



**Application Notes for Configuring the Esna Office-LinX
Version 8.2 with Avaya Communication Server 1000E
Release 7.5 and Avaya Aura® Session Manager Release 6.1
- Issue 1.0**

Abstract

These Application Notes describe the procedure for configuring the Esna Office-LinX v8.2 to interoperate with the Avaya Communication Server 1000E Release 7.5 and Avaya Aura® Session Manager Release 6.1.

The Office-LinX Enterprise Edition server connects to the Avaya Communication Server 1000E via SIP Trunk connectivity and provides unified communications features such as greeting menu, user mailbox services, wake up services and transfer functionalities.

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 procedure for configuring the Esna Office-LinX v8.2 (Office-LinX) to interoperate with Avaya Communication Server 1000E R7.5 (CS1000E) and Avaya Aura® Session Manager R6.1. The objective of this compliance testing is to verify that Office-LinX connects to the CS1000E via SIP trunks providing unified communication services such as greetings, messaging and transfer functionalities.

2. General Test Approach and Test Results

The general test approach was to place calls to Esna Office-LinX server, and the main objectives were to verify that the user can:

- Successfully establish calls to Office-LinX from/to the CS1000E end points.
- Successfully transfer from Office-LinX.
- Successfully leave messages for subscribers and to retrieve the same.

The Esna Office-LinX server was tested for serviceability and the objectives were to verify that:

- Office-LinX can successfully re-establish a connection to the CS1000E after the Ethernet cable has been disconnected and reconnected.
- Office-LinX can successfully recover after a reboot.

2.1. Interoperability Compliance Testing

The interoperability compliance test included features and serviceability that operate via SIP connectivity. The focus of the compliance testing was primarily on verifying the interoperability between Office-LinX v8.2 and the CS1000E so that the following features operate:

- CS1000E end points can access the Office-LinX pilot number.
- Office-LinX can access the CS1000E end points.
- Office-LinX provides messaging services to the CS1000E end points.
- Office-LinX can conduct transfer operations for the end points.

2.2. Test Results

All the relevant tests were verified and found to be passed.

2.3. Support

Technical support for the Office-LinX solution can be obtained by contacting Esna:

- URL – techsupp@esna.com
- Phone – (905) 707-1234

3. Reference Configuration

Figure 1 illustrates the configuration used in these Application Notes. The solution configuration shows a setup of a CS1000E communicating to the Office-LinX via Session Manager using SIP trunks. The CS1000E has UNISTim IP and SIP telephones connected as endpoints.

For interoperability, Office-LinX requires the use of the G.711MU codec, and transmission of DTMF tones using RFC2833.

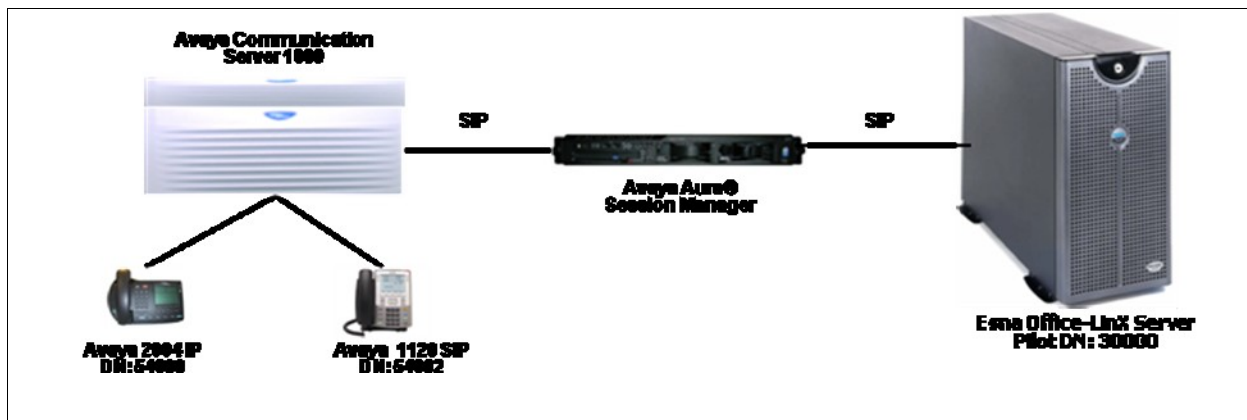


Figure 1: Solution Configuration

4. Equipment and Software Validated

The following equipment and software/firmware were used for the solution configuration provided:

Equipment	Software/Firmware
Avaya Communication Server 1000E	SW Version : 7.50 Q
Avaya Telephones: i2004 (UNISTim IP) 1120E (SIP)	0602B76 02.02.21.00
Avaya Aura® Session Manager	SW Version : 6.1
Avaya Aura® System Manager	SW Version : 6.1
Office-LinX Application Server	Windows XP Professional SP3
Office-LinX	SW Version 8.2.11.1714

5. Configure Avaya Communication Server 1000E R7.5

This section describes the procedure for setting up CS1000E. The steps include setting up

- Node properties.
- Route, Route List Block (RLB) and Distant Steering Code (DSC).
- Endpoints/Telephones.

The values used in this guide may be unique to the example shown. User will have to use values unique to their site, where this solution is being deployed e.g. site's IP address, extension numbers, etc.

CS1000E configurations are performed through Unified Communications Manager (UCM), Element Manager (EM) and Command Line Interface (CLI) via a telnet session to the Call Server.

5.1. Node IP (SIP Gateway) Configuration

This section only describes the configuration of the SIP Gateway application running on the CS1000E signaling server. In the solution test, Node ID **511** is configured, that has the SIP Gateway application enabled on it. For additional information on Nodes configuration refer to **Section 10**

To configure the SIP Gateway from EM, navigate to **System → IP Network → Nodes: Servers, Media Cards** and click on the **Node ID 511** as shown in **Figure 2** below.

The screenshot shows the Avaya CS1000 Element Manager interface. The left sidebar contains a navigation tree with the following items: - UCM Network Services, - Home, - Links, - Virtual Terminals, - System (circled in red), + Alarms, - Maintenance, + Core Equipment, - Peripheral Equipment, - IP Network, - Nodes: Servers, Media Cards (circled in red), - Maintenance and Reports, - Media Gateways, - Zones, - Host and Route Tables. The main content area is titled 'IP Telephony Nodes' and shows a table of nodes. The table has columns for Node ID, Components, Enabled Applications, and ELAN IP. The table contains two rows: Node ID 511 with 1 component (LTPS, Gateway (SIPGw, H323Gw)) and Node ID 512 with 1 component (SIP Line). The 'Node ID 511' cell is circled in red. Below the table, there are checkboxes for 'Show: Nodes' (checked), 'Component servers and cards' (unchecked), and 'IPv6 address' (checked).

<input type="checkbox"/> Node ID ^	Components	Enabled Applications	ELAN IP
<input type="checkbox"/> 511	1	LTPS, Gateway (SIPGw, H323Gw)	-
<input type="checkbox"/> 512	1	SIP Line	-

Figure 2: Accessing IP Telephony Nodes

Click on the link **Gateway (SIPGw & H323Gw)** link as shown in **Figure 3** below.

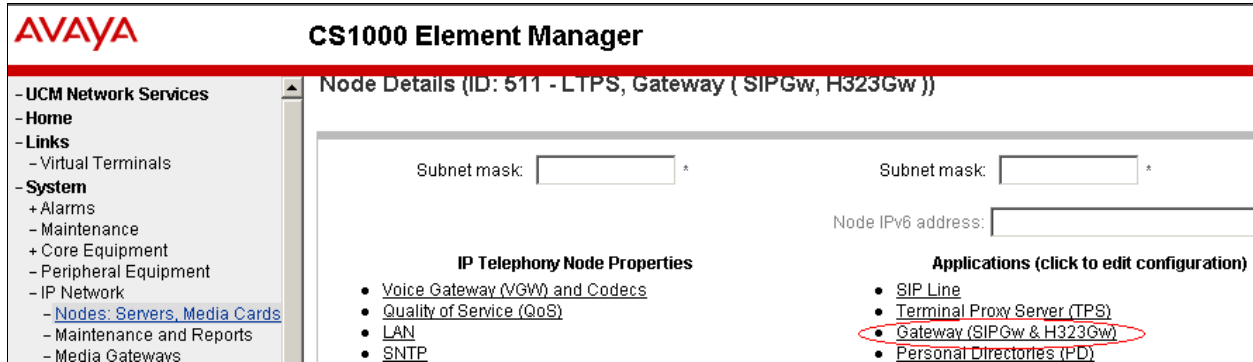


Figure 3: Accessing SIP and H323 Gateway

In the General section enter the **SIP domain name** as **bwvdev.com** as configured in **Figure 19** below, **Local SIP port** as **5060**, **Gateway endpoint name** as **cppm1** as configured in **Figure 23a** below and **Application node ID** as **511** as configured in **Figure 2** above. Refer to **Figure 4** below.

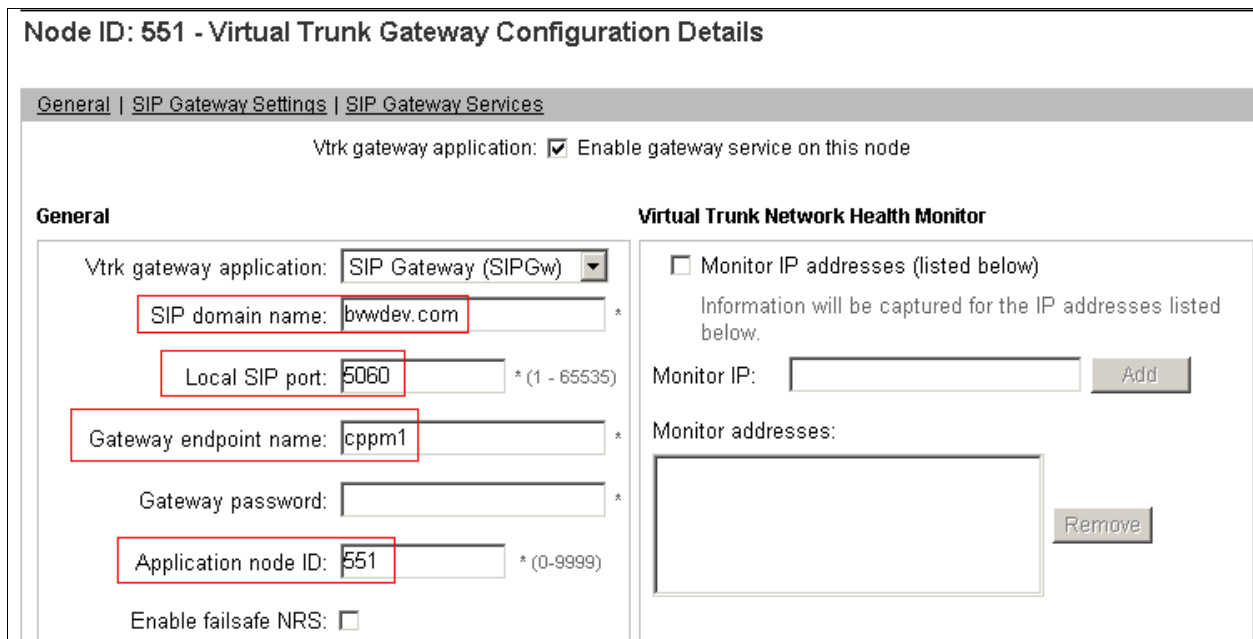


Figure 4: Configuration of General Fields

Figure 5 shows the **Primary TLAN IP address**, which is the IP address of the Session Manager. Rest of the fields is left at default.

The screenshot displays the configuration page for Node ID: 551 - Virtual Trunk Gateway Configuration Details. The page has a breadcrumb trail: General | SIP Gateway Settings | SIP Gateway Services. Under the heading "Proxy Server Route 1:", there is a form with the following fields:

- Primary TLAN IP address: 110.10.10.108 (highlighted with a red box)
- Port: 5060 (range 1 - 65535)
- Transport protocol: UDP (dropdown menu)
- Options: Support registration, Primary CDS proxy

A tooltip below the IP address field reads: "The IP address can have either IPv4 or IPv6 format based on the value of 'TLAN address type'".

Figure 5: Configuring the Session Manager IP Address

Figure 6 shows the **SIP URI Map** configuration where the **UDP** field is configured as **udp**. The rest of the fields are left as default.

The screenshot displays the configuration page for Node ID: 551 - Virtual Trunk Gateway Configuration Details. The page has a breadcrumb trail: General | SIP Gateway Settings | SIP Gateway Services. Under the heading "SIP URI Map:", there are two columns of fields:

- Public E.164 domain names:** National, Subscriber, Special number, Unknown.
- Private domain names (highlighted with a red box):** UDP: udp, CDP, Special number: PrivateSpecial, Vacant number: PrivateUnknown, Unknown.

Figure 6: Configuring SIP URI Map Fields

Figure 7 shows the **Microsoft Unified Messaging** configuration where the **MWI application DN** is configured as **30000**. This is the pilot DN being used to reach the Office-LinX during the solution testing. **CDP** is the selected **MWI dialing plan**.

Node ID: 551 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Microsoft Unified Messaging:

MWI application DN: 30000

MWI dialing plan: CDP

Options: Enable softkeys
 Enable secure media

Figure 7: Configuring Microsoft Unified Messaging Fields

Figure 8 shows the **Subscriber Access Service** number and **Auto Attendant Service** number configured. The values **30000** and **30005** are example values used during solution testing.

Node ID: 551 - Virtual Trunk Gateway Configuration Details

General | SIP Gateway Settings | SIP Gateway Services

Enable secure media

Subscriber Access Service

Add Remove

Access Number	Access Number Use	Insert Number
30000	Access Number is DN	

Auto Attendant Service

Add Remove

Auto Number	Auto Number Use	Insert Number
30005	Auto Number is DN	

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

Figure 8: Configuring Subscriber Access Service and Auto Attendant Service Fields

Note: Configurations explained in **Figures 7** and **8** are important. If these fields are not configured, the Office-LinX receives the SIP Message Header with Content-Type: multipart/mixed which Office-LinX does not currently support. This will therefore cause the solution to fail to accept calls with multipart/mixed message bodies. Office-LinX requires Content-Type: application/sdp.

5.2. Route, RLB and DSC Configuration

This section explains the steps to configure a routing entry that will access the Office-LinX server from the CS1000E using the RLB and DSC values.

After logging into the UCM, click on the EM link of the respective CS1000E (Not Shown). In the EM navigate to **Routes and Trunks > Routes and Trunks**. Click on **Add route** as shown in **Figure 9**.

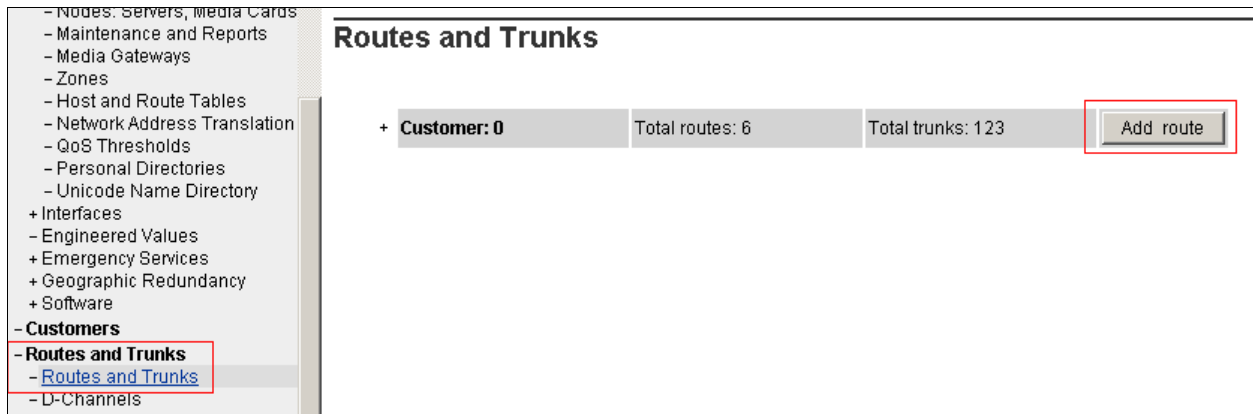


Figure 9: Adding Route

Figure 10 shows the configuration of the route being added. The values that are circled in red are to be configured by the user. The values shown are examples used during the solution testing.

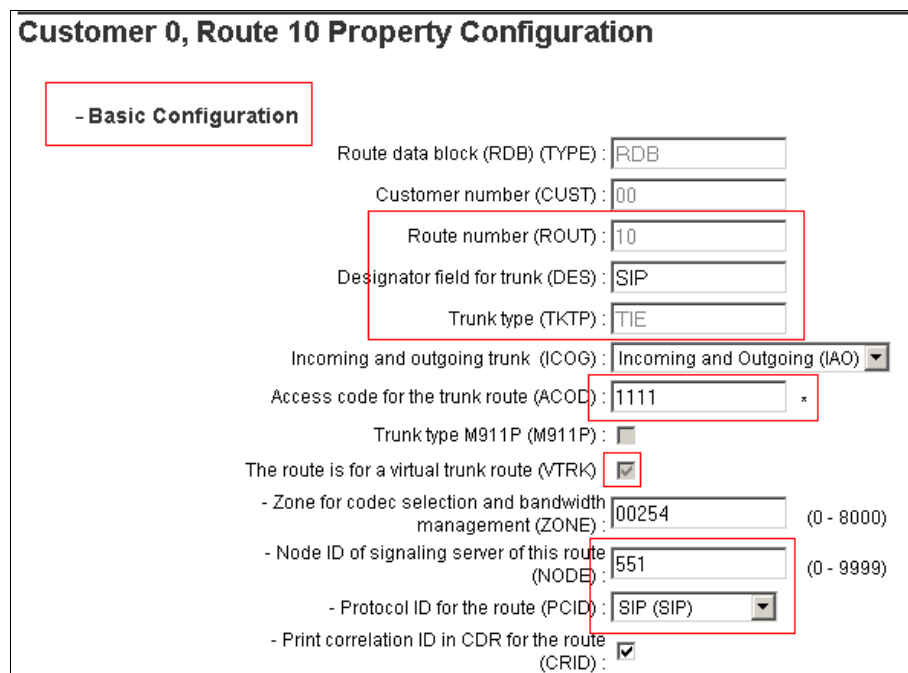


Figure 10: Route Configuration

To configure the RLB using EM navigate to **Dialing and Numbering Plans** → **Electronic Switched Network** → **Network Control & Services** → **Route List Block (RLB)** as shown in **Figure 11**.

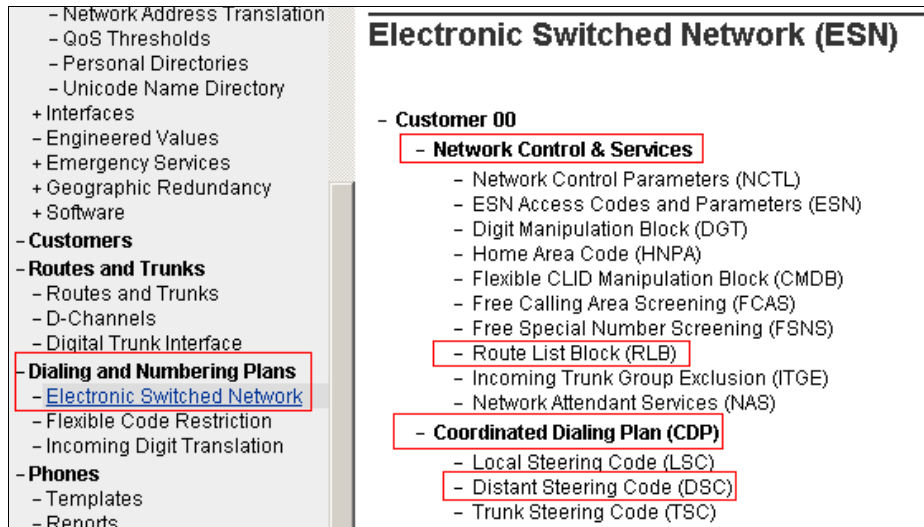


Figure 11: Accessing RLB

Enter the value of the route list index and click on **to Add** button to continue the configuration as shown in **Figure 12**. During the solution testing the value of **10** was added.

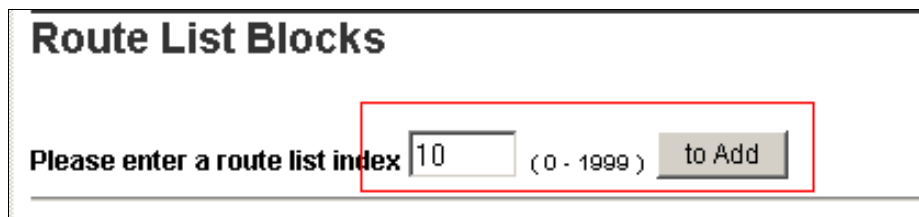


Figure 12: Adding RLB

Figure 13 shows the **Route Number 10** being selected to the RLB created. Route **10** is selected since it was the route number assigned while adding a route as shown in **Figure 10** above.

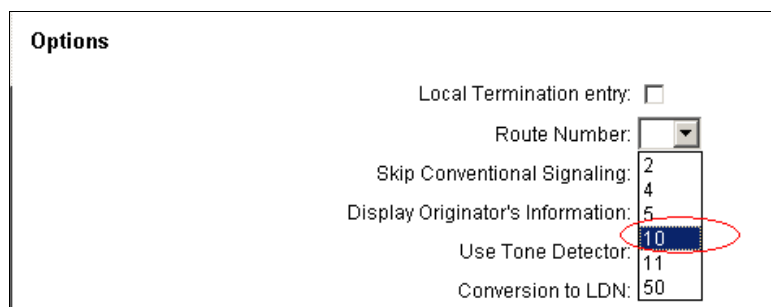


Figure 13: Selecting the configured Route to RLB

To configure the DSC using EM navigate to **Dialing and Numbering Plans > Electronic Switched Network > Coordinated Dialing Plan (CDP) > Distant Steering Code (DSC)** as shown in **Figure 11** above.

In the Distant Steering Code List page, select **Add** from the drop down list as shown in **Figure 14**.



Figure 14: Adding a new DSC

Enter the value of the DSC and click on the **to Add** button (Not Shown). As shown in **Figure 15** below DSC value of **3** was added during the solution testing. The value **3** was configured since the pilot DN of the Office-LinX system was **30000**. **Flexible Length number of digits** identifies length of the directory number (DN). During solution testing value of **5** was configured. **Route List to be accessed for trunk steering code** is selected as **10** from the drop down list. This value is selected based on the RLB created as shown in **Figure 12** above.

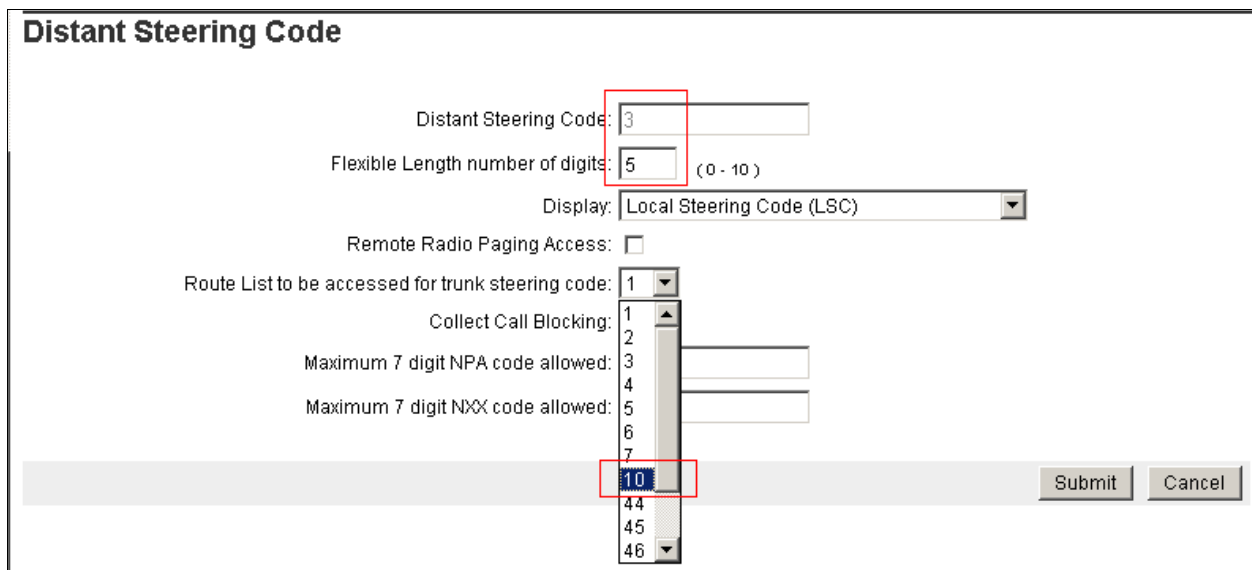


Figure 15: DSC configuration

For additional information on Route, RLB and DSC configuration, refer to **Section 10** of these Application Notes.

5.3. Endpoint/Telephone Configuration

This section explains the provisioning of an endpoint/telephone that was configured for the solution testing. Endpoint/Telephone can be configured using the CLI of the CS1000E from overlay LD 11/20. Refer to **Section 10** for further information regarding add/configuration of endpoints/telephones.

In **Figure 16**, values that are shown in red are to be configured by the user. The **FDN** and **HUNT** value of **30000** was used during the solution testing as the pilot DN of the Office-LinX system.

```
FDN 30000
TGAR 1
LDN NO
NCOS 1
SGRP 0
RNPG 3
SCI 0
SSU
LNRS 16
XLST
SCPW 1234
SFLT NO
CAC_MFC 0
CLS UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
POD SLKD CCSD SWD LNA CNDA
CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
ICDA CDMA LLCN MCTD CLBD AUTU
GPUD DPUD DND A CFXA ARHD CLTD ASCD
CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
UDI RCC HBTD AHD IPND DDGA NAMA MIND PRSD NRWD NRCN NROD
DRDD EXRO
USMD USRD ULAD CCBP RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DMO3 MCBM
FDSO NOVU VOLA VOUD CDMR PRED RECA MCDD T87A SBMD
KEM3 MSNV FRA PKCH MUTA MWTD DVLD CROD ELCD
CPND_LANG ENG
RCO 0
HUNT 30000
```

Figure 16: Configuring an Endpoint/ Telephone

6. Configure Routing using Avaya Aura® System Manager

This section provides the procedures for configuring routing using Avaya Aura ® System Manager. The procedures include the following areas:

- Logging into the Avaya Aura® System Manager.
- Adding Domain.
- Adding Location.
- Adding SIP entities.
- Adding Entity links.
- Adding Routing Policies.
- Adding Dial Patterns.

6.1. Logging into the Avaya Aura® System Manager

This section explains the steps to launch the login screen of the System Manager and accessing the Network Routing Policy.

To launch the System Manager Login screen, open a web browser session and type the IP address of the System Manager in the URL (not shown). **Figure 17** below shows the Log on Screen. Type the required **User ID** and **Password** credentials and click on **Log On** to continue.

AVAYA Avaya Aura® System Manager 6.1

Home / Log On

Log On

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 Clear

Figure 17: Avaya Aura® System Manager Login Screen

From the main screen of System Manager access the **Network Routing Policy** by selecting **Routing** as shown in **Figure 18** below.

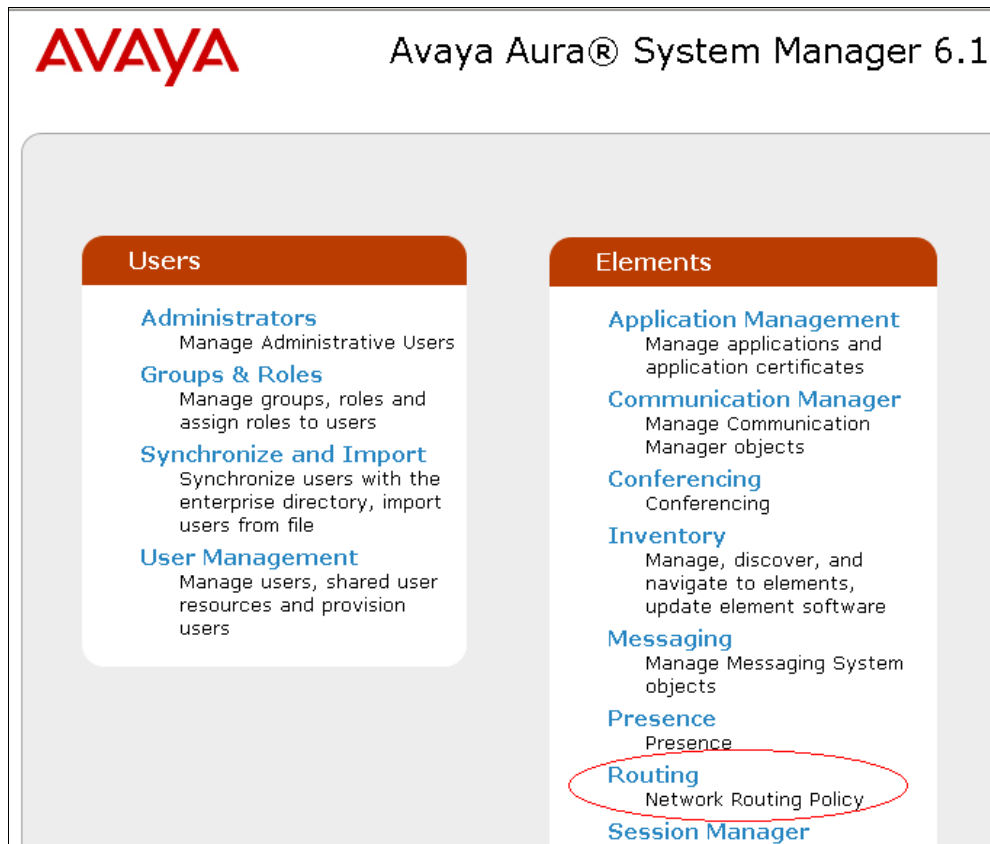


Figure 18: Avaya Aura® System Manager Main Screen

6.2. Adding Domain

To add a domain, select **Domains** from the left hand window of the Routing screen and click on **New** (not shown). Configure the **Name** as shown in **Figure 19** below and click on **Commit** to complete adding a domain. During compliance testing a domain name of **bwvdev.com** was used. Additional domains can be added in a similar fashion.

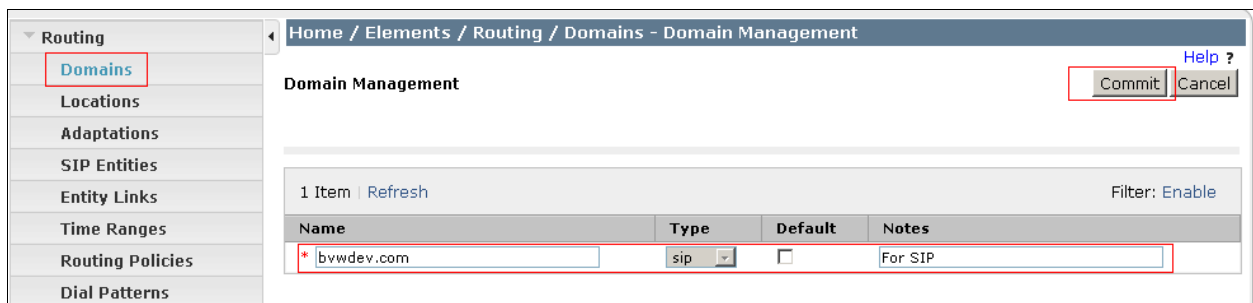


Figure 19: Domain Management

6.3. Adding Location

To add a location, select **Locations** from the left hand window of the Routing screen and click on **New** (not shown). Configure the **Name** as shown in **Figure 20** below and click on **Commit** to add a Domain. During compliance testing a location name of **Belleville, Ont, Ca** was used. Click on **Commit** to complete adding a location. Additional locations can be added in a similar fashion.

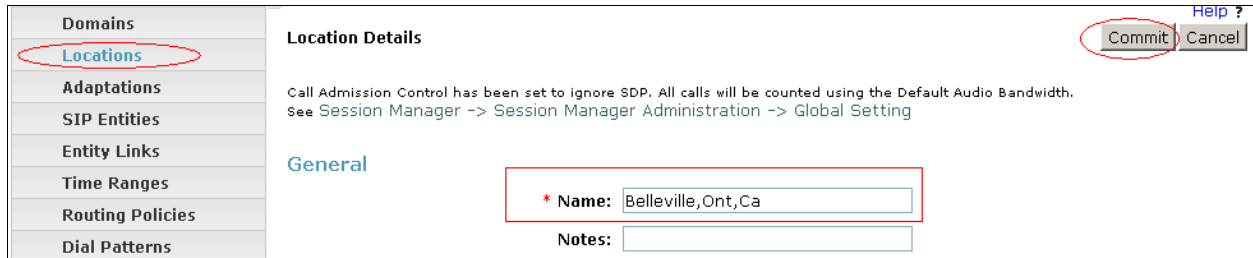


Figure 20: Location Details

6.4. Adding SIP Entities

This section explains the adding of SIP entities for the Session Manager, Office-LinX and the CS1000E system routing. To add SIP Entities, select **SIP Entities** from the left hand window of the Routing screen and click on **New** (not shown).

Figures 21a and **21b** show the SIP Entity Details for the Session Manager routing. The **FQDN or IP Address** of **110.10.10.108** is the IP address of the Session Manager. Click on **Commit** to complete adding the SIP Entity.

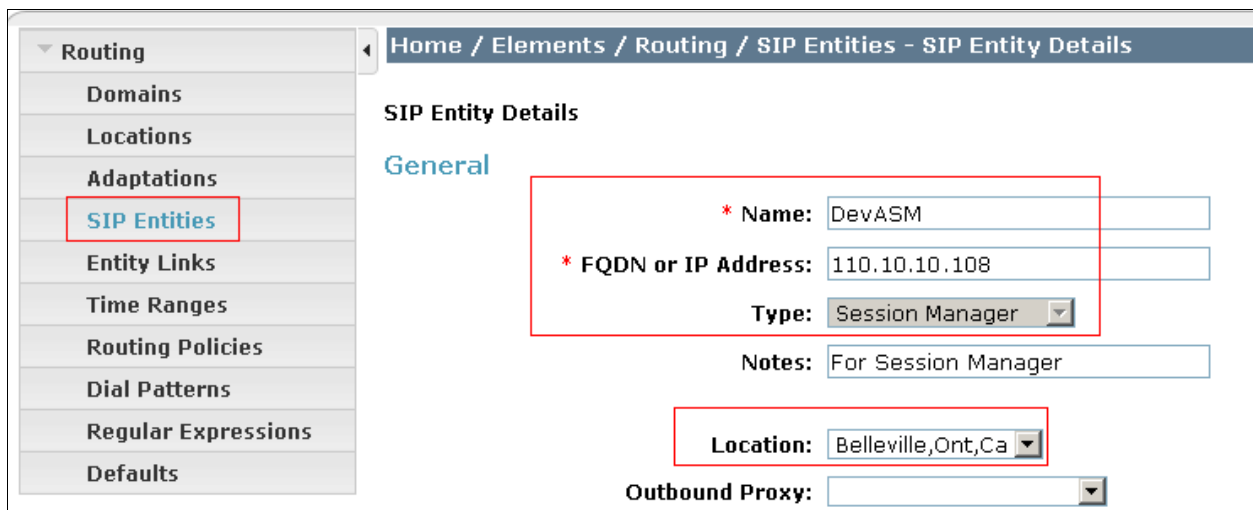


Figure 21a: SIP Entity Details for Session Manager

<input type="checkbox"/>	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
<input type="checkbox"/>	DevASM	UDP	* 5060	DevCM	* 5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	DevASM	TCP	* 5060	ESNA	* 5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	DevASM	UDP	* 5060	ESNA	* 5060	<input checked="" type="checkbox"/>

Select : All, None < Previous | Page 5 of 7 | Next >

Port

3 Items | Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	5060	TCP	bvwdev.com	
<input type="checkbox"/>	5060	UDP	bvwdev.com	

Figure 21b: SIP Entity Details for Session Manager (cont'd)

Figures 22a and 22b show the SIP Entity Details for Office-LinX routing. The **FQDN or IP Address** of **110.10.10.70** is the IP address of Office-LinX. Click on **Commit** to complete adding the SIP Entity.

Home / Elements / Routing / SIP Entities - SIP Entity Details

SIP Entity Details

General

*** Name:**

*** FQDN or IP Address:**

Type:

Notes:

Adaptation:

Location:

Time Zone:

Figure 22a: SIP Entity Details for Office-LinX System

<input type="checkbox"/>	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
<input type="checkbox"/>	DevASM	TCP	* 5060	ESNA	* 5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	DevASM	UDP	* 5060	ESNA	* 5060	<input checked="" type="checkbox"/>

Select : All, None

* Input Required

Figure 22b: SIP Entity Details for Office-LinX System (cont'd)

Figures 23a and 23b show the SIP Entity Details for the CS1000E System routing. The **FQDN or IP Address** of **110.10.10.130** is the Node IP address of the SIP Signaling Gateway of the CS1000E System. Click on **Commit** to complete adding the SIP Entity.

SIP Entity Details

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

Figure 23a: SIP Entity Details for CS1000E System

Entity Links

2 Items | Refresh Filter: Enable

<input type="checkbox"/>	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
<input type="checkbox"/>	DevASM	TCP	* 5060	cppm1	* 5060	<input checked="" type="checkbox"/>
<input type="checkbox"/>	DevASM	UDP	* 5060	cppm1	* 5060	<input checked="" type="checkbox"/>

Select : All, None

* Input Required

Figure 23b: SIP Entity Details for CS1000E System (cont'd)

6.5. Adding Entity Links

This section explains the adding of Entity links between the Session Manager and the CS1000E. To add an entity link, select **Entity Links** from the left hand window of the Routing screen and click on **New** (not shown).

Figure 24 below shows the Entity Link added between the Session Manager and CS1000E.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port
* DevASM_cppm1_5C	* DevASM	TCP	* 5060	* cppm1	* 5060

Figure 24: Adding Entity Link

Similarly an Entity Link is added between the Session Manager and the Office-LinX server.

6.6. Adding Routing Policies

This section explains the Routing Policy configuration for Office-LinX and CS1000E Systems. To add a routing policy, select **Routing Policies** from the left hand window of the Routing screen and click on **New** (not shown).

Figures 25a and **25b** show the Routing Policy Details for Office-LinX. Select the Office-LinX System as the SIP Entity Destination and add the dial pattern associated with it. A dial pattern can be added once it has been configured as explained in **Section 6.7** below. Click on **Commit** to complete adding a routing policy.

General

* Name: ESNA_Routing

Disabled:

Notes: Routing to the Office Linx Server

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
ESNA	110.10.10.70	Other	For Office Linx Testing

Figure 25a: Routing Policy Details for Office-LinX System

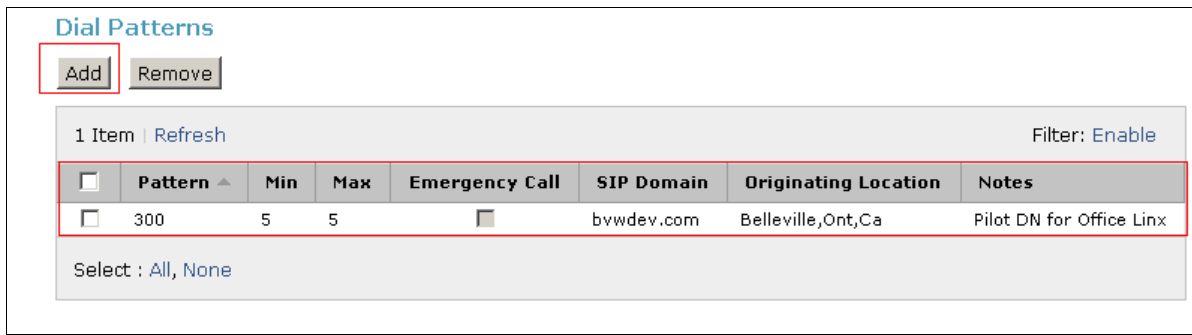


Figure 25b: Routing Policy Details for Office-LinX System (cont'd)

Above steps can be repeated with the required fields to configure routing policy for CS1000E. During compliance testing dial pattern of **58** was configured for CS1000E since extension series on CS1000E are in 58xxx range.

6.7. Adding Dial Patterns

This section explains the steps to add a dial pattern for Office-LinX and CS1000E systems. To add a dial pattern, select **Dial Patterns** from the left hand window of the Routing screen and click on **New** (not shown).

Figure 26 shows the Dial Pattern Details for Office-LinX. During compliance testing extensions range on Office-LinX starts with 3xxxx and therefore **300** are used in the **Pattern** field. The minimum and maximum size of the extension is defined as **5**. Add the **ESNA_Routing** policy as configured in **Section 6.6** above. Click on **Commit** to complete adding the dial pattern. Additional dial patterns can be configured as required in a similar fashion.

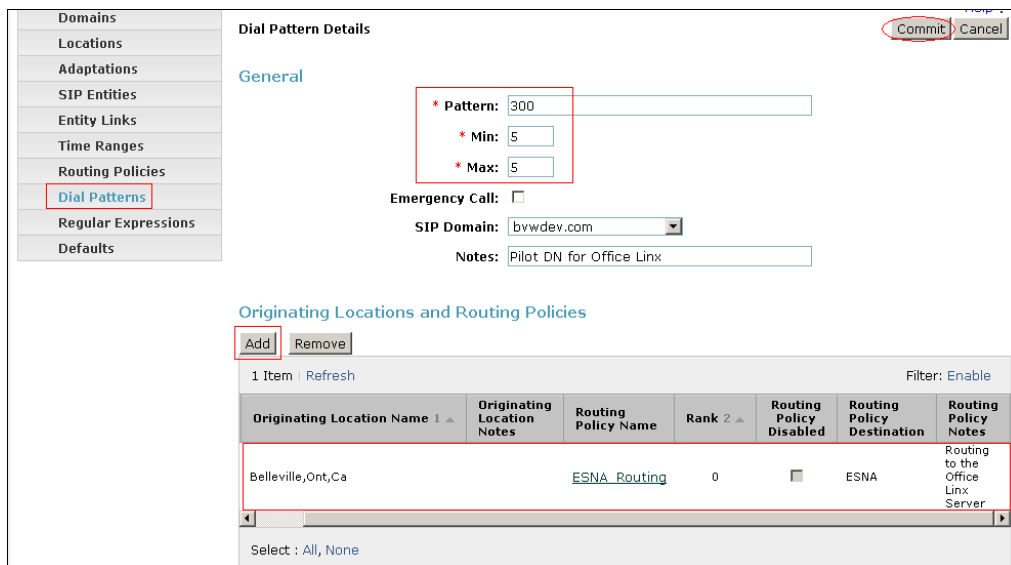


Figure 26: Dial Pattern Details

7. Configuring the Esna Office-LinX server

Office-LinX installation is covered in referenced product documentation, during the install the PBX template called **Nortel** is selected providing a pre-defined configuration. This section only describes the interface configuration as a reference for verification, so that the Office-LinX can communicate to the CS1000E. For further details on the Office-LinX configuration steps not covered in this document, refer to **Section 10**

The configuration of Office-LinX to integrate successfully with CS1000E is done from the Office-LinX's **SIP Configuration Tool**. Access the **SIP Configuration Tool** from the Office-LinX server by navigating to **Start → All Programs → Office-LinX → SIP Configurator** (Not shown).

Figure 27 shows the **SIP Configuration** tool where a New PBX is added. During the solution testing the template **Nortel** was selected as the PBX.

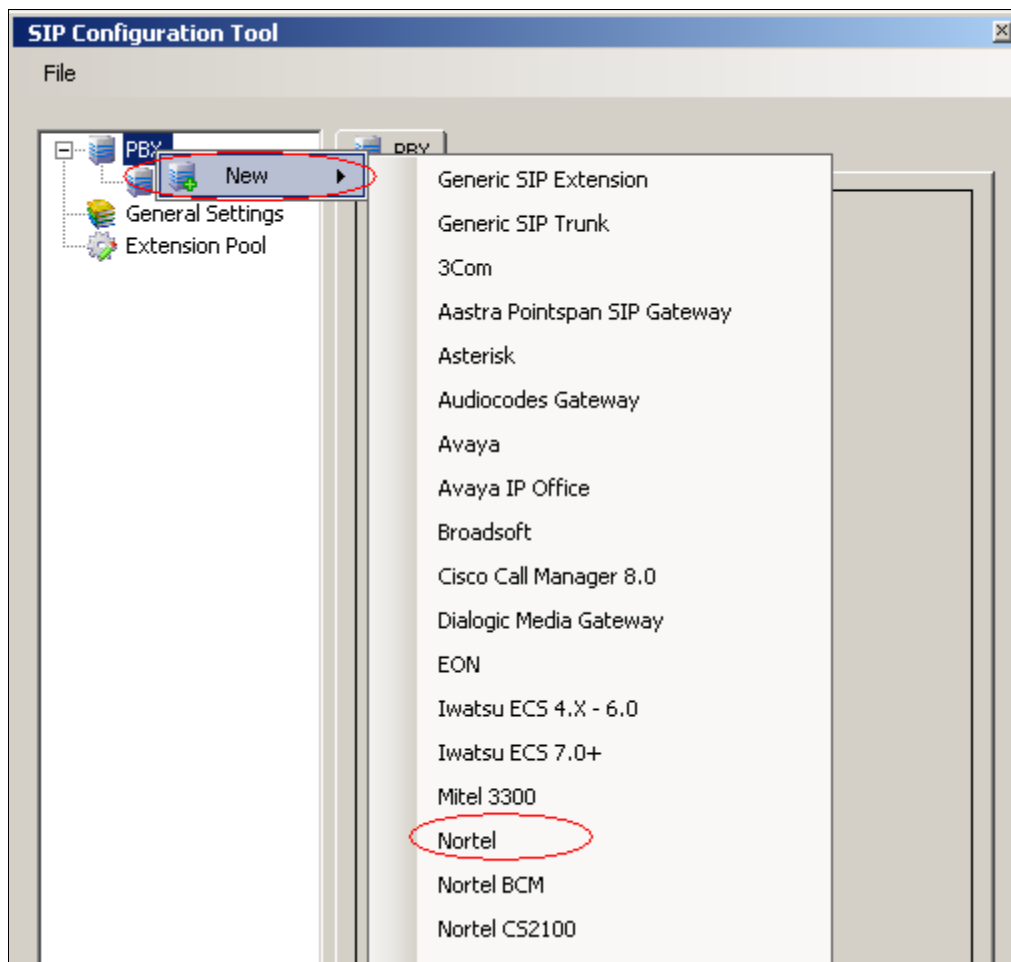


Figure 27: Adding new PBX

Figure 28 shows the **General** tab of the **SIP Configuration Tool**. Fields circled in red are to be populated by user. The **IP Address** field is populated with the IP address of the Session Manager.

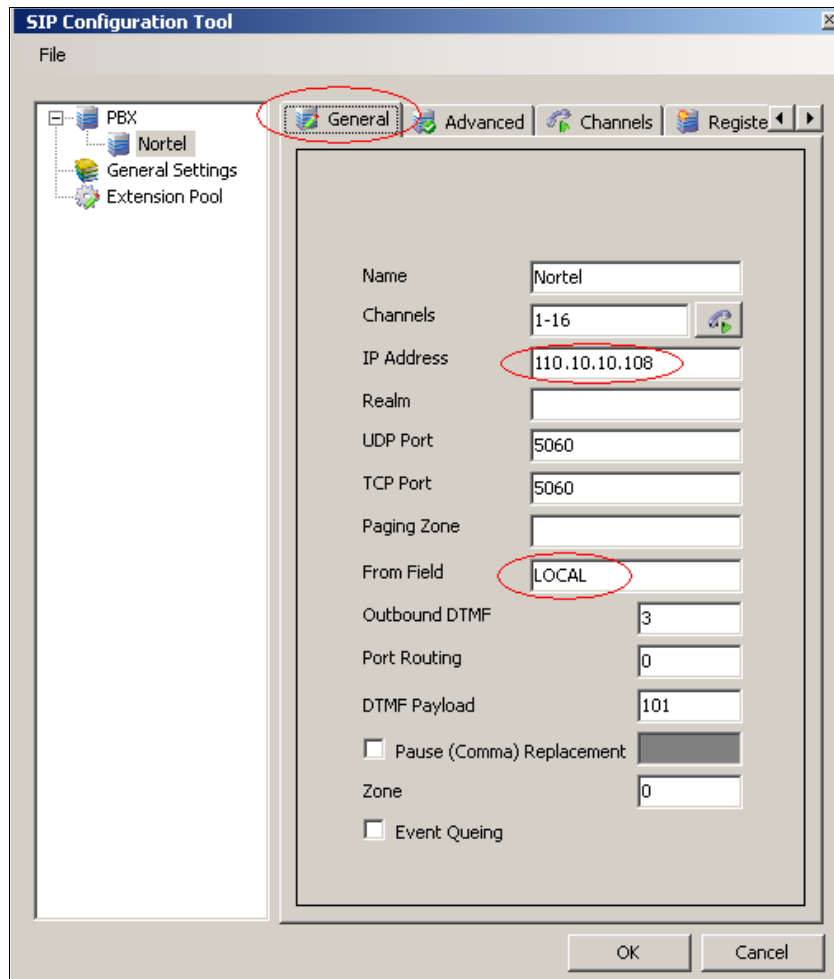


Figure 28: General Configuration

Figure 29 shows the **Advanced** tab configuration of the **SIP Configuration Tool**. Check the boxes shown in the **Figure 29** below.

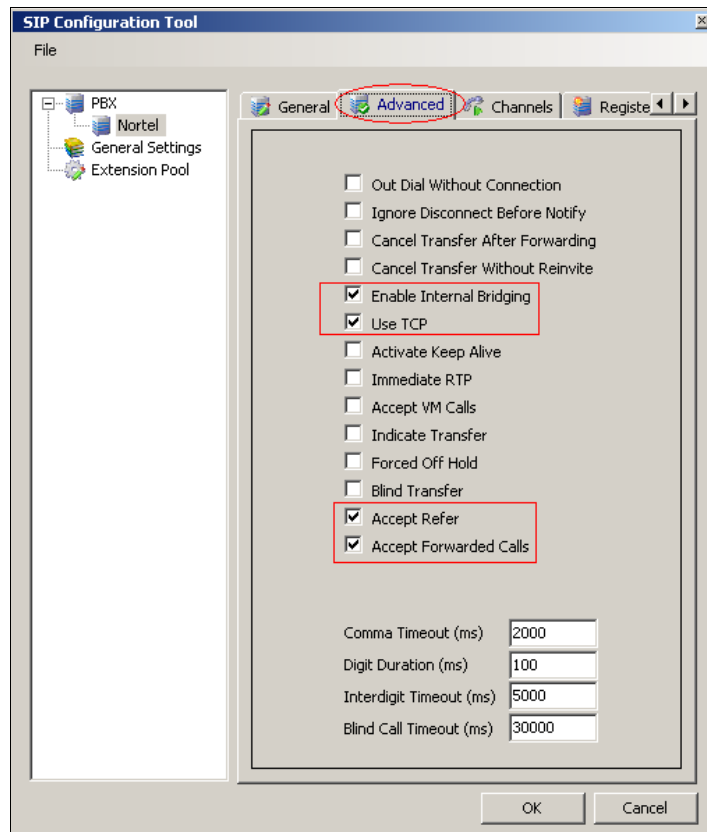


Figure 29: Advanced Configuration

Figure 30 shows the pilot DN of the Office-LinX being configured in the **Channels** tab of the **SIP Configuration Tool**. During solution testing, **30000** were configured as the Office-LinX's pilot DN.

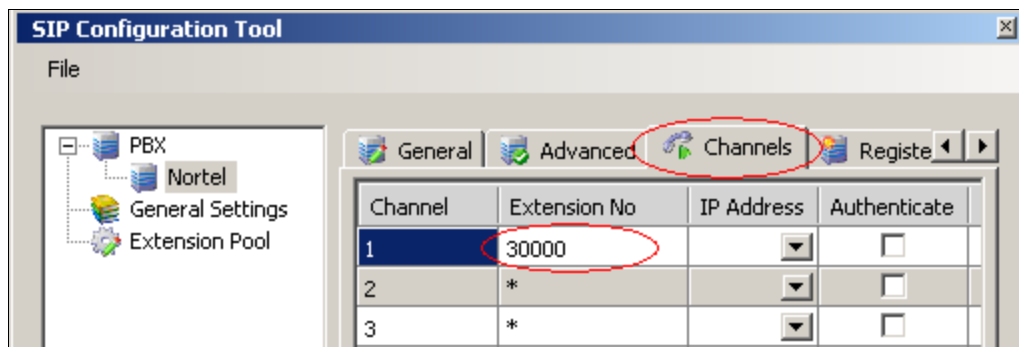


Figure 30: Channels Configuration

Figure 31 shows the **Force MWI** box checked under the **MWI** tab of the **SIP Configuration Tool**. The rest of the values are left at default.

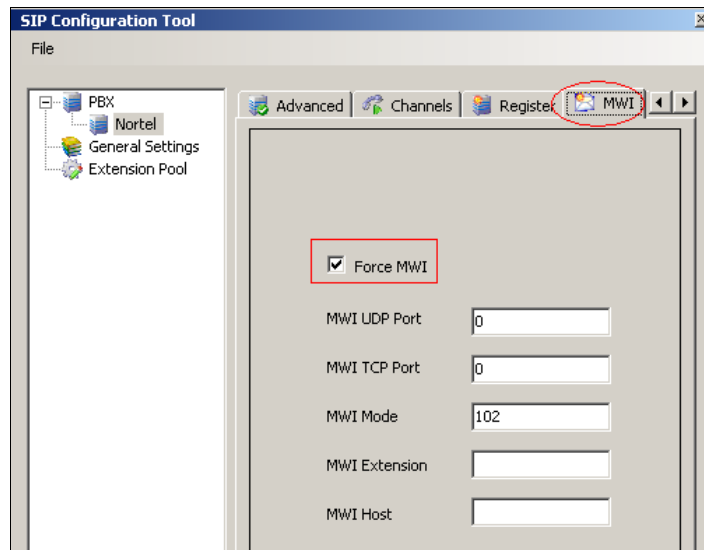


Figure 31: MWI Configuration

The integration mode number for Office-LinX to communicate with CS1000E is **4**. This **Integration Mode** field can be configured from the **ANI** tab of the **SIP Configuration Tool** as shown in **Figure 32** below. Click on **OK** to complete the configuration.

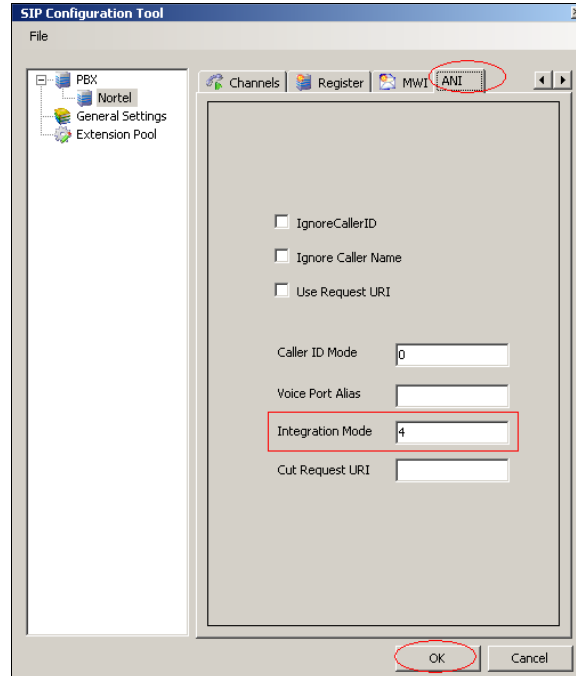


Figure 32: ANI Configuration

Figure 33 shows the **General Settings** tab where the values are left at default.

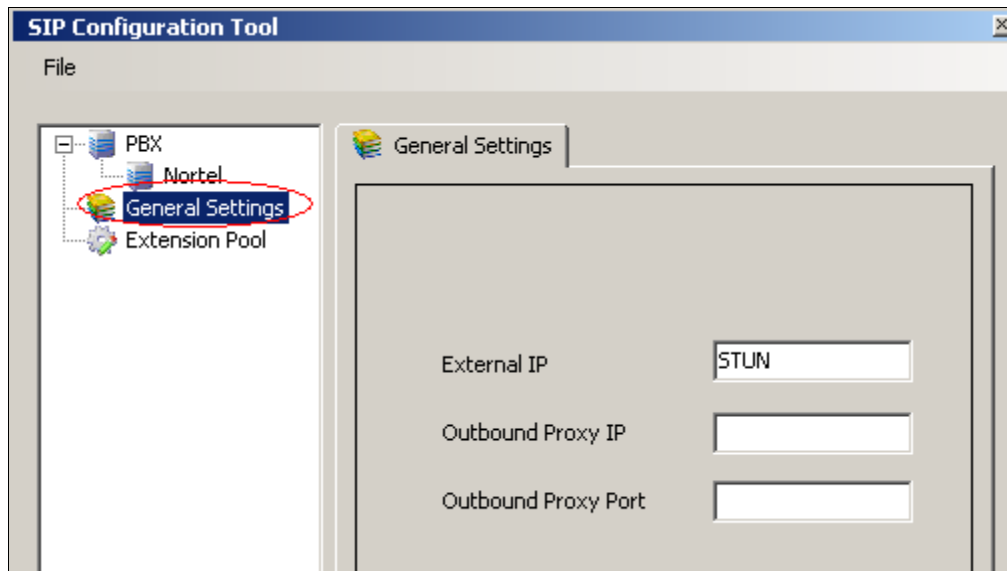


Figure 33: Outbound Proxy IP configuration

The Office-LinX needs to be configured for SIP to send Message Waiting Light indication to the PBX. To configure this feature access the **Office Linx Admin** screen by navigating **Start → All Programs → Office-LinX → Office Linx Admin** (Not Shown). Access the **Properties** of the PBX as shown in **Figure 34**.

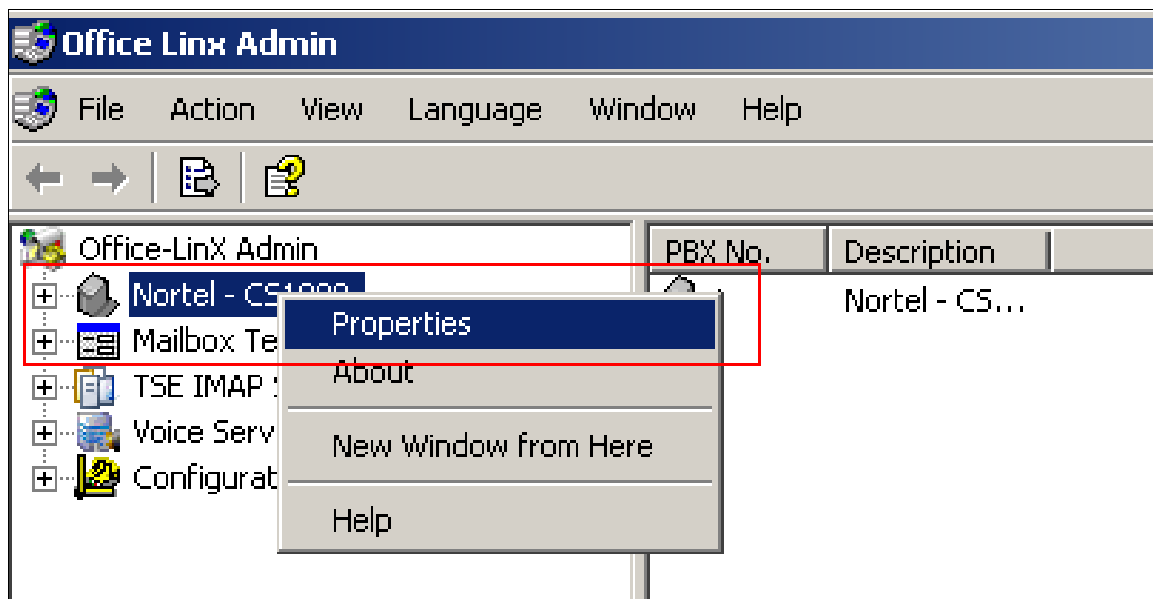


Figure 34: Accessing PBX Properties Screen

Select the **SIP** radio button from the **Message Light** tab as shown in **Figure 35** below.

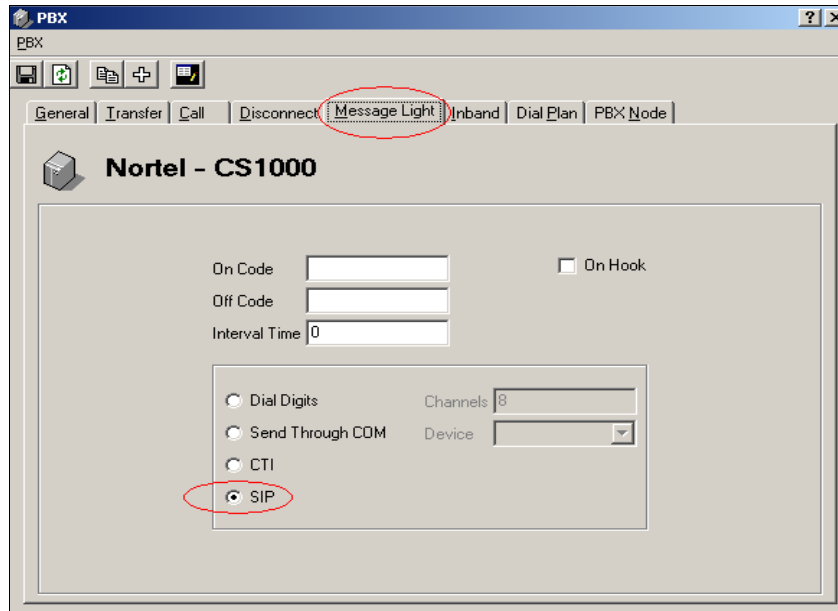


Figure 35: Message Light Configuration

Due to the large packet size of SIP messages from the Session Manager, additional settings need to be modified from default values of the Office-LinX server. Under the **C:\Windows\ETSIPService.ini** file, add the following line under the general section header:

```
[General settings]
Buffer Size = 4096
```

This modifies the default TCP packet size from 2kb to 4kb.

8. Verification Steps

The following steps may be used to verify the integration:

- From the CS1000E end point call the Office-LinX pilot DN 40000 and verify if general greeting is played.
- From Office-LinX server verify if a CS1000E endpoint receives a wakeup call.
- Verify if a call from a CS1000E endpoint to another CS1000E endpoint can be transferred via Office-LinX server.
- Verify if correct Office-LinX greeting messages are played depending on the status of the CS1000E endpoints.
- Verify if a message/s can be left for a CS1000E endpoint and retrieved via Office-LinX server.

9. Conclusion

All of the executed test cases have passed and met the objectives outlined in **Section 2**. The Esna Office-LinX v8.2 software is considered compliant with Avaya Communication Server 1000E Release 7.5 and Avaya Aura® Session Manager Release 6.1.

10. Additional References

Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Software Input Output Reference – Administration – Avaya Communication Server 1000, R7.5 NN43001-611, 05.09 Sept 2011*

Product documentation for Esna Office-LinX may be found at:

http://www.esnatech.com/support/tech_index.htm

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