

Avaya Solution & Interoperability Test Lab

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 <u>techsupp@esna.com</u>
- 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)	0602B76
1120E (SIP)	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 \rightarrow IP Network \rightarrow Nodes: Servers, Media Cards and click on the Node ID 511 as shown in Figure 2 below.

Αναγα	CS1000 Element Manager			
- UCM Network Services	Managing: System »	Username: ad IP Network » IP Tele	dmin phony Nodes	
-Links	IP Telephony	Nodes		
– Virtual Terminals	Click the Node ID to) view or edit its pi	roperties.	
System				
+ Alarms - Maintenance + Core Equinment	Add Import Export Delete			
- Peripheral Equipment	□ Node ID ▲	Components	Enabled Applications	ELAN IP
- Network - Nodes: Servers, Media Cards	511	1	LTPS, Gateway (SIPGw, H323Gw)	-
 Maintenance and Reports Media Gateways 	<u>□ 512</u>	1	SIP Line	-
- Zones - Host and Route Tables	Show: 🔽 Nodes	Compone	nt servers and cards	✓ IPv6 address

Figure 2: Accessing IP Telephony Nodes

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

Αναγα	CS1000 Element Manager	
- UCM Network Services	Node Details (ID: 511 - LTPS, Gateway (SI	PGw, H323Gw))
- Links - Virtual Terminals - System + Alarms - Maintenance	Subnet mask: *	Subnet mask: * Node IPv6 address:
+ Core Equipment - Peripheral Equipment - IP Network - <u>Nodes: Servers, Media Cards</u> - Maintenance and Reports - Media Cateways	IP Telephony Node Properties Voice Gateway (VGW) and Codecs Quality of Service (QoS) LAN SNTP	Applications (click to edit configuration) <u>SIP Line</u> <u>Terminal Proxy Server (TPS)</u> <u>Gateway (SIPGw & H323Gw)</u> <u>Personal Directories (PD)</u>

Figure 3: Accessing SIP and H323 Gateway

In the General section enter the **SIP domain name** as **bvwdev.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.

Node ID: 551 - Virtual Trunk Gateway Configuration Details			
General SIP Gateway Settings SIP Gateway Services			
Vtrk gateway application: 🔽 Enab	Vtrk gateway application: 🔽 Enable gateway service on this node		
General	Virtual Trunk Network Health Monitor		
Vtrk gateway application: SIP Gateway (SIPGw) 💌	☐ Monitor IP addresses (listed below)		
SIP domain name: bwwdev.com *	Information will be captured for the IP addresses listed below.		
Local SIP port: 5060 * (1 - 65535)	Monitor IP: Add		
Gateway endpoint name: cppm1 *	Monitor addresses:		
Gateway password: *	Remove		
Application node ID: 551 * (0-9999)			
Enable failsafe NRS: 🗖			

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.

Node ID: 551 - Virtua	l Trunk Gateway Configuration Details	
<u>General</u> <u>SIP Gateway Set</u>	tings <u>SIP Gateway Services</u>	
Proxy Server Route	Primary TLAN IP address: 110.10.10.108 The IP address can have either IPv4 or IPv6 format based on the value of "TLAN address type" Port: 5060 Transport protocol: UDP I Options: Support registration Primary CDS proxy	

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.

ails
Private domain names
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		
Ì	<u>General SIP Gateway Settings S</u>	IP Gateway Services	
ĺ	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 I SIP Gateway Setting	gs I SIP Gateway Services			
	L Enable s	ecure media		
Subscriber Access Service				
				_
Add Remove				
C Access Number	Access Number Use	Insert Number		
30000	Access Number is DN			
			1	
Auto Attendant Service				
Add Remove				
Auto Number	Auto Number Use Ins	ert Number		
32005	Auto Number is DN			
				-
* Required Value	Note: Changes made on t	this page will NOT be	Save	Cancel
Neguireu value.	transmitted until the No	ode is also saved.	- Ouro	o anoor

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.**

- Nodes: Servers, Media Cards				
 Maintenance and Reports 	Routes and Trunks			
– Media Gateways	Routoo and Franko			
- Zones				
- Host and Route Tables				
- Network Address Translation	Customory 0	Total vautaa: C	Total trunks: 100	Relative uter
- Oog Thresholds	+ Customer: 0	Total routes, 6	Total trunks. 123	
- Boreonal Directoriae				
- Fersonal Directories				
- Officiale Name Directory				
+ Interfaces				
- Engineered values				
+ Emergency Services				
+ Geographic Redundancy				
+ Software				
- Customers				
- Routes and Trunks				
- Routes and Trunks				
- D-Channels				

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.

Customer 0, Route 10 Pro	Customer 0, Route 10 Property Configuration		
- Basic Configuration			
Ro	ute data block (RDB) (TYPE) : RDB		
	Customer number (CUST) : 00		
	Route number (ROUT) : 10		
De	signator field for trunk (DES) : SIP		
	Trunk type (TKTP) : TIE		
Incoming	and outgoing trunk (ICOG) : Incoming and Outgoing (IAO) 💌		
Access coo	le for the trunk route (ACOD) : 1111 *		
	Trunk type M911P (M911P) : 🔲		
The route is fo	r a virtual trunk route (VTRK)		
- Zone for co	dec selection and bandwidth 00254 (0 - 8000) management (ZONE) :		
- Node ID of	signaling server of this route 551 (0 - 9999) (NODE):		
- Pro	otocol ID for the route (PCID) : SIP (SIP)		
- Print corre	elation ID in CDR for the route (CRID) :		

Figure 10: Route Configuration

RS; Reviewed: SPOC 3/8/2012 Solution & Interoperability Test Lab Application Notes ©2012 Avaya Inc. All Rights Reserved. To configure the RLB using EM navigate to **Dialing and Numbering Plans** \rightarrow **Electronic** Switched Network \rightarrow Network Control & Services \rightarrow 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.

Route List Blocks	
Please enter a route list index	10 (0 - 1999) to Add

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

RS; Reviewed: SPOC 3/8/2012

Solution & Interoperability Test Lab Application Notes ©2012 Avaya Inc. All Rights Reserved. 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.

Distant Steering Code List		
Add Add Display		
Please enter a distant steering codet	o Add	

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** indentifies 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
ຣຣບ	
LNRS	16
XLST	
SCPW	1234
SFLT	NO
CAC_1	NFC O
CLS	UNR FBA WTA LPR PUA MTD FNA HTA TDD HFA CRPD
	NWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
	POD SLKD CCSD SWD LNA CNDA
	CFTD <mark>SFA</mark> MRD DDV CNID CDCA MSID DAPA BFED RCBD
	ICDA CDMA LLCN MCTD CLBD AUTU
	GPUD DPUD DNDA CFXA ARHD CLTD ASCD
	CPFA CPTA ABDA CFHD FICD NAID BUZZ AGRD MOAD
	UDI RCC HBTD AHD IPND DDGA NAMA MIND PRSD NRWD NRCD NROD
	DRDD EXRO
	USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
	FDSD NOVD VOLA VOUD CDMR PRED RECA MCDD T87A SBMD
	KEN3 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 solel authorized users for legitimati purposes only. The actual or unauthorized access, use, or of this system is strictly prohit Unauthorized users are subje company disciplinary procedu criminal and civil penalties un federal, or other applicable du foreign laws.	ly to e business attempted modification bited. User ID: ures and or der state, omestic and 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.

Αναγα	Avaya Aura® System Manager 6.:
Users	Elements Application Management
Manage Administrati Groups & Roles Manage groups, role assign roles to users Synchronize and Im Synchronize users w enterprise directory, users from file User Management Manage users, share resources and provis users	Application Management Manage applications and application certificates as and s Communication Manager Manage Communication Manager objects Manager objects Conferencing Inventory Manage, discover, and navigate to elements, sion Manage discover, and navigate to elements, update element software
	Manage Messaging System objects Presence Presence Routing Network Routing Policy Session Manager

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 **bvwdev.com** was used. Additional domains can be added in a similar fashion.

Routing Home / Elements / Routing / Domains - Domain Management						
Domains	Domain Management				Help ? Commit Cancel	
Locations						
Adaptations						
SIP Entities						
Entity Links	1 Item Refresh				Filter: Enable	
Time Ranges	Name	Туре	Default	Notes		
Routing Policies	* bvwdev.com	sip 💌		For SIP		
Dial Patterns						



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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.

Domains Locations	Location Details
Adaptations	Call Admission Control has been set to ignore SDP. All calls will be counted using the Default Audio Bandwidth.
SIP Entities	See Session Manager -> Session Manager Administration -> Global Setting
Entity Links	General
Time Ranges	
Routing Policies	
Dial Patterns	Notes:

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.

Routing	∢ Home / Ele	ments / Routing / SIP E	ntities - SIP Entity Details
Domains	SID Entity De	tailc	
Locations	SIF Linty De	cuits	
Adaptations	General		
SIP Entities		* Name:	DevASM
Entity Links		* FQDN or IP Address:	110.10.10.108
Time Ranges		Туре:	Session Manager 🔄
Routing Policies		Notes:	For Session Manager
Dial Patterns			
Regular Expressions		Location:	Belleville,Ont,Ca 💌
Defaults		Outbound Proxy:	

Figure 21a: SIP Entity Details for Session Manager

	SIP Entity 1	Protocol	Port	SIP Entity 2		Port	Trusted
	DevASM	UDP 🗸	* 5060	DevCM	•	* 5060	
	DevASM	ТСР 🗸	* 5060	ESNA	•	* 5060	
	DevASM	UDP 💌	* 5060	ESNA	•	* 5060	
Selec	t : All, None				< Pr	evious Page 5	of 7 Next >
Port							
Add	Remove						
3 Itei	ms Refresh						Filter: Enable
	Port 🔺	Protocol	Default Domain		Notes		
	5060	ТСР 💌	bvwdev.com	•			
	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 Det	ails						
General							
	* Name:	ESNA					
	* FQDN or IP Address:	110.10.10.70					
	Type:	Other 🔽					
	Notes:	For Office Linx Testing					
	Adaptation:						
	Location:	Belleville,Ont,Ca					
	Time Zone:	America/New_York					

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

	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Trusted
	DevASM 💽	TCP 💌	* 5060	ESNA 💌	* 5060	
	DevASM 💽	UDP 💌	* 5060	ESNA	* 5060	
Selec	t : All, None					
* Inpu	t Required				Q	Commit Cancel

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 De	etails	
General	r	
	* Name:	cppm1
	* FQDN or IP Address:	110.10.130
	Туре:	Other 🔽
	Notes:	Connectivity to CS1K 7.5 Enterpri
	Adaptation:	
	Location:	Belleville,Ont,Ca 💌
	Time Zone:	America/Toronto

Figure 23a: SIP Entity Details for CS1000E System

SIP Entity 2 Port Trusted cppm1 * 5060 Image: Component of the second o	_					Filter: Enable
cppm1 ▼ \$5060 ▼ cppm1 ▼ \$5060 ▼	1 1	SIP Entity 1 Protocol	Port	SIP Entity 2	Port	Trusted
cppm1 💌 * 5060	•	DevASM TCP 💌	* 5060	cppm1 🔹	* 5060	~
		DevASM VDP -	* 5060	cppm1 💽	* 5060	v
		Select : All, None				
		Select : All, None				

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).

Locations	Entity Links					Commit	Cancel
Adaptations							
SIP Entities							
Entity Links							
Time Ranges	1 Item Refresh					Filter	r: Enable
Routing Policies	Name	SIP Entity 1	Protocol	Port	SIP Entity 2		Port
Dial Patterns	* DevASM_cppm1_50	* DevASM 🗾	TCP 💌	* 5060	* cppm1	v	* 5060
Regular Expressions							F
Defaults							
	* Input Required					Commit	Cancel

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

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.

Domains	- Routing Bolicy D	otailc			0	Help ?
Locations	Routing Foney D	Kouling Policy Details				
Adaptations	General				7	
SIP Entities		* NI:	me: ESNA Pouting			
Entity Links						
Time Ranges		Disat	oled:			
Routing Policies		No	ites: Routing to the O	ffice Linx Server		
Dial Patterns						
Regular Expressions	SIP Entity as	Destination				
Defaults	Select					
	Name	FQDN or IP Addr	ess	Туре	Notes	
	ESNA	110.10.10.70		Other	For Office Linx Testing	

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

	Dial P Add	Patterns Remove						
	1 Itei	m∣Refresh						Filter: Enable
E I								
		Pattern 🔺	Min	Мах	Emergency Call	SIP Domain	Originating Location	Notes
		Pattern 🔺 300	Min 5	Max 5	Emergency Call	SIP Domain	Originating Location Belleville,Ont,Ca	Notes Pilot DN for Office Linx

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.

Domains	Dial Pattern Details						Comm	it Cancel
Locations	Dian ruttern Details						Comm	Cancer
Adaptations	General							
SIP Entities		Dattorn: 2	200					
Entity Links		Pattern. 3	,00					
Time Ranges		* Min: 5	5					
Routing Policies		* Max: 5	5					
Dial Patterns	Emerge	ency Call: 🛛						
Regular Expressions	SIP	Domain: t	ovwdev.c	om 💌				
Defaults		Notes: P	ilot DN f	or Office Linx	-			
	Originating Locations ar Add Remove 1 Item Refresh	nd Routing) Policie	25			Filter	: Enable
	Originating Location Name	1 🔺 Origin Locati Notes	ating ion	Routing Policy Name	Rank 2 🔺	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
	Belleville,Ont,Ca			ESNA Routing	0	Г	ESNA	Routing to the Office Linx Server
	Select : All, None							

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** \rightarrow **All Programs** \rightarrow **Office-LinX** \rightarrow **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.

File PBX General Settings General Settings Extension Pool Name Nortel Channels I-16 General PAddress II0.10.10.108 Realm UDP Port 5060 TCP Port 5060 Paging Zone From Field LOCAL Outbound DTMF 3 Port Routing 0 DTMF Payload I01 Pause (Comma) Replacement Zone 0 Event Queing	SIP Configuration Tool	
PBX General Settings Extension Pool Name Name Name Name Name IP Address 110.10.10.108 Realm UDP Port 5060 TCP Port 5060 Paging Zone From Field LOCAL Outbound DTMF 3 Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement Zone 0 Event Queing	File	
Name Nortel Channels 1-16 IP Address 110.10.10.108 Realm UDP Port UDP Port 5060 TCP Port 5060 Paging Zone Paging Zone From Field LOCAL Outbound DTMF 3 Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement 2one Zone 0 Event Queing	PBX PBX Portel General Settings Extension Pool	General 😹 Advanced 🌾 Channels 📓 Registe 💶
Channels 1-16		Name Nortel
IP Address 110.10.10.108 Realm UDP Port 5060 TCP Port 5060 Paging Zone From Field LOCAL Outbound DTMF 3 Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement 2 Zone 0 Event Queing		Channels 1-16
Realm UDP Port 5060 TCP Port 5060 Paging Zone From Field LOCAL Outbound DTMF 3 Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement Zone 0 Event Queing		IP Address 110.10.10.108
UDP Port 5060 TCP Port 5060 Paging Zone From Field LOCAL Outbound DTMF 3 Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement 2 Zone 0 Event Queing		Realm
TCP Port 5060 Paging Zone		UDP Port 5060
Paging Zone From Field Outbound DTMF 3 Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement Zone 0 Event Queing		TCP Port 5060
From Field LOCAL Outbound DTMF 3 Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement 2 Zone 0 Event Queing		Paging Zone
Outbound DTMF 3 Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement 101 Zone 0 Event Queing		From Field LOCAL
Port Routing 0 DTMF Payload 101 Pause (Comma) Replacement 2 Zone 0 Event Queing		Outbound DTMF 3
DTMF Payload 101 Pause (Comma) Replacement 200 Cone 0 Event Queing		Port Routing
Pause (Comma) Replacement Zone Event Queing		DTME Payload
Zone 0		
Event Queing		
		Event Queing
	· ·	

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.

PBX	General Advanced Channels 🕽 Registe
General Settings	 Out Dial Without Connection Ignore Disconnect Before Notify Cancel Transfer After Forwarding Cancel Transfer Without Reinvite Enable Internal Bridging Use TCP Activate Keep Alive Immediate RTP Accept VM Calls Indicate Transfer Forced Off Hold Blind Transfer Accept Refer Accept Forwarded Calls
	Comma Timeout (ms) 2000 Digit Duration (ms) 100 Interdigit Timeout (ms) 5000 Blind Call Timeout (ms) 30000

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.

S	IP Configuration Tool					×
	File					
	PBX	🤯 General 🛛	😸 Advanced 💏	Channels	Registe 🚺	<u>-</u>
	😪 General Settings	Channel	Extension No	IP Address	Authenticate	
	Extension Pool	1 🤇	30000	•		
		2	*	_		
		3	*	-		

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.

SIP Configuration Tool		×
File		
PBX Nortel General Settings Extension Pool	Advanced S Channel	s 🎽 Register MWI 🔹 🕨
	MWI UDP Port	0
	MWI TCP Port	0
	MWI Mode	102
	MWI Extension	
	MWI Host	

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.

SIP Configuration Tool		×
File		
File	Channels Register MWI ANI	
		ancel

Figure 32: ANI Configuration

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

SIP Configuration Tool		×
File		
PBX General Settings Extension Pool	General Settings External IP Outbound Proxy IP Outbound Proxy Port	STUN

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** \rightarrow **All Programs** \rightarrow **Office-LinX** \rightarrow **Office Linx Admin** (Not Shown). Access the **Properties** of the PBX as shown in **Figure 34**.

🔝 Office Linx Adı	nin				
🍠 File Action	View Language	Window	Help		
← → 🖻 🖆	?				
🏂 Office-LinX Adn	nin	PBX	Ng.	Description	
 	Properties About New Window from	n Here		Nortel - CS	
	Help				

Figure 34: Accessing PBX Properties Screen

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

🖏 PBX	? ×
<u>P</u> BX	
	1
Nortel - CS1000	
On Code 🔽 On Hook	
Off Code	
Interval Time 0	
C Dial Diaita	
O Send Through LUM Device	
ССТІ	

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 Release7.5 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 Sever 1000, R7.5 NN43001-611, 05.09 Sept 2011

Product documentation for Esna Office-LinX may be found at: <u>http://www.esnatech.com/support/tech_index.htm</u>

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