



Avaya Solution & Interoperability Test Lab

Configuring Juniper SRX210 Switch to provide Power over Ethernet to Avaya 9600, 1600 and 4600 Series IP and SIP Telephones with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services running on Avaya Aura™ Midsize Enterprise Single Server - Issue 1.0

Abstract

These Application Notes describes the configuration steps required to connect Avaya 9600, 1600 and 4600 Series IP and SIP Telephones to a Juniper SRX210 Power over Ethernet Switch with Avaya Aura™ Communication Manager and Avaya Aura™ SIP Enablement Services running on Avaya Aura™ Midsize Enterprise Single Server. The Application Notes identifies how to configure a vlan on a Juniper SRX210 switch and to administer a SIP Trunk between Avaya Aura™ Communication Manager and SIP Enablement Services to carry voice calls between Avaya IP and SIP endpoints.

The validation test of the sample configuration was conducted at the Avaya Solution and Interoperability Test Lab at the request of the Avaya Solutions and Marketing Team.

1. Introduction

Power over Ethernet (PoE) technology describes a system to safely pass electrical power, along with data, on Ethernet cabling. Power over Ethernet requires category 5 cable or higher for high power levels, but can operate with category 3 cable for low power levels. Power can come from a power supply within a Power over Ethernet enabled networking device such as an Ethernet switch or from a device built for injecting power onto the Ethernet cabling, dubbed Midspan. The IEEE 802.3af 2003 Power over Ethernet standard provides up to 15.4 Watts of DC power to each device. Only 12.95 Watts is assured to be available at the powered device as some power is dissipated in the cable. The IEEE 802.3at 2009 Power over Ethernet standard provides up to 25.5 Watts of power. Some vendors have announced products that claim to comply with the new 802.3at standard and offer up to 51 Watts of power over a single cable by utilizing all 4 pairs in the category 5 cable. Numerous non-standard schemes had been used prior to Power over Ethernet standardization to provide Power over Ethernet cabling.

1.1. Interoperability Compliance Testing

The objective of this interoperability test is to verify that the Juniper SRX210 can provide Power over Ethernet capability to Avaya 9600, 1600 and 4600 Series IP and SIP telephones and interoperate with Avaya AuraTM Communication Manager 5.2.1 and Avaya AuraTM SIP Enablement Services 5.2 running on Avaya AuraTM Midsize Enterprise Single Server. It also includes procedures for configuring a vlan in router and Juniper SRX210. Testing was carried out on codec support and negotiation supported by Avaya 9600, 1600 and 4600 IP and SIP telephones and as well as supplementary features such as Call Hold, Forward, Transfer and Conference between the Avaya IP and SIP endpoints.

1.2. Configuration

The configuration used in these Application Notes is shown in **Figure 1**. The Avaya Aura™ Midsize Enterprise software is installed and configured on Avaya Aura™ System Platform on a S8500C Media Server. The Avaya Aura™ Midsize Enterprise Single Server is a template running software applications. These software applications include Avaya Aura™ Communication Manager, Avaya Aura™ SIP Enablement Services The Avaya Aura™ Midsize Enterprise Media Server is connected to a 24P data switch and is configured in a separate vlan. All IP and SIP telephones are physically connected to a single Juniper SRX210 switch and are administered in a single subnet. The 9600, 1600 and 4600 IP telephones register to Avaya Aura™ Communication Manager running on the Avaya Aura™ Midsize Enterprise Single Server and are administered as H.323 stations. The 9600 SIP telephones register to Avaya Aura™ SIP Enablement Services running on the Avaya Aura™ Midsize Enterprise Single Server and are administered as an OPS station on Avaya Aura™ Communication Manager. Both the 24P data switch and the Juniper SRX210 switch are connected to a router. Each of the two switches is configured with an uplink trunk port to connect to the router.

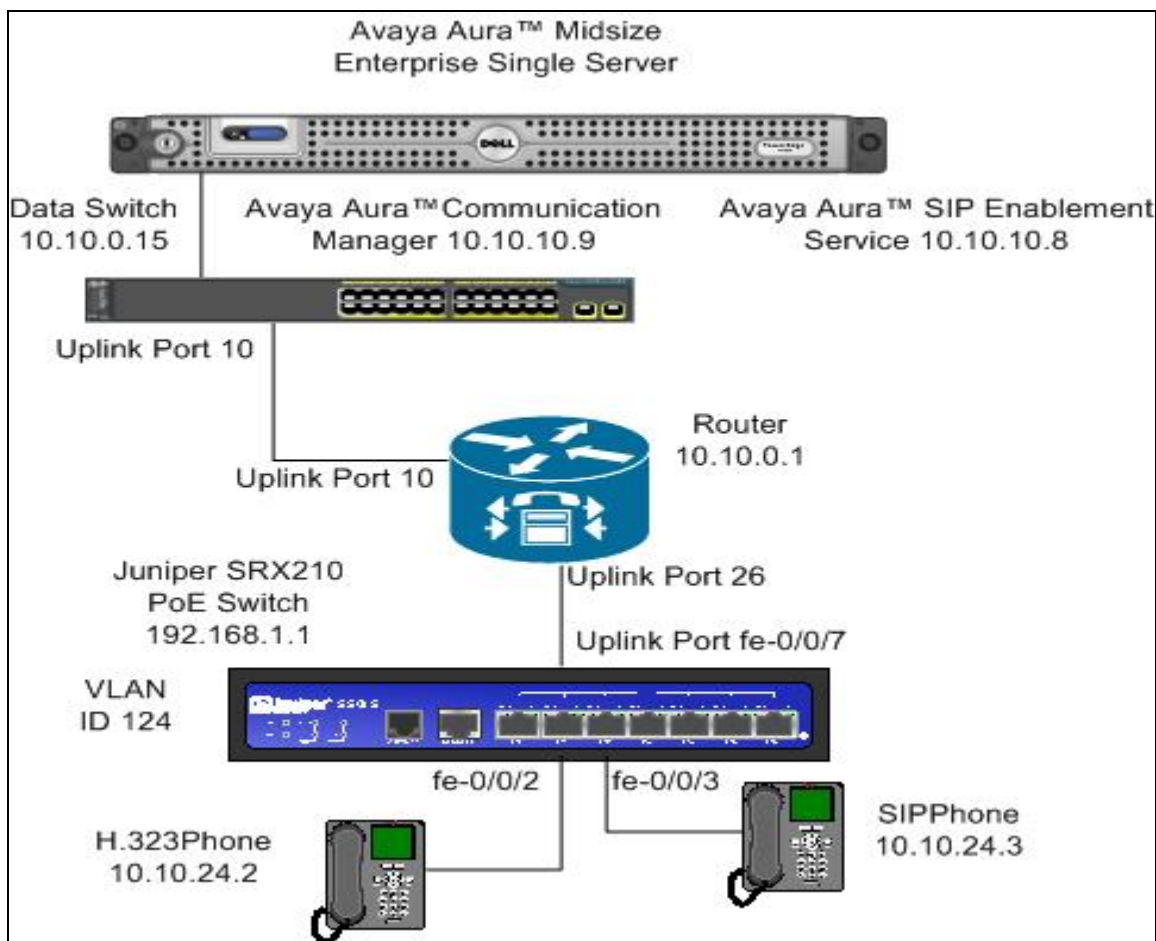


Figure 1: Avaya Aura™ Midsize Enterprise Single Server with Juniper SRX210 Switch

2. Equipment and Software Validated

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

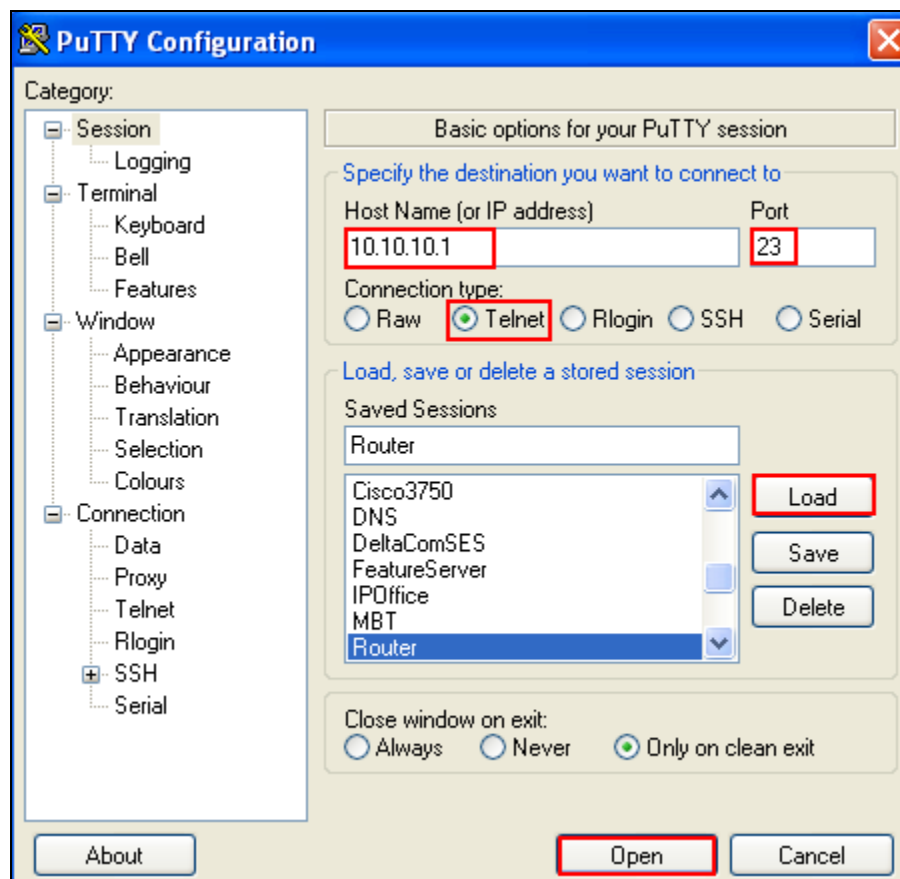
Avaya Aura™	Software
Avaya Avaya™ Midsize Enterprise Single Server on a S8500C Media Server	Avaya Avaya™ Midsize Enterprise R5.2 Release 5.2.1.2.5 Avaya Aura™ Communication Manager R5.2.1 Release 5.2.1 R15x.02.1.016.4 Update: Service Pack 0 Avaya Aura™ SIP Enablement Services R5.2 Release SES05.2.1.016.4 Update: Service Pack 0
Avaya one-X® 9600 Series IP Telephones (SIP)	Release 2.5
Avaya one-X® 9600 Series IP Telephones (H.323)	Release 3.1
Avaya one-X® 1600 Series IP Telephones (H.323)	Release 1.22
Avaya one-X® 4600 IP Telephones (H.323)	Release 2.9
Juniper SRX210	Junos 10.0 R1
Data Switch	Release 12.0.3.16
Router	Release 12.0.3.16

3. Configure Vlan in a Router

This section describes the steps needed to configure vlans in the router. In this sample configuration vlan 124 was assigned IP address range 10.10.24.1/24 with the name vlan p124 and the Communication Manager vlan 220 was assigned the IP address range 10.10.10.1/24 the name vlan s220.

3.1. Access the Router

To access the router open **PuTTY Configuration** and input the IP Address of the router and use the **Telnet** connection type with **port 23**. The **IP Address** of the router was **10.10.10.1**. Load the following information and press the **Open** button.



Enter the router **Login: admin** and **password** and hit the return key. This brings the user to the command line interface of the router shown as **X450e-48p.1 #**.

```
telnet session telnet0 on /dev/ptyb2

login: admin
password:

Press the <tab> or '?' key at any time for completions.
Remember to save your configuration changes.

router #
```

3.2. Create Vlan 124

To create vlan 124 the **create vlan p124** was issued from the command line interface of the router. To configure voice vlan 124 and assign vlan tag 124 to the vlan the following command **configure vlan p124 tag 124** was issued. The subnet range 10.10.24.1/24 for voice vlan 124 was used. From the command line interface the **configure vlan p124 ip address 10.10.24.1/24** was issued to assign this range to vlan 124. When new ip interfaces are added to the router IP forwarding is disabled by default and must be enabled. To enable IP forwarding on voice vlan p124 the command **enable ip forwarding vlan p124** was issued from the command line interface of the router.

```
router #create vlan p124
router #configure vlan p124 tag 124
router #configure vlan p124 ip address 10.10.24.1/24
router #enable ip forwarding vlan p124
```

3.3. Add Uplink Interface to Vlan 124

This sample configuration uses **port 26** on the router as the uplink interface that would connect to the Juniper SRX210 switch. This port needed to be added to voice vlan 124 so voice vlan 124 could communicate with Communication Manager in vlan 220. From the command line interface of the router the **configure vlan p124 add port 26 tagged** performed this function. The uplink interface port 26 was added to vlan s220 also with the command **configure vlan s220 add port 26 tagged**.

```
router #configure vlan p124 add port 26 tagged
router #configure vlan s220 add port 26 tagged
```

3.4. Create Vlan 220

To create vlan 220 the **create vlan s220** was issued from the command line interface of the router. To configure vlan 220 and assign vlan tag 220 to the vlan the following command **configure vlan s220 tag 220** was issued. The subnet range 10.10.10.1/24 for vlan 220 was used. From the command line interface the **configure vlan s220 ip address 135.64.186.68/24** was issued to assign this range to vlan 220. When new ip interfaces are added to the router IP forwarding is disabled by default and must be enabled. To enable IP forwarding on vlan s124 the command **enable ip forwarding vlan s220** was issued from the command line interface of the router.

```
router #create vlan s220
router #configure vlan s220 tag 220
router #configure vlan s220 ip address 135.64.186.68/24
router #enable ip forwarding vlan s220
```

3.5. Add Uplink Interface to Vlan 220

This sample configuration uses **port 10** on the router as the uplink interface that would connect to the switch. This port needed to be added to vlan s220 so vlan 220 could communicate with vlan 124. From the command line interface of the router the **configure vlan s220 add port 10 tagged** performed this function. The uplink interface port 10 was added to vlan p124 also with the command **configure vlan p124 add port 10 tagged**.

```
router #configure vlan s220 add port 10 tagged
router #configure vlan p124 add port 10 tagged
```

4. Configure Vlan in Juniper SRX210

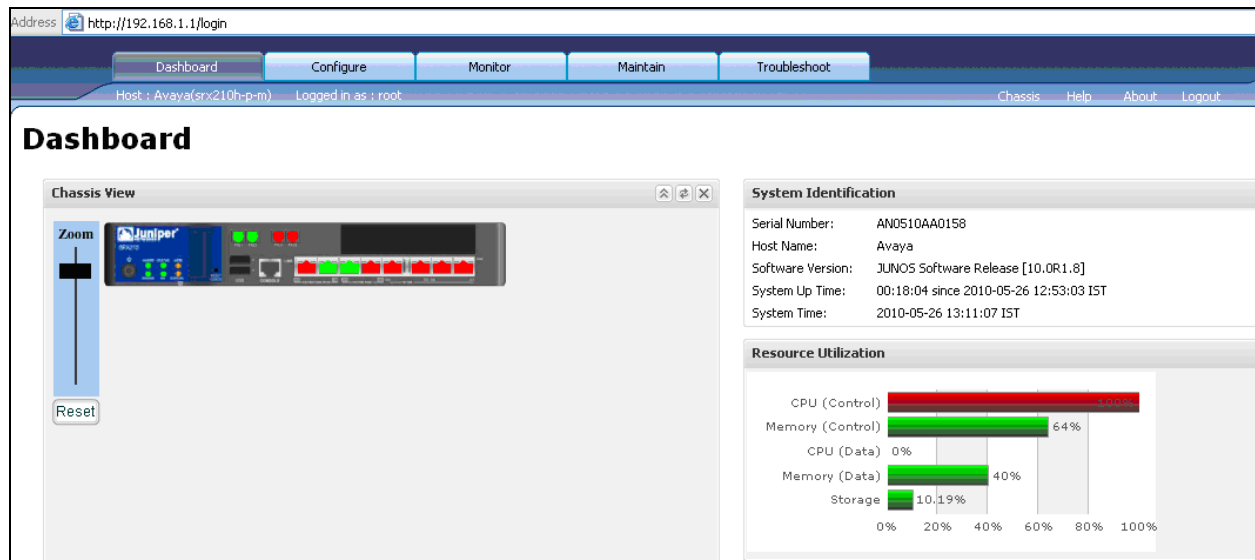
This section describes steps needed to configure a vlan in the Juniper SRX210. This sample configuration uses vlan with IP address range 10.10.24.1/24.

4.1. Access Juniper SRX210

To access the Juniper SRX210 browse to the management IP Address of the Juniper SRX210 which was **192.168.1.1**. The following screenshot appears and enter the **username** and **password** and press the **Login** button.



The following screenshot is displayed showing the graphical user interface of the Juniper SRX210.



It was decided to configure the Juniper SRX210 through the command line interface rather than use the graphical user interface. To access the command line interface input the login and password for the Juniper SRX210. This brings the user to the command line interface. At the level **root@Avaya%** type **cli**. This takes the user to the **root@Avaya>** prompt which is privileged mode. Then type **configure** to enter configuration mode.

```
Login:
Password:
root@Avaya%cli
root@Avaya>configure
root@Avaya#
```

4.2. Enable Power over Ethernet on Interfaces

On the Juniper SRX210 Power over Ethernet is enabled through the command line interface. The PoE interface must be enabled in order for the port to provide power to a connected powered device. The **set poe interface all** command enables power over ethernet to all interfaces on the Juniper SRX210.

```
root@Avaya>configure
root@Avaya#set poe interface all
```

4.3. Assign Interfaces to Vlan 124

To create vlan 124 in the Juniper SRX210 switch the user must enter configuration mode from privileged mode. It was decided to use interface fe-0/0/2 as the interface the Avaya 9600 IP telephone would plug into and interface fe-0/0/3 as the interface the Avaya 9600 SIP telephone would plug into. Create vlan124 with vlan-id 124 with the command **set vlans vlan124 vlan-id 124**. Assign vlan 124 to interface fe-0/0/2 with the command **set interfaces fe-0/0/2 unit 0 vlan-id 124**.

```
root@Avaya#configure
root@Avaya#set vlans vlan124 vlan-id 124
root@Avaya#set interfaces fe-0/0/2 unit 0 vlan-id 124
```

4.4. Assign IP Address to Interface fe-0/0/2

The IP address 10.10.24.2 was assigned to interface fe-0/0/2 by issuing the following command **set interfaces fe-0/0/2 unit 0 family inet address 10.10.24.2** from the command line interface of the Juniper SRX210 switch. Voice vlan was assigned to the interface fe-0/0/2 with the command **set interfaces fe-0/0/2 unit 0 description voice vlan**.

```
root@Avaya#set interfaces fe-0/0/2 unit 0 description voice vlan
root@Avaya#set interfaces fe-0/0/2 unit 0 family inet address 10.10.24.2
```

4.5. Assign IP Address to Interface fe-0/0/3

The IP address 10.10.24.3 was assigned to interface fe-0/0/3 by issuing the following command **set interfaces fe-0/0/3 unit 0 family inet address 10.10.24.3** from the command line interface of the Juniper SRX210 switch. Voice vlan was assigned to the interface fe-0/0/3 with the command **set interfaces fe-0/0/3 unit 0 description voice vlan**.

```
root@Avaya#set interfaces fe-0/0/3 unit 0 description voice vlan
root@Avaya#set interfaces fe-0/0/3 unit 0 family inet address 10.10.24.3
```

4.6. Assign Security Zones to Interface fe-0/0/2 and fe-0/0/3

In this sample configuration the interfaces fe-0/0/2 and fe-0/0/3 was assigned to a trusted security zone using the commands **set security zones security-zones trust interfaces fe-0/0/2** and **set security zones security-zones trust interfaces fe-0/0/3**.

```
root@Avaya#set security zones security-zone trust interfaces fe-0/0/2
root@Avaya#set security zones security-zone trust interfaces fe-0/0/3
```

4.7. Configure Uplink Interface on the Juniper SRX210

Interface **fe-0/0/7** on the Juniper SRX210 was used as the uplink interface to the router. It was configured as a trunking port to carry traffic between the Juniper SRX210 and the router. The **vlan124** was allowed across interface **fe-0/0/7** with the command **set interfaces fe-0/0/7 unit 0 ethernet-switching vlan members vlan124**. The interface **fe-0/0/7** was set to 802.1q trunk mode by issuing the command **set interfaces fe-0/0/7 unit 0 family ethernet-switching port-mode trunk**.

```
root@Avaya#configure
root@Avaya#set interfaces fe-0/0/7 unit 0 ethernet-switching vlan members vlan124
root@Avaya#set interfaces fe-0/0/7 unit 0 family ethernet-switching port-mode trunk
```

4.8. Assign Security Zones to Interface fe-0/0/7

In this sample configuration the interface **fe-0/0/7** was assigned to a trusted security zone using the command **set security zones security-zones trust interfaces fe-0/0/7**.

```
root@Avaya#set security zones security-zone trust interfaces fe-0/0/2
```

5. Administer Avaya Aura™ Communication Manager

This section highlights the important commands for registering Avaya IP telephones within Communication Manager and administering an IP network region and IP codecs to carry calls between Avaya IP and SIP endpoints. It also highlights the important commands for defining Avaya SIP telephones as an Off-PBX Station (OPS) and administering a SIP Trunk and Signaling Group to carry calls between Avaya IP and SIP telephones.

5.1. Verify OPS Capacity

Use the **display system-parameters customer-options** command to verify that **Maximum Off-PBX Telephones** has been set to the value that has been licensed, and that this value will accommodate addition of the SIP telephones. If a required feature is not enabled or there is insufficient capacity, contact an authorized Avaya Sales representative to obtain additional capacity.

display system-parameters customer-options		Page	1 of 11
OPTIONAL FEATURES			
G3 Version: V15	Software Package: Standard		
Location: 2	RFA System ID (SID): 1		
Platform: 25	RFA Module ID (MID): 1		
			USED
Platform Maximum Ports:			44000 181
Maximum Stations:			2400 9
Maximum XMOBILE Stations:			2400 0
Maximum Off-PBX Telephones - EC500:			2400 2
Maximum Off-PBX Telephones - OPS:			2400 5
Maximum Off-PBX Telephones - PBFMC:			2400 2
Maximum Off-PBX Telephones - PVFMC:			2400 0

Verify that **Maximum Concurrently Registered IP Stations** has been set to the value that has been licensed, and that this value will accommodate addition of IP telephones. Verify that the **Maximum Administered SIP Trunks** has been set to accommodate addition of SIP Trunks.

display system-parameters customer-options		Page	2 of 11
OPTIONAL FEATURES			
IP PORT CAPACITIES			USED
Maximum Administered H.323 Trunks:			8000 12
Maximum Concurrently Registered IP Stations:			18000 5
Maximum Administered Remote Office Trunks:			8000 0
Maximum Concurrently Registered Remote Office Stations:			18000 0
Maximum Concurrently Registered IP eCons:			128 0
Max Concur Registered Unauthenticated H.323 Stations:			100 0
Maximum Video Capable Stations:			2400 0
Maximum Video Capable IP Softphones:			100 3
Maximum Administered SIP Trunks:			5000 80
Maximum Administered Ad-hoc Video Conferencing Ports:			8000 0
Maximum Number of DS1 Boards with Echo Cancellation:			522 0
Maximum TN2501 VAL Boards:			10 1
Maximum Media Gateway VAL Sources:			250 0
Maximum TN2602 Boards with 80 VoIP Channels:			128 0
Maximum TN2602 Boards with 320 VoIP Channels:			128 0
Maximum Number of Expanded Meet-me Conference Ports:			300 0

5.2. Administer Dial Plan Analysis

This section describes the **Dial Plan Analysis** screen. This is Communication Manager's way of translating digits dialed by the user. The user can determine the beginning digits and total length for each type of call that Communication Manager needs to interpret. The **Dialed String** beginning with the number **4** and with a **Total Length** of **5** digits will be used to administer the **extension** range used for the IP telephones.

display dialplan analysis						Page 1 of 12					
DIAL PLAN ANALYSIS TABLE											
Location: all						Percent Full: 0					
	Dialed String	Total Length	Call Type		Dialed String	Total Length	Call Type		Dialed String	Total Length	Call Type
0		1	fac								
1		5	ext								
2		5	aar								
209		5	ext								
3		5	aar								
4		5	ext								
5		5	ext								
6		5	ext								
7		5	aar								
8		5	aar								
9		1	fac								
*		3	dac								
#		3	dac								

5.3. Administer IP Node-Name

This section describes **IP Node-Name** form. This is where Communication Manager assigns the IP Address and node-name to the SIP Enablement Services server. The node-name of the SIP Enablement Services server is **ses1** and the IP Address of the SIP Enablement Services server is **135.64.186.89**. Communication Manager automatically populates a processor node name to the IP Address of Communication Manager. This node name is **procr** with IP Address **135.64.186.81**.

```
list node-names all
```

NODE NAMES		
Type	Name	IP Address
IP	AES1	135.64.186.88
IP	CMM	135.64.186.82
IP	MedSvcsMedpro1	135.64.186.84
IP	MedSvcsMedpro2	135.64.186.85
IP	MedSvcsMedpro3	135.64.186.86
IP	MedSvcsMedpro4	135.64.186.87
IP	procr	135.64.186.81
IP	ses1	135.64.186.89

5.4. Administer Signaling Group

This section describes the **Signaling Group** screen. The **Group Type** was set to **sip** and the **Transport Method** was set to **tls**. Since the sip trunk is between Communication Manager and SIP Enablement Services the **Near-end Node Name** is the node name of Communication Manager, **procr**. The **Far-end Node Name** is the node name of SIP Enablement Services. This is **ses1**. The **Near-end Listen Port** and **Far-end Listen Port** are both set to port number **5061**. The **Far-end Network-Region** was set to **1**. The **Far-end Domain** is **silstack.com**, the domain name of the SIP Enablement Services Server.

```
display signaling-group 3
```

SIGNALING GROUP	
Group Number: 3	Group Type: sip
	Transport Method: tls
IMS Enabled? n	
IP Video? n	
Near-end Node Name: procr	Far-end Node Name: ses1
Near-end Listen Port: 5061	Far-end Listen Port: 5061
	Far-end Network Region: 1
Far-end Domain: silstack.com	
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y
Enable Layer 3 Test? n	IP Audio Hairpinning? n
	Direct IP-IP Early Media? n

5.5. Administer Trunk Group

This section describes **Trunk Group** used to carry calls between SIP telephones. Trunk Group 3 was configured as a SIP Trunk with the **Group Type** set as **sip**. The trunk **Group Name** was set to **SIP Trunk to SES**. The **Direction** of the calls was set to **two-way** as there will be calls to and from Avaya IP and SIP telephones. The **Service Type** was set to **tie** since the trunk is configured as an internal trunk between Communication Manager and SIP Enablement Services. The **Signaling Group** number assigned to this trunk is **3**. The **Number of Members** assigned to this trunk group is **100**. All other fields on this page are left as default.

```
display trunk-group 3                                     Page 1 of 21

                                TRUNK GROUP

Group Number: 3                      Group Type: sip          CDR Reports: y
  Group Name: SIP Trunk to SES        COR: 1                TN: 1          TAC: *03
  Direction: two-way                 Outgoing Display? n
Dial Access? n                               Night Service:
Queue Length: 0
Service Type: tie                      Auth Code? n

                                Signaling Group: 3
                                Number of Members: 100
```

5.6. Administer IP Network Region

This section describes **IP Network Region** screen. The sample configuration places all IP and SIP endpoints in the one network region. The **Authoritative Domain** must mirror the domain name of the SIP Enablement Services Server. This was **silstack.com**. The codecs used on the IP and SIP endpoints were placed in **Codec Set 1**. IP Shuffling was turned on so both **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** were set to **yes**.

```
display ip-network-region 1                               Page 1 of 19

                                IP NETWORK REGION

Region: 1
Location: 1      Authoritative Domain: silstack.com
Name:
MEDIA PARAMETERS                                Intra-region IP-IP Direct Audio: yes
  Codec Set: 1                                Inter-region IP-IP Direct Audio: yes
  UDP Port Min: 2048                          IP Audio Hairpinning? n
  UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                        RTCP Reporting Enabled? y
  Call Control PHB Value: 46                  RTCP MONITOR SERVER PARAMETERS
  Audio PHB Value: 46                        Use Default Server Parameters? y
  Video PHB Value: 26
802.1P/Q PARAMETERS
  Call Control 802.1p Priority: 6
  Audio 802.1p Priority: 6
  Video 802.1p Priority: 5                    AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                RSVP Enabled? n
  H.323 Link Bounce Recovery? y
Idle Traffic Interval (sec): 20
Keep-Alive Interval (sec): 5
Keep-Alive Count: 5
```

5.7. Administer IP Codec Set

This section describes the **IP Codec Set** screen. It was decided to use IP Codec **G.711MU**, **G.711A** and **G.729** for testing purposes with the IP and SIP endpoints.

display ip-codec-set 1

Page 1 of 2

IP Codec Set

Codec Set: 1

Audio	Silence	Frames	Packet
Codec	Suppression	Per Pkt	Size(ms)
1: G.711MU	n	2	20
2: G.711A	n	2	20
3: G.729	n	2	20
4:			

5.8. Administer Off PBX Telephone Station Mapping

This section shows the **off-pbx-telephone station-mapping** form. The Avaya SIP telephone extension **40126** uses off pbx **Application OPS** which is used for SIP enabled telephones. The **SIP Trunk Selection** is **3** as Trunk Group 3 was configured. The **Config Set** which is the desired call treatment was set to **1**.

display off-pbx-telephone station-mapping 40126								Page	1 of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION										
Station	Application	Dial	CC	Phone Number	Trunk	Config	Dual			
Extension		Prefix			Selection	Set	Mode			
40126	OPS	-		40126	3	1				

On **Page 2**, the **Call Limit** is set to **6** as shown below. This is the maximum amount of simultaneous calls for extension 40126. The **Mapping Mode** field was set to **both** in this configuration setup. This is used to control the degree of integration between SIP telephones. The **Calls Allowed** field was set to **all**. This identifies the call filter type for a SIP Phone. The **Bridged Calls** field was set to **none** as it was not needed for testing purposes.

display off-pbx-telephone station-mapping 40126								Page	2 of	3
STATIONS WITH OFF-PBX TELEPHONE INTEGRATION										
Station	Appl	Call	Mapping	Calls	Bridged					
Extension	Name	Limit	Mode	Allowed	Calls	Location				
40126	OPS	6	both	all	none					

5.9. Administer Station Screen

This screen describes the **station** form used to define the SIP telephone in Communication Manager. The **Extension** used was **40126** with phone **Type 9630**. The **Name** of the phone was set to **PoE test** and all other values on **Page 1** of the **station** form were left as default.

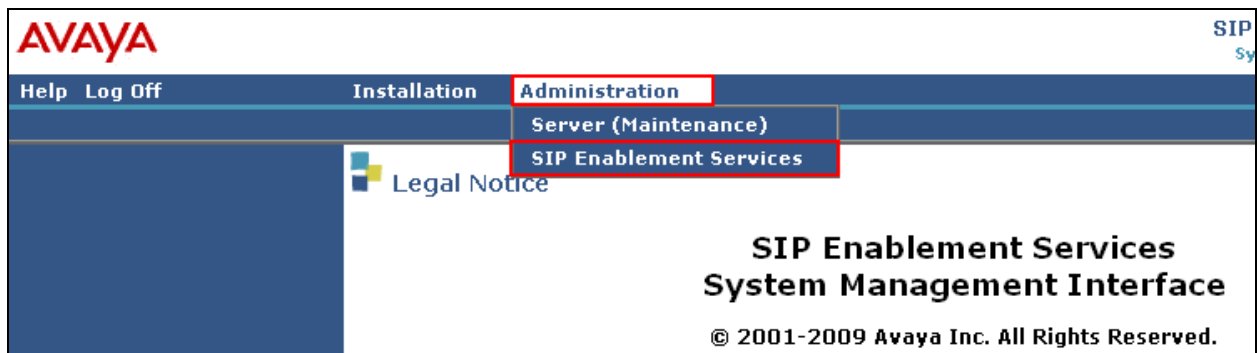
display station 40126		Page 1 of 5
STATION		
Extension: 40126	Lock Messages? n	BCC: 0
Type: 9630	Security Code:	TN: 1
Port: S00010	Coverage Path 1:	COR: 1
Name: PoE test	Coverage Path 2:	COS: 1
	Hunt-to Station:	
STATION OPTIONS		
Loss Group: 19	Time of Day Lock Table:	
	Personalized Ringing Pattern: 1	
Speakerphone: 2-way	Message Lamp Ext: 40030	
Display Language: english	Mute Button Enabled? y	
Survivable GK Node Name:	Expansion Module? n	
Survivable COR: internal	Media Complex Ext:	
Survivable Trunk Dest? y	IP SoftPhone? n	
	IP Video? n	

6. Administer Avaya Aura™ SIP Enablement Services

The following steps describe configuration of SIP Enablement Services to allow Avaya SIP telephones to register to SIP Enablement Services.

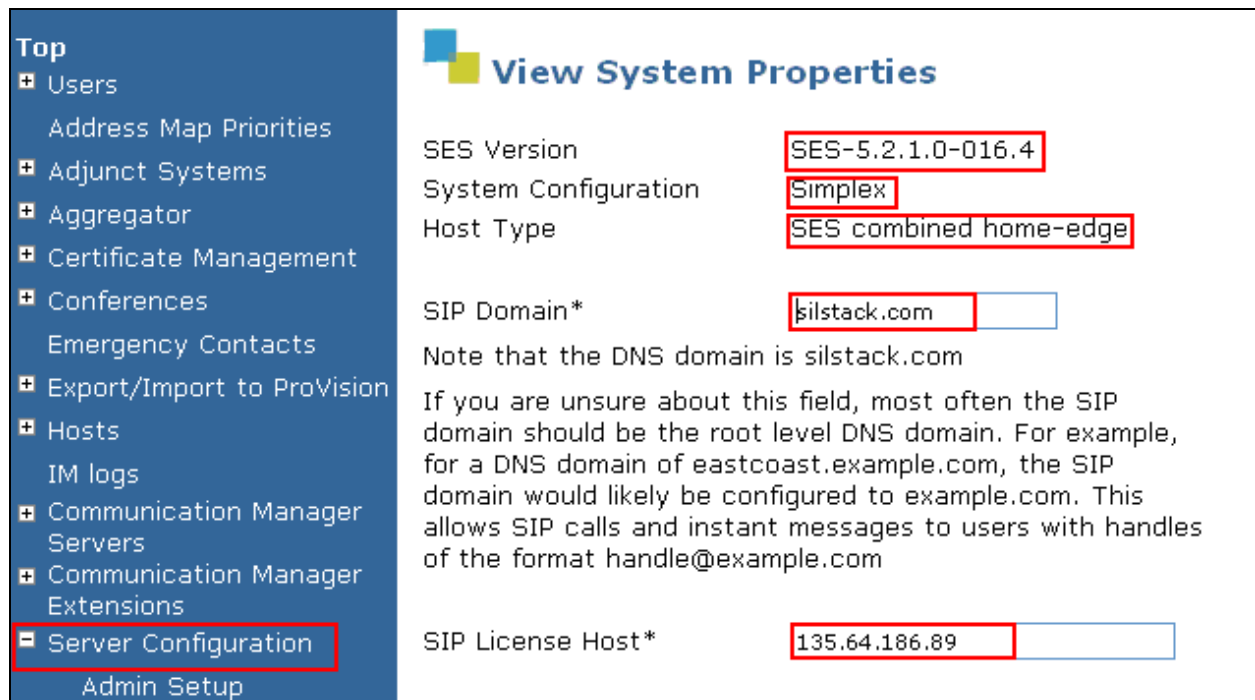
6.1. Access Avaya Aura™ SIP Enablement Services

Access the SES Administration web interface, by entering **http://<ip-addr>/admin** as the URL in Internet browser, where **<ip-addr>** is the IP address of the SIP Enablement Services server. Log in with the appropriate credentials and then select the **Administration** link and then **SIP Enablement Services** from the main screen.



6.2. System Properties

The **View System Properties** screen defines the server's type and domain. The **SES Version** field displays the release number, the current load and build number of the Avaya software that is running on this SES server. The **System Configuration** field identifies the SES server as being a **Simplex** machine. The **Host Type** field identifies the SES server as a **SES combined home-edge** type server. The **SIP Domain** field indicates the domain name assigned to the SIP Enablement Services Configuration. This was set to **silstack.com**. The **SIP License Host** field requires the IP address of the SES server that is running the WebLM application and has the associated license file installed. This entry shows the IP address of the SIP Enablement Services server was entered as **135.64.186.89**.



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- IM logs
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- Communication Manager Extensions
- Server Configuration**
- Admin Setup

View System Properties

SES Version: SES-5.2.1.0-016.4

System Configuration: Simplex

Host Type: SES combined home-edge

SIP Domain*: silstack.com

Note that the DNS domain is silstack.com

If you are unsure about this field, most often the SIP domain should be the root level DNS domain. For example, for a DNS domain of eastcoast.example.com, the SIP domain would likely be configured to example.com. This allows SIP calls and instant messages to users with handles of the format handle@example.com

SIP License Host*: 135.64.186.89

6.3. Add Host Screen

The **Host IP Address** field contains the IP address for this combined home/edge server. This was **135.64.186.89**. The **Profile Service Password** is for permissions between SES hosts. This is not used by the administrator; it is used by internal software components for secure communication between SES servers and the master administration system. The **Host Type** functions as a **SES combined home-edge** server. In the **Listen Protocol** fields **UDP** and **TLS** were selected. The **Link Protocols** field refers to the trunk signaling between SIP Enablement Services and Communication Manager. Typically, the selection here matches the Signal Group value on Communication Manager. This was **TLS**. For third-party proxy servers you may select to link to SES with TLS, TCP or UDP.

Edit Host

Host IP Address*	135.64.186.89
Profile Service Password*	••••••••
Host Type	SES combined home-edge
Parent	none
Listen Protocols	<input checked="" type="checkbox"/> UDP <input checked="" type="checkbox"/> TCP <input checked="" type="checkbox"/> TLS
Link Protocols	<input type="radio"/> UDP <input type="radio"/> TCP <input checked="" type="radio"/> TLS

6.4. Administer Avaya SIP Telephones

This screen allows Avaya SIP telephone users to be added to the SES. Users are added one at a time with this screen. A handle identifies the user on the SES system. Users **Primary Handle** must be the same as the **User ID**'s. In this example the **Primary Handle** and **User ID** is **40126**. The **Password** needs to be six characters long and was set to **123456**. This password is needed when the Avaya SIP telephone registers to the SES server after the extension of the SIP phone is inputted. The **Host** IP address is populated automatically to **135.64.186.89**. The name of the Avaya SIP telephone was **PoE SIP (First Name, Last Name)**. Check the **Add Communication Manager Extension**. Press the **Add** button at the bottom of the screen. The SIP Phone extension 40126 must be added to Communication Manager also.

Address Map Priorities	Primary Handle*	40126
+ Adjunct Systems	User ID	40126
+ Aggregator	Password*	••••••
+ Certificate Management	Confirm Password*	••••••
+ Conferences	Host*	135.64.186.89
Emergency Contacts	First Name*	QoS
+ Export/Import to ProVision	Last Name*	SIP
+ Hosts	Address 1	
IM logs	Address 2	
+ Communication Manager Servers	Office	
+ Communication Manager Extensions	City	
+ Server Configuration	State	
+ SIP Phone Settings	Country	
+ Survivable Call Processors	Zip	
System Status	Survivable Call Processor	none
+ Trace Logger	Add Communication Manager Extension	<input checked="" type="checkbox"/>
+ Trusted Hosts	Fields marked * are required.	
	<input type="button" value="Add"/>	

Once the **Add Communication Manager Extension** field is checked the screen below appears. Confirm that extension **40126** is the **Communication Manager Extension** and press **Add**.

Top

- Users
 - Add
 - Default Profile
 - Delete
 - Edit
 - List
 - Password
 - Search
 - Manage All Registered Users

Add Communication Manager Extension

Extension: 40126

Communication Manager: CM

Server:

Fields marked * are required.

Add

7. Verification Steps

The following verification steps were tested using the sample configuration.

From the Juniper SRX web interface, access the **Configure** tab, then the **Interfaces** heading on the left hand side of the graphical user interface. Verify the interfaces fe-0/0/2, fe-0/0/3 are in service for the Juniper SRX210 as shown below.

Address: http://192.168.1.1/login

Dashboard **Configure** Monitor Maintain Troubleshoot

Host: Avaya(srx210h-p-m) Logged in as: root Commit Compare Discard Chassis Help About

Interfaces

Link Aggregation

Authentication

NAT

Security

IPSec VPN

Wireless LAN

Switching

Routing

Class of Service

System Properties

Convergence Services

Configure

Interfaces

Interface Name	Link State	Configured	Description
ge-0/0/0	Down	Yes	Gigabit Ethernet Interface 'ge-0/0/0'
ge-0/0/0.0	Down	Yes	Logical Unit 0 on Gigabit Ethernet Interface 'ge-0/0/0'
lsq-0/0/0	Up	No	Adaptive Services with link-services mode 'lsq-0/0/0'
ge-0/0/1	Down	No	Gigabit Ethernet Interface 'ge-0/0/1'
ge-0/0/1.0	Down	No	Logical Unit 0 on Gigabit Ethernet Interface 'ge-0/0/1'
fe-0/0/2	Up	Yes	Fast Ethernet Interface 'fe-0/0/2'
fe-0/0/2.0	Up	Yes	Logical Unit 0 on Fast Ethernet Interface 'fe-0/0/2'
fe-0/0/3	Up	Yes	Fast Ethernet Interface 'fe-0/0/3'
fe-0/0/3.0	Up	No	Logical Unit 0 on Fast Ethernet Interface 'fe-0/0/3'
fe-0/0/4	Down	No	Fast Ethernet Interface 'fe-0/0/4'
fe-0/0/4.0	Down	No	Logical Unit 0 on Fast Ethernet Interface 'fe-0/0/4'
fe-0/0/5	Down	No	Fast Ethernet Interface 'fe-0/0/5'
fe-0/0/5.0	Down	No	Logical Unit 0 on Fast Ethernet Interface 'fe-0/0/5'
fe-0/0/6	Down	No	Fast Ethernet Interface 'fe-0/0/6'
fe-0/0/6.0	Down	No	Logical Unit 0 on Fast Ethernet Interface 'fe-0/0/6'

From Communication Manager, verify the IP telephones registered to Communication Manager as shown below.

```
list registered-ip-stations
```

REGISTERED IP STATIONS				
Station Ext or Orig Port	Set Type/ Net Rgn	Prod ID/ Release	TCP Skt Gatekeeper	Station IP Address/ IP Address
40020	9640 1	IP_Phone 3.1000	y	10.10.99.11 135.64.186.81
40124	9640 1	IP_Phone 3.1000	y	10.10.24.2 135.64.186.81
40125	4621 1	IP_Phone 2.9	y	10.10.24.4 135.64.186.81
40127	1608 1	IP_Phone 1.2200	y	10.10.24.5 135.64.186.81
40128	9630 1	IP_Phone 3.1000	y	10.10.24.6 135.64.186.81

To see what endpoints are registered to the SIP Enablement Services server access the **Search Registered Users** on the left hand side of the System Management Interface menu. The screenshot below shows the Avaya SIP telephone 40126 registered to SIP Enablement Services.

Top

Users

Add

Default Profile

Delete

Edit

List

Password

Search

Manage All Registered Users

Search Registered Devices

Search Registered Users

Address Map Priorities

Adjunct Systems

Registered Devices on 135.64.186.89

[Registered and Provisioned Devices](#)
[Registered Devices](#)
[Provisioned Devices](#)
[Search](#)
[Refresh](#)

Showing 1 to 2 of 2 registered contacts.

Handle	Program Version	MAC Address	Phone Type	Timestamp
<input type="checkbox"/> 40001	R5.2100-SP1-19397	00:00:00:00:00:00	one-X Communicator	2010-03-31 17:08:17
	2.5.0	00:04:0d:ec:a6:9e	one-X Deskphone	2010-04-15 16:59:35
<input type="checkbox"/> 40126	2.5.0	00:04:0d:ec:a3:ec	one-X Deskphone	2010-04-16 16:33:27

Verify calls can be made with clear audio from the Avaya IP telephone to the SIP telephone.
Verify the calls are seen to be active within Communication Manager.

list trace station 40125		
LIST TRACE		
time	data	
11:24:15	active station 40125 cid 0x473	
11:24:15	G711MU ss:off ps:20	
	rgn:1 [10.10.24.2]:2340	
	rgn:1 [135.64.186.86]:2980	
11:24:18	dial 40124	
11:24:18	term station 40124 cid 0x473	
11:24:20	active station 40124 cid 0x473	
11:24:20	G711MU ss:off ps:20	
	rgn:1 [10.10.24.3]:5004	
	rgn:1 [10.10.24.2]:2340	
11:24:20	G711MU ss:off ps:20	
	rgn:1 [10.10.24.2]:2340	
	rgn:1 [10.10.23.3]:5004	
11:25:30	idle station 40125 cid 0x473	

Verify supplementary features such as Call Hold, Call Forward, Conference and Transfer can be completed between the Avaya IP and SIP endpoints. All test calls were successful.

8. Conclusion

These Application Notes describe the administration steps required to configure: a vlan on a router and Juniper SRX210, administer Avaya IP and SIP telephones within Avaya AuraTM Communication Manager and Avaya AuraTM SIP Enablement Services to result in H.323 and SIP registration, configure IP network-region and IP codecs, and administration of a SIP Trunk and Signaling Group to carry calls between Avaya IP and SIP endpoints. The power consumption for each IP and SIP telephone was within the required specification of 15.4Watts. All of these functions were verified successfully.

9. Additional References

This section references the documentation relevant to these Application Notes. Additional Avaya product documentation is available at <http://support.avaya.com>.

- [1] Avaya AuraTM SIP Enablement Services (SES) Implementation Guide, May 2009, Document Number 16-300140
- [1] Administering Avaya AuraTM Communication Manager, Document Number 03-300509
- [3] SRX210 Hardware Guide, Release 10.0, available at <http://www.juniper.com>
- [4] Interface and Routing Configuration Guide for J-Series Services Routers and SRX-series Services Gateways, Release 10.0, available at <http://www.juniper.com>

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