Avaya Solution & Interoperability Test Lab

Application Notes for Noble Systems Contact Center Solution 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 Noble Systems Contact Center Solution to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks.

Noble Systems Contact Center Solution is a unified customer interaction management solution. In the compliance testing, Noble Systems Contact Center Solution used SIP trunks to Avaya Aura® Session Manager for dedicated connections with agents, and for calls with the PSTN.

Readers should pay attention to Section 2, in particular the scope of testing as outlined in Section 2.1 as well as any observations noted in Section 2.2, to ensure that their own use cases are adequately covered by this scope and results.

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 Noble Systems Contact Center Solution to interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager using SIP trunks.

Noble Systems Contact Center Solution is a unified customer interaction management solution for multimedia business environments that combines outbound predictive dialing and inbound with blended call management. In the compliance testing, Noble Systems Contact Center Solution used SIP trunks to Avaya Aura® Session Manager for dedicated connections with agents, and for calls with the PSTN.

Noble Systems Contact Center Solution agents are administered as regular station users on Avaya Aura® Communication Manager, with desktop computers running the web-based or client version of Noble Systems Composer to perform ACD related activities such as login/logout and answer/drop calls. All ACD functionalities are provided by Noble Systems Contact Center Solution.

Noble Systems Contact Center Solution can support direct trunk connection to the PSTN or via a PBX. In the compliance testing, the connection with the PSTN for inbound/outbound calls was accomplished via Avaya Aura® Communication Manager. Inbound calls were routed by Avaya Aura® Communication Manager to Avaya Aura® Session Manager and then to Noble Systems Contact Center Solution. Noble Systems Contact Center Solution delivered the inbound calls to available agents by merging the talk paths of the inbound calls from the PSTN with the dedicated connections to the agents. Outbound calls were initiated by Noble System Contact Center Solution to Avaya Aura® Communication Manager via Avaya Aura® Session Manager, and Noble Systems Contact Center Solution delivered the answered outbound calls to available agents by merging the talk paths.

2. General Test Approach and Test Results

The feature test cases were performed both automatically and manually. Outbound calls were automatically launched by Contact Center Solution, whereas the inbound calls were manually made. Call controls were performed from the agent desktops or telephones to verify the various call scenarios.

The serviceability test cases were performed manually by disconnecting and reconnecting the Ethernet cables to Contact Center Solution.

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.

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing.

The feature testing included G.711MU, G.729, codec negotiation, DTMF, blind/attended transfer, blind/attended conference, inbound, outbound, and multiple agents.

The serviceability testing focused on verifying the ability of Contact Center Solution to recover from adverse conditions, such as disconnecting/reconnecting the Ethernet connections to Contact Center Solution.

2.2. Test Results

All test cases were executed and verified. The following were the observations on Contact Center Solution from the compliance testing.

- Contact Center Solution does not support media shuffling. Therefore corresponding parameters must be disabled on the relevant signaling group and network region.
- The transfer-to and conference-to agents do not receive screen updates associated with the call. Furthermore, there isn't a way for the conference-to agent to initiate a drop from the active conference call.
- The conference-from agent will see a "hang up during transfer" pop-up message, whenever the user or agent drops first from a conference call.
- PSTN user in the conference call with 2 Noble agents, if PSTN user hangs up the call, the conference call will be disconnected on all agents.
- Agent will see a "hang up during transfer" pop-up message whenever PSTN or Agent drop the call during call is put on hold.
- If Agent hangs up a call that is on hold, after reconnect agent deskphone with Composer, agent need to manually change status from Paused to Connected on Composer.

- No blind transfer support to internal or external number. Blind transfer only support for call transferred from Agent to Agent.
- Unplug agent's PC, NobleWinAgent will not response to the mouse and pop-up a window say: "Unable to connect appserver or dbserver" after plug LAN cable back, need to reboot and everything is work as normal.

2.3. Support

Technical support on Contact Center Solution can be obtained through the following:

- **Phone:** (888) 966-2539
- **Web:** http://www.noblesys.com/contact.aspx
- **Email:** info@noblesys.com

3. Reference Configuration

Contact Center Solution consists of multiple servers, and the compliance testing used a two-server configuration with the Composer Web Server component running on a separate server.

SIP trunks are used from Contact Center Solution to Session Manager, to reach users on Communication Manager and on the PSTN.

A five digit Uniform Dial Plan (UDP) was used to facilitate dialing with Contact Center Solution. Unique extension ranges were associated with Communication Manager Users (56xxx), and Contact Center Solution (52xxx).

The detailed administration of basic connectivity between Communication Manager and Session Manager is not the focus of these Application Notes and will not be described.

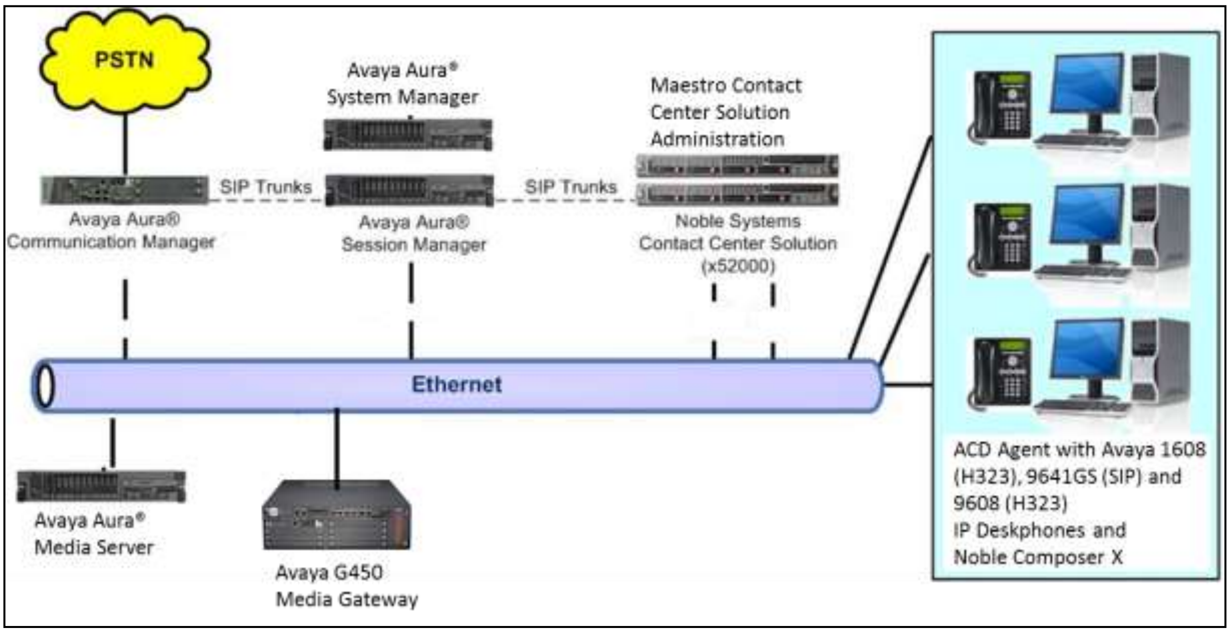


Figure 1: Noble Systems Contact Center Solution with Avaya Aura® Communication Manager and Avaya Aura® Session Manager

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 in Virtual Environment	R017x.00.0.441.0 7.0.1.0.0-FP1
Avaya G450 Media Gateway	37.19.0
Avaya Aura® Media Server in Virtual Environment	7.7.019 (FP1)
Avaya Aura® System Manager running on Virtualized Environment	7.0.1.0
Avaya Aura® Session Manager running on Virtualized Environment	7.0.1.0.701007
Avaya 9641G, IP Deskphone (SIP)	7.0.1
Avaya 9608 IP Deskphone (H.323)	6.6029
Avaya 1608-I IP Deskphones (H.323)	1.3 Release 9
The Noble Enterprise Contact Solution on CentOS	Version 10 6.7
Maestro Contact Center Solution Administration on Windows Server 2012 R2	Version 8.1 R2 64bit
Composer X Agent Desktop and Designer on Windows 10 Pro	version 3.1 2015 32Bit

5. Configure Avaya Aura® Communication Manager

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

- Verify license
- Administer system parameters features
- Administer SIP trunk group
- Administer SIP signaling group
- Administer SIP trunk group members
- 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, a separate set of codec, network region, trunk group, and signaling group were used for integration with Noble Systems.

5.1. Verify License

Log into 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.

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

IP PORT CAPACITIES                                              USED
                     Maximum Administered H.323 Trunks: 12000 10
           Maximum Concurrently Registered IP Stations: 18000 3
             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 System Parameters Features

Use the "change system-parameters features" command to allow for trunk-to-trunk transfers.

For ease of interoperability testing, the **Trunk-to-Trunk Transfer** field was set to "all" to enable all trunk-to-trunk transfers on a system wide basis. Note that this feature poses significant security risk, and must be used with caution. For alternatives, the trunk-to-trunk feature can be implemented on the Class Of Restriction or Class Of Service levels. Refer to [1] for more details.

```
change system-parameters features                               Page   1 of  19
                            FEATURE-RELATED SYSTEM PARAMETERS
                               Self Station Display Enabled? n
                                    Trunk-to-Trunk Transfer: all
               Automatic Callback with Called Party Queuing? n
    Automatic Callback - No Answer Timeout Interval (rings): 3
                       Call Park Timeout Interval (minutes): 10
        Off-Premises Tone Detect Timeout Interval (seconds): 20
                                 AAR/ARS Dial Tone Required? y

              Music (or Silence) on Transferred Trunk Calls? no
                      DID/Tie/ISDN/SIP Intercept Treatment: attd
    Internal Auto-Answer of Attd-Extended/Transferred Calls: transferred
                  Automatic Circuit Assurance (ACA) Enabled? n


             Abbreviated Dial Programming by Assigned Lists? n
       Auto Abbreviated/Delayed Transition Interval (rings): 2
                    Protocol for Caller ID Analog Terminals: Bellcore
    Display Calling Number for Room to Room Caller ID Calls? n
```

5.3. Administer SIP Trunk Group

Use the “add trunk-group n” command, where “n” is an available trunk group number, in this case “52”. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** “sip”
- **Group Name:** A descriptive name.
- **TAC:** An available trunk access code.
- **Service Type:** “tie”

```
add trunk-group 52                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 52                   Group Type: sip          CDR Reports: y
  Group Name: Noble Systems              COR: 1        TN: 1        TAC: 1052
   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:
                                                     Number of Members: 0
```

Navigate to Page 3, and enter “private” for Numbering Format.

```
add trunk-group 52                                              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
```

5.4. Administer SIP Signaling Group

Use the "add signaling-group n" command, where "n" is an available signaling group number, in this case "52". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Group Type:** "sip"
- **Transport Method:** "tls"
- **Near-end Node Name:** An existing C-LAN node name or "procr" in this case.
- **Far-end Node Name:** The existing Session Manager node name.
- **Near-end Listen Port:** An available port for integration with Noble Systems.
- **Far-end Listen Port:** The same port number as in **Near-end Listen Port**.
- **Far-end Network Region:** An existing network region to use with Noble Systems.
- **Far-end Domain:** The applicable domain name for the network.

For **Direct IP-IP Audio Connections**, enter "n" since Noble Systems does not support shuffling.

```
add signaling-group 52                                          Page   1 of   1
                                SIGNALING GROUP

 Group Number: 52              Group Type: sip
  IMS Enabled? n         Transport Method: tls
        Q-SIP? n                                                 SIP Enabled LSP? n
     IP Video? n                                       Enforce SIPS URI for SRTP? y
  Peer Detection Enabled? y  Peer Server: Others



   Near-end Node Name: procr                    Far-end Node Name: SM-VM
 Near-end Listen Port: 5061                   Far-end Listen Port: 5061
                                           Far-end Network Region: 7
                                    Far-end Secondary Node Name:
Far-end Domain: bvwdev.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? n
Session Establishment Timer(min): 3                          IP Audio Hairpinning? n
         Enable Layer 3 Test? y
                                                       Alternate Route Timer(sec): 6
```

5.5. Administer SIP Trunk Group Members

Use the “change trunk-group n” command, where “n” is the trunk group number from **Section 5.3**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Signaling Group:** The signaling group number from **Section 5.4**.
- **Number of Members:** The desired number of members, in this case “10”.

```
add trunk-group 52                                              Page   1 of  21
                                TRUNK GROUP

Group Number: 52                   Group Type: sip          CDR Reports: y
  Group Name: Noble Systems               COR: 1        TN: 1        TAC: 1052
   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: 52
                                                     Number of Members: 10
```

5.6. 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 5.4** (For this testing 7 was used).

For **Authoritative Domain**, enter the applicable domain for the network. Enter a descriptive **Name**. Enter "no" for **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio**, as shown below. For **Codec Set**, enter an available codec set number for integration with Noble Systems.

```
change ip-network-region 7                                      Page   1 of  20
                               IP NETWORK REGION
  Region: 7
Location: 1       Authoritative Domain: bvwdev.com
    Name: Noble Systems
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 7              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 Call Control PHB Value: 46
        Audio PHB Value: 46
        Video PHB Value: 26
```

Navigate to **Page 4**, and specify this codec set to be used for calls with the network region used by the Avaya endpoints and with the PSTN. There are other network regions in the compliance testing; network region "1" is used by the Avaya endpoints, and network region "4" is used with the trunk to the PSTN.

```
change ip-network-region 7                                      Page   4 of  20

 Source Region: 7     Inter Network Region Connection Management     I        M
                                                                     G  A     t
 dst codec direct   WAN-BW-limits   Video      Intervening      Dyn  A  G     c
 rgn  set   WAN  Units    Total Norm  Prio Shr Regions          CAC  R  L     e
 1    7     y    NoLimit                                             n        t
 2
 3
 4    7
 5
 6
 7    7                                                                 all
 8
```

5.7. Administer IP Codec Set

Use the "change ip-codec-set n" command, where "n" is the codec set number from **Section 5.6**. Update the audio codec types in the **Audio Codec** fields as necessary. Note that Noble Systems supports the G.711 and G.729 codec variants. The codec shown below were used in the compliance testing.

```
change ip-codec-set 7                                               Page   1 of   2

                          IP Codec Set

    Codec Set: 7

    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:
```

5.8. Administer Route Pattern

Use the "change route-pattern n" command, where "n" is an existing route pattern number to be used to reach Noble Systems, in this case "52". Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Pattern Name:** A descriptive name.
- **Grp No:** The SIP trunk group number from **Section 5.3**.
- **FRL:** A level that allows access to this trunk, with 0 being least restrictive.

```
change route-pattern 52                                             Page   1 of   3
                    Pattern Number: 52  Pattern Name: Noble Systems
                             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: 52   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
```

5.9. Administer Private Numbering

Use the "change private-numbering 0" command, to define the calling party number to send to Noble Systems. Add an entry for the trunk group defined in **Section 5.3**. In the example shown below, all calls originating from a 5-digit extension beginning with 4 and routed to trunk group 52 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  4                52                           5     Total Administered: 1
                                                           Maximum Entries: 540
```

5.10. Administer Uniform Dial Plan

This section provides a sample AAR routing used for routing calls with dialed digits 52xxx to Noble Systems. 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 52xxx, 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

 52              5   0                  aar  n
```

5.11. Administer AAR Analysis

Use the "change aar analysis 0" command, and add an entry to specify how to route calls to 52xxx. In the example shown below, calls with digits 52xxx will be routed as an AAR call using route pattern "52" from **Section 5.8**.

```
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
    52                   5    5     52       unku         n
```

5.12. Administer ISDN Trunk Group

Use the “change trunk-group n” command, where “n” is the existing trunk group number used to reach the PSTN, in this case “450”.

Navigate to **Page 3**. For **Modify Tandem Calling Number**, enter “tandem-cpn-form” to allow for the calling party number from Noble Systems to be modified.

```
change trunk-group 450                                          Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                 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
                                              UUI IE Treatment: service-provider

                                                  Replace Restricted Numbers? n
                                                 Replace Unavailable Numbers? n
                                                        Send Connected Number: y
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


 DSN Term? n
```

5.13. 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 Noble Systems.

In the example shown below, all calls originating from a 5-digit extension beginning with 5 and routed to trunk group 450 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   4                 450                90884            pub-unk
 5   5                 450                90884            pub-unk
```

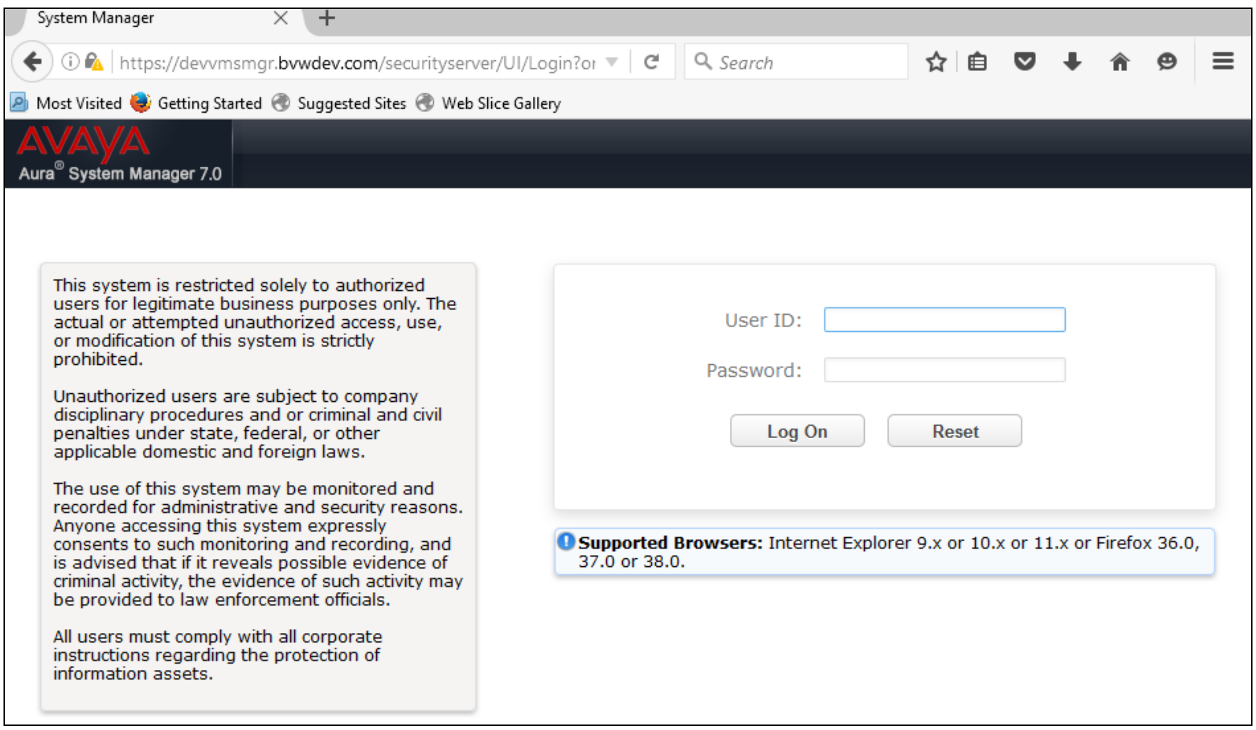
6. Configure Avaya Aura® Session Manager

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

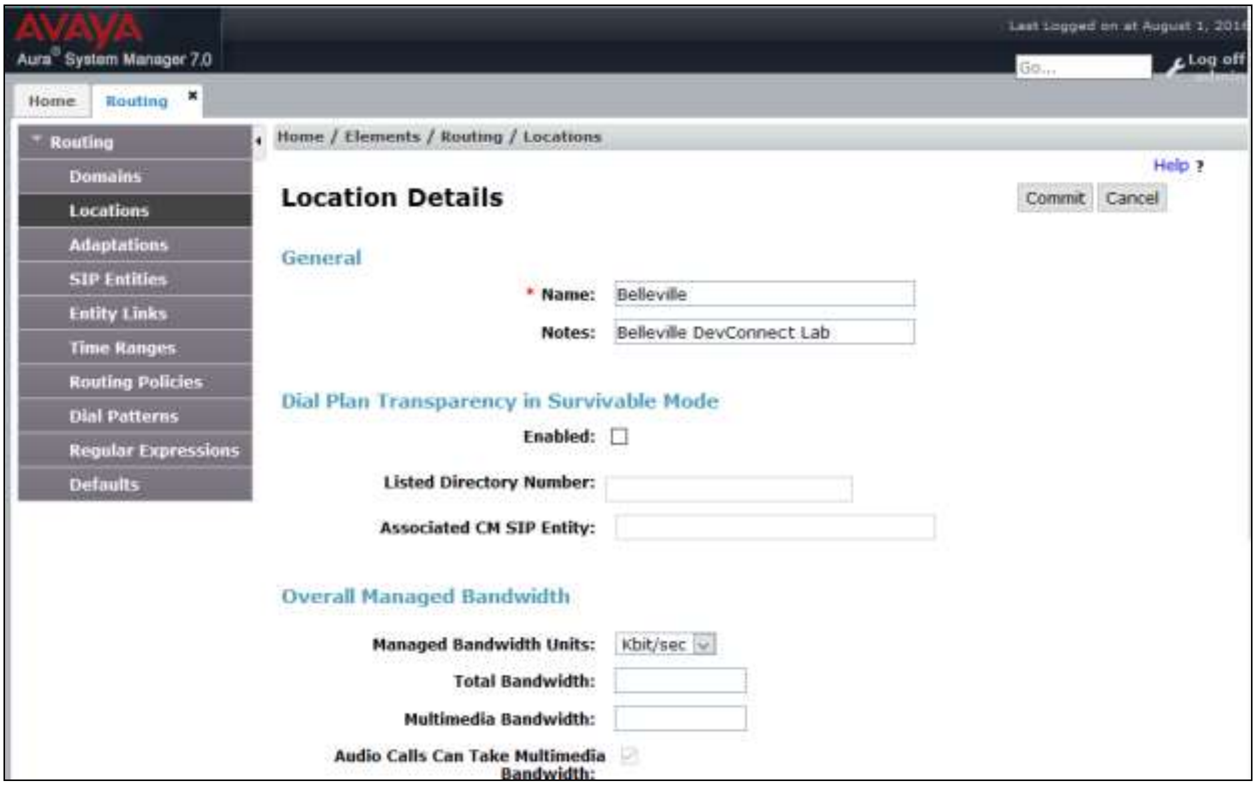
Access the System Manager web interface by using the URL "https://ip-address" in an Internet browser window, where "ip-address" is the IP address of the System Manager server. Log in using the appropriate credentials.



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 Noble Systems.

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.

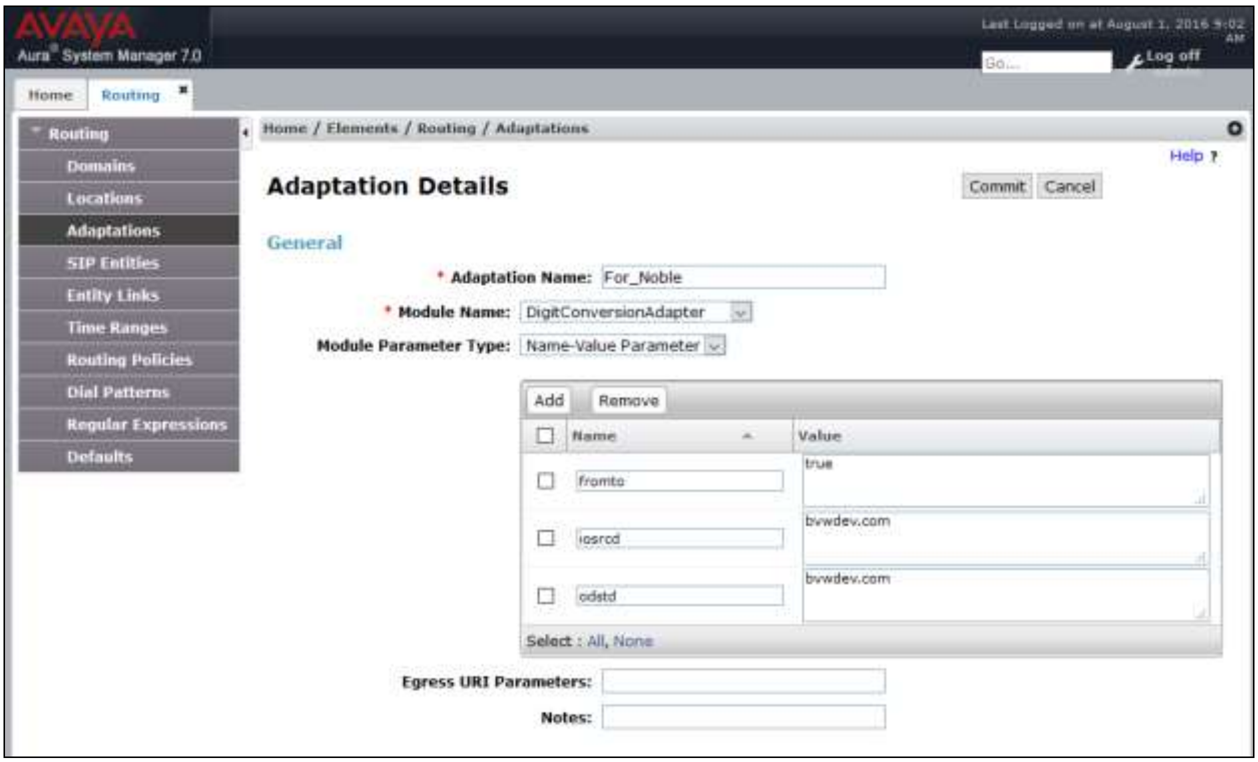


6.3. Administer Adaptations

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

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 "osrcd=bvwdev.com odstd=bvwdev.com, where "bvwdev.com" is the applicable domain. This will set the source and destination domains for all incoming and outgoing calls for Noble Systems.



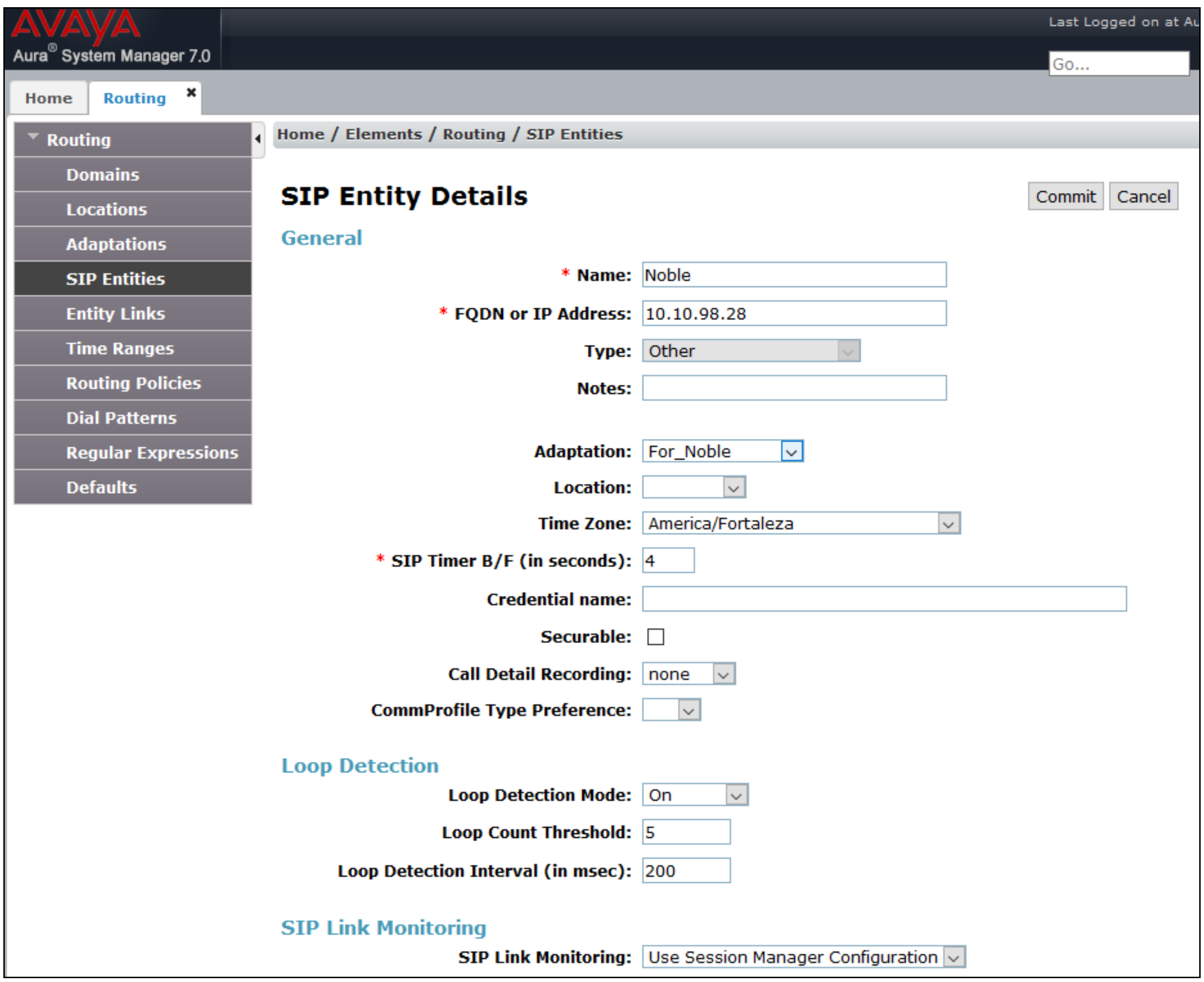
6.4. Administer SIP Entities

Add new SIP entity for Noble Systems.

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

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 the Contact Center Solution server.
- **Type:** "Other"
- **Adaptation:** Select the Noble Systems adaptation name from **Section 6.3**.
- **Location:** Select the Noble Systems location name from **Section 6.2**.
- **Time Zone:** Select the applicable time zone.



6.5. Administer Entity Links

Add new entity link for Noble Systems.

Select **Routing > Entity Links** from the left pane, and click **New** in the subsequent screen (not shown) to add a new entity link for IPC. 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 "SM_Noble".
- **Protocol:** "UDP"
- **Port:** "5060"
- **SIP Entity 2:** The Noble Systems entity name from **Section** .
- **Port:** "5060"
- **Connection Policy:** "Trusted"

6.6. Administer Routing Policies

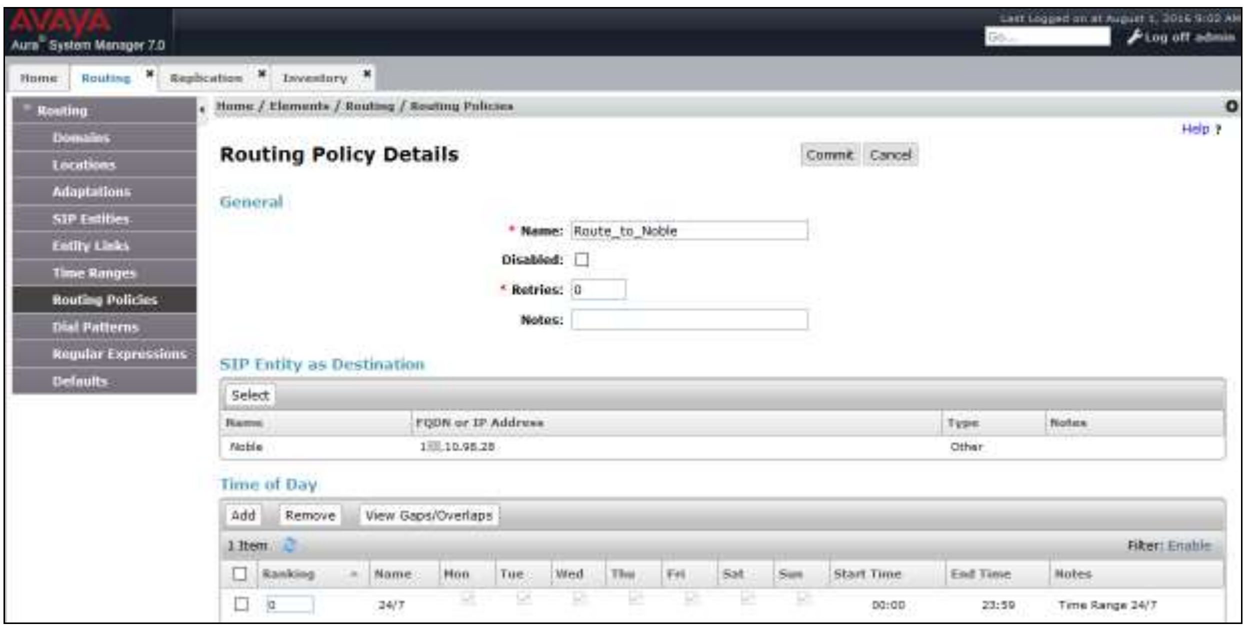
Add new routing policy for Noble Systems.

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

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 Noble Systems entity name from **Section 6.4** in the listing (not shown).

Retain the default values in the remaining fields.



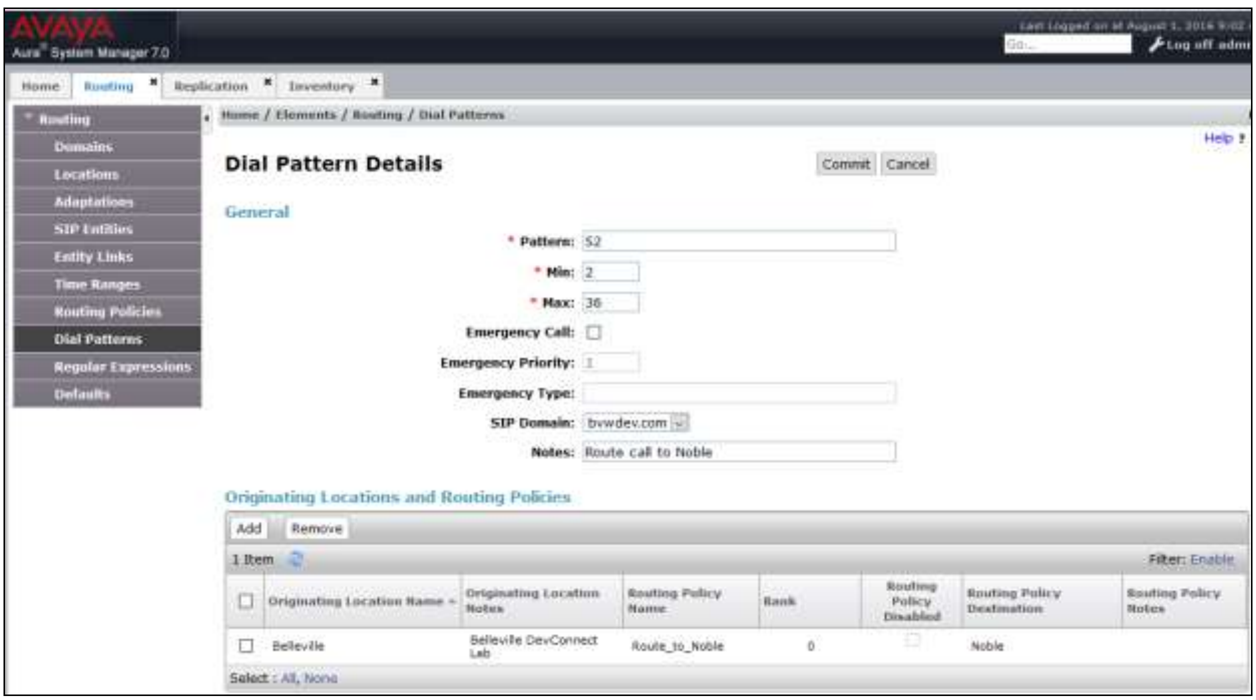
6.7. Administer Dial Patterns

Add a new dial pattern for Noble Systems.

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 Noble Systems. 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 5.4**.
- **Notes:** Any desired description.

In the **Originating Locations and Routing Policies** sub-section, click **Add** and create a new policy for reaching Noble Systems as shown below.



7. Configure Noble Systems Contact Center Solution

This section provides the procedures for configuring Contact Center Solution. The procedures include the following areas:

- Administer domain resolution
- Administer mappings
- Launch Maestro
- Administer calling number
- Administer routing

The configuration of Contact Center Solution is typically performed by Noble Systems technicians. The procedural steps are presented in these Application Notes for informational purposes.

7.1. Administer Domain Resolution

Log in to the Linux shell of the Contact Center Solution server with the appropriate credentials. Navigate to the **/etc** directory. Open the **hosts** file, and add an entry to resolve the network domain of Noble Contact Center Solution Linux server as shown below.

```
# Do not remove the following line, or various programs
# that require network functionality will fail.
127.0.0.1               localhost
20.32.39.170            sipfort
10.10.97.28            avayafdev1 avayafdev1.noblesys.com
```

7.2. Administer Mappings

Navigate to the **/etc/asterisk** directory. Open the **hannibal.xml** file, and navigate to the stations mapping entry. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **Map name:** "AVAYAStations"
- **technology:** "SIP"
- **pattern:** "\b\d{x}\b" where "x" is the number of digits in the station extensions.
- **suffix:** The applicable network domain, in this case "bvwdev.com".
- **format:** The desired codec, in this case "G729" followed by "ULAW".

In the compliance testing, the agent station extensions on Communication Manager were "4xxxx".

```
<?xml version="1.0"?>

<TechnologyMap>
      <!-- The following is a list of examples for the various technology types -->

      <Map name="AVAYAStations" technology="SIP" pattern="\b.*\b" prefix=""
suffix="@135.10.97.228" formats="ULAW|ALAW|GSM" maxNumberOfUses="50"
hannibalChanBeg="-1" hannibalChanEnd="-1" beginningChannelNumber="-1"
endingChannelNumber="-1" supportsInbound="true" supportsOutbound="true" />

</TechnologyMap>
```

7.3. Launch Maestro

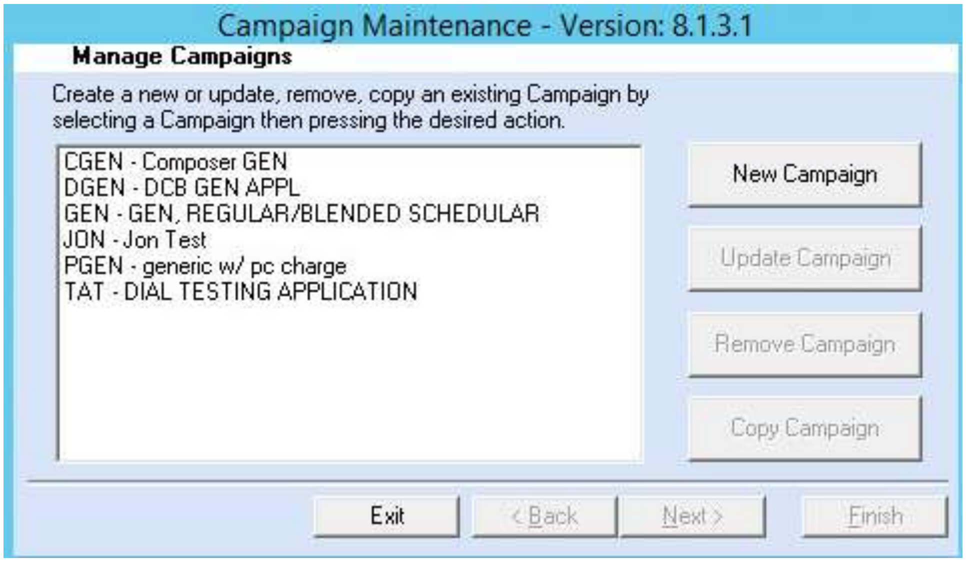
From the Contact Center Solution server, launch the Maestro application by double-clicking the **Maestro** icon shown below, which was created as part of installation.

The screen below is displayed. Enter the appropriate credentials.

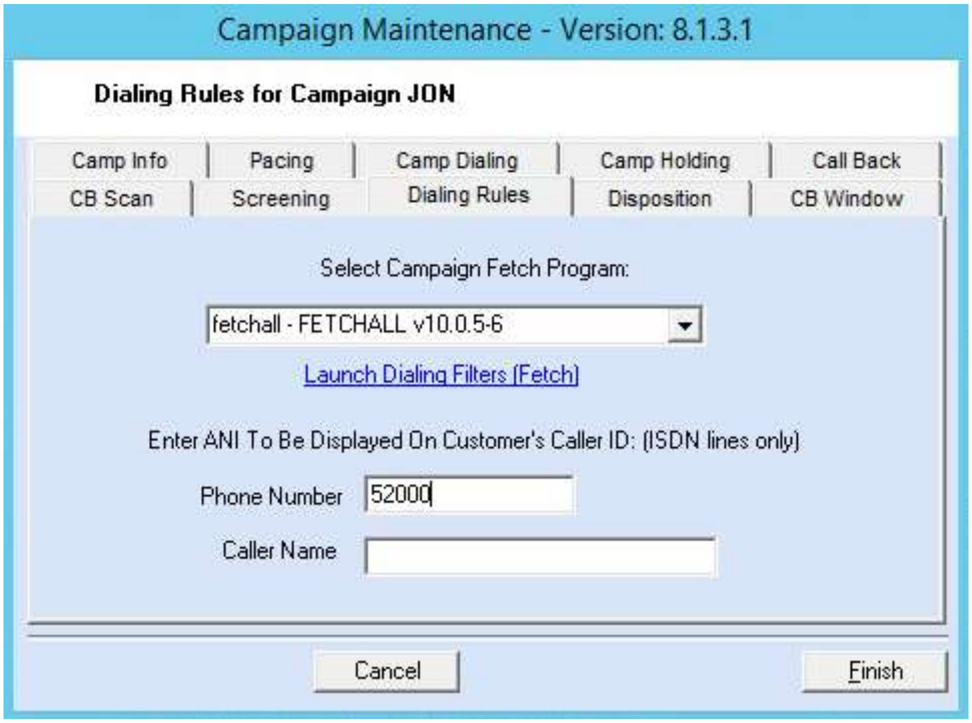
7.4. Administer Calling Number

The **MANAGER PORTAL** screen is displayed next. Double click on **Campaign Setup > Campaign Maintenance** in the left pane.

The **Campaign Maintenance** screen is displayed. Select **JON- Jon Test** and click **Update Campaign**.

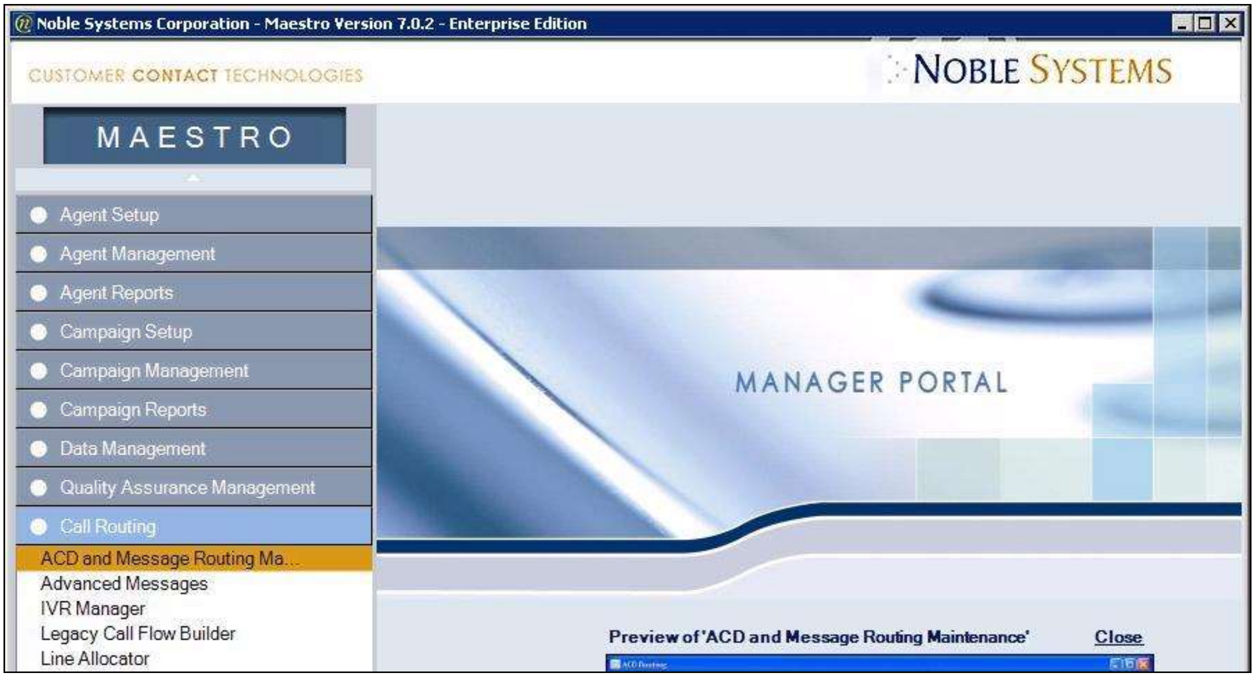


The **Campaign Maintenance** screen is updated. Select **Dialing Rules** to display the screen below. For **Phone Number**, enter the applicable extension to be used as calling party extension for outbound calls from Noble Systems, in this case “52000”.



7.5. Administer Routing

From the **MANAGER PORTAL** screen, double-click on **Call Routing > ACD and Message Routing Maintenance** from the left pane.



The **ACD Routing** screen is displayed. Select **Add** from the bottom of the screen (not shown) to add a new entry. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **ListId:** A desired and unique value.
- **DNIS:** The assigned Contact Center Solution extension from **Section** □.
- **Group:** The applicable group number, in this case '1".
- **Campaign:** "INB"
- **Description:** A desired description.

ACD Routing

ListId	DNIS	Group	Campaign	Open Message	Closed Message	Description	MaxHold	NextDNIS
11113	52000	1	JON	2 -	2 -	inbound to g1		
80012563	g2	256	JON	2 -	(None)	Hold Group		
80010991	5556	1	JON	(None)	(None)	DEFAULT OUT...		
80012560	g1	256	JON	2 -	(None)	Hold Group		
11111	5557	2	JON	2 -	(None)	DEFAULT OUT...		
80012562	5559	2	JON	(None)	(None)	DIAL NOW		
11110	5554	1	JON	(None)	(None)	DIAL NOW		
80012561	52001	2	JON	2 -	2 -	inbound to g2		
80020990	g2	99	JON	2 -	(None)	hold		
80010990	g1	99	JON	2 -	(None)	hold		

8. Verification Steps

This section provides tests that can be performed to verify proper configuration of Communication Manager, Session Manager, and Contact Center Solution.

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 5.3**. Verify that all trunks are in the "in-service/idle" state as shown below.

```
status trunk 52

                            TRUNK GROUP STATUS

Member   Port     Service State      Mtce Connected Ports
                                     Busy

0052/001 T00021   in-service/idle    no
0052/002 T00022   in-service/idle    no
0052/003 T00023   in-service/idle    no
0052/004 T00024   in-service/idle    no
0052/005 T00025   in-service/idle    no
0052/006 T00026   in-service/idle    no
0052/007 T00027   in-service/idle    no
0052/008 T00028   in-service/idle    no
0052/009 T00029   in-service/idle    no
0052/010 T00030   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 5.4**. Verify that the signaling group is "in-service" as indicated in the **Group State** field shown below.

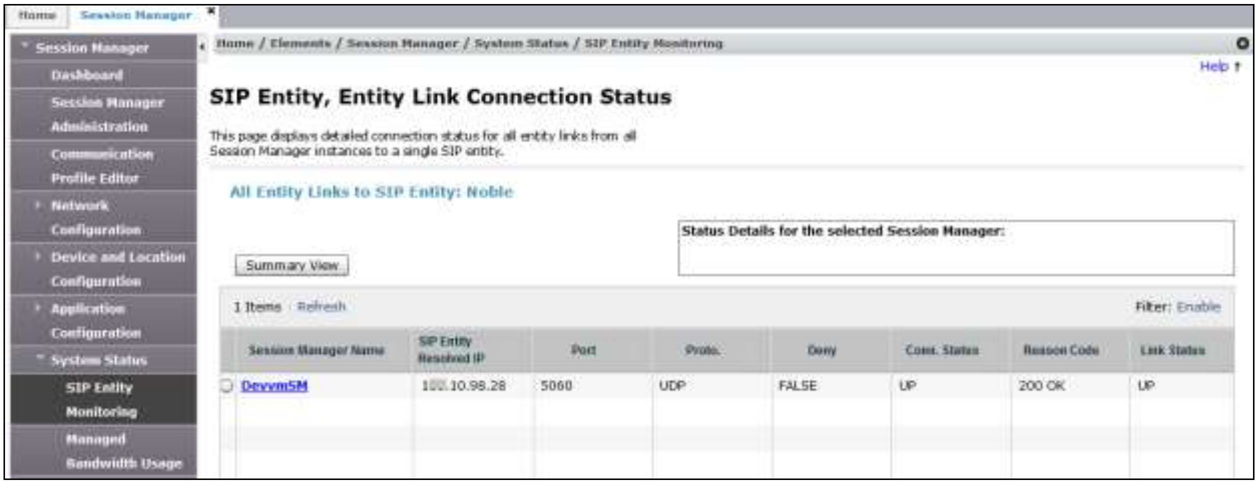
```
status signaling-group 52
                        STATUS SIGNALING GROUP

        Group ID: 52
      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.The **SIP Entity, Entity Link Connection Status** screen is displayed. Verify that **Conn Status** and **Link Status** are "Up", as shown below.

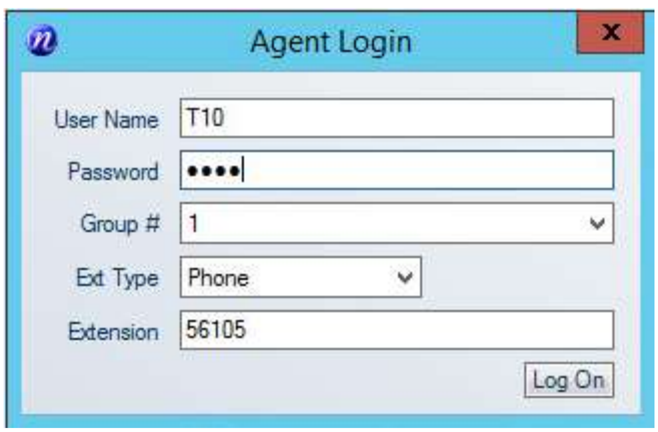


8.3. Verify Noble Systems Contact Center Solution

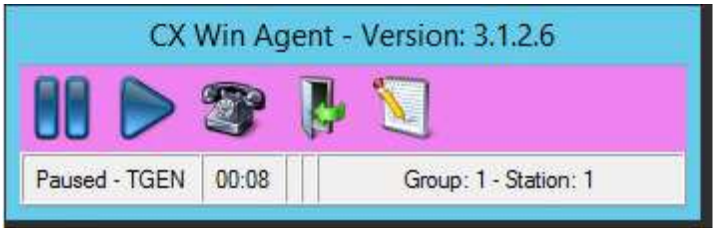
Prior to verification, start an outbound campaign on Contact Center Solution.

From the agent PC, access the Composer web-based interface by using the URL "http://ip-address/NobleWebAgent" in an Internet browser window, where "ip-address" is the IP address of the Composer Web Server. The **Welcome to Composer 9** screen is displayed. Click **Login**.

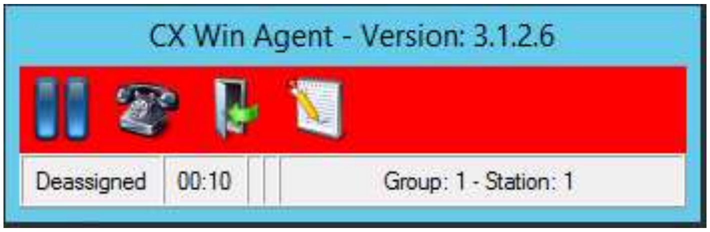
The pop-screen below is displayed. For **User Name** and **Password**, enter the appropriate agent credentials. For **Group**, select the applicable group number, in this case "1". Select "Other" for **Ext Type**. For **Extension**, enter an available agent station extension from **Section** □, and click **Log On**.



The screen is updated as shown below. Click on the **Resume** icon to log into Contact Center Solution. Verify that Contact Center Solution initiates a dedicated connection to the agent, with the call ringing at the agent's telephone.



Answer the call at the agent's telephone. Verify that the screen is updated to reflect agent successfully logged into Contact Center Solution, and is waiting for a call, as shown below.



Verify that Contact Center Solution successfully placed an outbound call to a PSTN user, with the call ringing at the PSTN user.

Answer the call at the PSTN user. Verify that the agent is connected to the PSTN user with two-way talk paths, and that the agent screen is updated to reflect the connected call, as shown below.

9. Conclusion

These Application Notes describe the configuration steps required for Noble Systems Contact Center Solution to successfully interoperate with Avaya Aura® Communication Manager using Avaya Aura® Session Manager. All feature and serviceability test cases were completed with observations noted in **Section 2.2**.

10. Additional References

This section references the product documentation that is relevant to these Application Notes. Documentation for Avaya products may be obtained via http://support.avaya.com.

1. Administering Avaya Aura® Communication Manager, Release 7.0.3, Document 03-300509, Issue 10, June 2016.
2. Administering Avaya Aura® Session Manager, Release 7.0, Issue 7, Jan 2016.
3. *Noble Systems Composer 9 version 2011.1.1 User Manual*, Revised June 27, 2011, available at http://nobleusersgroup.noblesys.com.

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.