



Avaya Solution & Interoperability Test Lab

Application Notes for Voice4net ePBX and EBS with Avaya Aura® Communication Manager and Avaya Aura® Session Manager – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate the Voice4net ePBX and EBS solutions with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Voice4net ePBX is an interactive voice response (IVR) system that can run custom applications. Voice4net EBS (Event Broadcasting System) provides call centers and public agencies the capability to broadcast messages via the telephone. These Voice4net solutions interface to Avaya Aura® Session Manager using SIP trunks.

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 configuration steps required to integrate the Voice4net ePBX and EBS solutions with Avaya Aura® Communication Manager and Avaya Aura® Session Manager. Voice4net ePBX is an interactive voice response (IVR) system that can run custom applications. Voice4net EBS (Event Broadcasting System) provides call centers and public agencies the capability to broadcast messages via the telephone. These Voice4net solutions interface to Avaya Aura® Session Manager using SIP trunks.

The Voice4net ePBX IVR Platform and EBS Automated Dialer are flexible, fully programmable telephony solutions which allow the delivery custom applications. For the compliance test, a custom IVR application that performed blind transfers was used. The EBS Automated Dialer is used to make outbound calls for purposes such as broadcast, notifications, reminders, telemarketing campaigns, appointments, etc. EBS is a software module that may be configured with the standard ePBX platform to give the system this functionality.

2. General Test Approach and Test Results

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

The feature test cases were performed manually. For the ePBX testing, blind transfers to the PSTN and local stations were exercised. For the EBS testing, a broadcast message was sent to PSTN and local stations.

The serviceability test cases were performed manually by disconnecting/reconnecting the Ethernet cable to the EBS/ePBX server and by rebooting the server.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing focused on verifying the following on Voice4net ePBX/EBS:

- Inbound calls to ePBX from the PSTN and local stations. ePBX ran a customer application that performed blind transfers.
- Blind transfers from ePBX to the PSTN and to local stations.
- Call transfers from ePBX to call center agents (i.e., hunt group members).
- Call transfers from ePBX to busy station.
- Outbound calls from EBS to the PSTN and local stations. EBS called out to the specified number and delivered a broadcast message.
- Outbound calls using the Element Dashboard to PSTN and local stations.
- Verification of ANI/DNIS.

- Verification of ePBX call logs.
- G.711 mu-law codec support.
- DTMF using RFC 2833.

The serviceability testing focused on verifying the ability of Voice4net ePBX/EBS to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet cable and rebooting the server.

2.2. Test Results

All test cases passed. One observation noted was that a call transfer to a busy station resulted in the call being disconnected without the caller hearing busy tone. ePBX assumes that there will be a forwarding path on any blind transfer so that the call would either go another extension, voicemail, or an available call center agent.

2.3. Support

Contact Voice4net at (214) 237-7600 (option 2) for ePBX/EBS technical support or submit a support request through the Voice4net website at <http://www.voice4net.com/voice4net-support.html>.

3. Reference Configuration

Figure 1 illustrates a sample configuration with an Avaya SIP-based network that includes the following Avaya products:

- Communication Manager running on an Avaya S8300 Server with a G450 Media Gateway.
- Session Manager connected to Communication Manager via a SIP trunk and serving SIP telephones and Voice4net ePBX/EBS. Session Manager was configured using Avaya Aura® System Manager (not shown).
- Avaya H.323 and SIP telephones.

In the compliance testing, the Avaya IP Phones had extensions in the range of 4xxxx and Voice4net ePBX/EBS was assigned extension 49000. Voice4net ePBX/EBS interfaced to Avaya Aura® Session Manager via SIP trunks.

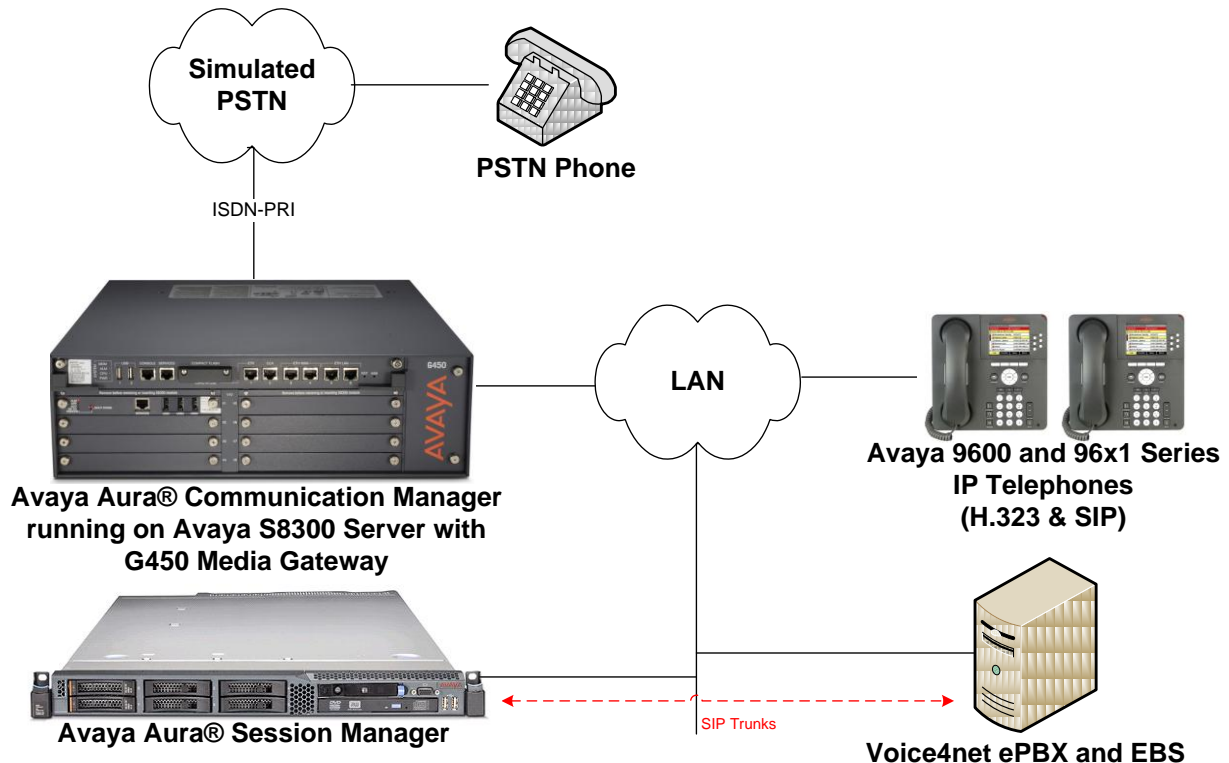


Figure 1: Voice4net ePBX/EBS with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunks

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Communication Manager	6.2 SP 3 (R016x.02.0.823.0 with Patch 19926)
Avaya Aura® System Manager	6.2.0 SP 3 (Build No. 6.2.0.0.15669-6.2.12.307) (System Update Revision No: 6.2.15.1.1959)
Avaya Aura® Session Manager	6.2 (6.2.3.0.623006)
Avaya 9600 Series IP Telephones	3.1 SP 4 (H.323) 2.6.7.0 (SIP)
Avaya 96x1 Series IP Telephones	6.2119 (H.323) 6.2.0.72 (SIP)
Voice4net ePBX/EBS	6.1.5.2
Voice4net SIP Stack	2.2.7.0

5. Configure Avaya Aura® Communication Manager

This section describes the steps for configuring a SIP trunk to Session Manager, which in turn routes calls to Voice4net ePBX/EBS via a SIP trunk. Administration of Communication Manager was performed using the System Access Terminal (SAT). The SAT is accessed by establishing a telnet session to Communication Manager using a terminal emulation application.

Note: The SAT commands that were entered are displayed at the top of each screenshot.

This section covers the following configuration:

- Verify SIP Trunk Capacity
- **IP Node Names** to associate names with IP addresses.
- **IP Network Region** to specify the domain name, IP codec set, and enable IP-IP direct audio (i.e., Shuffling).
- **IP Codec Set** to specify the codec type used for calls to Voice4net ePBX.
- **SIP trunks** for incoming/outgoing calls to/from Voice4net ePBX/EBS.
- **Private Numbering** to allow the caller's extension to be sent to Voice4net ePBX/EBS.
- **Call Routing** to route calls to Voice4net ePBX using AAR.

5.1. Verify OPS and SIP Trunk Capacity

Using the SAT, verify that the SIP Trunks options is enabled on the **system-parameters customer-options** form. The license file installed on the system controls these options. If a required feature is not enabled, contact an authorized Avaya sales representative.

On **Page 2** of the **system-parameters customer-options** form, verify that the number of **Maximum Administered SIP Trunks** supported by the system is sufficient.

display system-parameters customer-options		Page 2 of 11
OPTIONAL FEATURES		
IP PORT CAPACITIES		USED
Maximum Administered H.323 Trunks:	4000	30
Maximum Concurrently Registered IP Stations:	2400	4
Maximum Administered Remote Office Trunks:	4000	0
Maximum Concurrently Registered Remote Office Stations:	2400	0
Maximum Concurrently Registered IP eCons:	68	0
Max Concur Registered Unauthenticated H.323 Stations:	100	0
Maximum Video Capable Stations:	2400	0
Maximum Video Capable IP Softphones:	2400	0
Maximum Administered SIP Trunks:	4000	90
Maximum Administered Ad-hoc Video Conferencing Ports:	4000	0
Maximum Number of DS1 Boards with Echo Cancellation:	80	0
Maximum TN2501 VAL Boards:	10	0
Maximum Media Gateway VAL Sources:	50	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
(NOTE: You must logoff & login to effect the permission changes.)		

5.2. Configure IP Node Names

In the **IP Node Names** form, assign an IP address and host name for the S8300 in the G450 Media Gateway (*procr*) and Session Manager (*lz-asm*). The host names will be used in other configuration screens of Communication Manager.

change node-names ip		Page 1 of 2
IP NODE NAMES		
Name	IP Address	
default	0.0.0.0	
devcon13	10.32.24.20	
lz-asm	192.168.100.235	
procr	192.168.100.10	
procr6	::	
(5 of 5 administered node-names were displayed)		
Use 'list node-names' command to see all the administered node-names		
Use 'change node-names ip xxx' to change a node-name 'xxx' or add a node-name		

5.3. Configure IP Network Region

In the **IP Network Region** form, the **Authoritative Domain** field is configured to match the domain name configured on Session Manager. In this configuration, the domain name is *devcon.com*. By default, **IP-IP Direct Audio** (shuffling) is enabled to allow audio traffic to be sent directly between IP endpoints without using media resources in the Avaya G650 Media Gateway. The **IP Network Region** form also specifies the **IP Codec Set** to be used for calls routed over the SIP trunk to Session Manager. This codec set is used when its corresponding network region (i.e., IP Network Region 1) is specified in the SIP signaling group.

change ip-network-region 1		Page 1 of 20
IP NETWORK REGION		
Region: 1		
Location: 1		Authoritative Domain: devcon.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? y
UDP Port Max: 65535		
DIFFSERV/TOS PARAMETERS		
Call Control PHB Value: 46		
Audio PHB Value: 46		
Video PHB Value: 26		
802.1P/Q PARAMETERS		
Call Control 802.1p Priority: 6		
Audio 802.1p Priority: 6		
Video 802.1p Priority: 5		
H.323 IP ENDPOINTS		AUDIO RESOURCE RESERVATION PARAMETERS
H.323 Link Bounce Recovery? y		RSVP Enabled? n
Idle Traffic Interval (sec): 20		
Keep-Alive Interval (sec): 5		
Keep-Alive Count: 5		

5.4. Configure IP Codec Set

In the **IP Codec Set** form, select the audio codec type supported for calls routed over the SIP trunk to Voice4net ePBX/EBS. The form is accessed via the **change ip-codec-set 1** command. Testing was performed with G.711mu-law.

change ip-codec-set 1

Page 1 of 2

IP Codec Set

Codec Set: 1

	Audio Codec	Silence Suppression	Frames Per Pkt	Packet Size (ms)
1:	G.711MU	n	2	20
2:				
3:				
4:				
5:				
6:				
7:				

5.5. Configure SIP Trunk

Prior to configuring a SIP trunk group for communication with Session Manager, a SIP signaling group must be configured. Configure the **Signaling Group** form as follows:

- Set the **Group Type** field to *sip*.
- Set the **IMS Enabled** field to *n*.
- The **Transport Method** field was set to *tcp*.
- Specify the S8300 and the Session Manager as the two ends of the signaling group in the **Near-end Node Name** field and the **Far-end Node Name** field, respectively. These field values are taken from the **IP Node Names** form.
- Ensure that the TCP port value of *5060* is configured in the **Near-end Listen Port** and the **Far-end Listen Port** fields.
- The preferred codec for the call will be selected from the IP codec set assigned to the IP network region specified in the **Far-end Network Region** field.
- Enter the domain name of Session Manager in the **Far-end Domain** field. In this configuration, the domain name is *devcon.com*.
- The **Direct IP-IP Audio Connections** field was enabled on this form.
- The **DTMF over IP** field should be set to the default value of *rtp-payload*.
- Disable **Initial IP-IP Direct Media**.

Communication Manager supports DTMF transmission using RFC 2833. The default values for the other fields may be used.

add signaling-group 60		Page 1 of 2
SIGNALING GROUP		
Group Number: 60	Group Type: sip	
IMS Enabled? n	Transport Method: tcp	
Q-SIP? n		
IP Video? n	Enforce SIPS URI for SRTP? y	
Peer Detection Enabled? y	Peer Server: SM	
Near-end Node Name: procr	Far-end Node Name: lz-asm	
Near-end Listen Port: 5060	Far-end Listen Port: 5060	
	Far-end Network Region: 1	
Far-end Domain: devcon.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? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

Configure the **Trunk Group** form as shown below. This trunk group is used for calls to SIP endpoints and Voice4net ePBX/EBS. Set the **Group Type** field to *sip*, set the **Service Type** field to *tie*, specify the signaling group associated with this trunk group in the **Signaling Group** field, and specify the **Number of Members** supported by this SIP trunk group. Configure the other fields in bold and accept the default values for the remaining fields.

change trunk-group 60		Page 1 of 21	
TRUNK GROUP			
Group Number: 60	Group Type: sip	CDR Reports: y	
Group Name: To lz-asm	COR: 1	TN: 1	TAC: 1060
Direction: two-way	Outgoing Display? n		
Dial Access? n	Night Service:		
Queue Length: 0			
Service Type: tie	Auth Code? n		
	Member Assignment Method: auto		
	Signaling Group: 60		
	Number of Members: 40		

On **Page 3** of the trunk group form, set the **Numbering Format** field to *private*. This field specifies the format of the calling party number sent to the far-end.

add trunk-group 60		Page 3 of 21	
TRUNK FEATURES			
ACA Assignment? n	Measured: none	Maintenance Tests? y	
Numbering Format: private			
UUI Treatment: service-provider			
Replace Restricted Numbers? n			
Replace Unavailable Numbers? n			
Modify Tandem Calling Number: no			
Show ANSWERED BY on Display? Y			

5.6. Configure Private Numbering

Configure the **Numbering – Private Format** form to send the calling party number to the far-end. Add an entry so that local stations with a 5-digit extension beginning with **4** whose calls are routed over any trunk group, including SIP trunk group 60, have the extension sent to the far-end.

change private-numbering 0					Page 1 of 2
NUMBERING - PRIVATE FORMAT					
Ext Len	Ext Code	Trk Grp(s)	Private Prefix	Total Len	
5	4			5	Total Administered: 1 Maximum Entries: 540

5.7. Configure Call Routing

In this configuration, AAR was used to route calls to Voice4net ePBX. The extension assigned to ePBX was 49000. This extension was included in the **Uniform Dial Plan Table** so that when it is dialed it is routed using AAR. The **AAR analysis** table then routed calls destined for ePBX over the SIP trunk configured in **Section 5.5** as specified on the **Route Pattern** form. For information in configuring AAR or ARS, refer to [1]. The route pattern form is shown below.


change route-pattern 60													Page 1 of 3
Pattern Number: 60 Pattern Name: To lz-asm													
SCCAN? n Secure SIP? n													
Grp No	FRL	NPA	Pfx	Hop	Toll	No.	Inserted	DCS/	IXC				
			Mrk	Lmt	List	Del	Digits	QSIG					
							Dgts	Intw					
1:	60	0						n	user				
2:								n	user				
3:								n	user				
4:								n	user				
5:								n	user				
6:								n	user				
BCC	VALUE	TSC	CA-TSC	ITC	BCIE	Service/Feature	PARM	No.	Numbering	LAR			
0	1	2	M	4	W	Request		Dgts	Format				
										Subaddress			
1:	y	y	y	y	y	n	n	rest	unk-unk	none			
2:	y	y	y	y	y	n	n	rest		none			
3:	y	y	y	y	y	n	n	rest		none			
4:	y	y	y	y	y	n	n	rest		none			
5:	y	y	y	y	y	n	n	rest		none			
6:	y	y	y	y	y	n	n	rest		none			

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include adding the following items:

- SIP domain
- Logical/physical Locations that can be occupied by SIP Entities
- SIP Entities corresponding to Session Manager, Communication Manager, and Voice4net ePBX/EBS
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities
- Routing Policies, which control call routing between the SIP Entities
- Dial Patterns, which govern to which SIP Entity a call is routed
- Session Manager, corresponding to the Session Manager Server to be managed by System Manager

Configuration is accomplished by accessing the browser-based GUI of System Manager using the URL “https://<ip-address>/SMGR”, where <ip-address> is the IP address of System Manager. Log in with the appropriate credentials. The initial screen is displayed as shown below. The configuration in this section will be performed under **Routing** and **Session Manager** listed within the **Elements** box.

 Avaya Aura® System Manager 6.2 Last Logged on at November 30, 2012 2:41 PM
[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Users

- Administrators**
Manage Administrative Users
- Directory Synchronization**
Synchronize users with the enterprise directory
- Groups & Roles**
Manage groups, roles and assign roles to users
- UCM Roles**
Manage UCM Roles, assign roles to users
- User Management**
Manage users, shared user resources and provision users

Elements

- B5800 Branch Gateway**
Manage B5800 Branch Gateway 6.2 elements
- Communication Manager**
Manage Communication Manager 5.2 and higher elements
- Conferencing**
Manage Conferencing Multimedia Server objects
- Inventory**
Manage, discover, and navigate to elements, update element software
- Meeting Exchange**
Manage Meeting Exchange and Avaya Aura Conferencing 6.0 elements
- Messaging**
Manage Avaya Aura Messaging, Communication Manager Messaging, and Modular Messaging
- Presence**
Presence
- Routing**
Network Routing Policy
- Session Manager**
Session Manager Element Manager
- SIP AS 8.1**
SIP AS 8.1

Services

- Backup and Restore**
Backup and restore System Manager database
- Bulk Import and Export**
Manage Bulk Import and Export of Users, User Global Settings, Roles, Elements and others
- Configurations**
Manage system wide configurations
- Events**
Manage alarms, view and harvest logs
- Licenses**
View and configure licenses
- Replication**
Track data replication nodes, repair replication nodes
- Scheduler**
Schedule, track, cancel, update and delete jobs
- Security**
Manage Security Certificates
- Templates**
Manage Templates for Communication Manager, Messaging System and B5800 Branch Gateway elements
- UCM Services**
Manage UCM applications and navigation such as CS1000 deployment, patching, ISSS and SNMP

6.1. Specify SIP Domain

Select **Routing** under the **Elements** section. Add the SIP domain for which the communications infrastructure will be authoritative. Do this by selecting **Domains** on the left and clicking the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

- **Name:** The authoritative domain name (e.g., *devcon.com*)
- **Notes:** Descriptive text (optional).

Click **Commit**.

Since the sample configuration does not deal with any other domains, no additional domains need to be added.

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The top header includes the Avaya logo, the product name 'Avaya Aura® System Manager 6.2', and a user status bar indicating 'Last Logged on at November 30, 2012 2:41 PM' with links for 'Help', 'About', 'Change Password', and 'Log off admin'. The left sidebar shows a navigation menu with 'Routing' expanded and 'Domains' selected. The main content area is titled 'Domain Management' and features a breadcrumb trail 'Home / Elements / Routing / Domains'. Below the title are buttons for 'Edit', 'New', 'Duplicate', 'Delete', and 'More Actions'. A table lists 2 items, with the first item being 'devcon.com' of type 'sip'. The table has columns for 'Name', 'Type', 'Default', and 'Notes'. A 'Filter: Enable' link is present on the right. At the bottom of the table, there is a 'Select : All, None' option.

	Name	Type	Default	Notes
<input type="checkbox"/>	devcon.com	sip	<input type="checkbox"/>	

6.2. Add Locations

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management. To add a location, select **Locations** on the left and click on the **New** button (not shown) on the right. The following screen will then be shown. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **Notes:** Descriptive text (optional).

The screen below shows addition of the *Lincroft* location, which includes the Communication Manager and Session Manager.

AVAYA Avaya Aura® System Manager 6.2 Last Logged on at November 30, 2012 2:41 PM
Help | About | Change Password | Log off
admin

[Routing](#) [Home](#)

Home / Elements / Routing / Locations

Location Details [Help ?](#)
[Commit](#) [Cancel](#)

General

* **Name:**

Notes:

Overall Managed Bandwidth

Managed Bandwidth Units:

Total Bandwidth:

Multimedia Bandwidth:

Audio Calls Can Take Multimedia Bandwidth: ☒

Per-Call Bandwidth Parameters

Maximum Multimedia Bandwidth (Intra-Location): Kbit/Sec

Maximum Multimedia Bandwidth (Inter-Location): Kbit/Sec

* **Minimum Multimedia Bandwidth:** Kbit/Sec

* **Default Audio Bandwidth:** Kbit/sec

Alarm Threshold

Overall Alarm Threshold: %

Multimedia Alarm Threshold: %

* **Latency before Overall Alarm Trigger:** Minutes

* **Latency before Multimedia Alarm Trigger:** Minutes

Under *Location Pattern*:

- **IP Address Pattern:** A pattern used to logically identify the location.
- **Notes:** Descriptive text (optional).

Click **Commit** to save the Location definition.

The screenshot shows a web-based configuration interface for 'Location Pattern'. At the top, there are 'Add' and 'Remove' buttons. Below them is a table with one item. The table has two columns: 'IP Address Pattern' and 'Notes'. The first row shows a checkbox, a text input field containing '* 192.168.100.*', and a text input field containing 'devcon14 (CM) and lz-asm (SM)'. Below the table, there is a 'Select : All, None' dropdown. At the bottom, there is a red asterisk followed by 'Input Required' and 'Commit' and 'Cancel' buttons.

IP Address Pattern	Notes
* 192.168.100.*	devcon14 (CM) and lz-asm (SM)

6.3. Add SIP Entities

In the sample configuration, a SIP Entity is added for Session Manager, the S8300 running Communication Manager, and Voice4net ePBX/EBS. Session Manager will communicate with each SIP entity directly.

6.3.1. Session Manager

A SIP Entity must be added for Session Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface on Session Manager.
- **Type:** Select *Session Manager*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Under *Port*, click **Add**, and then edit the fields in the resulting new row as shown below:

- **Port:** Port number on which the system listens for SIP requests.
- **Protocol:** Transport protocol to be used to send SIP requests.
- **Default Domain** The domain used for the enterprise (e.g., *devcon.com*).

Defaults can be used for the remaining fields. Click **Commit** to save each SIP Entity definition.



Routing x Home

- Routing
- Domains
- Locations
- Adaptations
- SIP Entities**
- Entity Links
- Time Ranges
- Routing Policies
- Dial Patterns
- Regular Expressions
- Defaults

Home / Elements / Routing / SIP Entities

[Help ?](#)

SIP Entity Details

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Location:

Outbound Proxy:

Time Zone:

Credential name:

SIP Link Monitoring

SIP Link Monitoring:

Entity Links

Entity Links can be modified after SIP Entity is committed.

Port

TCP Failover port:

TLS Failover port:

3 Items Refresh

Filter: Enable

<input type="checkbox"/>	Port	Protocol	Default Domain	Notes
<input type="checkbox"/>	<input type="text" value="5060"/>	<input type="text" value="UDP"/>	<input type="text" value="devcon.com"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5060"/>	<input type="text" value="TCP"/>	<input type="text" value="devcon.com"/>	<input type="text"/>
<input type="checkbox"/>	<input type="text" value="5061"/>	<input type="text" value="TLS"/>	<input type="text" value="devcon.com"/>	<input type="text"/>

Select : All, None

6.3.2. Communication Manager

A SIP Entity must be added for the Communication Manager. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., S8300) on the telephony system.
- **Type:** Select *CM*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

Defaults may be used for the remaining fields. Click **Commit** to save the SIP Entity definition.

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.2", and a user status bar indicating "Last Logged on at November 30, 2012 2:41 PM" with links for "Help", "About", "Change Password", and "Log off admin". A breadcrumb trail shows "Home / Elements / Routing / SIP Entities". On the left, a sidebar menu lists various configuration options, with "SIP Entities" highlighted. The main content area is titled "SIP Entity Details" and contains a "General" tab. The form fields are as follows: "Name" (devcon14), "FQDN or IP Address" (192.168.100.10), "Type" (CM), "Notes" (empty), "Adaptation" (empty), "Location" (Lincroft), "Time Zone" (America/New_York), "Override Port & Transport with DNS SRV" (unchecked), "SIP Timer B/F (in seconds)" (4), "Credential name" (empty), "Call Detail Recording" (none), "SIP Link Monitoring" (Use Session Manager Configuration), "Supports Call Admission Control" (unchecked), "Shared Bandwidth Manager" (unchecked), "Primary Session Manager Bandwidth Association" (empty), and "Backup Session Manager Bandwidth Association" (empty). At the bottom, there is a section for "Entity Links" with a note: "Entity Links can be modified after SIP Entity is committed." Buttons for "Commit" and "Cancel" are located at the top right of the form area.

6.3.3. Voice4net ePBX/EBS

A SIP Entity must be added for Voice4net ePBX/EBS. To add a SIP Entity, select **SIP Entities** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

- **Name:** A descriptive name.
- **FQDN or IP Address:** IP address of the signaling interface (e.g., *Voice4net*) on the telephony system.
- **Type:** Select *SIP Trunk*.
- **Location:** Select the location defined previously.
- **Time Zone:** Time zone for this location.

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The top navigation bar includes the Avaya logo, the title "Avaya Aura® System Manager 6.2", and user information: "Last Logged on at November 30, 2012 2:41 PM", "Help | About | Change Password | Log off", and "admin". A breadcrumb trail shows "Home / Elements / Routing / SIP Entities". The left sidebar contains a menu with "Routing" expanded, showing sub-items: Domains, Locations, Adaptations, SIP Entities (highlighted), Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled "SIP Entity Details" with a "General" tab selected. It contains several form fields: "Name" (Voice4net), "FQDN or IP Address" (192.168.100.160), "Type" (SIP Trunk), "Notes" (empty), "Adaptation" (empty), "Location" (Lincroft), "Time Zone" (America/New_York), "Override Port & Transport with DNS SRV" (checkbox), "SIP Timer B/F (in seconds)" (4), "Credential name" (empty), "Call Detail Recording" (egress), "SIP Link Monitoring" (Use Session Manager Configuration), "Supports Call Admission Control" (checkbox), "Shared Bandwidth Manager" (checkbox), "Primary Session Manager Bandwidth Association" (empty), and "Backup Session Manager Bandwidth Association" (empty). At the bottom, there is a section for "Entity Links" with a red warning message: "Entity Links can be modified after SIP Entity is committed." Buttons for "Commit", "Cancel", and "Help ?" are located at the top right of the form area.

6.4. Add Entity Links

In the sample configuration, two Entity links were added, one for Communication Manager and another one for Voice4net ePBX/EBS.

6.4.1. Communication Manager

The SIP trunk from Session Manager to Communication Manager is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *lz-asm to devcon14*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the name of Communication Manager.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Selected *Trusted*. *Note: If the link is not trusted, calls from the associated SIP Entity specified in Section 6.3.2 will be denied.*

Click **Commit** to save the Entity Link definition.

Avaya Aura® System Manager 6.2

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Routing * Home

Home / Elements / Routing / Entity Links

Entity Links

Commit Cancel Help ?

1 Item | Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* lz-asm to devcon14	* lz-asm	TCP	* 5060	* devcon14	* 5060	Trusted	

* Input Required

Commit Cancel

6.4.2. Voice4net ePBX/EBS

The SIP trunk from Session Manager to Voice4net ePBX/EBS is described by an Entity link. To add an Entity Link, select **Entity Links** on the left and click on the **New** button (not shown) on the right. Fill in the following fields in the new row that is displayed:

- **Name:** A descriptive name (e.g., *Voice4net Link*).
- **SIP Entity 1:** Select the Session Manager.
- **Protocol:** Select the appropriate protocol.
- **Port:** Port number to which the other system sends SIP requests.
- **SIP Entity 2:** Select the *Voice4net* SIP entity.
- **Port:** Port number on which the other system receives SIP requests.
- **Connection Policy:** Selected *Trusted*. *Note: If the link is not trusted, calls from the associated SIP Entity specified in Section 6.3.3 will be denied.*

Click **Commit** to save the Entity Link definition.

Avaya Aura® System Manager 6.2

Last Logged on at November 30, 2012 2:41 PM
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[Routing](#) * [Home](#)

Home / Elements / Routing / Entity Links

Entity Links

[Commit](#) [Cancel](#) [Help ?](#)

1 Item Refresh Filter: Enable

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
* Voice4net Link	* lz-asm	UDP	* 5060	* Voice4net	* 5060	Trusted	

* Input Required

[Commit](#) [Cancel](#)

6.5. Add Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities specified in **Section 6.3**. Two routing policies were added – one for Communication Manager and one for Voice4net. To add a routing policy, select **Routing Policies** on the left and click on the **New** button (not shown) on the right. The following screen is displayed. Fill in the following:

Under *General*:

Enter a descriptive name in **Name**.

Under *SIP Entity as Destination*:

Click **Select**, and then select the appropriate SIP entity to which this routing policy applies.

Defaults can be used for the remaining fields. Click **Commit** to save each Routing Policy definition. The following screen shows the Routing Policy for Communication Manager.

The screenshot displays the Avaya Aura System Manager 6.2 web interface. The left sidebar shows a navigation menu with 'Routing Policies' selected. The main content area is titled 'Routing Policy Details' and includes a breadcrumb trail 'Home / Elements / Routing / Routing Policies'. The 'General' tab is active, showing fields for 'Name' (devcon14 Policy), 'Disabled' (unchecked), 'Retries' (0), and 'Notes'. The 'SIP Entity as Destination' section has a 'Select' button and a table listing the selected entity 'devcon14' with FQDN '192.168.100.10' and Type 'CM'. The 'Time of Day' section includes 'Add', 'Remove', and 'View Gaps/Overlaps' buttons, a table with one item for 'Ranking 1' (24/7), and a 'Filter: Enable' option. The 'Dial Patterns' section has 'Add' and 'Remove' buttons.

Avaya Aura® System Manager 6.2

Last Logged on at November 30, 2012 2:41 PM
Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Routing Policies

Routing Policy Details

General

* Name: devcon14 Policy

Disabled: ☐

* Retries: 0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
devcon14	192.168.100.10	CM	

Time of Day

Add Remove View Gaps/Overlaps

1 Item Refresh Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	24/7	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

Dial Patterns

Add Remove

The following screen shows the Routing Policy for Voice4net.

AVAYA

Avaya Aura® System Manager 6.2

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[Help](#) | [About](#) | [Change Password](#) | [Log off admin](#)

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Home / Elements / Routing / Routing Policies

Routing Policy Details

Commit

Cancel

Help ?

General

* Name:

Voice4net Policy

Disabled:

☐

* Retries:

0

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
Voice4net	192.168.100.160	SIP Trunk	

Time of Day

Add

Remove

View Gaps/Overlaps

1 Item | Refresh

Filter: Enable

<input type="checkbox"/>	Ranking	1 ▲	Name	2 ▲	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
<input type="checkbox"/>	0		24/7		<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	Time Range 24/7

Select : All, None

Dial Patterns

Add

Remove

6.6. Add Dial Patterns

Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “4” reside on Communication Manager, extension “49000” is assigned to Voice4net ePBX/EBS. To add a dial pattern, select **Dial Patterns** on the left and click on the **New** button (not shown) on the right. Fill in the following:

Under *General*:

- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern.

Under *Originating Locations and Routing Policies*:

Click **Add**, and then select the appropriate location and routing policy from the list.

Default values can be used for the remaining fields. Click **Commit** to save this dial pattern. The following screen shows the dial pattern definitions for local extensions on Communication Manager.

Avaya Aura® System Manager 6.2

Last Logged on at November 30, 2012 2:41 PM
Help | About | Change Password | Log off admin

Routing * Home

Home / Elements / Routing / Dial Patterns

Dial Pattern Details

General

* Pattern: 4

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority:

Emergency Type:

SIP Domain: devcon.com

Notes: Avaya CM

Originating Locations and Routing Policies

Add Remove

1 Item Refresh Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Lincroft	DevConnect Network	devcon14 Policy	0	<input type="checkbox"/>	devcon14	

Select : All, None

The following screen shows the dial pattern definition for Voice4net.

AVAYA Avaya Aura® System Manager 6.2

Last Logged on at November 30, 2012 2:41 PM
Help | About | Change Password | **Log off admin**

Routing * Home

Home / Elements / Routing / Dial Patterns

Routing

Domains

Locations

Adaptations

SIP Entities

Entity Links

Time Ranges

Routing Policies

Dial Patterns

Regular Expressions

Defaults

Dial Pattern Details

Commit Cancel

Help ?

General

* Pattern: 49000

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: devcon.com

Notes: Voice4net Extension

Originating Locations and Routing Policies

Add Remove

1 Item Refresh

Filter: Enable

<input type="checkbox"/>	Originating Location Name 1 ▲	Originating Location Notes	Routing Policy Name	Rank 2 ▲	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Lincroft	DevConnect Network	Voice4net Policy	0	<input type="checkbox"/>	Voice4net	

Select : All, None

6.7. Add Session Manager

To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Click **Add** (not shown), and fill in the fields as described below and shown in the following screen:

Under *Identity*:

- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

Under *Security Module*:

- **Network Mask:** Enter the network mask corresponding to the IP address of Session Manager
- **Default Gateway:** Enter the IP address of the default gateway for Session Manager

Avaya Aura® System Manager 6.2

Last Logged on at November 30, 2012 2:41 PM
Help | About | Change Password | Log off admin

Session Manager * Routing * Home

Home / Elements / Session Manager / Session Manager Administration

Edit Session Manager Commit Cancel

General | Security Module | NIC Bonding | Monitoring | CDR | Personal Profile Manager (PPM) - Connection Settings | Event Server |
Expand All | Collapse All

General

SIP Entity Name: lz-asm

Description:

*Management Access Point Host Name/IP: 192.168.100.233

*Direct Routing to Endpoints:

VMware Virtual Machine: ☐

Security Module

SIP Entity IP Address: 192.168.100.235

*Network Mask: 255.255.255.0

*Default Gateway: 192.168.100.1

*Call Control PHB: 46

*QOS Priority: 6

*Speed & Duplex:

VLAN ID:

The following screen shows the **Monitoring** section, which determines how frequently Session Manager sends SIP OPTIONS messages to Voice4net ePBX/EBS. Use default values for the remaining fields. Click **Commit** to add this Session Manager.

Monitoring ▾

Enable Monitoring ☒

*Proactive cycle time (secs)

900

*Reactive cycle time (secs)

120

*Number of Retries

1

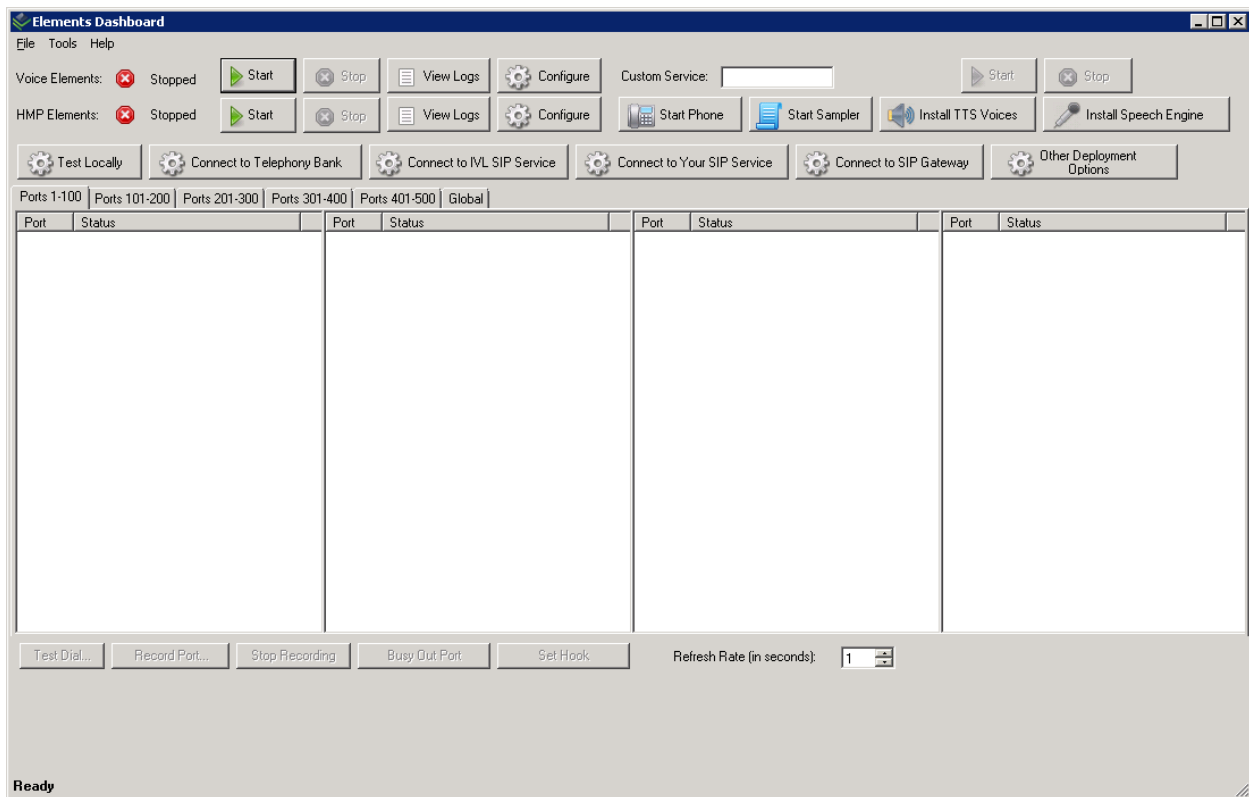
7. Configure Voice4net ePBX and EBS

This section provides the procedures for configuring Voice4net ePBX and EBS. The procedures include the following areas:

- Administer Site Configuration
- Administer SIP parameters
- Restart the ePBX for the SIP settings to take effect
- Specify the Voice4net ePBX custom application
- Configure Voice4net EBS broadcast event

7.1. Administer Site Configuration

To configure Voice4net ePBX, launch the **Elements Dashboard**, which is shown below.



The site configuration settings are configured in the C:\Program Files\Inventive Labs\Voice Elements Platform\Voice Elements Server\VoiceElementsServer.exe.config file. To access the file, click on the **Configure** button by **Voice Elements**. In the **applicationSettings** section, that the parameters in **bold** are set to either the Voice4net server IP address (e.g., 192.168.100.160), the Session Manager Signaling IP address (e.g., 192.168.100.235), the port (e.g., 5060), the name assigned to Voice4net ePBX (e.g., Voice4net), and the extension assigned to Voice4net ePBX (e.g., 49000). Voice4net uses UDP as the transport protocol. No additional configuration is required to set the transport protocol.

```
<applicationSettings>
  <VoiceElements.Properties.Settings>
    <setting name="PerfectCall" serializeAs="String">
      <value>17</value>
    </setting>
    <setting name="ServerConnectionString" serializeAs="String">
      <value>gtcp://192.168.100.160:54331</value>
    </setting>
    <setting name="ServerListeningPort" serializeAs="String">
      <value>54331</value>
    </setting>
    <setting name="ServerListeningIp" serializeAs="String">
      <value>192.168.100.160</value>
    </setting>
    <setting name="GlobalCall" serializeAs="String">
      <value>True</value>
    </setting>
    <setting name="IgnoreResources" serializeAs="String">
      <value />
    </setting>
    <setting name="ISDN" serializeAs="String">
      <value>True</value>
    </setting>
    <setting name="ExplicitBoards" serializeAs="String">
      <value />
    </setting>
    <setting name="T1DnisAniMask" serializeAs="String">
      <value />
    </setting>
    <setting name="CustomAuthenticationDll" serializeAs="String">
      <value />
    </setting>
    <setting name="CustomAuthenticationType" serializeAs="String">
      <value />
    </setting>
    <setting name="IpMediaServers" serializeAs="String">
      <value>
      </value>
    </setting>
    <setting name="Robodog" serializeAs="String">
      <value>False</value>
    </setting>
    <setting name="CallDetailConnectionString" serializeAs="String">
      <value />
    </setting>
    <setting name="GlobalCallProtocol" serializeAs="String">
      <value>DM3</value>
    </setting>
  </VoiceElements.Properties.Settings>
</applicationSettings>
```

```

    <setting name="SetChannelState" serializeAs="String">
      <value>False</value>
    </setting>
    <setting name="HmpElements" serializeAs="String">
      <value>True</value>
    </setting>
    <setting name="StaticVoiceResourceAssignment" serializeAs="String">
      <value>False</value>
    </setting>
    <setting name="HmpElementsPortCount" serializeAs="String">
      <value>0</value>
    </setting>
  </VoiceElements.Properties.Settings>
  <HmpElements.Properties.Settings>
    <setting name="HmpElementsUrl" serializeAs="String">
      <value>gtcp://192.168.100.160:55245</value>
    </setting>
  </HmpElements.Properties.Settings>
  <CTI32NetLib.Properties.Settings>
    <setting name="HmpDefaultDestinationHost" serializeAs="String">
      <value>192.168.100.235</value>
    </setting>
    <setting name="HmpDefaultSourceDisplayName" serializeAs="String">
      <value>Voice4net</value>
    </setting>
    <setting name="HmpDefaultSourceUser" serializeAs="String">
      <value>49000</value>
    </setting>
    <setting name="HmpDefaultSourceHost" serializeAs="String">
      <value>192.168.100.160</value>
    </setting>
    <setting name="HmpDefaultDestinationPort" serializeAs="String">
      <value>5060</value>
    </setting>
    <setting name="HmpDefaultSourcePort" serializeAs="String">
      <value>5060</value>
    </setting>
    <setting name="HmpDestinationHostOverrides" serializeAs="String">
      <value>
    </value>
    </setting>
    <setting name="TtsDefaultVoice" serializeAs="String">
      <value>
    </value>
    </setting>
  </CTI32NetLib.Properties.Settings>
</applicationSettings>

```

7.2. Administer SIP Settings

The SIP settings are configured in the C:\Program Files\Inventive Labs\Voice Elements Platform\HMP Elements Server\HMPElementsServer.exe.config file. To access the file, click on the **Configure** button by **HMP Elements** in the **Elements Dashboard**. In the **applicationSettings** section, the parameters in bold are set to either the Voice4net server IP address (e.g., 192.168.100.160), the Session Manager Signaling IP address (e.g., 192.168.100.235), and the port (e.g., 5060). Note that Voice4net uses UDP as the transport protocol. No additional configuration is required to set the transport protocol. Note that the **CodecOrder** parameter is set '0', which specifies G.711mu-law.

```
<applicationSettings>
  <HmpElements.Properties.Settings>
    <setting name="LoggingFileHistoryCount" serializeAs="String">
      <value>2</value>
    </setting>
    <setting name="LoggingFileSizeMB" serializeAs="String">
      <value>10</value>
    </setting>
    <setting name="MappedDrives" serializeAs="String">
      <value />
    </setting>
    <setting name="HmpElementsUrl" serializeAs="String">
      <value>gtcp://192.168.100.160:55245</value>
    </setting>
    <setting name="Robodog" serializeAs="String">
      <value>False</value>
    </setting>
    <setting name="RtpMediaIp" serializeAs="String">
      <value>192.168.100.160</value>
    </setting>
    <setting name="RtpMediaPortBase" serializeAs="String">
      <value>49152</value>
    </setting>
    <setting name="ExternalIp" serializeAs="String">
      <value>192.168.100.160</value>
    </setting>
    <setting name="AdditionalTranslations" serializeAs="String">
      <value />
    </setting>
    <setting name="AdditionalLocalTraffic" serializeAs="String">
      <value>192.168.1.</value>
    </setting>
    <setting name="HmpIp" serializeAs="String">
      <value>192.168.100.160</value>
    </setting>
    <setting name="HmpPort" serializeAs="String">
      <value>5060</value>
    </setting>
    <setting name="RtpMediaPortCount" serializeAs="String">
      <value>0</value>
    </setting>
    <setting name="AuthUsername" serializeAs="String">
      <value>
      </value>
    </setting>
```

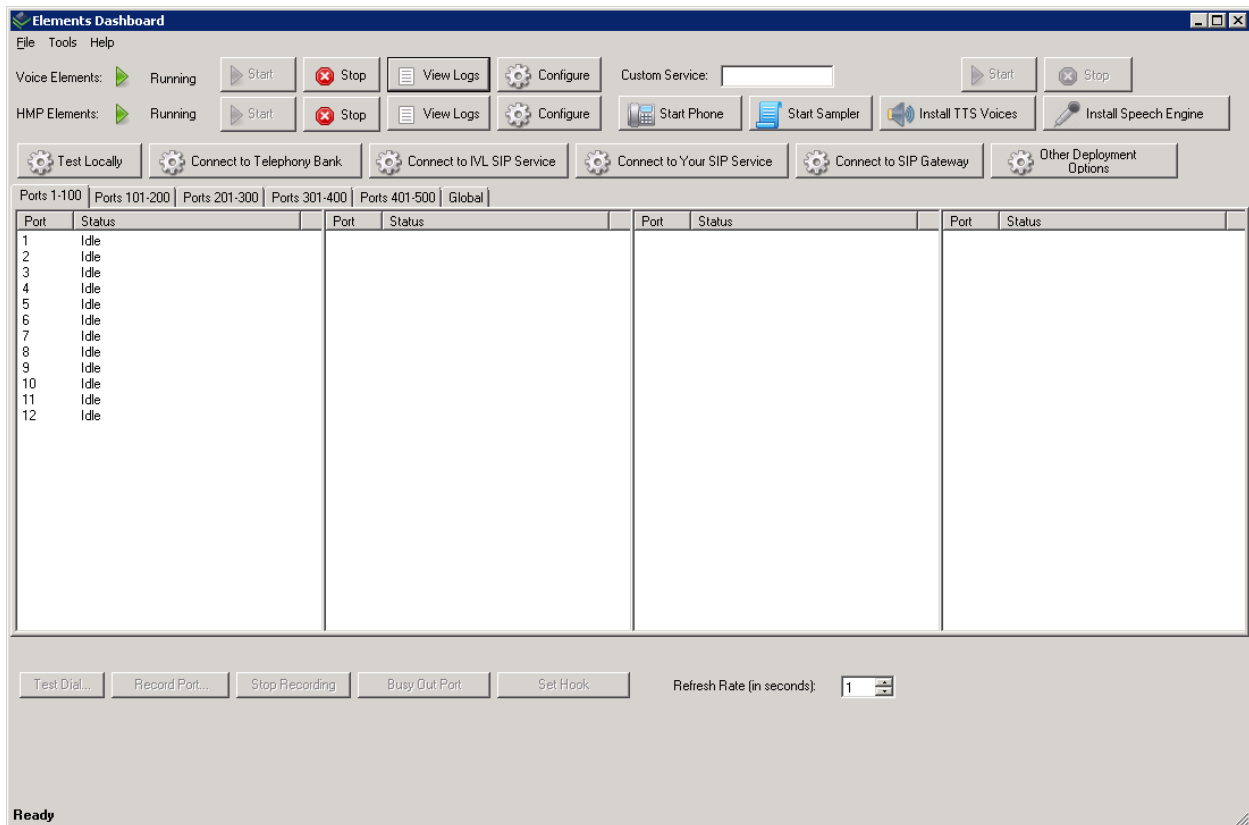
```

    <setting name="AuthPassword" serializeAs="String">
      <value>
      </value>
    </setting>
    <setting name="AuthUri" serializeAs="String">
      <value>
      </value>
    </setting>
    <setting name="InbandDtmf" serializeAs="String">
      <value>False</value>
    </setting>
    <setting name="AnalyzeCallLogLevel" serializeAs="String">
      <value>0</value>
    </setting>
    <setting name="AnalyzeCallRecordingPath" serializeAs="String">
      <value />
    </setting>
    <setting name="CodecOrder" serializeAs="String">
      <value>0</value>
    </setting>
    <setting name="PacketCaptureMode" serializeAs="String">
      <value>Legacy</value>
    </setting>
    <setting name="SpeechRecognitionDll" serializeAs="String">
      <value>
      </value>
    </setting>
    <setting name="SpeechRecognitionType" serializeAs="String">
      <value>HmpElements.Server.MicrosoftSpeech</value>
    </setting>
    <setting name="SpeechRecognitionNumberOfPorts" serializeAs="String">
      <value>0</value>
    </setting>
    <setting name="SpeechRecognitionLicenseType" serializeAs="String">
      <value>en-US</value>
    </setting>
    <setting name="SpeechEngineIpAddress" serializeAs="String">
      <value>127.0.0.1</value>
    </setting>
  </HmpElements.Properties.Settings>
</applicationSettings>
</configuration>

```

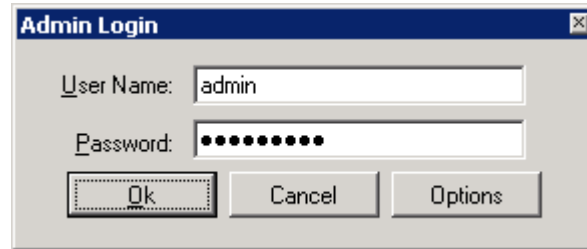
7.3. Restart Voice4net ePBX

After the SIP settings have been configured, restart ePBX for the SIP settings to take effect. Close the **Elements Dashboard** and then re-launch it. To restart ePBX, click the **Start** buttons by **Voice Elements** and **HMP Elements**. When ePBX has been started, the **Elements Dashboard** will appear as shown below along with the available ports. The number of ports is determined by the installed license. In this example, 12 SIP trunks are supported.



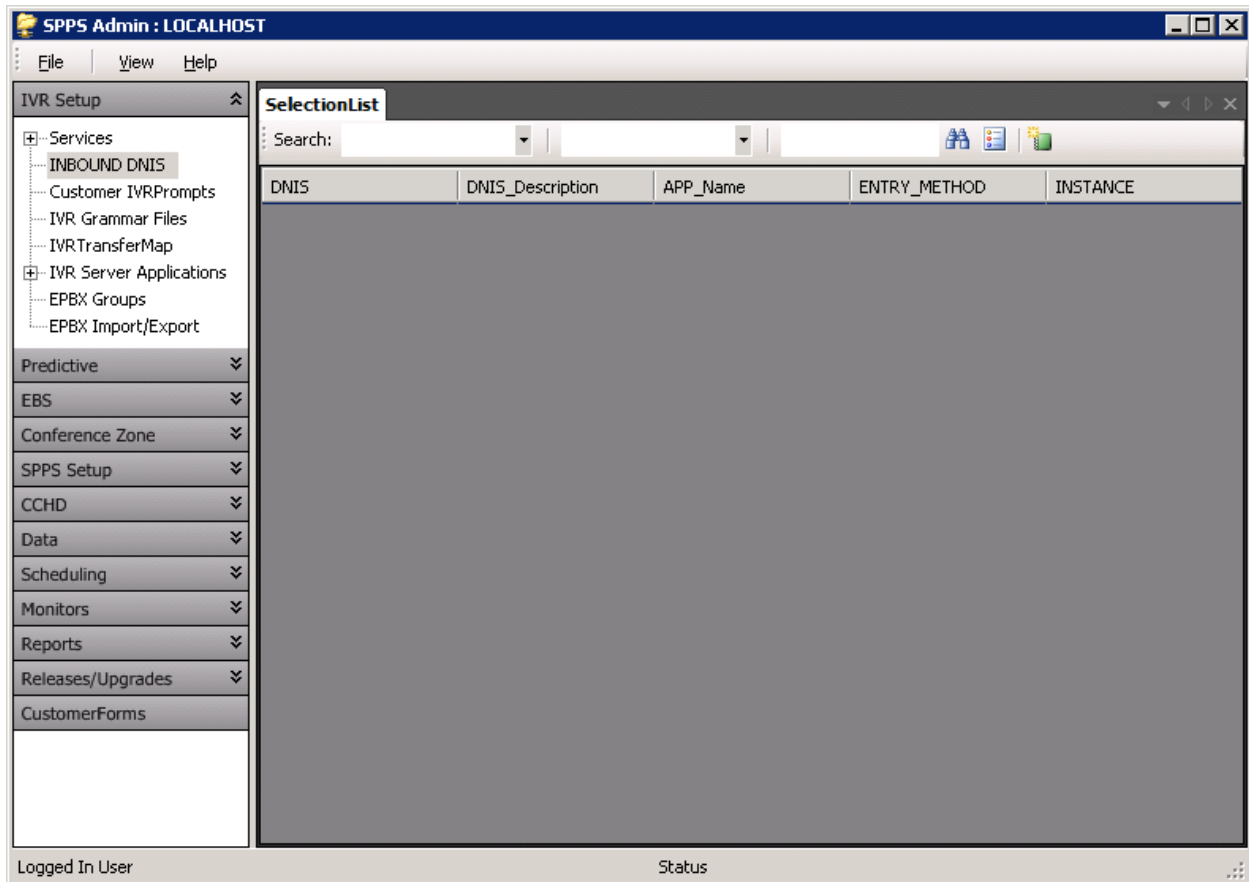
7.4. Specify the Voice4net ePBX Custom Application

To specify the custom application that ePBX will run based on the DNIS, launch the **SPPS Admin** application and login with the appropriate credentials as shown below.



The Admin Login dialog box has a title bar with 'Admin Login' and a close button. It contains two text input fields: 'User Name' with the value 'admin' and 'Password' with masked characters. Below the fields are three buttons: 'Ok', 'Cancel', and 'Options'.

Once logged in, the **SPPS Admin** window will be displayed as shown below. Expand **IVR Setup** in the left pane, and then click on **INBOUND DNIS**. Next, click on the **New Item** icon to map a DNIS to a custom application.



In the **IVR DNIS Definition** window, specify the **DNIS** and the appropriate custom application in the **IVR App**, **AfterHours App**, and **Holiday App** fields. In this example, the **DNIS** field is set to the wildcard “*”, which maps to any DNIS, and the customer application is called *Avaya Cert App*. The custom application provides the user with the ability to transfer to the specified number.

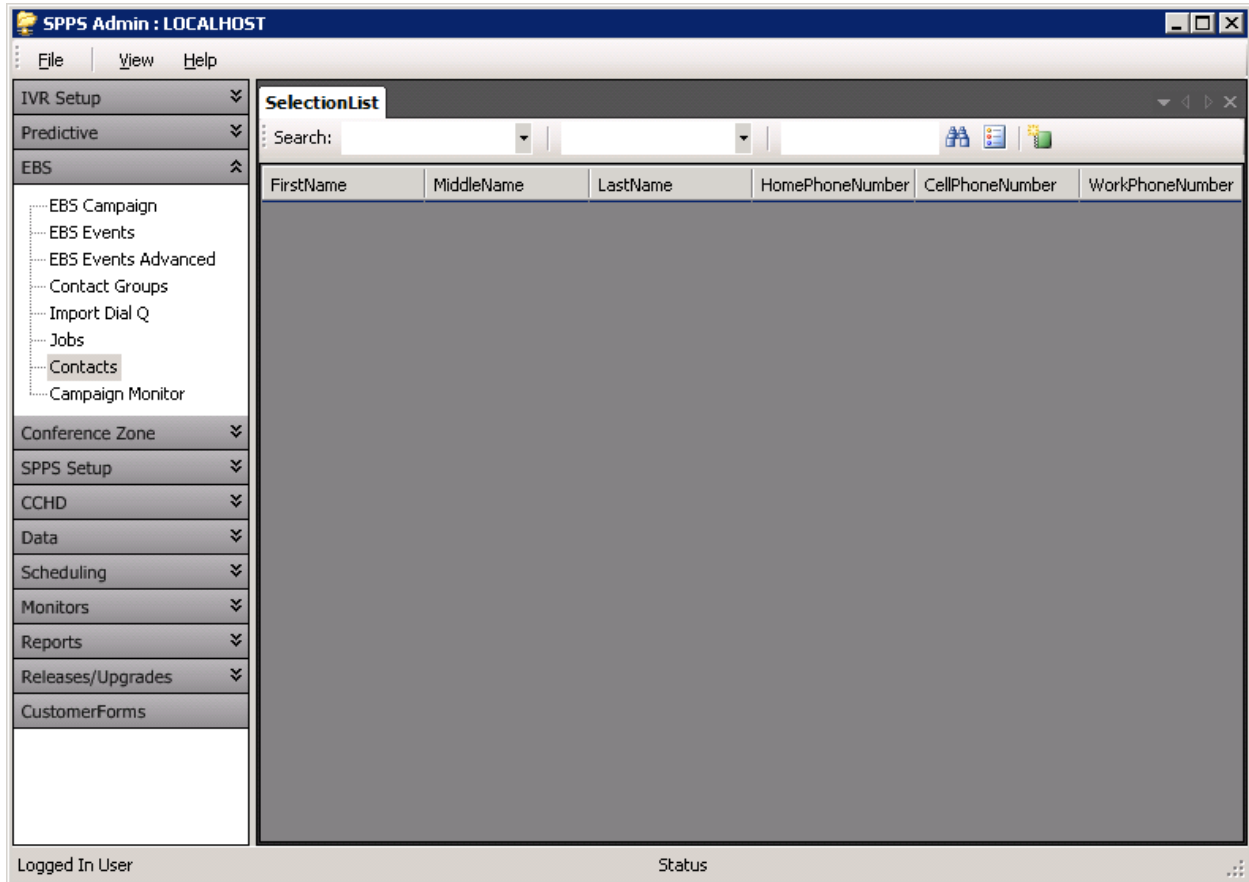
The screenshot shows the 'IVR DNIS Definition' window with the 'General' tab selected. The fields are configured as follows:

Field	Value
DNIS:	*
Description:	Register All
Customer Group:	[Empty]
IVR App:	Avaya Cert App
AfterHours App:	Avaya Cert App
Holiday App:	Avaya Cert App
Instance:	LOCALHOST

Buttons at the bottom: Ok, Cancel, Apply, Help.

7.5. Launch a Voice4net EBS Broadcast Event

Voice4net EBS can send a broadcast message to specified users. This configuration is performed via the **SPPS Admin** application under the **EBS** option in the left pane as shown below. This requires configuration of Contact Groups, EBS Events to schedule the events, and an EBS Campaign.



The detailed configuration of EBS events is outside the scope of these Application Notes, but the EBS Campaign used for the compliance test is shown below. The **Campaign** window shown below brings together the EBS contacts, event, and schedule. For example, in the sample campaign below, the **Service** field was set to *IVR Main* and a descriptive name was provided for the **CampaignName** and **Campaign Definition** fields. The remaining campaign fields specify the schedule to run the campaign, the contact group, and the event type. The EBS event can also be launched immediately by clicking on the **Submit Group** button.

The screenshot shows the 'Campaign' configuration window. The 'Settings' section includes the following fields and controls:

- Service:** A dropdown menu set to 'IVR Main'.
- Max Concurrent Events:** A dropdown menu set to '4'.
- Licensed:** A text field containing '4'.
- CampaignName:** A text field containing 'TestCampaign'.
- Start Option:** A dropdown menu set to 'StartOver'.
- Campaign Definition:** A text field containing 'TestCampaign'.
- Restrict Dialing:** An unchecked checkbox.
- Agents Available:** A section with three dropdown menus: 'If there are 0', 'agents available for 0 seconds', and 'allow 0 calls to be made from the dialer'.
- HuntGroups:** A list box showing a single entry 'HuntGroups'.
- Run Schedule:** A dropdown menu set to 'BusHours', with 'AddNew...' and 'Edit...' buttons.
- Contact Group:** A dropdown menu set to 'TestGroup', with 'AddNew...' and 'Edit...' buttons.
- Event Types:** A dropdown menu set to 'EBS'.
- Events:** A dropdown menu set to 'TestEvent', with 'AddNew...' and 'Edit...' buttons.
- Submit Group:** A button located to the right of the 'Contact Group' and 'Event Types' fields.

At the bottom of the window are four buttons: 'Ok', 'Cancel', 'Apply', and 'Help'.

8. Verification Steps

This section provides the tests that can be performed to verify proper configuration of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and Voice4net ePBX/EBS.

1. From **System Manager**, navigate to **Elements→Session Manager→System Status→SIP Entity Monitoring**. Next, click on the *Voice4net* SIP entity under the **All Monitored SIP Entities** section to display the page below. Verify that the **Conn. Status** and **Link Status** fields are *Up*.

The screenshot shows the Avaya Aura® System Manager 6.2 web interface. The top navigation bar includes the Avaya logo, the title 'Avaya Aura® System Manager 6.2', and a user status bar indicating 'Last Logged on at December 3, 2012 11:22 AM' with links for 'Help', 'About', 'Change Password', and 'Log off admin'. The left sidebar contains a menu with 'Session Manager' selected, showing sub-items like 'Dashboard', 'Session Manager Administration', 'Communication Profile Editor', 'Network Configuration', 'Device and Location Configuration', 'Application Configuration', and 'System Status'. The main content area displays the 'SIP Entity, Entity Link Connection Status' page. It includes a breadcrumb trail 'Home / Elements / Session Manager / System Status / SIP Entity Monitoring' and a 'Help ?' link. The page title is 'SIP Entity, Entity Link Connection Status'. Below the title, it states 'This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.' A section titled 'All Entity Links to SIP Entity: Voice4net' contains a 'Summary View' button. Below this, a table shows the connection status for one item. The table has columns: 'Details', 'Session Manager Name', 'SIP Entity Resolved IP', 'Port', 'Proto.', 'Conn. Status', 'Reason Code', and 'Link Status'. The data row shows 'lz-asm' as the Session Manager Name, '192.168.100.160' as the SIP Entity Resolved IP, '5060' as the Port, 'UDP' as the Protocol, 'Up' as the Connection Status, '200 OK' as the Reason Code, and 'Up' as the Link Status. A 'Filter: Enable' link is also present.

Details	Session Manager Name	SIP Entity Resolved IP	Port	Proto.	Conn. Status	Reason Code	Link Status
Show	lz-asm	192.168.100.160	5060	UDP	Up	200 OK	Up

2. Place a call to Voice4net ePBX by dialing the IVR extension and verify that the system greeting is heard.
3. From the **Campaign** window shown above, click on the **Submit Group** button to initiate the broadcast event. Verify that the specified contacts received the broadcast event. Alternatively, an outbound call can also be initiated directly from the **Elements Dashboard** by selecting a channel, clicking on **Test Dial** button, and then specifying the number to dial. The station associated with the dialed number should be called and ePBX should run the specified custom application.

9. Conclusion

These Application Notes describe the configuration steps required to integrate Voice4net ePBX/EBS with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks. The Voice4net ePBX IVR platform and the EBS broadcasting module were verified. All feature and serviceability test cases were completed and passed.

10. Additional References

This section references the product documentation relevant to these Application Notes.

- [1] *Administering Avaya Aura® Communication Manager*, Release 6.2, Issue 7.0, December 2012, Document Number 03-300509.
- [2] *Implementing Avaya Aura® Session Manager*, Release 6.2, Issue 1.0, June 2012.

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