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.

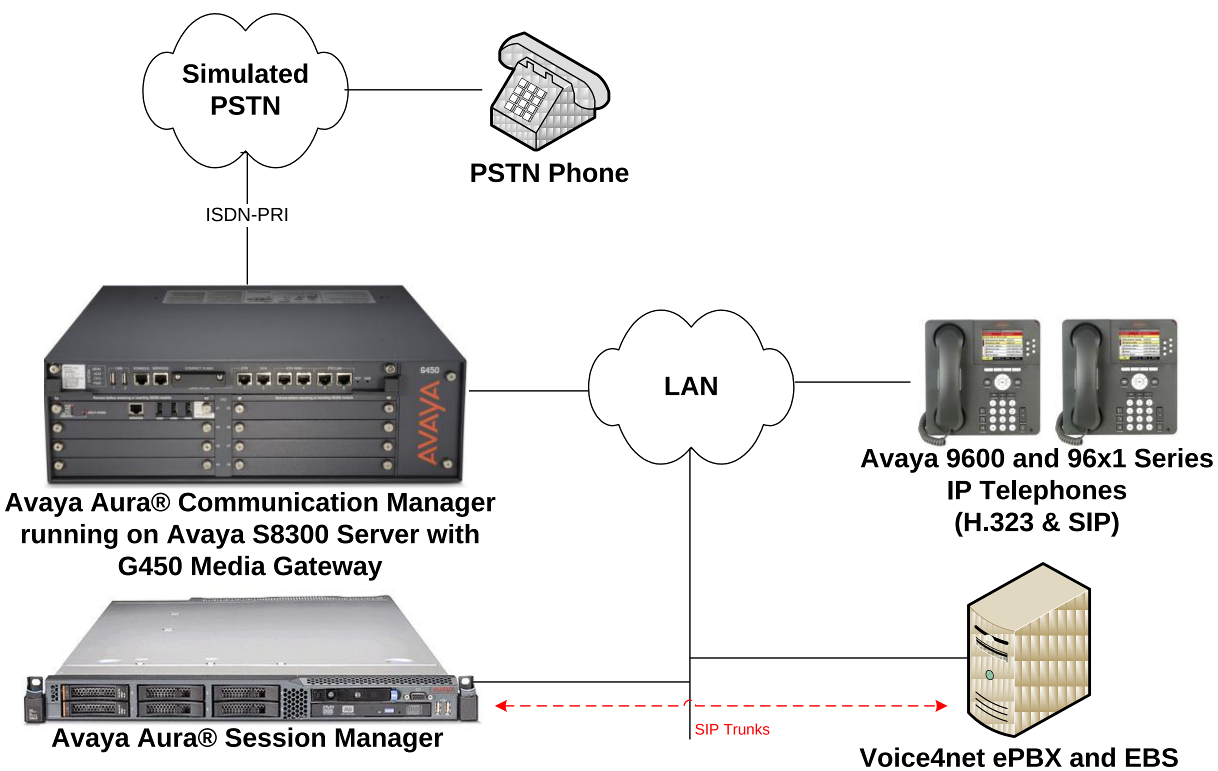


Figure 1: Voice4net ePBX/EBS with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunks

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      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
```

5.4. 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        Silence      Frames   Packet
    Codec        Suppression  Per Pkt  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
                                                Bypass If IP Threshold Exceeded? n
Incoming Dialog Loopbacks: eliminate                     RFC 3389 Comfort Noise? n
         DTMF over IP: rtp-payload               Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                        IP Audio Hairpinning? n
         Enable Layer 3 Test? y                      Initial IP-IP Direct Media? n
H.323 Station Outgoing 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 Ext              Trk         Private           Total
Len Code             Grp(s)      Prefix            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.

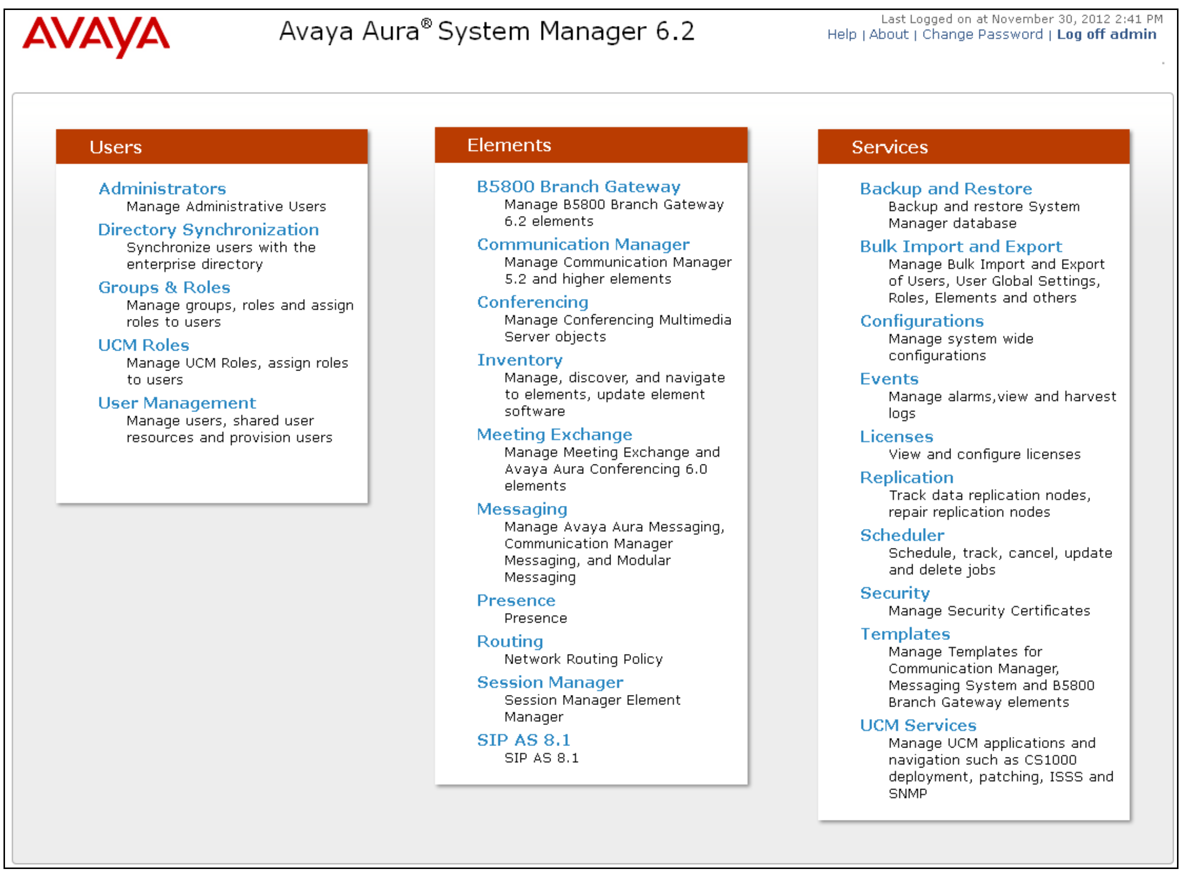
```
change route-pattern 60                                         Page   1 of   3
                    Pattern Number: 60  Pattern Name: To lz-asm
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          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.



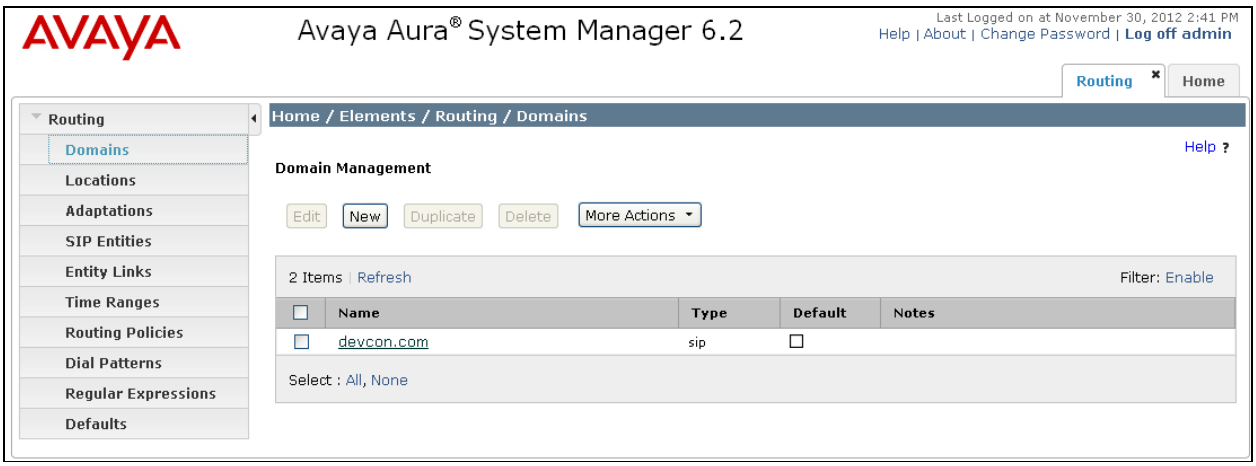
6.1. Specify SIP Domain

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.



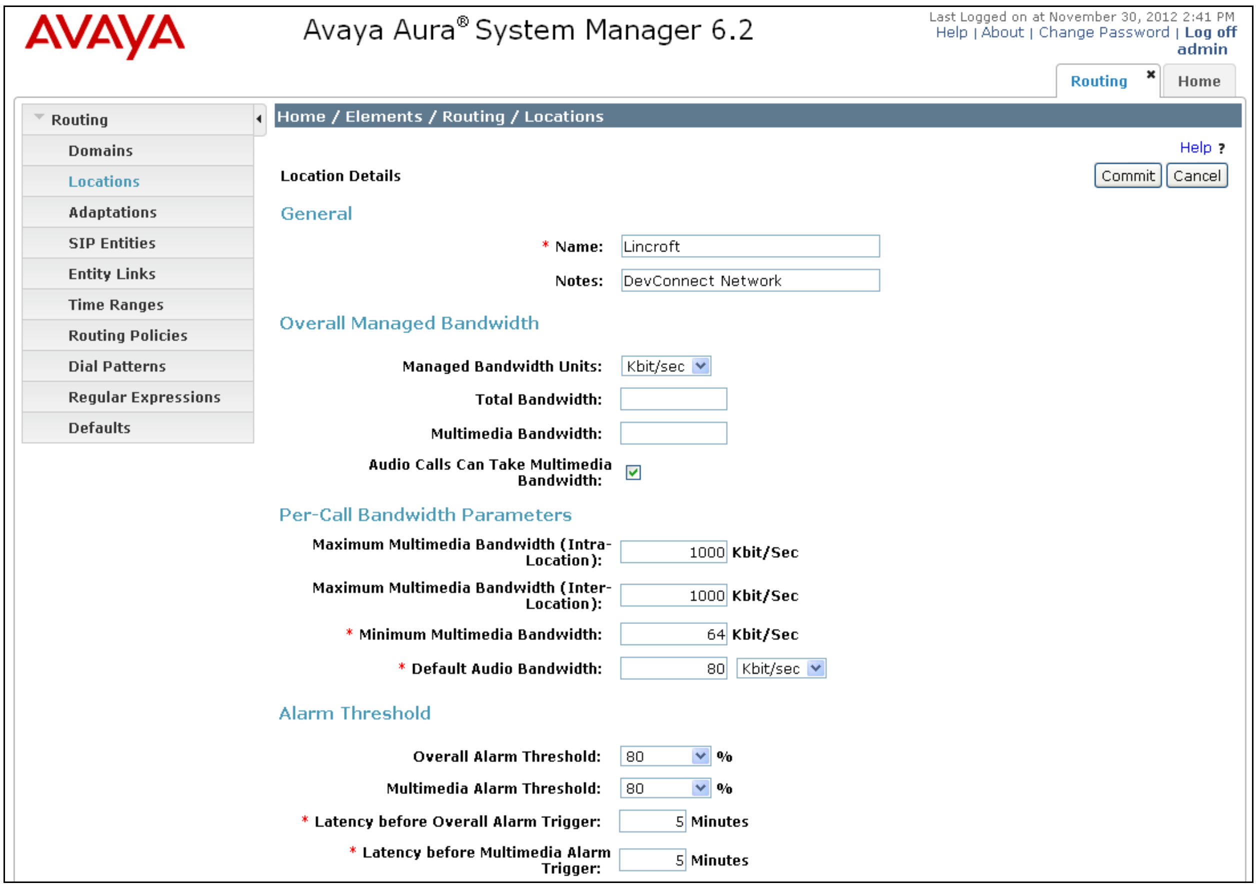
6.2. Add Locations

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).

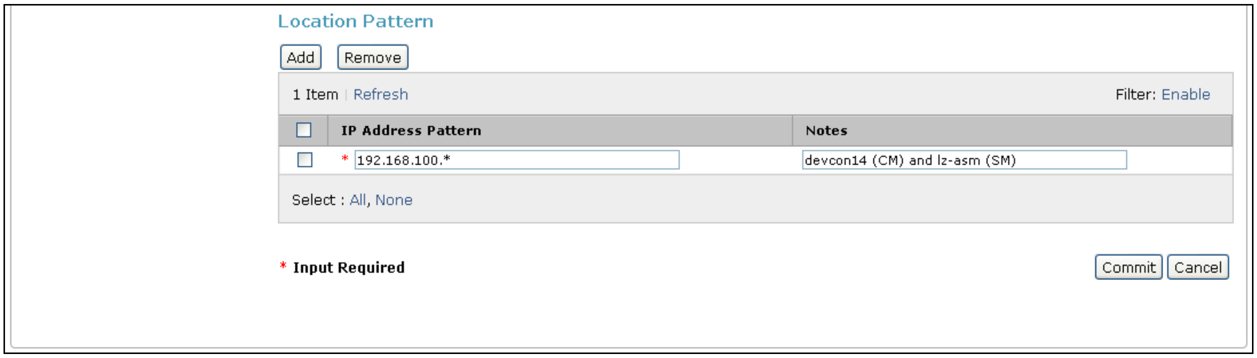
The screen below shows addition of the *Lincroft* location, which includes the Communication Manager and Session Manager.



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.



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:

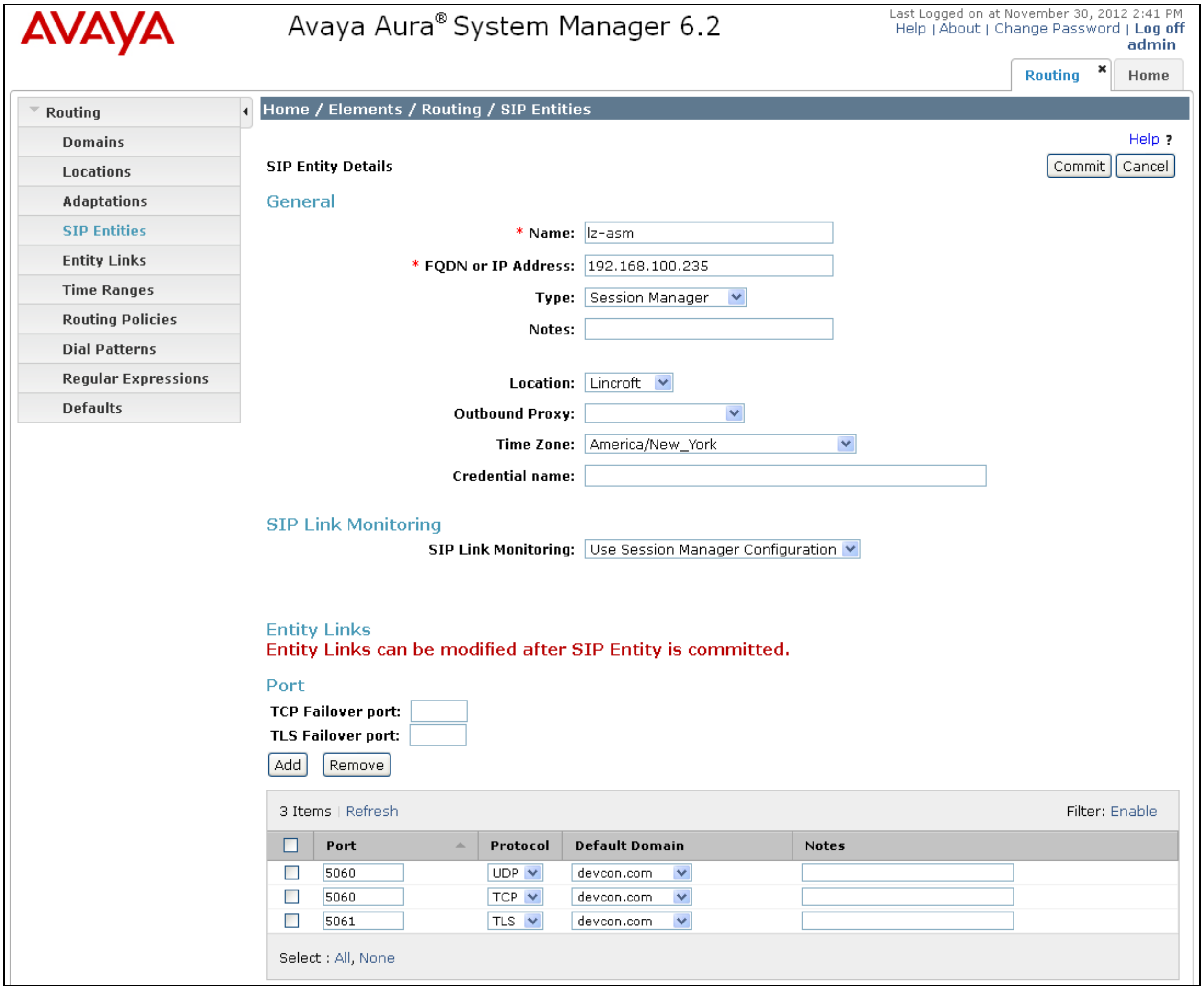
Under *General*:

- **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.


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
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
Commit
Cancel
General
* Name: lz-asm
* FQDN or IP Address: 192.168.100.235
Type: Session Manager
Notes:
Location: Lincroft
Outbound Proxy:
Time Zone: America/New_York
Credential name:
SIP Link Monitoring
SIP Link Monitoring: Use Session Manager Configuration
Entity Links
Entity Links can be modified after SIP Entity is committed.
Port
TCP Failover port:
TLS Failover port:
Add
Remove
3 Items | Refresh
Filter: Enable
Port
Protocol
Default Domain
Notes
5060 UDP devcon.com
5060 TCP devcon.com
5061 TLS devcon.com
Select : All, None

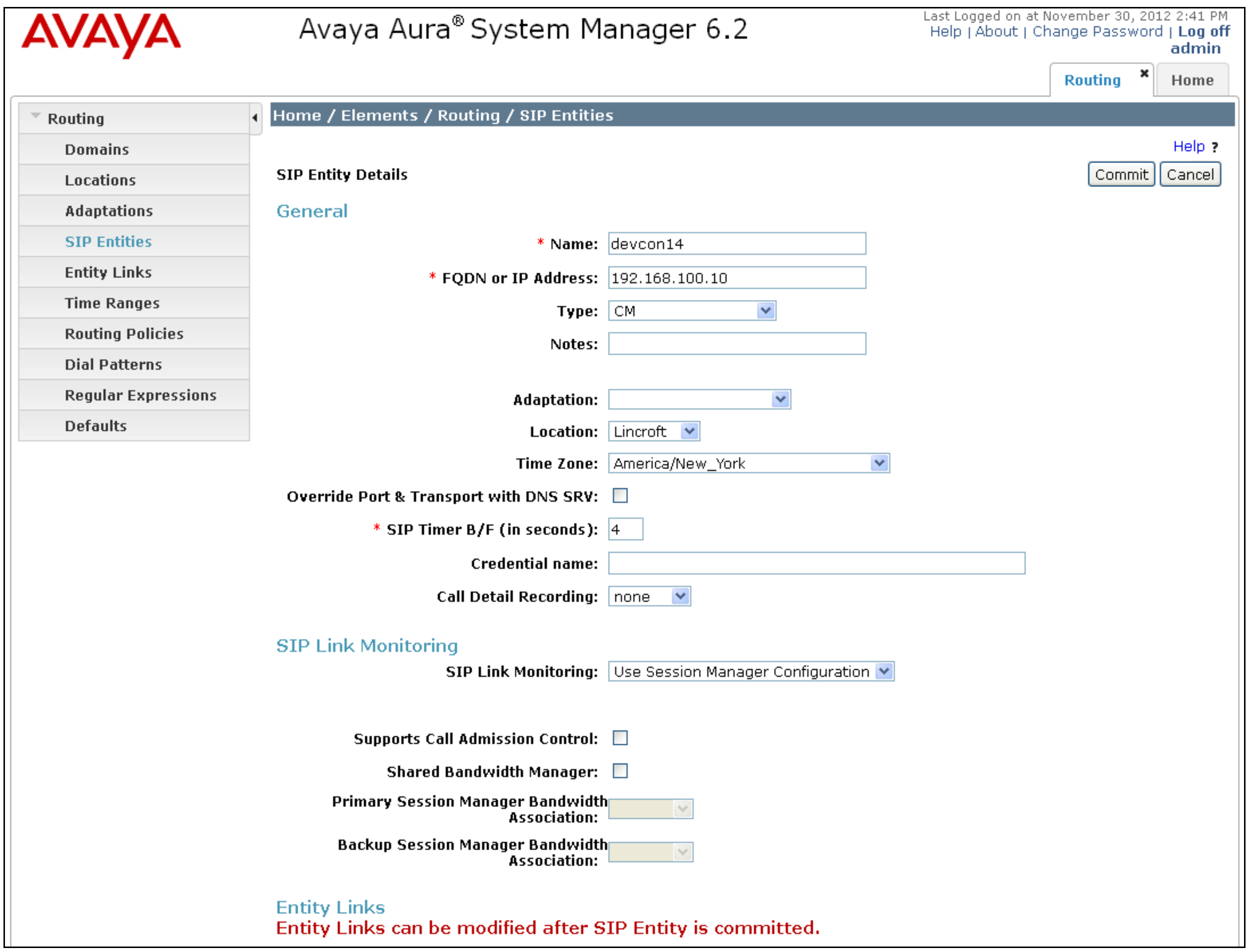
6.3.2. Communication Manager

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.

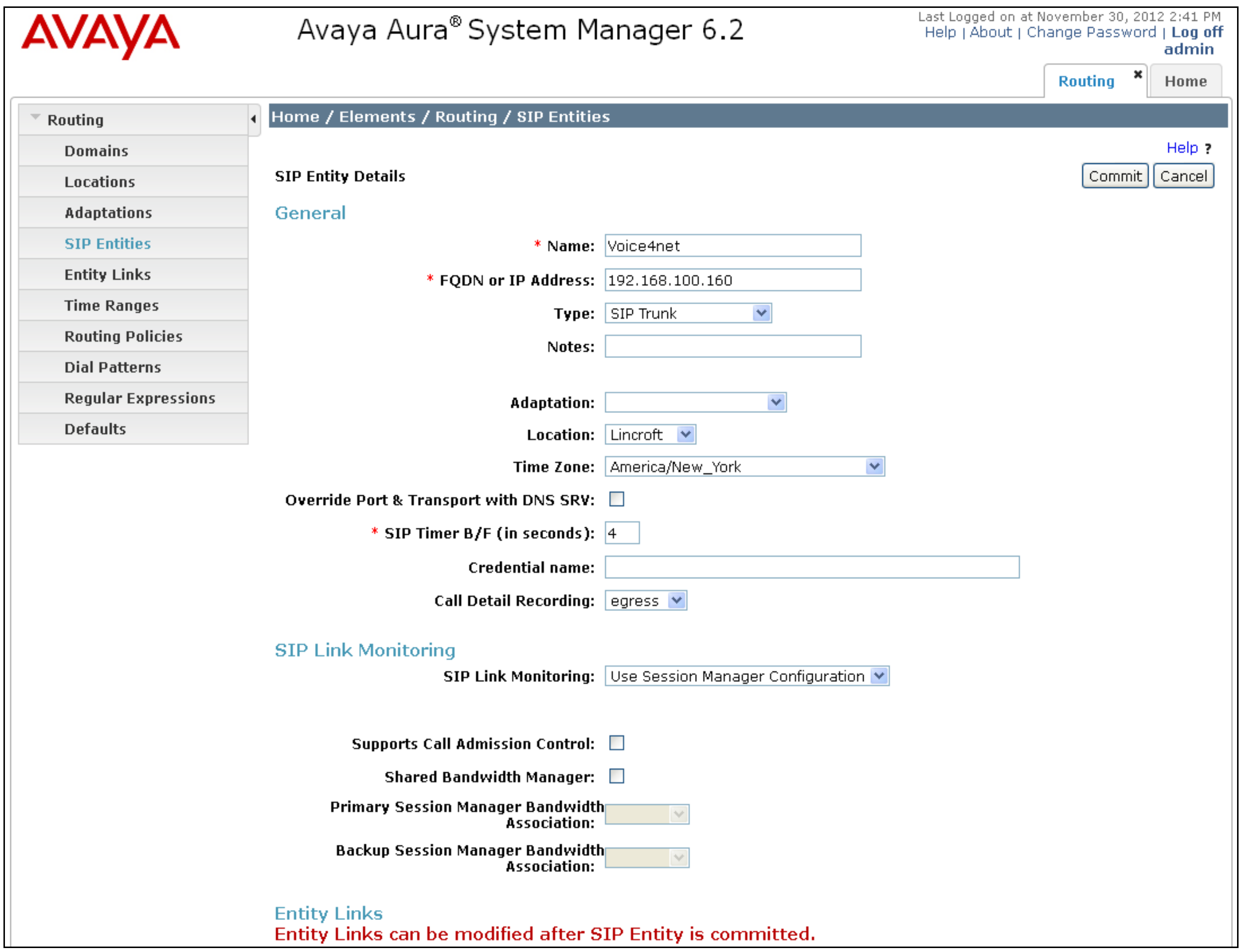


6.3.3. Voice4net ePBX/EBS

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.



6.4. Add Entity Links

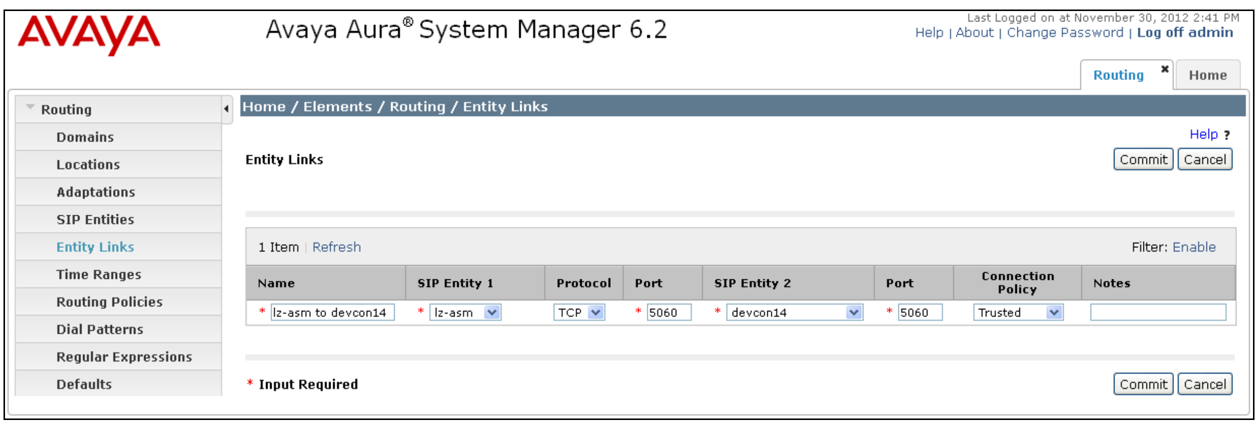
In the sample configuration, two Entity links were added, one for Communication Manager and another one for Voice4net ePBX/EBS.

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.

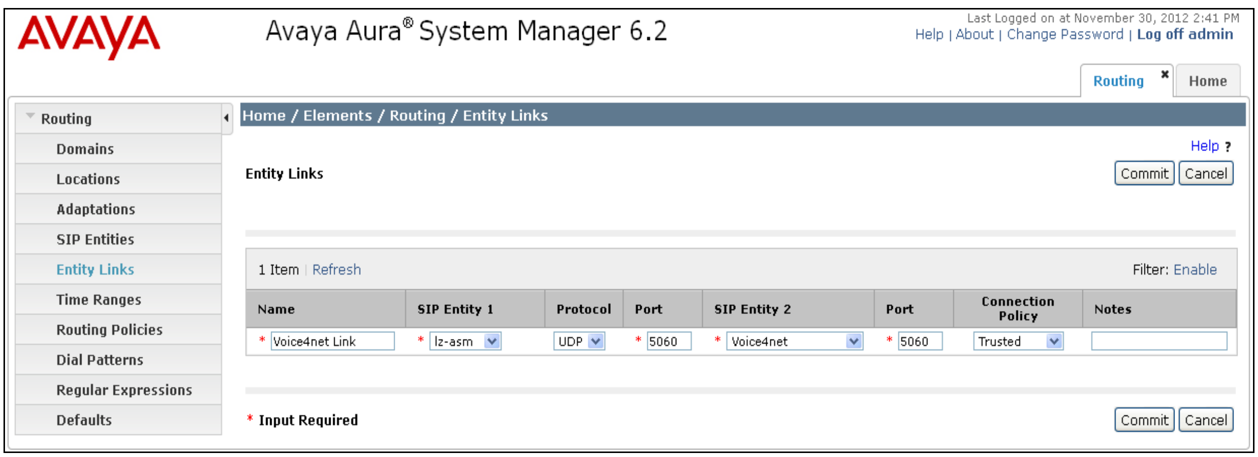


6.4.2. Voice4net ePBX/EBS

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.



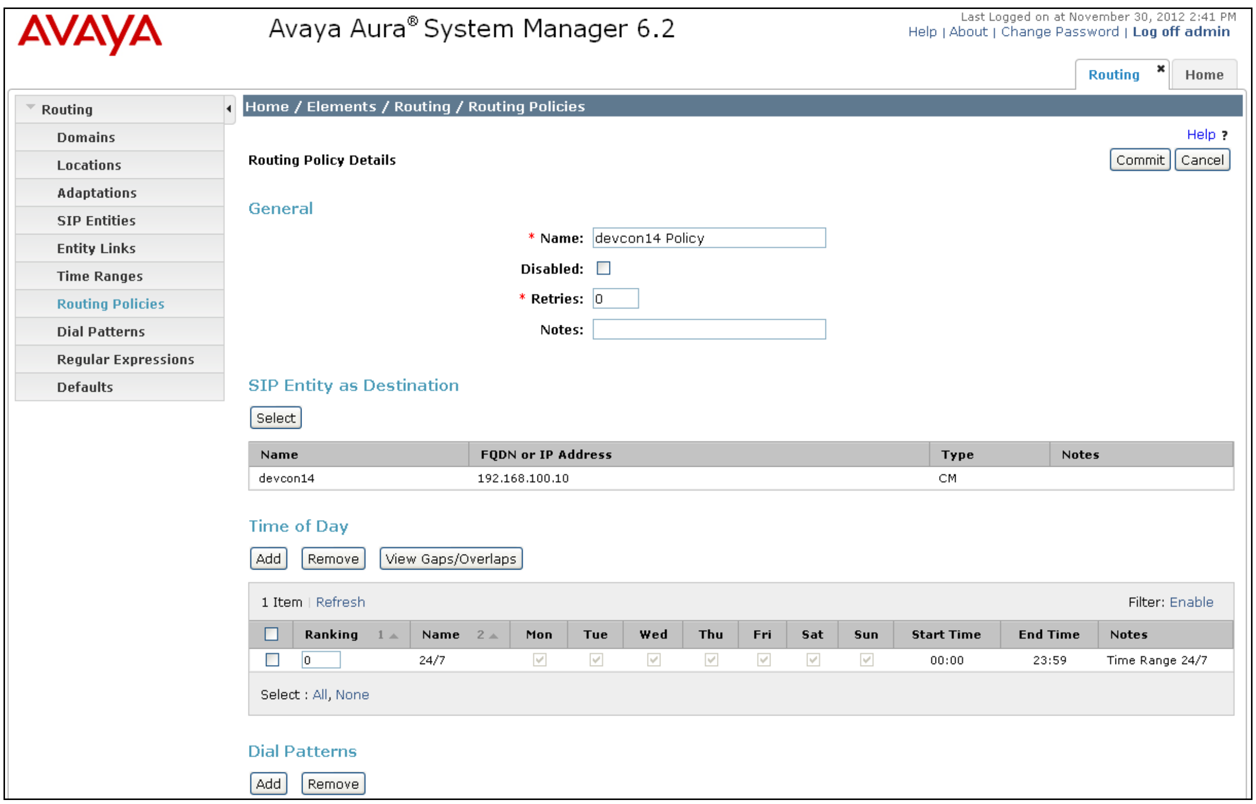
6.5. Add Routing Policies

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:

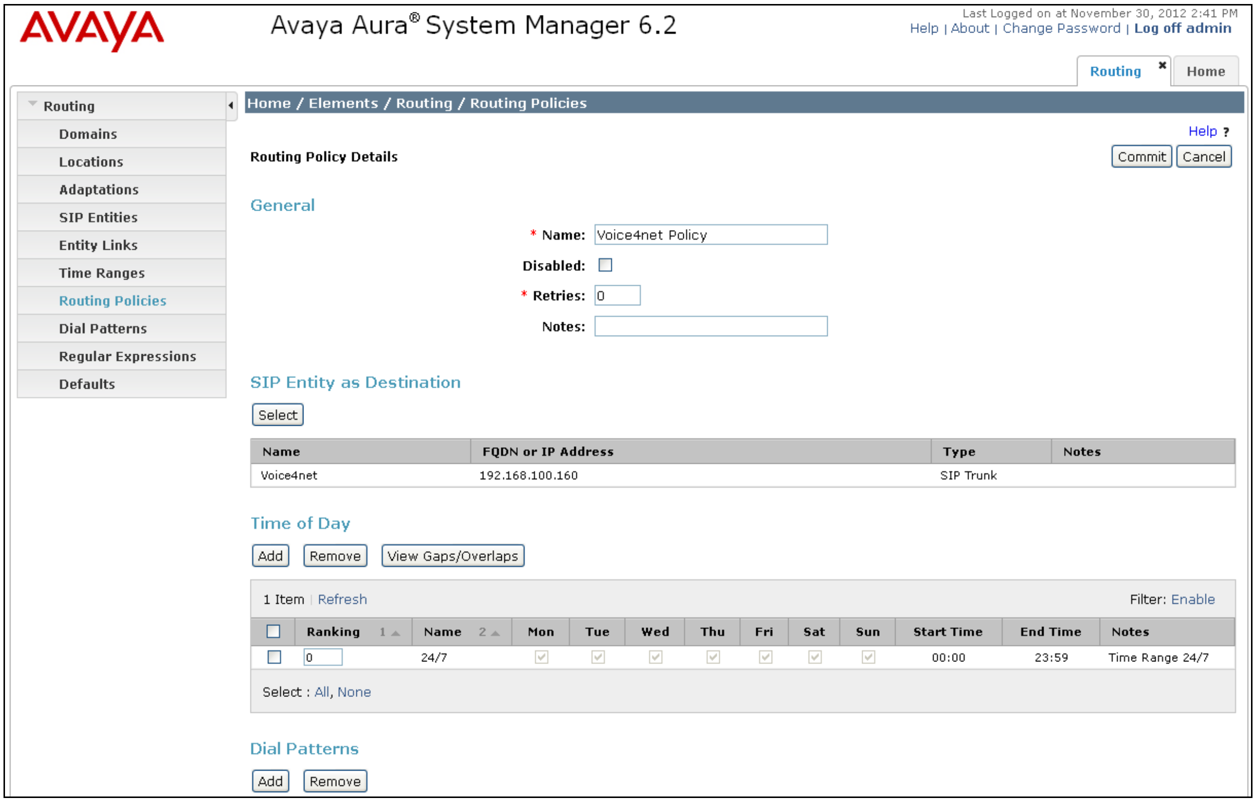
Under *General*:
Enter a descriptive name in **Name**.

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 following screen shows the Routing Policy for Voice4net.



6.6. Add Dial Patterns

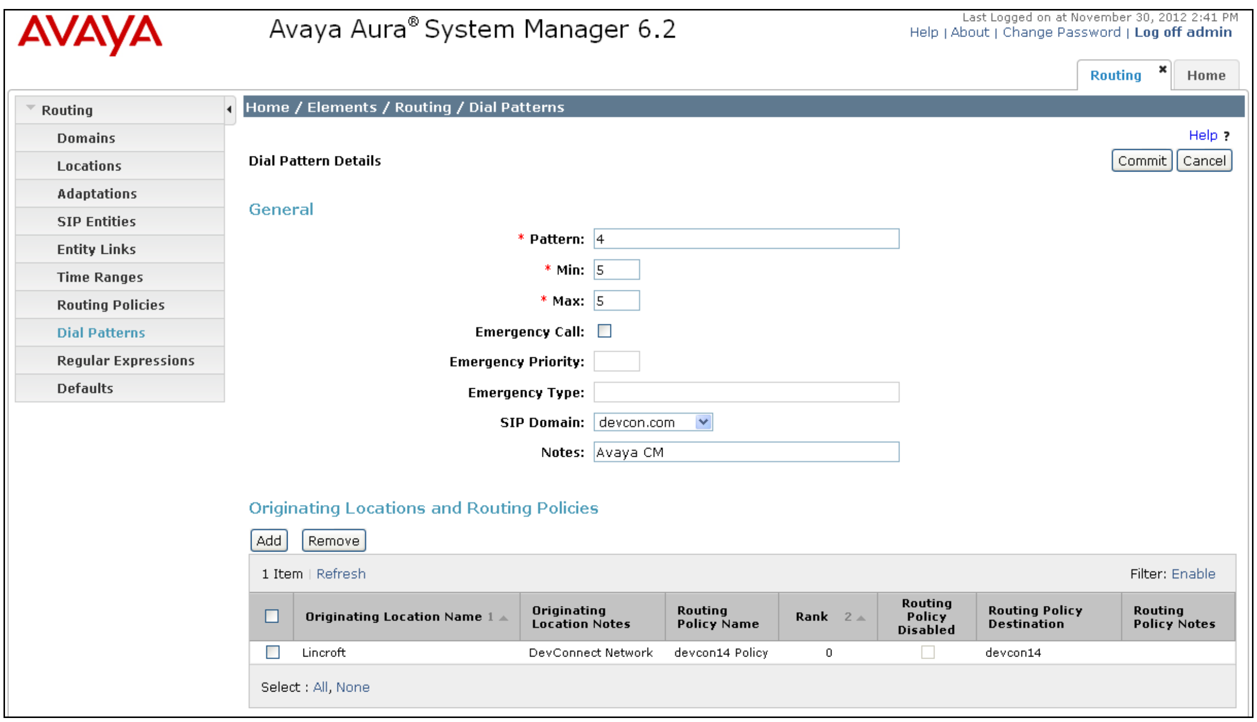
Dial patterns must be defined that will direct calls to the appropriate SIP Entity. In the sample configuration, 5-digit extensions beginning with “4” reside on Communication Manager, extension “49000” is assigned to Voice4net ePBX/EBS. To add a dial pattern, select **Dial Patterns** on the left and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

Under *General*:

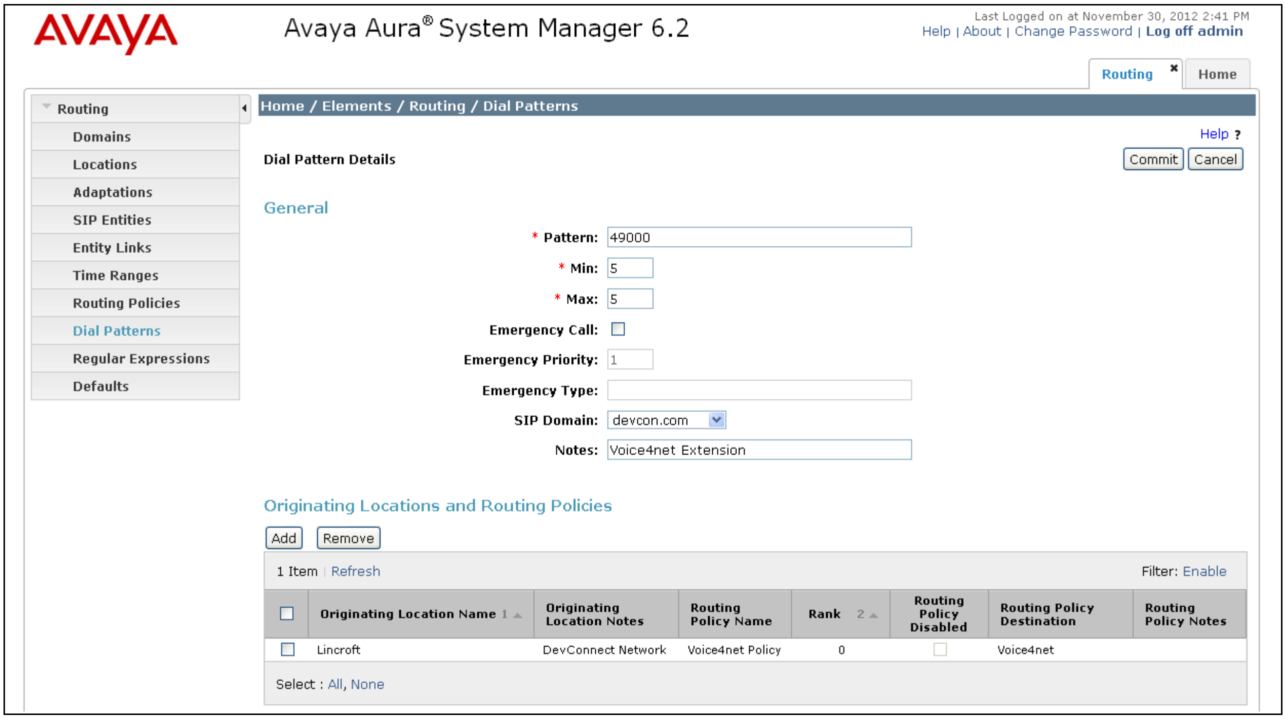
- **Pattern:** Dialed number or prefix.
- **Min** Minimum length of dialed number.
- **Max** Maximum length of dialed number.
- **SIP Domain** SIP domain of dial pattern.
- **Notes** Comment on purpose of dial pattern.

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.



The following screen shows the dial pattern definition for Voice4net.



6.7. Add Session Manager

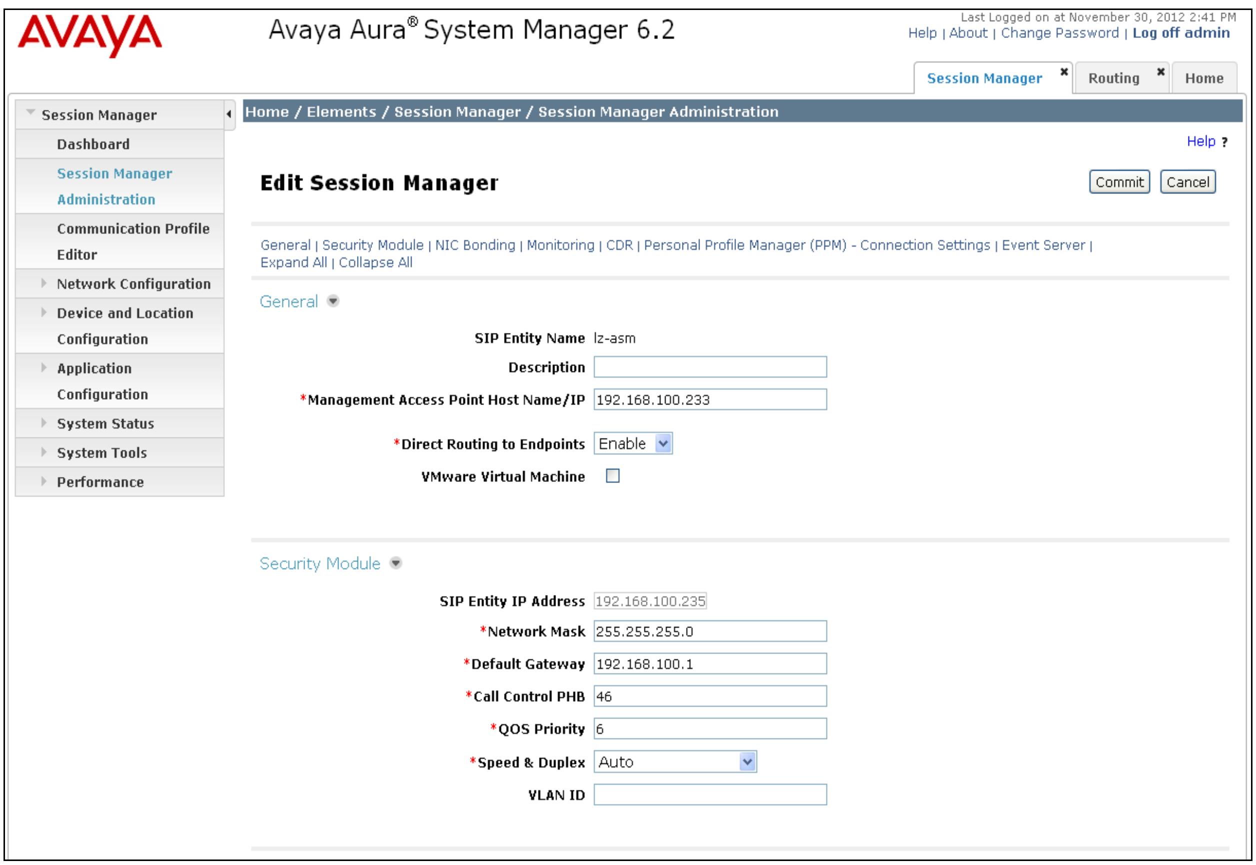
To complete the configuration, adding the Session Manager will provide the linkage between System Manager and Session Manager. Expand the **Session Manager** menu on the left and select **Session Manager Administration**. Click **Add** (not shown), and fill in the fields as described below and shown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

Under *Identity*:

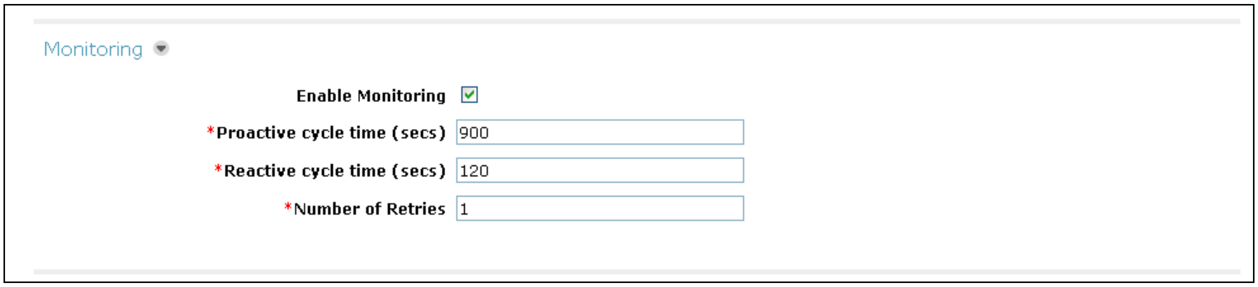
- **SIP Entity Name:** Select the name of the SIP Entity added for Session Manager
- **Description:** Descriptive comment (optional)
- **Management Access Point Host Name/IP:** Enter the IP address of the Session Manager management interface.

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



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.



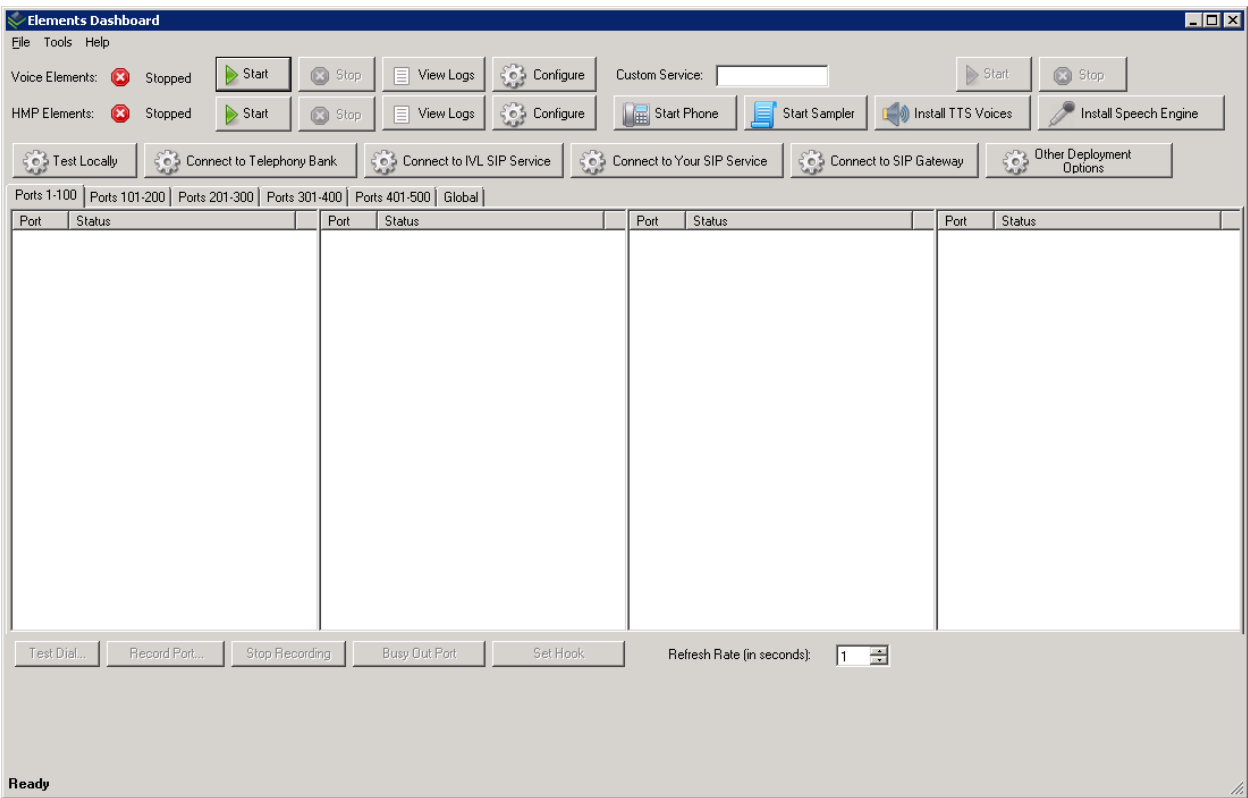
7. Configure Voice4net ePBX and EBS

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>
```

```
    <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

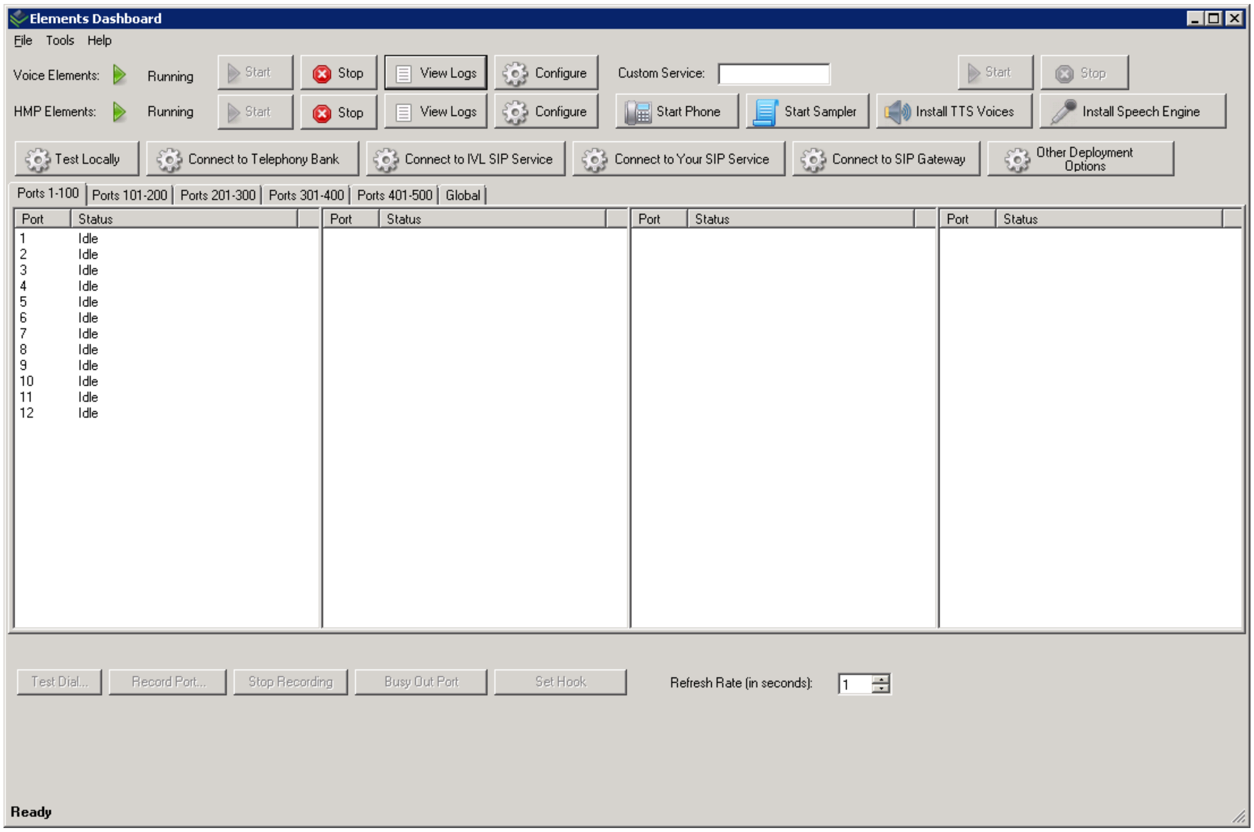
The SIP settings are configured in the `C:\Program Files\Inventive Labs\Voice Elements Platform\HMP Elements Server\HMPElementsServer.exe.config` file. To access the file, click on the **Configure** button by **HMP Elements** in the **Elements Dashboard**. In the **applicationSettings** section, the 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>
```

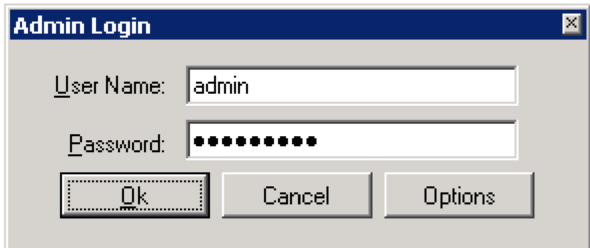
7.3. Restart Voice4net ePBX

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.

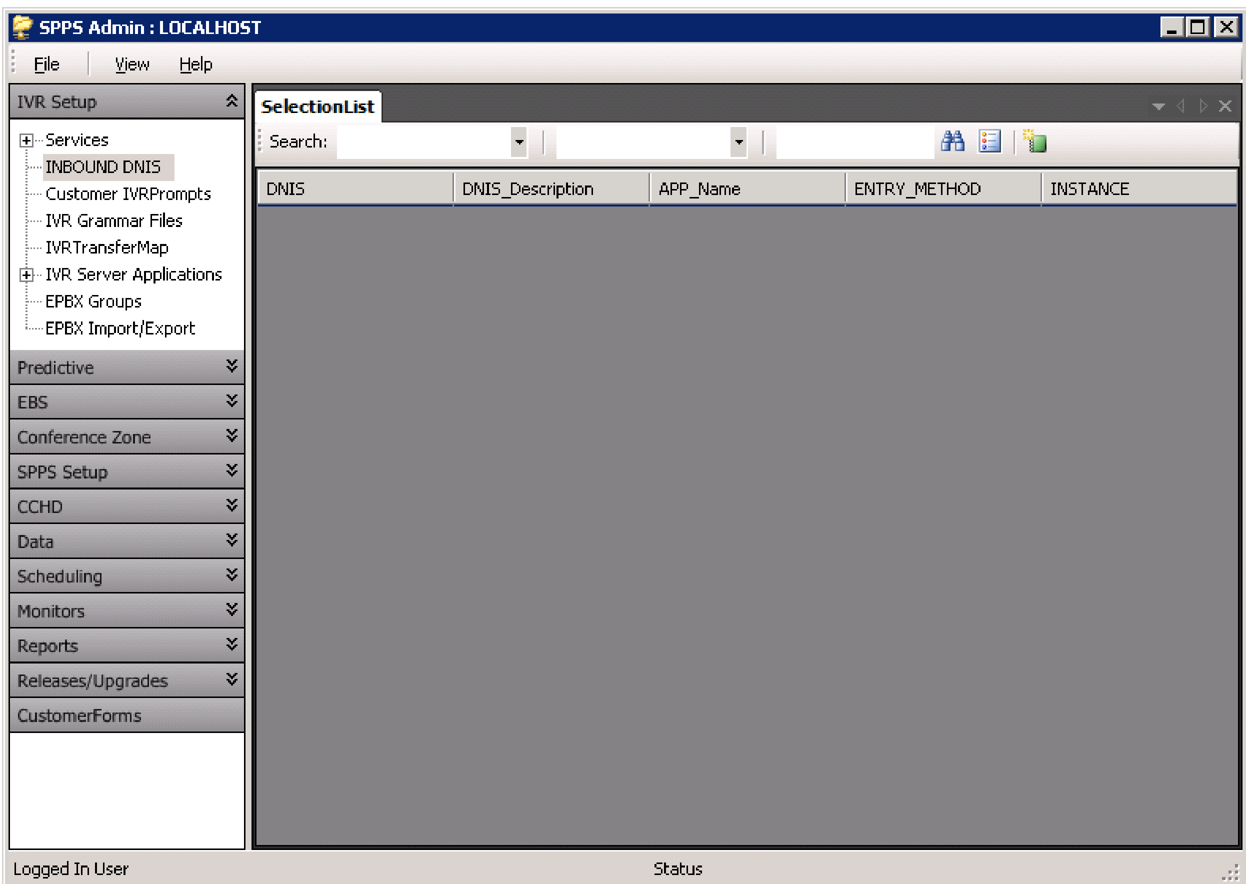


7.4. Specify the Voice4net ePBX Custom Application

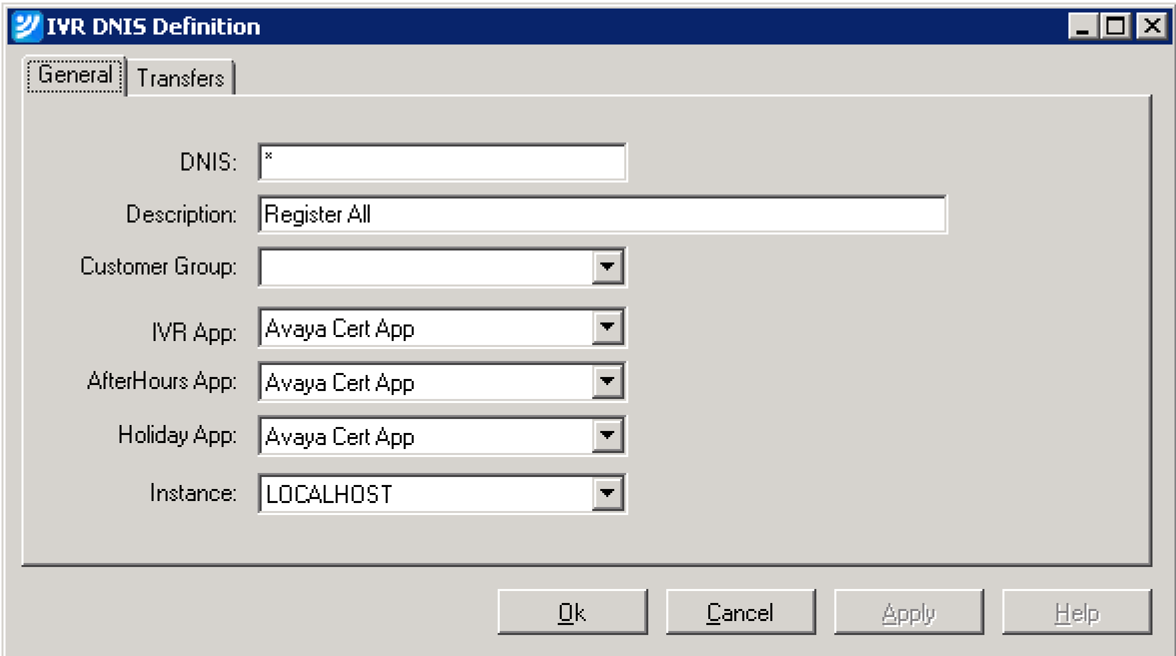
To specify the custom application that ePBX will run based on the DNIS, launch the **SPPS Admin** application and login with the appropriate credentials as shown below.



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.

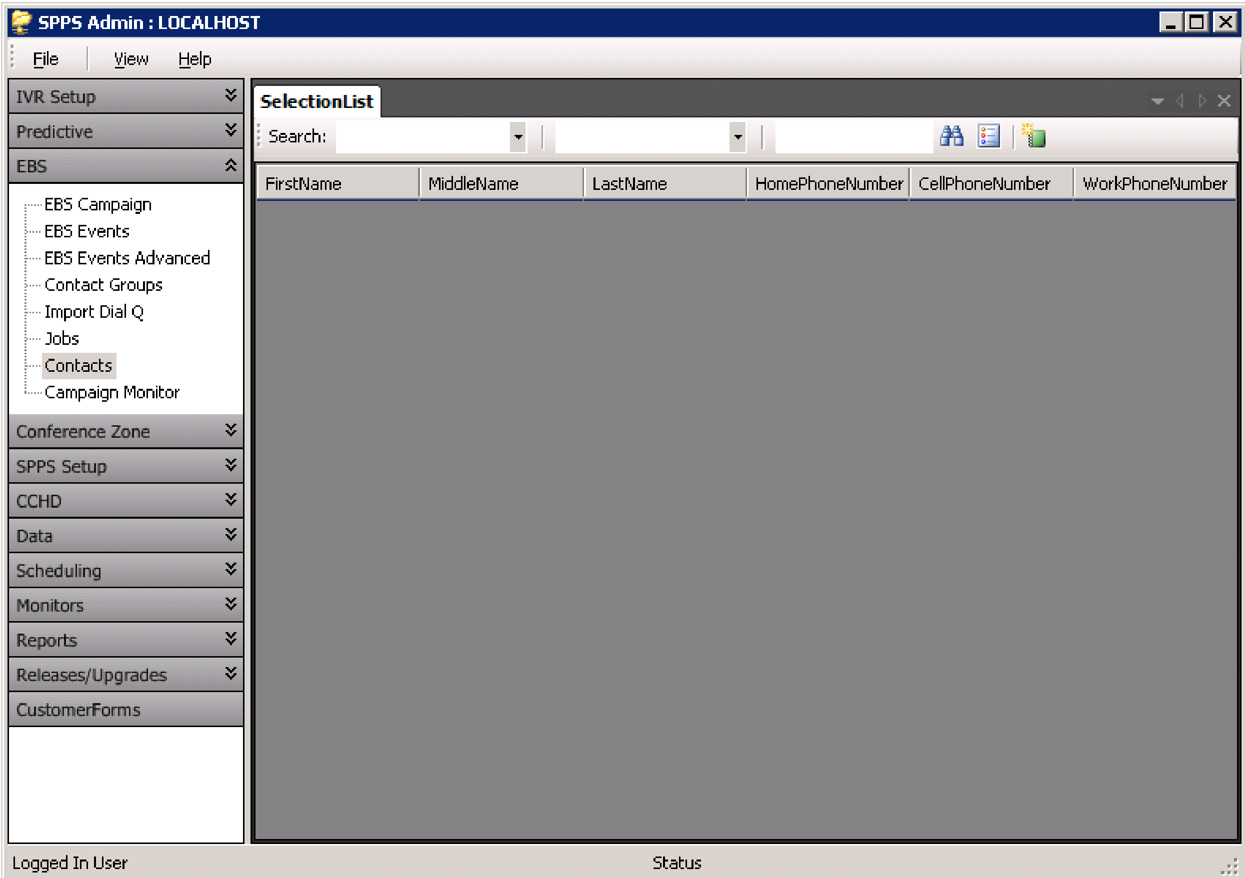


In the **IVR DNIS Definition** window, specify the **DNIS** and the appropriate custom application in the **IVR App**, **AfterHours App**, and **Holiday App** fields. In this example, the **DNIS** field is set to the wildcard "*", which maps to any DNIS, and the customer application is called *Avaya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

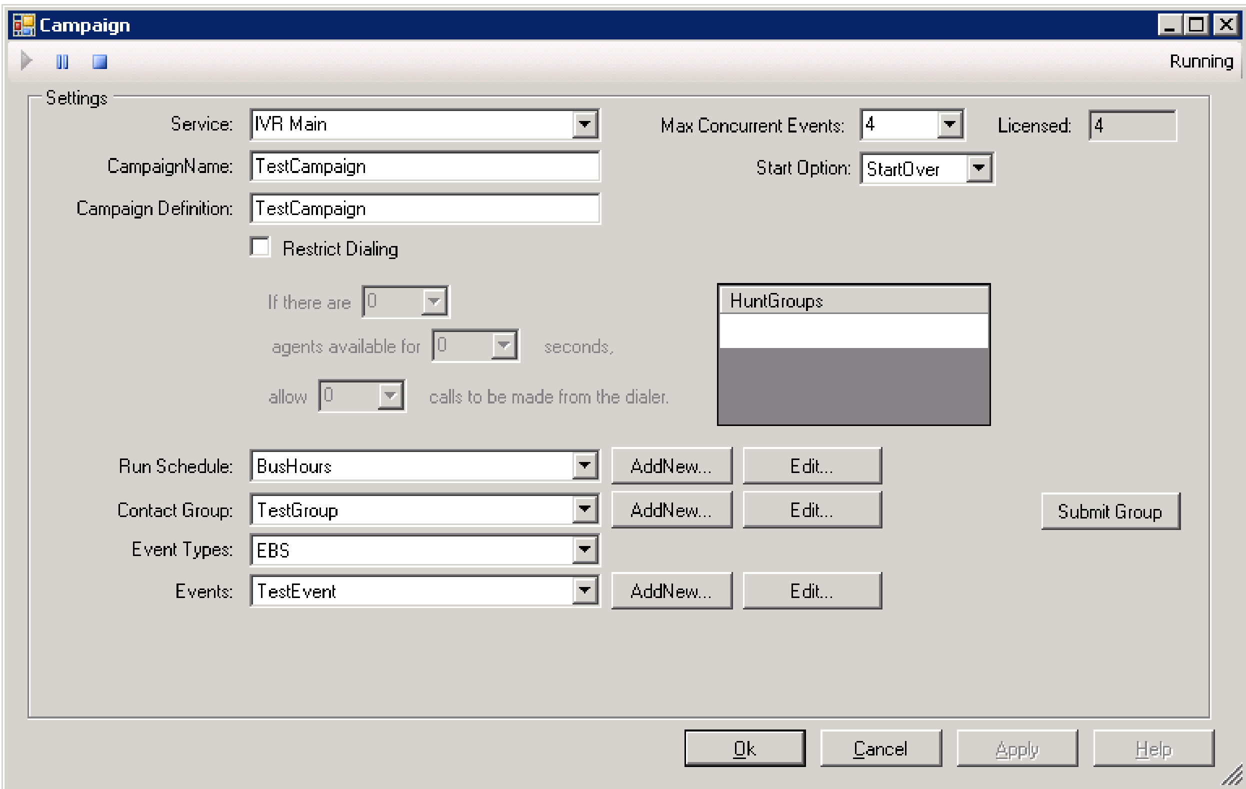


7.5. Launch a Voice4net EBS Broadcast Event

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.



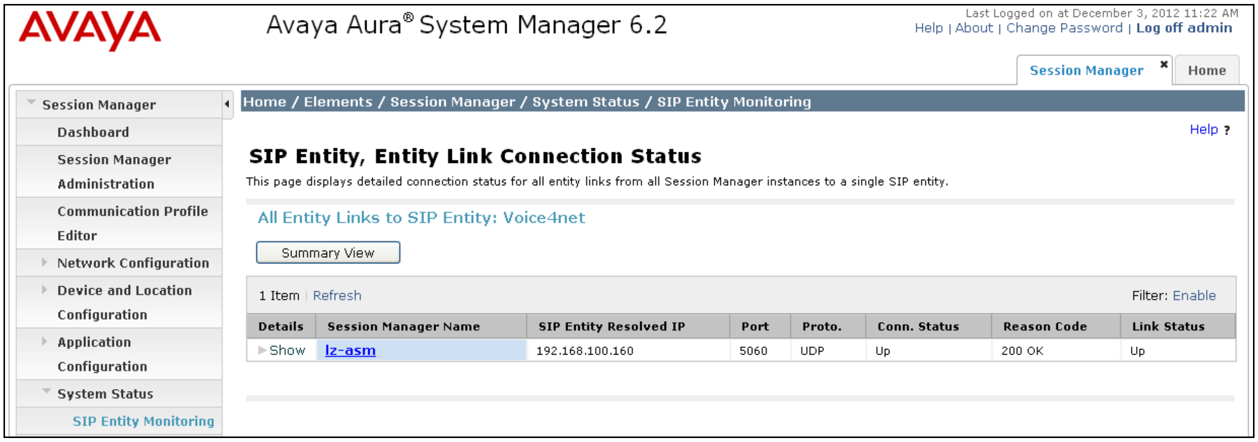
The detailed configuration of EBS events is outside the scope of these Application Notes, but the EBS Campaign used for the compliance test is shown below. The **Campaign** window shown below brings together the EBS contacts, event, and schedule. For example, in the sample campaign below, the **Service** 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.



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*.



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.

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