Avaya Solution & Interoperability Test Lab

Application Notes for Vocantas Utilities OnCall with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP Trunks – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Vocantas Utilities OnCall to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks.

Vocantas Utilities OnCall is a voice response solution designed for the requirements of utilities companies. In the compliance testing, Vocantas Utilities OnCall used SIP trunks to Avaya Aura® Session Manager for connections with the PSTN and for transfer of incoming calls to agents on Avaya Aura® Communication Manager.

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 for Vocantas Utilities OnCall to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks.

Vocantas Utilities OnCall is a voice response solution designed for the requirements of utilities companies. In the compliance testing, Vocantas Utilities OnCall used SIP trunks to Avaya Aura® Session Manager for connections with the PSTN and for transfer of incoming calls to agents on Avaya Aura® Communication Manager.

Incoming trunk calls destined for Vocantas Utilities OnCall are delivered by Avaya Aura® Communication Manager to Avaya Aura® Session Manager, and by Avaya Aura® Session Manager to Vocantas Utilities OnCall via SIP trunks. Vocantas Utilities OnCall answers the incoming call and plays the appropriate greeting, and uses DTMF tones from the calling party to determine the service to provide.

When requested by the calling party, Vocantas Utilities OnCall can perform blind transfer of the call to agents on Avaya Aura® Communication Manager. Vocantas Utilities OnCall can also initiate outbound calls to the PSTN, to notify customers with pertinent account information.

2. General Test Approach and Test Results

The feature test cases were performed manually. Calls were manually established between PSTN users and Utilities OnCall. Call controls were performed from the PSTN users to verify the various call scenarios.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet cable to Utilities OnCall.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included basic call, G.711, G.729, codec negotiation, media shuffling, drop, DTMF, blind transfer to internal agents for assistance, outbound to PSTN users for customer account notification, simultaneous calls, and reporting.

The serviceability testing focused on verifying the ability of Utilities OnCall to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connection to Utilities OnCall.

2.2. Test Results

All test cases were executed and verified.

2.3. Support

Technical support on Utilities OnCall can be obtained through the following:

- **Phone:** (877) 271-8853
- **Email:** info@vocantas.com

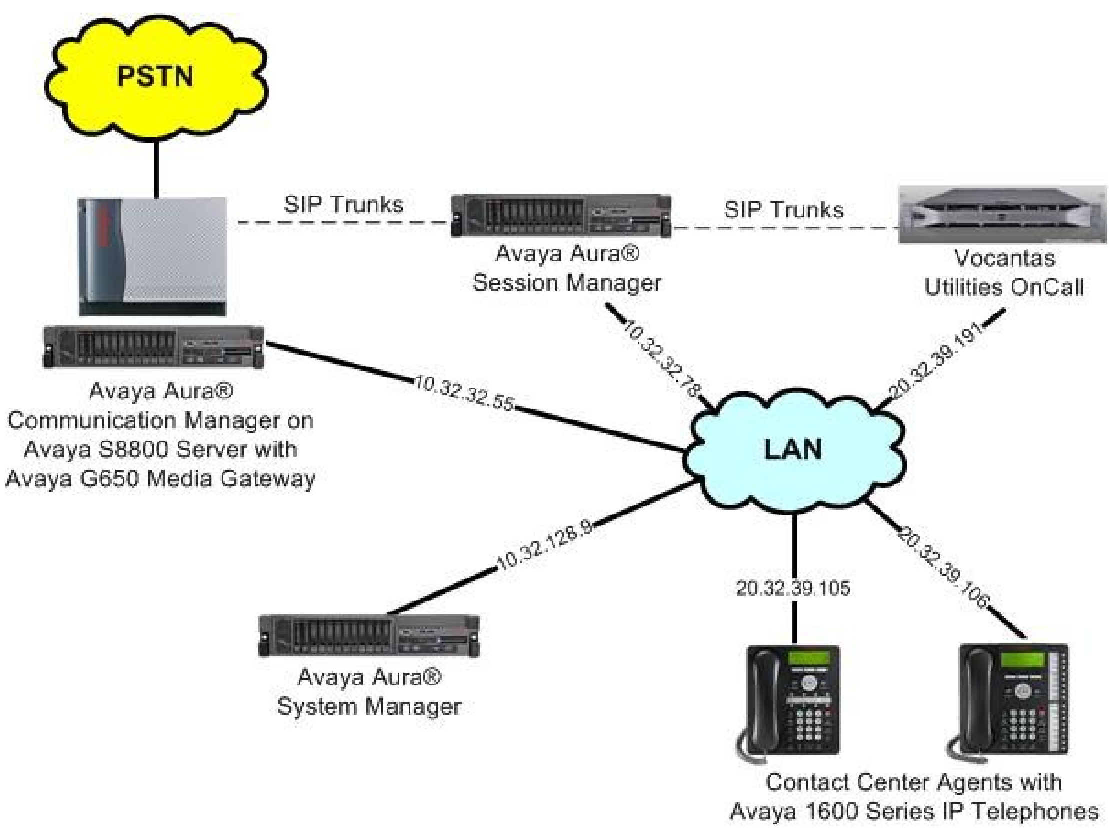
3. Reference Configuration

As shown in the test configuration below, SIP trunks are used between Utilities OnCall and Session Manager, to connect to users on the PSTN and to transfer to agents on Communication Manager. The Utilities OnCall server used the Dialogic Host Media Processing card for SIP messaging exchanges with Session Manager.

A five digit Uniform Dial Plan (UDP) was used to facilitate dialing. In the compliance testing, extensions of "61xxx" were associated with Utilities OnCall, and extensions of "62xxx-69xxx" were associated with resources on Communication Manager.

The detailed administration of basic connectivity between Communication Manager and Session Manager, and of contact center devices are not the focus of these Application Notes and will not be described.

The contact center devices used in the compliance testing consists of a skill group with extension "65555", and two agent extensions "65001-2".



4. Equipment and Software Validated

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

Equipment	Software
Avaya Aura® Communication Manager on Avaya S8800 Server	6.0.1 SP5.01 (R016x.00.1.510.1-19303)
Avaya G650 Media Gateway • TN799DP C-LAN Circuit Pack • TN2302AP IP Media Processor	 HW01 FW040 HW20 FW122
Avaya Aura® Session Manager	6.1 SP5
Avaya Aura® System Manager	6.1 SP5
Avaya 1600 Series IP Telephones (H.323)	1.3
Vocantas Utilities OnCall • Dialogic Host Media Processing	2.0 3.0 Service Update 307

5. Configure Avaya Aura® Communication Manager

This section provides the procedures for configuring Communication Manager. The procedures include the following areas:

- Verify license
- Administer SIP trunk group
- Administer SIP signaling group
- Administer IP network region
- Administer IP codec set
- Administer route pattern
- Administer private numbering
- Administer uniform dial plan
- Administer AAR analysis
- Administer ISDN trunk group
- Administer tandem calling party number

In the compliance testing, the existing SIP trunk group for communication with Session Manager and the associated signaling group, network region, and codec set were used for integration with Vocantas.

5.1. Verify License

Log in to the System Access Terminal (SAT) to verify that the Communication Manager license has proper permissions for features illustrated in these Application Notes. Use the "display system-parameters customer-options" command. Navigate to **Page 2**, and verify that there is sufficient remaining capacity for SIP trunks by comparing the **Maximum Administered SIP Trunks** field value with the corresponding value in the **USED** column.

The license file installed on the system controls the maximum permitted. If there is insufficient capacity, contact an authorized Avaya sales representative to make the appropriate changes.

```
change system-parameters customer-options                       Page   2 of  11
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 12000 7
           Maximum Concurrently Registered IP Stations: 18000 2
             Maximum Administered Remote Office Trunks: 12000 0
Maximum Concurrently Registered Remote Office Stations: 18000 0
              Maximum Concurrently Registered IP eCons: 414   0
  Max Concur Registered Unauthenticated H.323 Stations: 100   0
                        Maximum Video Capable Stations: 18000 1
                   Maximum Video Capable IP Softphones: 18000 0
                       Maximum Administered SIP Trunks: 24000 20
  Maximum Administered Ad-hoc Video Conferencing Ports: 24000 0
   Maximum Number of DS1 Boards with Echo Cancellation: 522   0
```

5.2. Administer SIP Trunk Group

Use the "change trunk-group n" command, where "n" is the existing SIP trunk group number used to reach Session Manager, in this case "5".

For **Group Name**, update as desired to reflect the same trunk group used to reach Session Manager and Vocantas. For **Number of Members**, enter sufficient number for simultaneous calls with Session Manager and Vocantas. Make a note of the **Signaling Group** number.

```
change trunk-group 5                                            Page   1 of  21
                                TRUNK GROUP

Group Number: 5                    Group Type: sip           CDR Reports: y
  Group Name: SIP Trunk to SM/Vocantas    COR: 1        TN: 1        TAC: 1005
   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: 5
                                                     Number of Members: 10
```

Navigate to **Page 3**, and enter "private" for **Numbering Format**.

```
change trunk-group 5                                            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
```

5.3. Administer SIP Signaling Group

Use the "change signaling-group n" command, where "n" is the existing SIP signaling group number used by the SIP trunk group from **Section 5.2**.

For **DTMF over IP**, enter "rtp-payload". For **Direct IP-IP Audio Connections**, enter "y". Make a note of the **Far-end Network Region** number, and the **Far-end Domain** value. Note that **Transport Method** is set to "tcp" for troubleshooting purposes, also note the values of **Near-end Listen Port** and **Far-end Listen Port**, which will be used later.

```
change signaling-group 5                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 5              Group Type: sip
  IMS Enabled? n        Transport Method: tcp
        Q-SIP? n                                            SIP Enabled LSP? n
     IP Video? n                                   Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: SM


   Near-end Node Name: Clan-1                Far-end Node Name: S8800-SM-SIG
 Near-end Listen Port: 5060                Far-end Listen Port: 5060
                                        Far-end Network Region: 1
                                   Far-end Secondary Node Name:
Far-end Domain: br110.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
```

5.4. Administer IP Network Region

Use the "change ip-network-region n" command, where "n" is the existing far-end network region number used by the SIP signaling group from **Section 5.3**.

For **Name**, update as desired to reflect the same network region used to reach Vocantas. Enter "yes" for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. In the compliance testing, the same network region was used for all Avaya users. Make a note of the **Codec Set** number.

```
change ip-network-region 1                                        Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: br110.com
    Name: Main/Vocantas
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
```

5.5. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the existing codec set number used by the IP network region from **Section 5.4**. Update the audio codec types in the **Audio Codec** fields as desired. The screenshot below shows the settings used in the compliance testing.

```
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.729              n          2        20
 2: G.711MU            n          2        20
 3:
 4:
 5:
 6:
 7:
```

5.6. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is the existing route pattern number to reach Session Manager, in this case "5". For **Pattern Name**, update as desired to reflect the same route pattern used to reach Session Manager and Vocantas. For **Secure SIP**, make certain the value is "n".

```
change route-pattern 5                                          Page   1 of   3
                    Pattern Number: 5   Pattern Name: To SM/Vocantas
                             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: 5    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                                        none
```

5.7. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to Vocantas. Add an entry for the trunk group defined in **Section 5.2**. In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 5 will result in a 5-digit calling number. The calling party number will be in the SIP "From" header.

```
change private-numbering 0                                     Page   1 of   2
                           NUMBERING - PRIVATE FORMAT

Ext Ext              Trk        Private          Total
Len Code             Grp(s)     Prefix           Len
 5  6                5                           5     Total Administered: 1
                                                          Maximum Entries: 540
```

5.8. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 61xxx to Vocantas. Note that other methods of routing may be used. Use the "change uniform-dialplan 0" command, and add an entry to specify the use of AAR for routing digits 61xxx, as shown below.

```
change uniform-dialplan 0                                      Page   1 of   2
                       UNIFORM DIAL PLAN TABLE
                                                             Percent Full: 0

  Matching                    Insert             Node
  Pattern       Len Del       Digits     Net Conv Num

 61              5   0                   aar  n
```

5.9. Administer AAR Analysis

Use the "change aar analysis 0" command, and add an entry to route calls to 61xxx. In the example shown below, calls with digits 61xxx will be routed using route pattern "5" from **Section 5.6**. Set the **Call Type** to "unku", to prevent "+" being added as a prefix.

```
change aar analysis 0                                          Page   1 of   2
                             AAR DIGIT ANALYSIS TABLE
                                   Location:  all          Percent Full:    2

          Dialed           Total      Route     Call   Node  ANI
          String          Min  Max   Pattern    Type   Num   Reqd
    61                    5    5      5         unku         n
```

5.10. Administer ISDN Trunk Group

Use the "change trunk-group n" command, where "n" is the existing ISDN trunk group number used to reach the PSTN, in this case "10". Navigate to **Page 3**.

For **Modify Tandem Calling Number**, enter "tandem-cpn-form" to allow for the calling party number from Vocantas to be modified.

```
change trunk-group 10                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none      Wideband Support? n
                                 Internal Alert? n       Maintenance Tests? y
                               Data Restriction? n     NCA-TSC Trunk Member:
                                      Send Name: y      Send Calling Number: y
            Used for DCS? n                            Send EMU Visitor CPN? n
   Suppress # Outpulsing? n    Format: public
 Outgoing Channel ID Encoding: preferred     UUI IE Treatment: service-provider

                                                 Replace Restricted Numbers? n
                                                Replace Unavailable Numbers? n
                                                      Send Connected Number: n
Network Call Redirection: none                    Hold/Unhold Notifications? n
             Send UUI IE? y     Modify Tandem Calling Number: tandem-cpn-form
               Send UCID? n
 Send Codeset 6/7 LAI IE? y                          Ds1 Echo Cancellation? n

    Apply Local Ringback? n              US NI Delayed Calling Name Update? n
 Show ANSWERED BY on Display? y
                             Network (Japan) Needs Connect Before Disconnect? n
 DSN Term? n
```

5.11. Administer Tandem Calling Party Number

Use the "change tandem-calling-party-num" command, to define the calling party number to send to the PSTN for tandem calls from Vocantas.

In the example shown below, all calls originating from a 5-digit extension beginning with 6 and routed to trunk group 10 will result in a 10-digit calling number. For **Number Format**, use an applicable format, in this case "pub-unk".

```
change tandem-calling-party-num                                 Page   1 of   8
                  CALLING PARTY NUMBER CONVERSION
                          FOR TANDEM CALLS
     CPN              Trk                              Number
 Len Prefix           Grp(s)     Delete  Insert        Format

 5   6                10                 90884         pub-unk
```

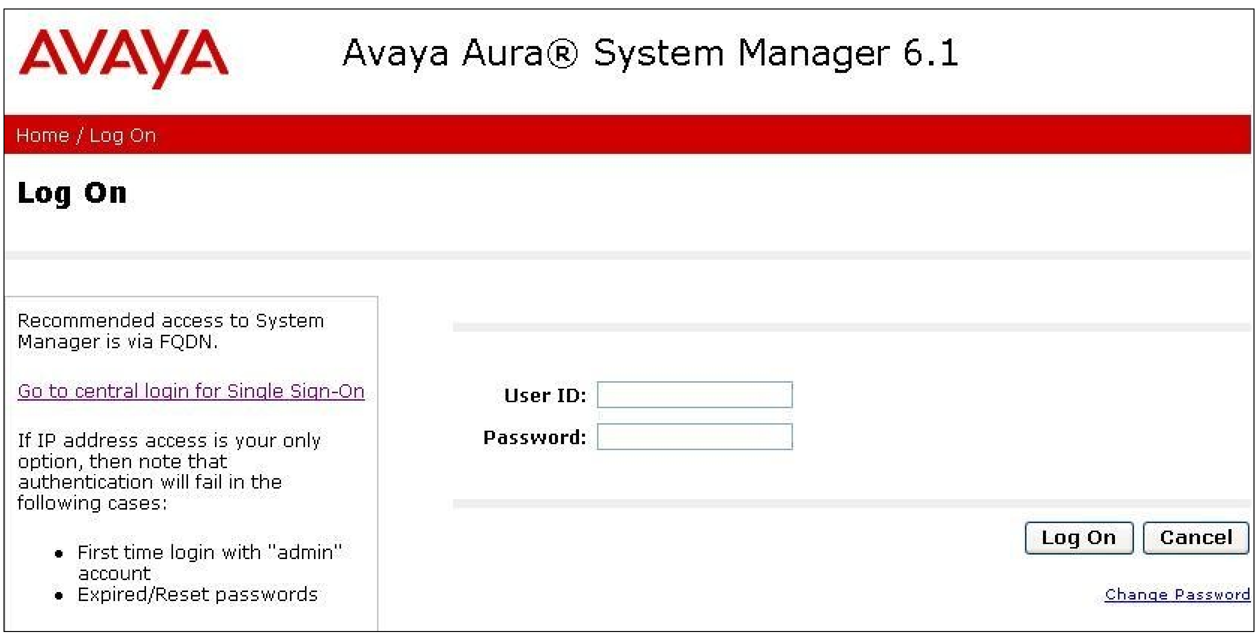
6. Configure Avaya Aura® Session Manager

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

- Launch System Manager
- Administer locations
- Administer adaptations
- Administer SIP entities
- Administer entity links
- Administer routing policies
- Administer dial patterns

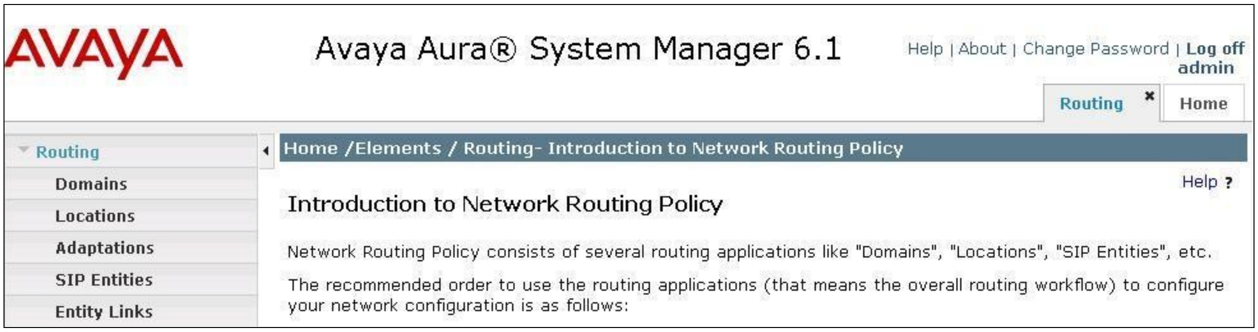
6.1. Launch System Manager

Access the System Manager web interface by using the URL “https://ip-address” in an Internet browser window, where “ip-address” is the IP address of the System Manager server. Log in using the appropriate credentials.

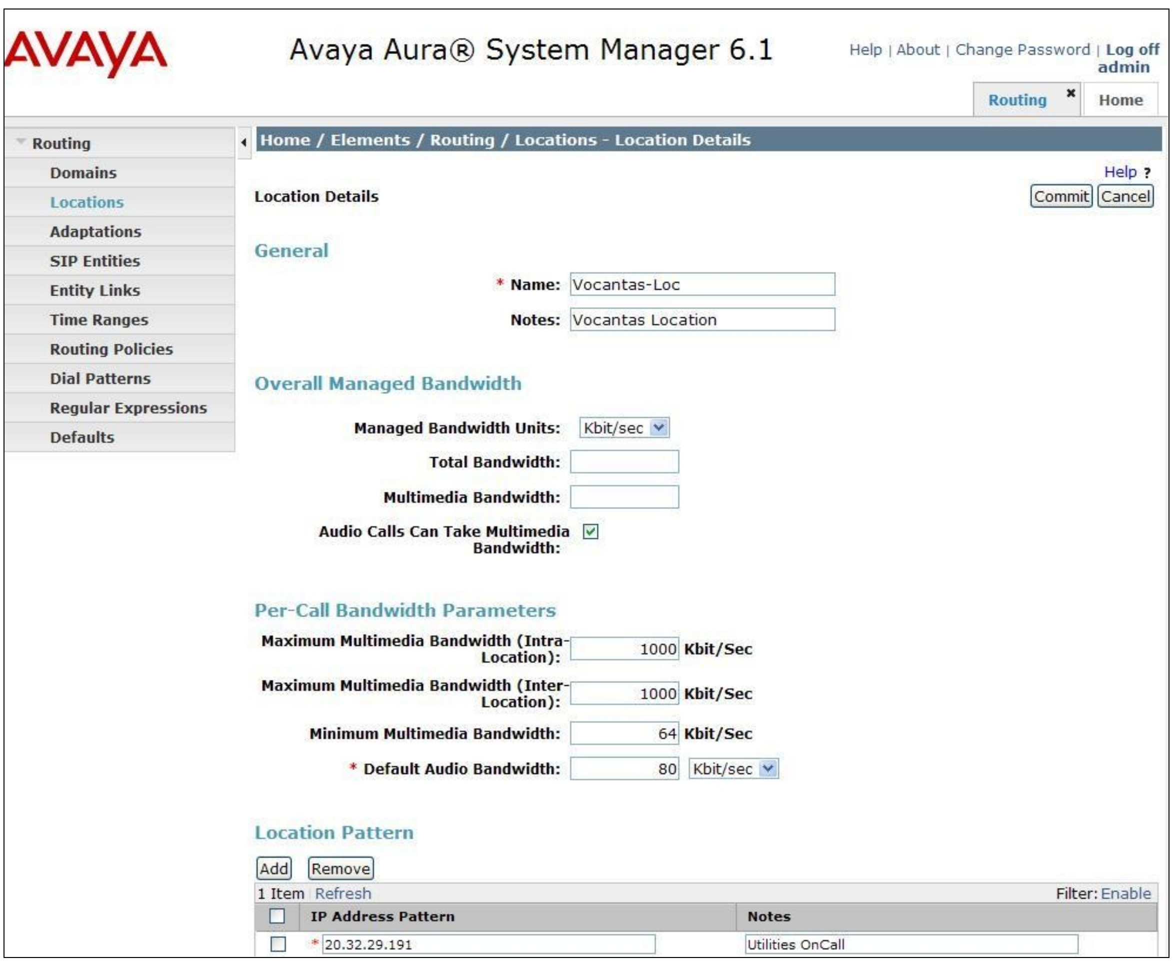


6.2. Administer Locations

In the subsequent screen (not shown), select **Elements > Routing** to display the **Introduction to Network Routing Policy** screen below. Select **Routing > Locations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new location for Vocantas.



The **Location Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name** and optional **Notes**. In the **Location Pattern** sub-section, click **Add** and enter the applicable **IP Address Pattern**, as shown below. Retain the default values in the remaining fields.

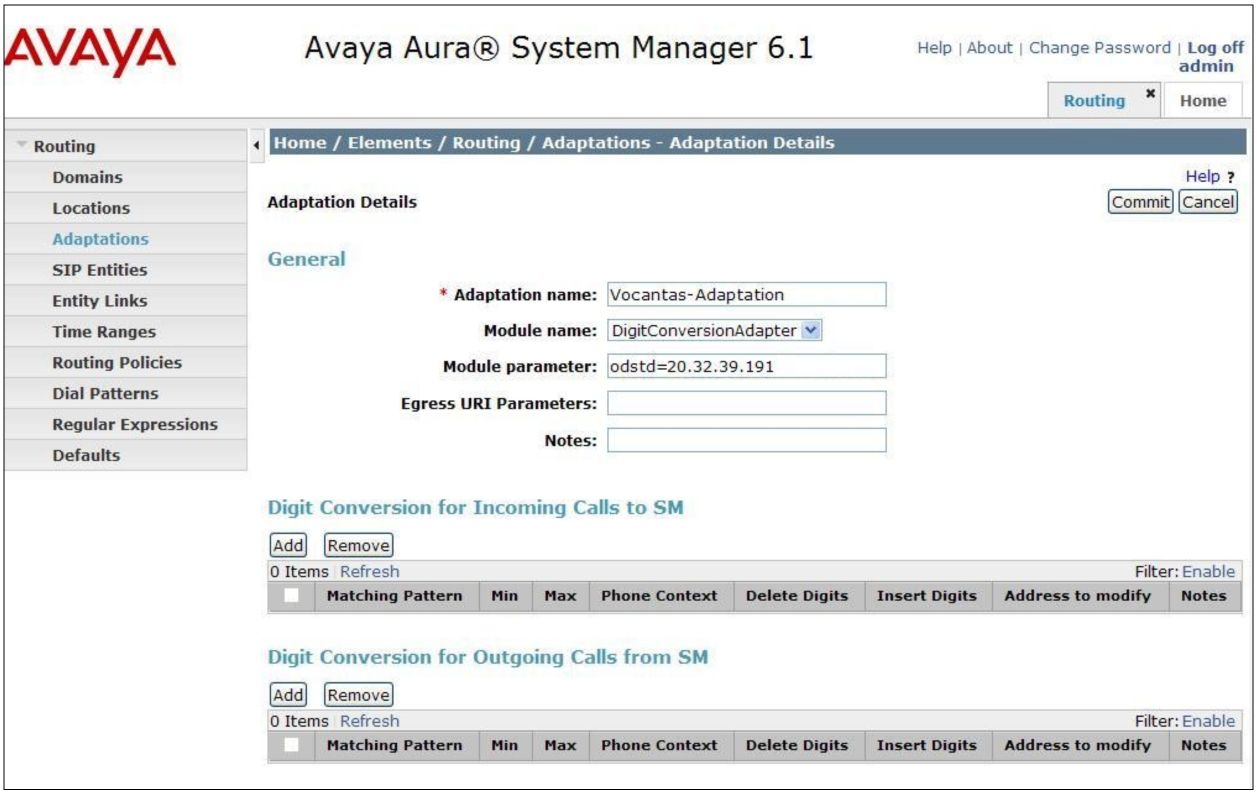


6.3. Administer Adaptations

Select **Routing > Adaptations** from the left pane, and click **New** in the subsequent screen (not shown) to add a new adaptation for Vocantas.

The **Adaptation Details** screen is displayed. In the **General** sub-section, enter a descriptive **Adaptation name**. For **Module name**, select "DigitConversionAdapter".

For **Module parameter**, enter "odstd=20.32.39.191", where "20.32.39.191" is the IP address of Vocantas. This will set the destination domain for outgoing calls from Session Manager to the IP address of Vocantas, as required by Vocantas.

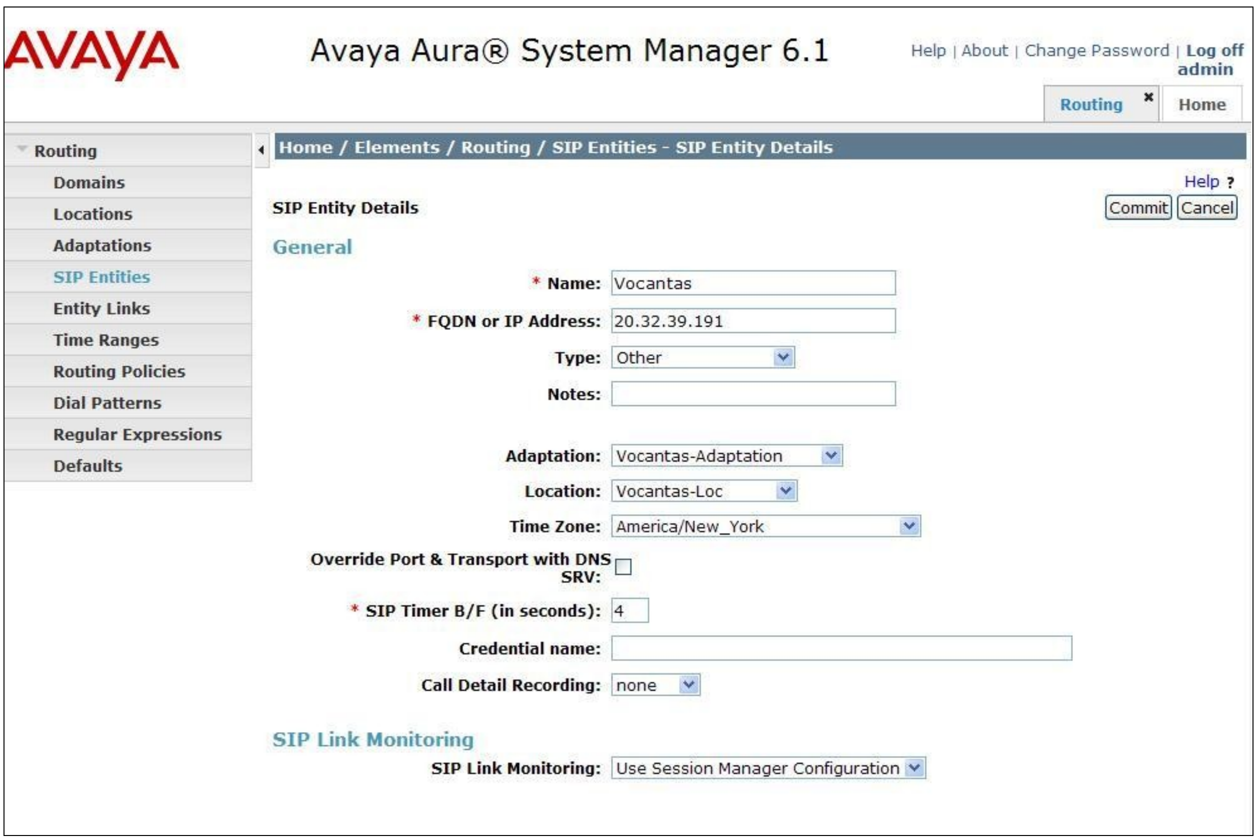


6.4. Administer SIP Entities

Select **Routing > SIP Entities** from the left pane, and click **New** in the subsequent screen (not shown) to add a new SIP entity for Vocantas.

The **SIP Entity Details** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **FQDN or IP Address:** The IP address of Vocantas.
- **Type:** "Other"
- **Adaptation:** Select the Vocantas adaptation name from **Section 6.3**.
- **Location:** Select the Vocantas location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.

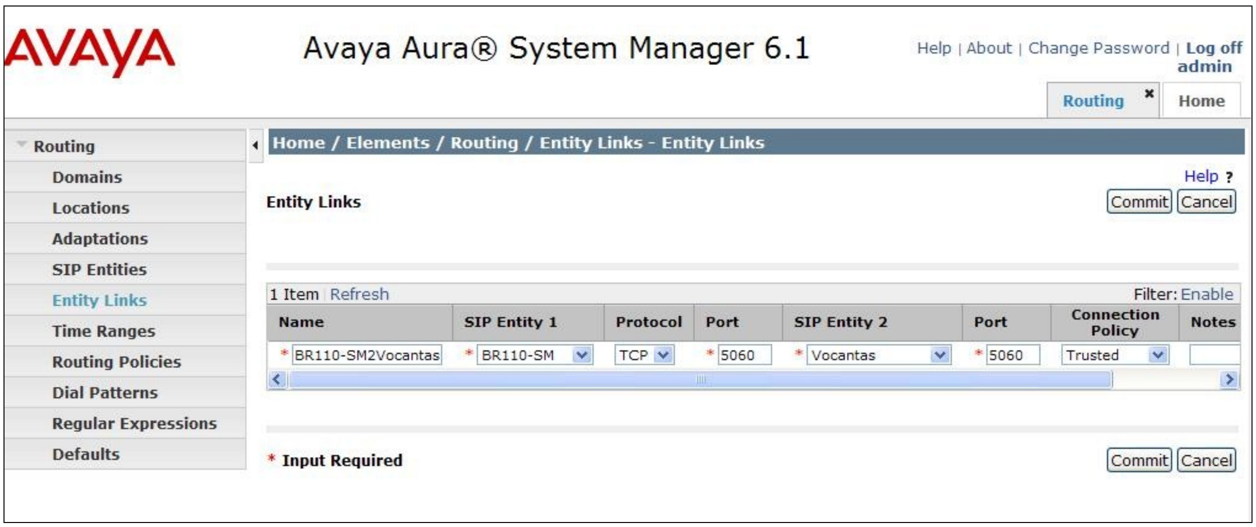


6.5. Administer Entity Links

Select **Routing > Entity Links** from the left pane, and click **New** in the subsequent screen (not shown) to add a new entity link for Vocantas.

The **Entity Links** screen is displayed. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Name:** A descriptive name.
- **SIP Entity 1:** The Session Manager entity name, in this case "BR110-SM".
- **Protocol:** The signaling group transport method from **Section 5.3**.
- **Port:** The signaling group listen port number from **Section 5.3**.
- **SIP Entity 2:** The Vocantas entity name from **Section 6.4**.
- **Port:** The signaling group listen port number from **Section 5.3**.



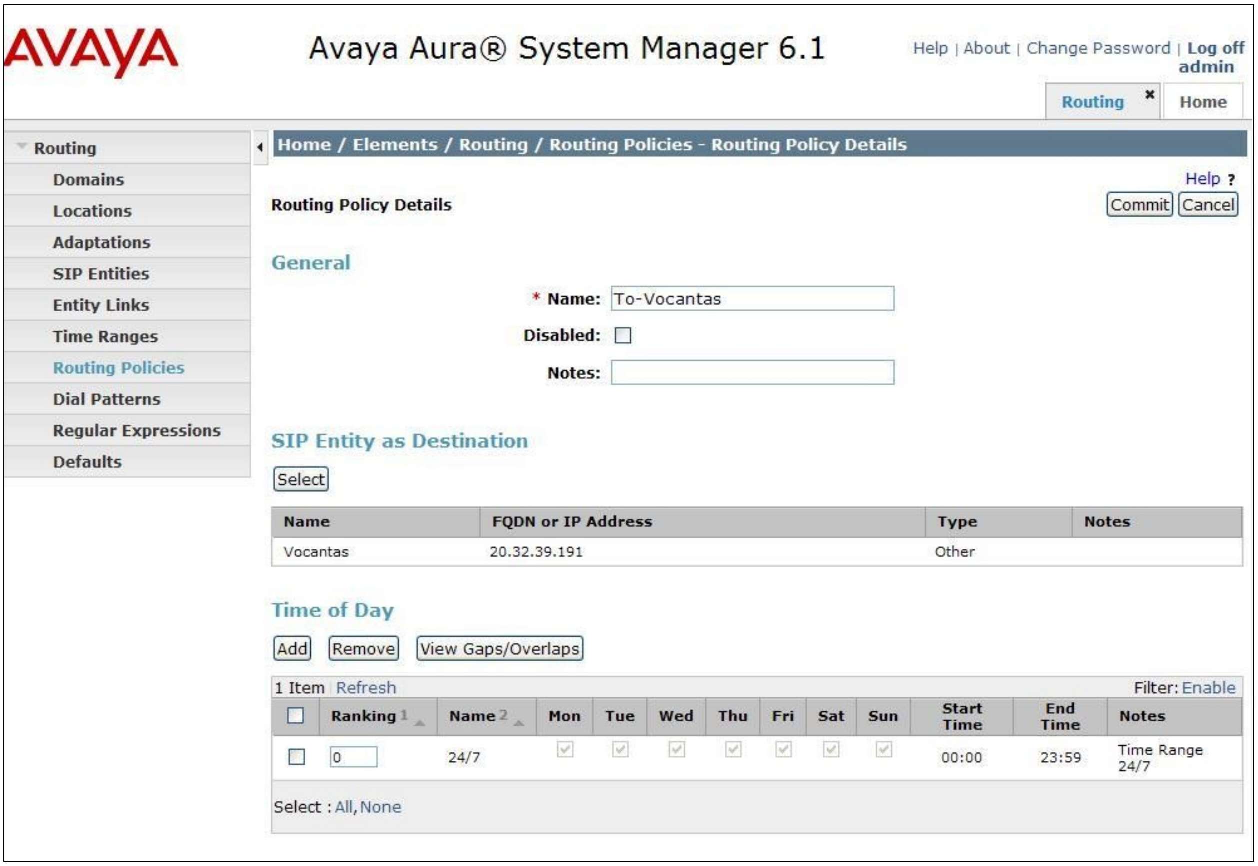
6.6. Administer Routing Policies

Select **Routing > Routing Policies** from the left pane, and click **New** in the subsequent screen (not shown) to add a new routing policy for Vocantas.

The **Routing Policy Details** screen is displayed. In the **General** sub-section, enter a descriptive **Name**.

In the **SIP Entity as Destination** sub-section, click **Select** and select the Vocantas entity name from **Section 6.4** in the listing (not shown).

Retain the default values in the remaining fields.



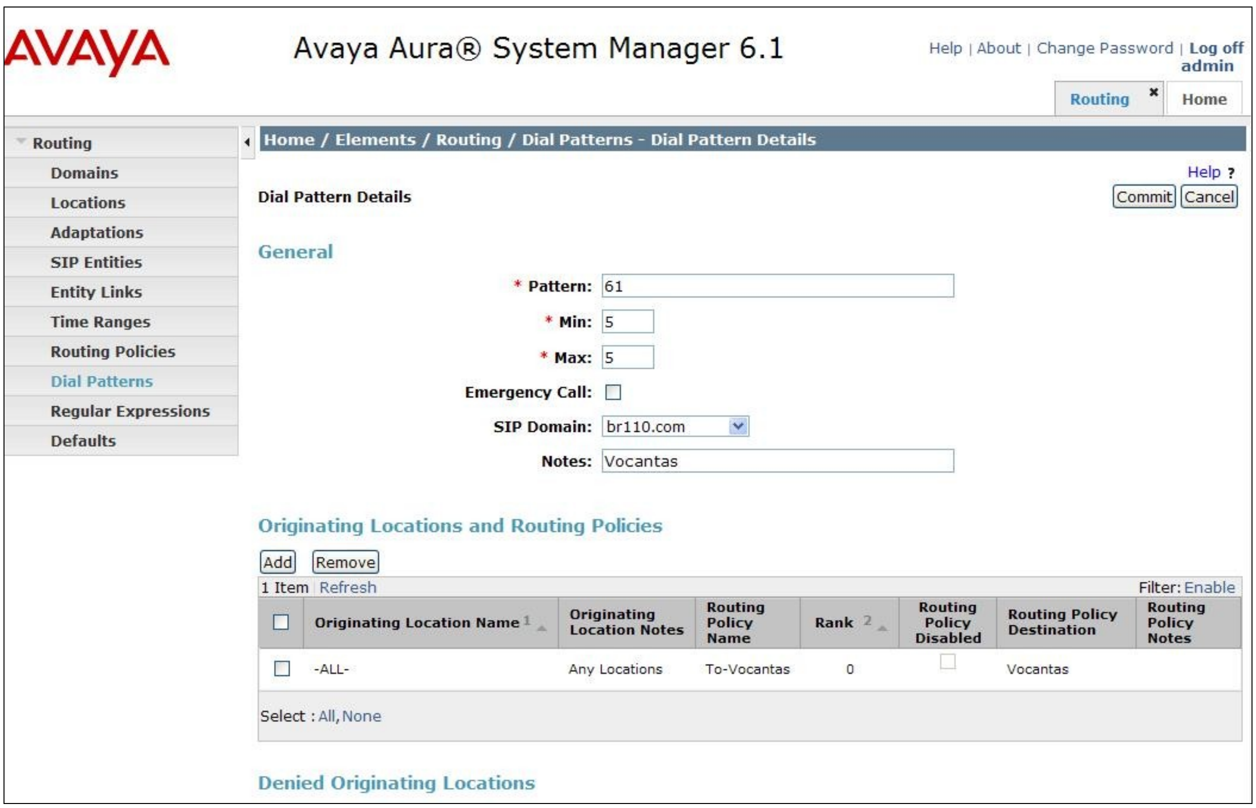
6.7. Administer Dial Patterns

Select **Routing > Dial Patterns** from the left pane, and click **New** in the subsequent screen (not shown) to add a new dial pattern to reach Vocantas.

The **Dial Pattern Details** screen is displayed. In the **General** sub-section, enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern:** A dial pattern to match.
- **Min:** The minimum number of digits to be matched.
- **Max:** The maximum number of digits to be matched.
- **SIP Domain:** The signaling group domain name from **Section 5.3**.
- **Notes:** Any desired description.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy for reaching Vocantas. In the compliance testing, the policy allowed for call origination from all locations, as shown below. Retain the default values in the remaining fields.



7. Configure Vocantas Utilities OnCall

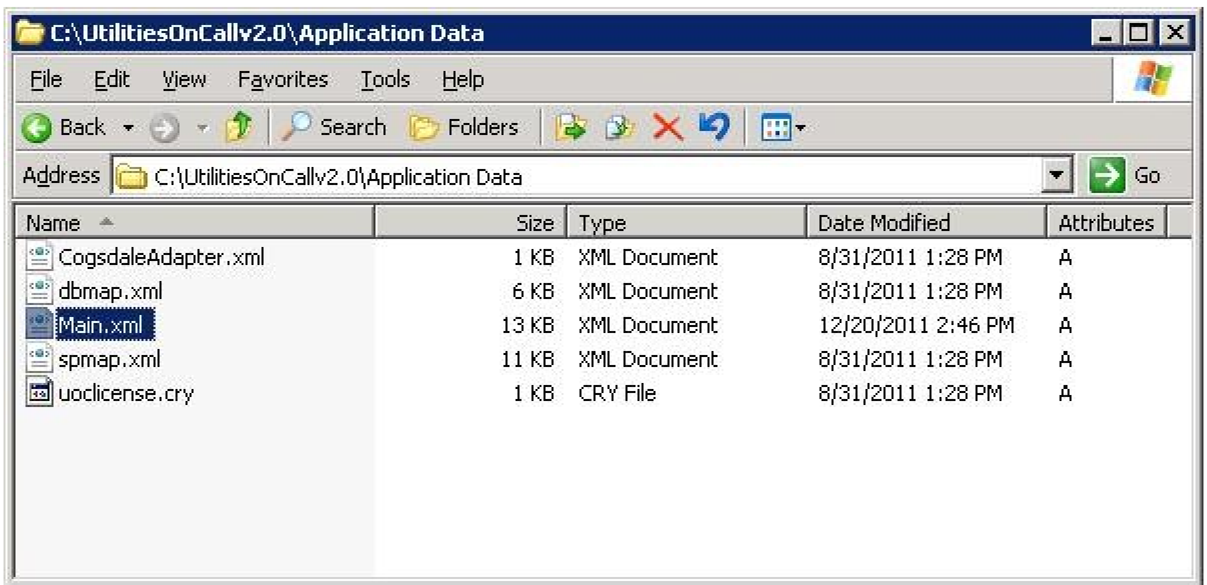
This section provides the procedures for configuring Utilities OnCall. The procedures include the following areas:

- Administer Main.xml
- Administer VBVoice

The configuration of Utilities OnCall is typically performed by Vocantas support engineers. The procedural steps are presented in these Application Notes for informational purposes.

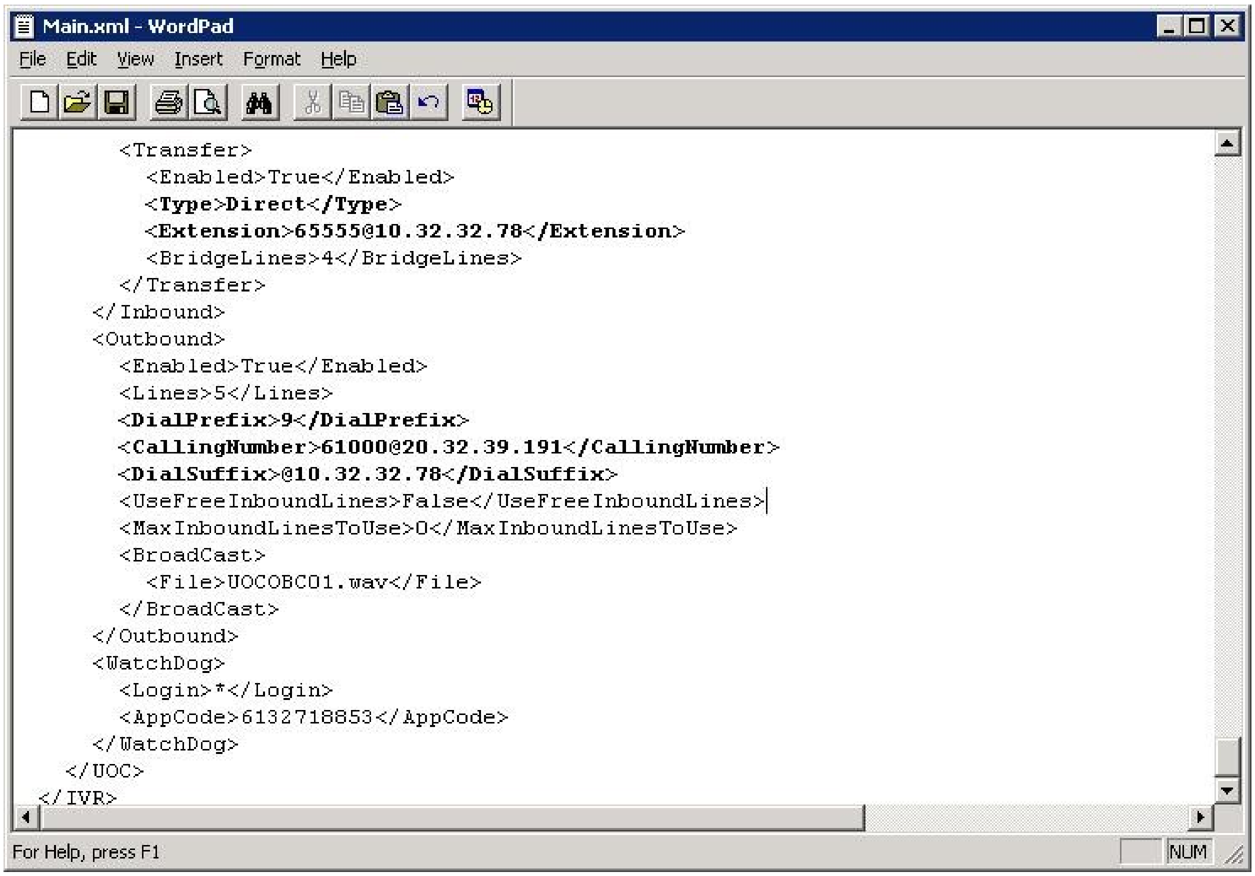
7.1. Administer Main.xml

From the Utilities OnCall server, navigate to the **C:\UtilitiesOnCallv2.0\Application Data** directory to locate the **Main.xml** file shown below.



Open the **Main.xml** file with the WordPad application. Scroll down to the bottom of the file. For transfer **Type**, enter “Direct”. For transfer **Extension**, enter “x@y” where “x” is the skill group extension from **Section 3**, and “y” is the IP address of Session Manager.

For outbound **DialPrefix**, enter the applicable ARS/AAR dialing prefix, in this case “9”. For outbound **CallingNumber**, enter “x@y” where “x” is an available extension assigned to Utilities OnCall, and “y” is the IP address of the Utilities OnCall server. For outbound **DialSuffix**, enter “@y” where “y” is the IP address of Session Manager.

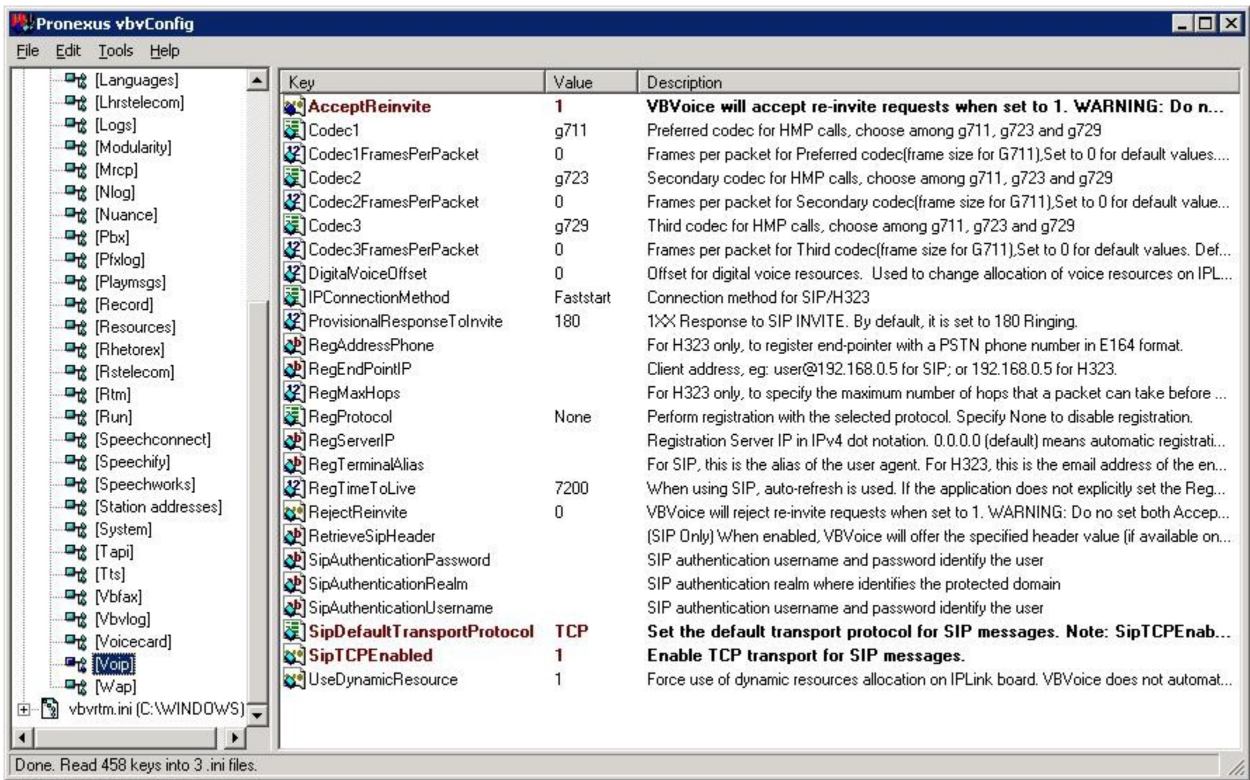


7.2. Administer VBVoice

Select **All Program > Pronexus > VBVConfig > Configure VBVoice**, to display the **Pronexus vbvConfig** screen below. Select **vbvoice.ini > Voip** in the left pane, to display a list of parameters in the right pane.

Right click on **AcceptReinvite**, and enable the parameter in the subsequent screen (not shown). Use similar procedure to enable **SipTCPEnabled**, and set **SipDefaultTransportProtocol** to "TCP". Retain the default values in the remaining fields.

The screenshot below shows the parameter settings used in the compliance testing.



8. Verification Steps

This section provides tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and Utilities OnCall.

8.1. Verify Avaya Aura® Communication Manager

From the SAT interface, verify the status of the SIP trunk groups by using the "status trunk n" command, where "n" is the trunk group number administered in **Section 5.2**. Verify that all trunks are in the "in-service/idle" state as shown below.

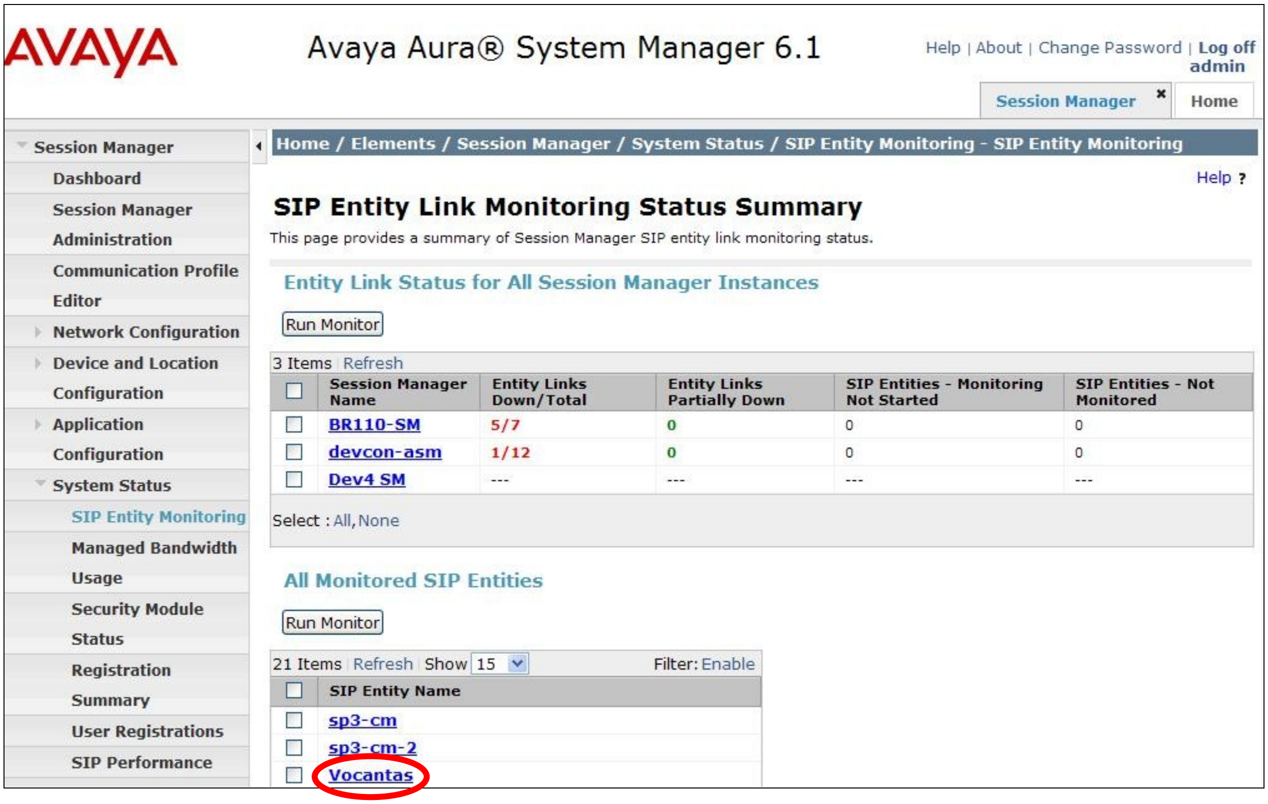
```
status trunk 5

                             TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0005/001 T00083   in-service/idle    no
0005/002 T00084   in-service/idle    no
0005/003 T00085   in-service/idle    no
0005/004 T00086   in-service/idle    no
0005/005 T00087   in-service/idle    no
0005/006 T00045   in-service/idle    no
0005/007 T00046   in-service/idle    no
0005/008 T00047   in-service/idle    no
0005/009 T00048   in-service/idle    no
0005/010 T00049   in-service/idle    no
```

Verify the status of the SIP signaling groups by using the "status signaling-group n" command, where "n" is the signaling group number administered in **Section 5.3**. Verify that the signaling group is "in-service" as indicated in the **Group State** field shown below.

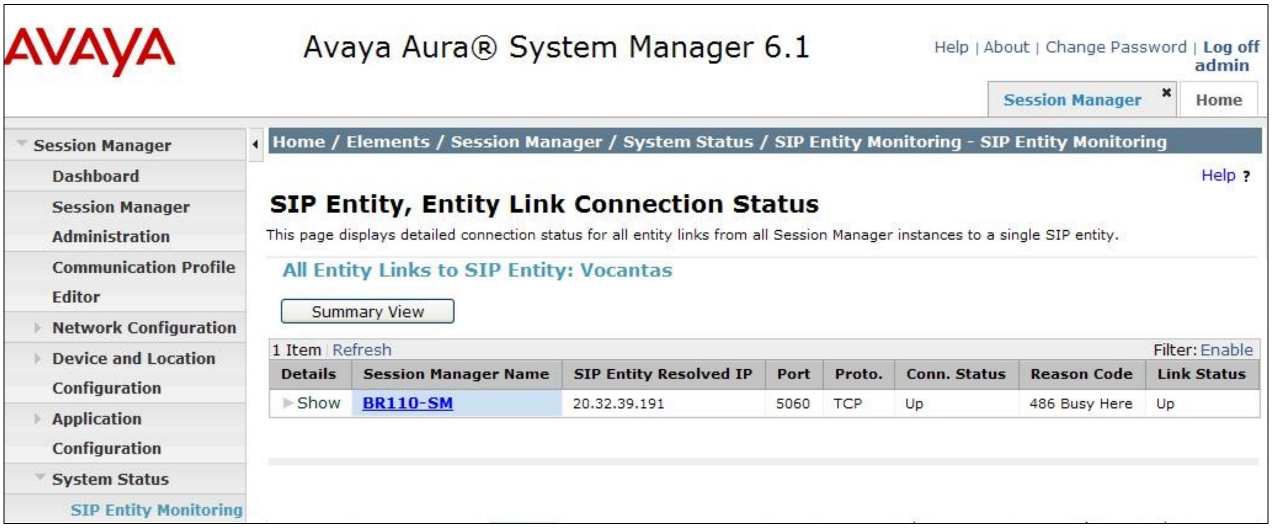
```
status signaling-group 5
                         STATUS SIGNALING GROUP

        Group ID: 5
      Group Type: sip

     Group State: in-service
```

8.2. Verify Avaya Aura® Session Manager

From the System Manager home page (not shown), select **Elements > Session Manager** to display the **Session Manager Dashboard** screen (not shown). Select **Session Manager > System Status > SIP Entity Monitoring** from the left pane to display the **SIP Entity Link Monitoring Status Summary** screen below. Click on the Vocantas entity name from **Section 6.4**.



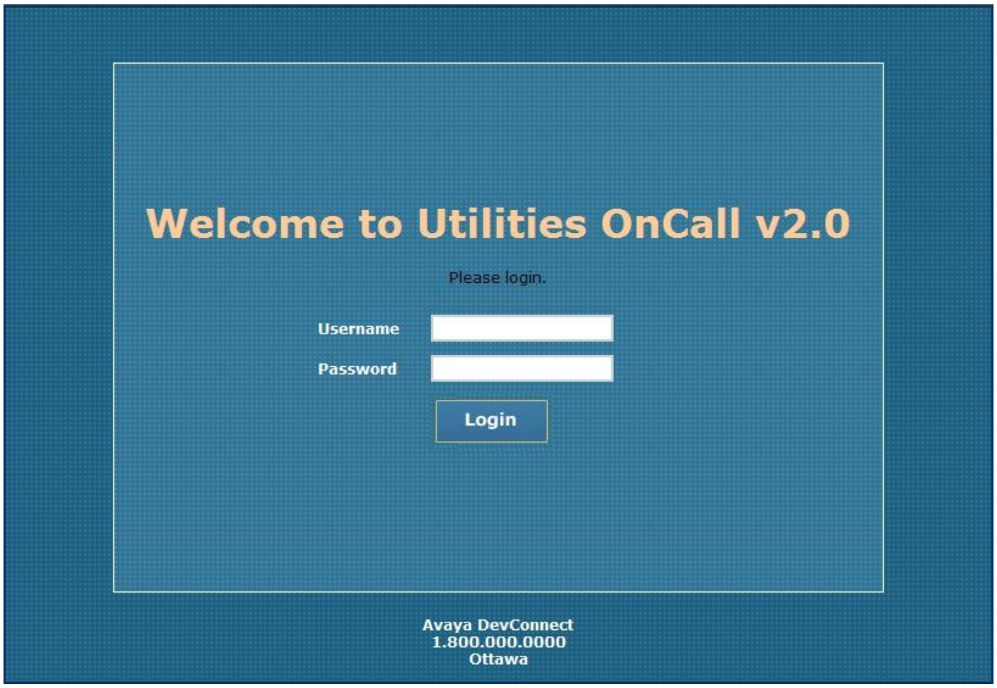
The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that **Conn. Status** and **Link Status** are "Up", as shown below.



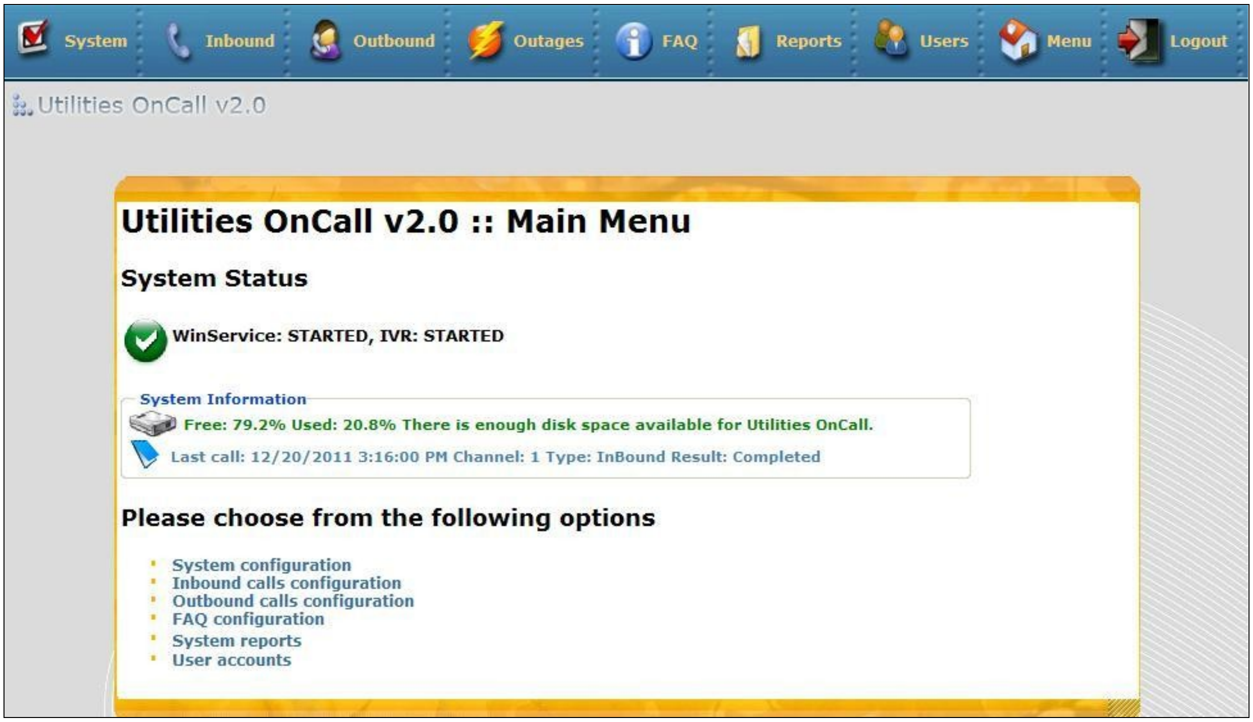
8.3. Verify Vocantas Utilities OnCall

Make and complete an incoming trunk call from the PSTN to Utilities OnCall.

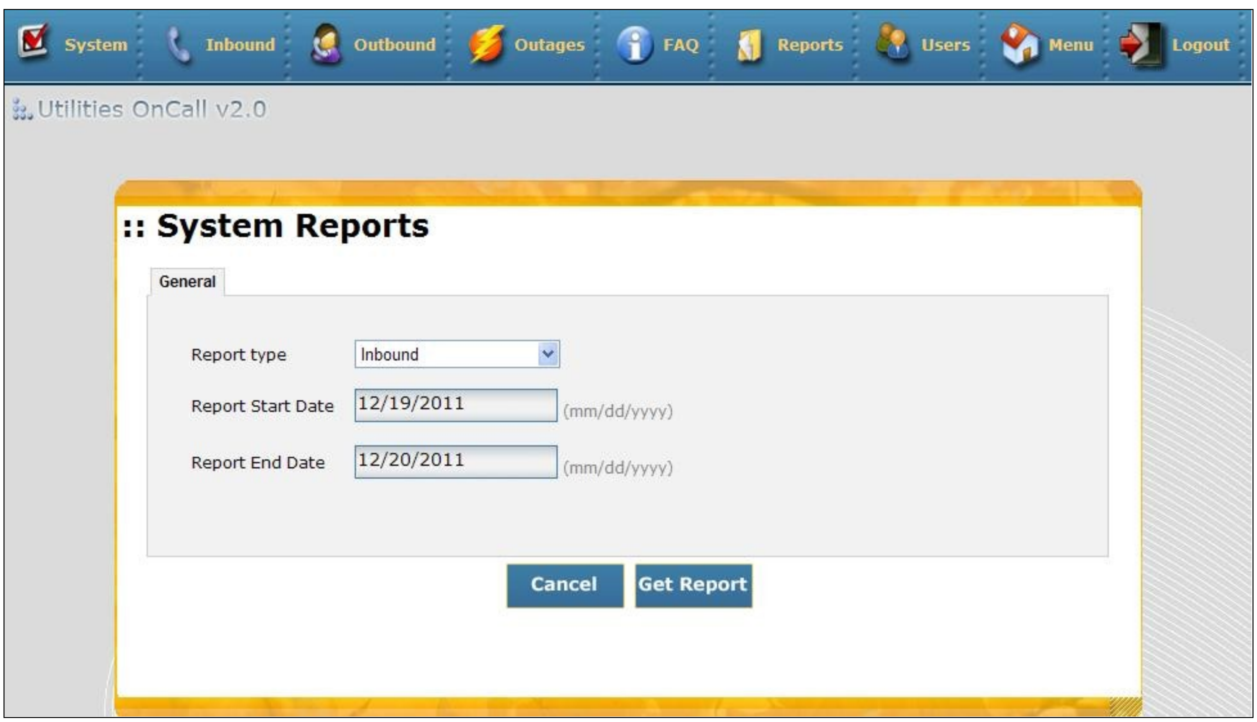
Access the Utilities OnCall web-based interface by using the URL "http://ip-address/uocgui/webgui" in an Internet browser window, where "ip-address" is the IP address of the Utilities OnCall server. The **Welcome to Utilities OnCall v2.0** screen is displayed. Log in using the appropriate credentials.



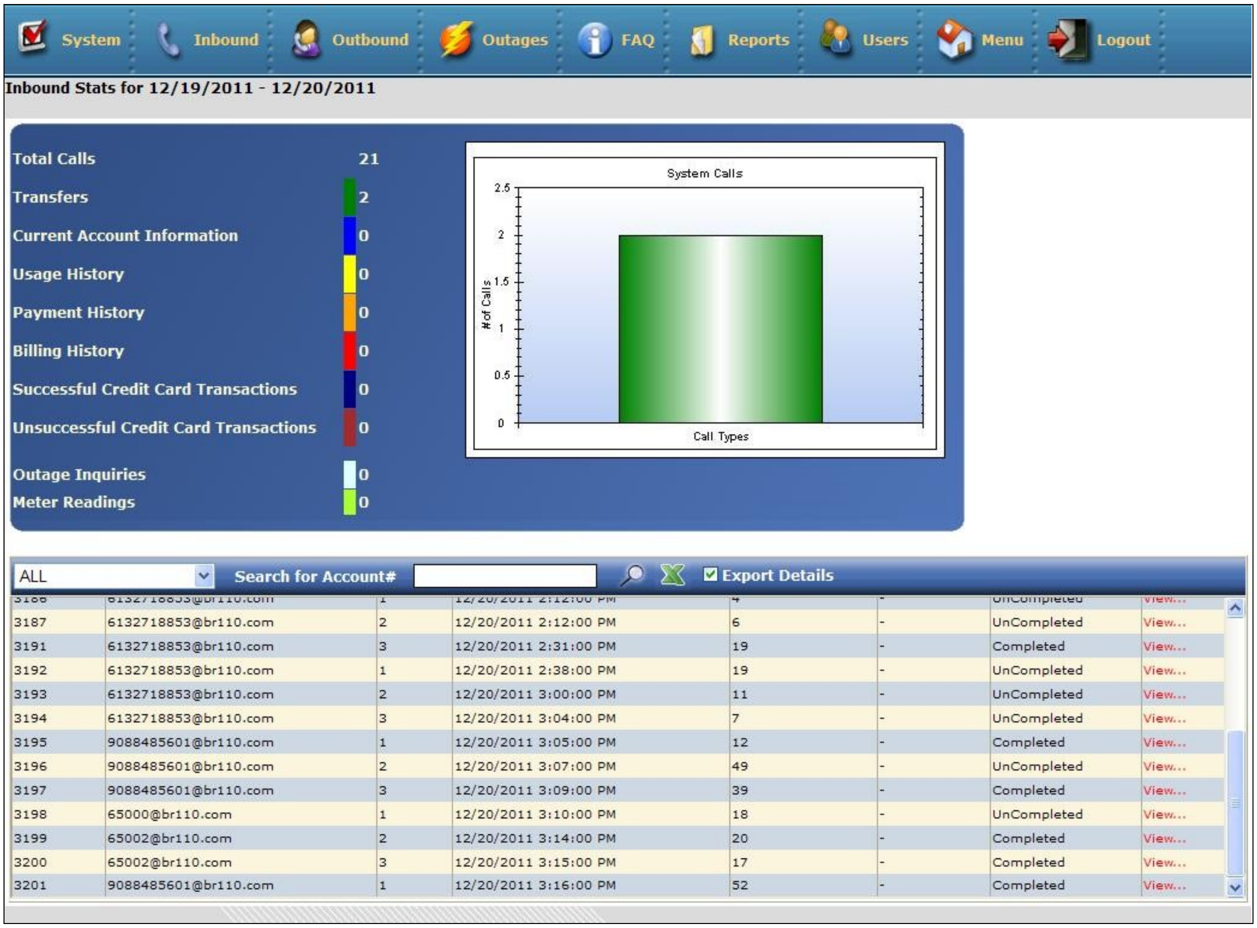
The **Utilities OnCall v2.0 Main Menu** screen is displayed. Select **Reports** from the top menu.



The **System Reports** screen is displayed next. Retain all default values and click **Get Report**.



The **Inbound Stats** report is displayed. Verify that there is an entry reflecting the last call, with proper values in the relevant fields, as shown below.



9. Conclusion

These Application Notes describe the configuration steps required for Vocantas Utilities OnCall to successfully interoperate with Avaya Aura® Communication Manager using Avaya Aura® Session Manager. All feature and serviceability test cases were completed.

10. Additional References

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

1. *Administering Avaya Aura™ Communication Manager*, Document 03-300509, Issue 6.0, Release 6.0, June 2010, available at http://support.avaya.com.

2. *Administering Avaya Aura™ Session Manager*, Document Number 03-603324, Issue 3, Release 6.0, August 2010, available at http://support.avaya.com.

3. *Vocantas Utilities OnCall Administrator and User Guide*, 2011, available upon request to Vocantas Support.

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.