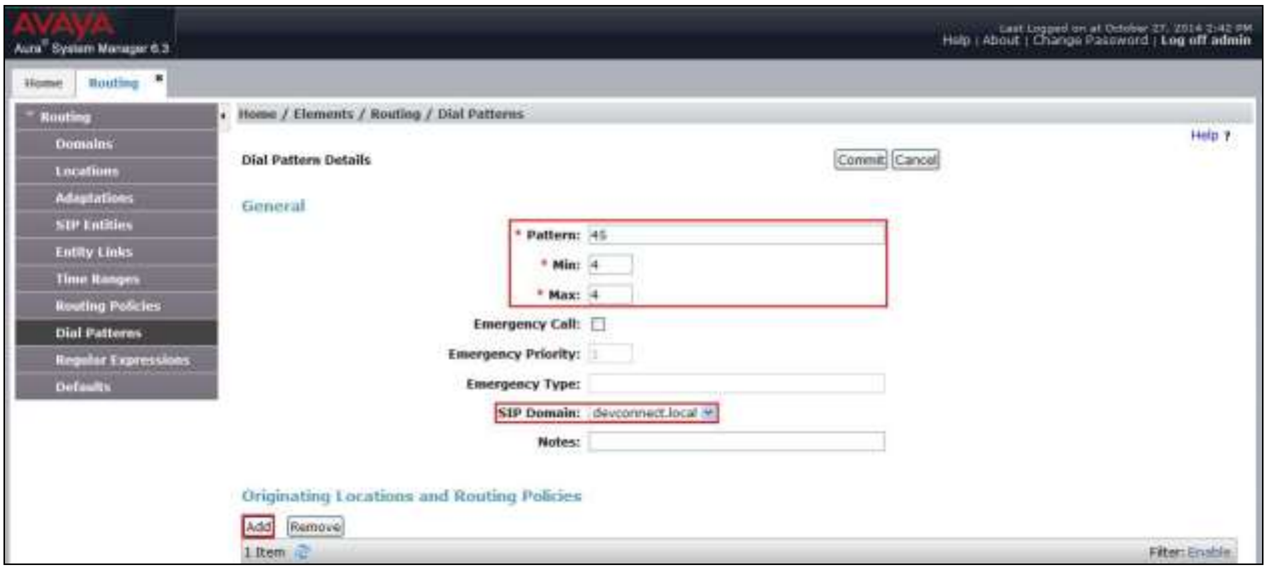
Once the SIP Entity List screen opens, check the **Trio** radio button. Click on the **Select** button to confirm the chosen options and then return to the **Routing Policies Details** screen and select **Commit** button (Not shown) to save.



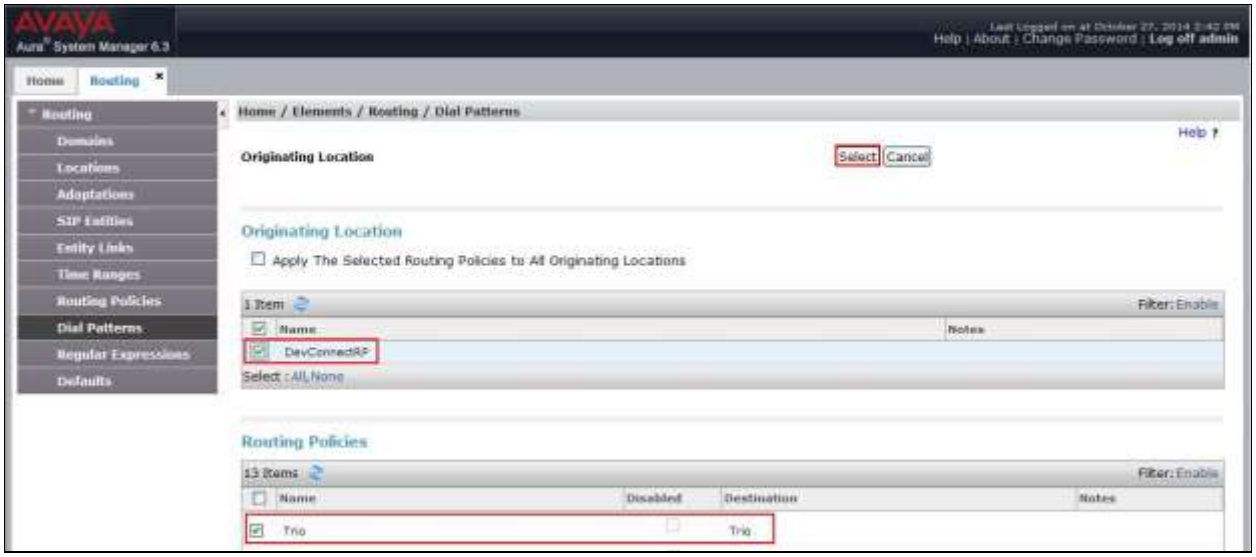
7.5. Create a Dial Pattern for Vision 8020HA Server

A dial pattern must be created on the Session Manager to route calls to and from the 8020HA Server. See **Section 5.1** for the numbers used during testing. A Dial Pattern of **45** was created which routed any number beginning with 45 and four digits long to the 8020HA Server. 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 45
- **Min** Enter 4 as the minimum length of dialled number
- **Max** Enter 4 as the maximum length of dialled number
- **SIP Domain** Select **devconnect.local** from the drop down box
- Click the **Add** button in **Originating Locations and Routing Policies**.



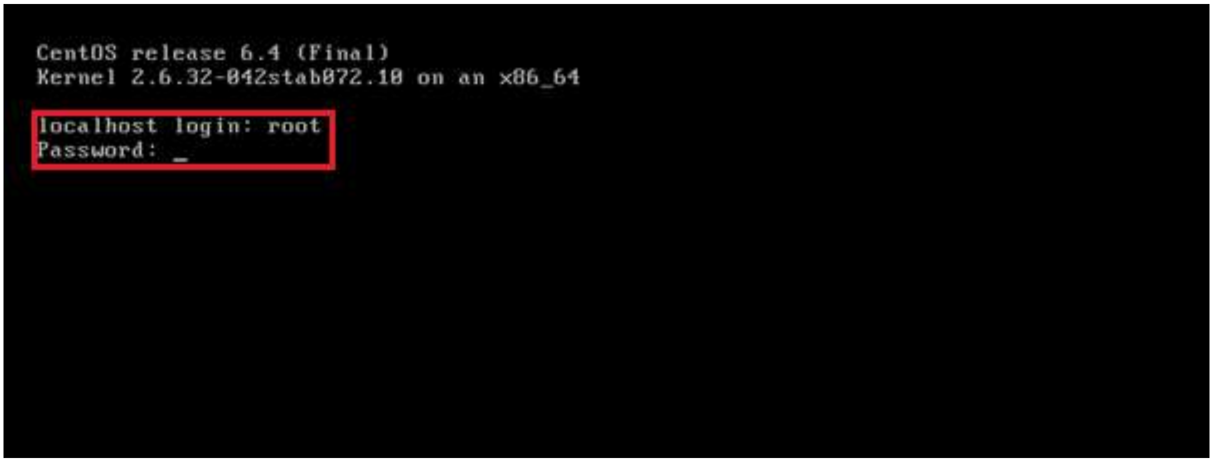
In **Originating Location page** check the **DevConnectRP** check box. Under **Routing Policies** check the **Trio** check box. Click on the **Select** button to confirm the chosen options to return to the Dial Pattern screen (shown previously), select **Commit** button to save (not shown).



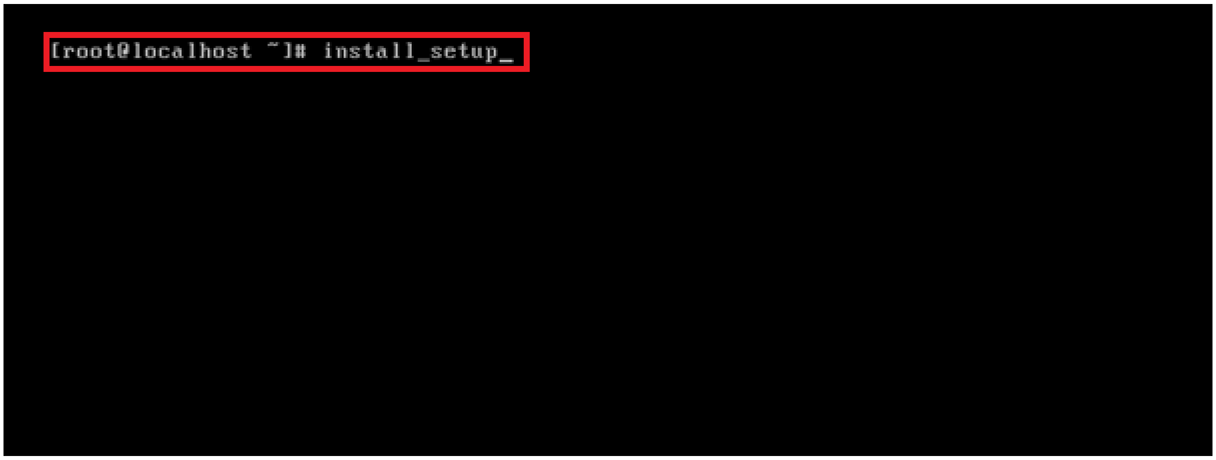
8. Configure Vision 8020HA Call Server for VoIP

This section shows how to configure Vision 8020HA to successfully connect to Session Manager. The installation of the Vision 8020HA software is assumed to be completed and the correct licence is installed.

Using SSH connect to the Vision 8020HA server and login as root with the appropriate password.



At the **root@**localhost prompt start the configuration program using the **install_setup** command.

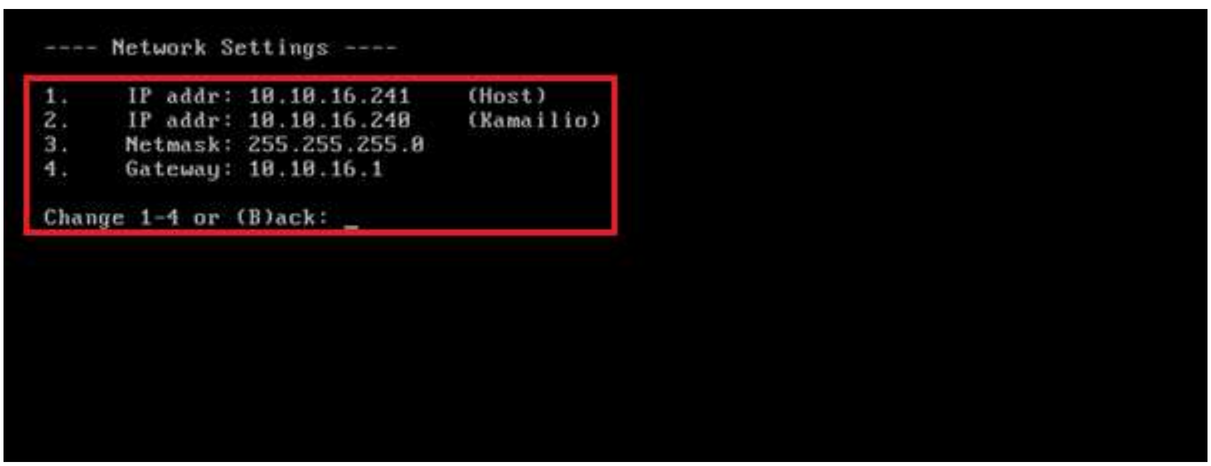


Select option **2** to configure the **Network**.

Using the menu options, set;

1. IP Address for the host (Centos Server)
2. IP Address for Kamailio (where the CS1000E is sending SIP traffic)
3. Netmask
4. Default gateway

Select **(B)ack** when completed.



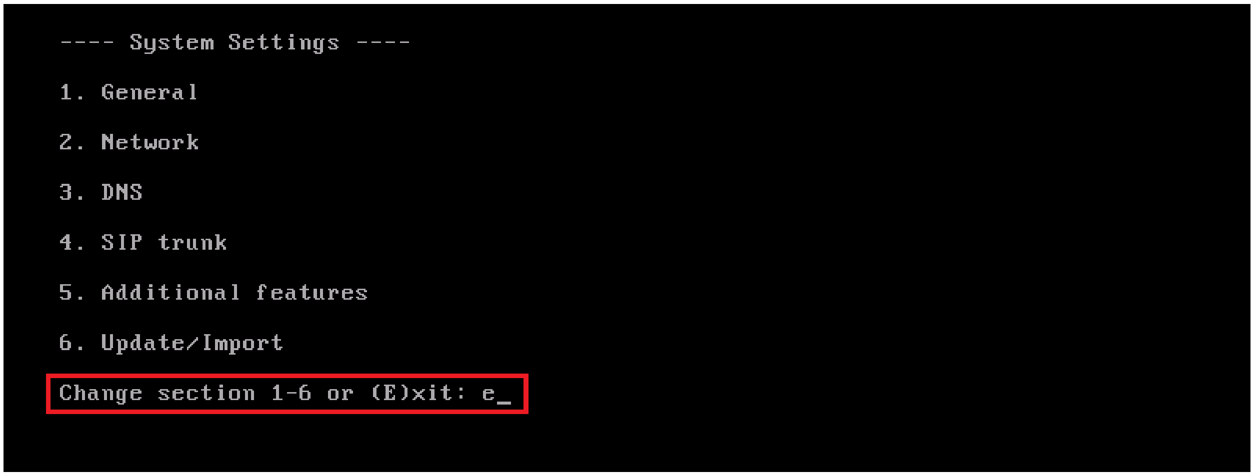
Select option 3 if using a DNS. During compliance testing no DNS was used.

Select option **4** to configure the **SIP trunk**.

Using the menu options, set:
1. IP Address for the Session Manager
2. the TCP port used on the Session Manager

Select **(B)ack** when completed

Select **(E)xit** when complete.



Run the command **shutdown –r now** to reboot the server.

9. Verification Steps

This section provides the tests that can be performed to verify correct configuration of the Avaya and Vision 8020HA solution.

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.

```
Prompt         Response             Description
>              LD 96                Enter Overlay 96
.              STAT DCH             Check status of all D-Channels

DCH 066        OPER EST             DES : to_Trio
```

10. Conclusion

A full and comprehensive set of feature and functional test cases were performed during Compliance testing. Vision 8020HA from Enghouse Interactive AB 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.

11. 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 your 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, Issue 3 October 2013

[4] Administering Avaya Aura® System Manager, Release 6.3, Issue 3, October, 2013

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

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