Avaya Solution & Interoperability Test Lab

Application Notes for DuVoice DV2000 VS6 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager – Issue 1.0

Abstract

These Application Notes contains instructions for configuring DuVoice DV2000 VS6 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager.

Readers should pay attention to section 2, in particular the scope of testing as outlined in Section 2.1 as well as the 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

This application notes contain instruction for configuring DuVoice DV2000 VS6 with Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and various Avaya endpoints. In the compliance testing, SIP trunks were used between the DuVoice DV2000 VS6 Voice Messaging System and Avaya Aura® Session Manager.

DuVoice DV2000 VS6 is a hospitality application that provides voicemail, automated attendant, and wake-up call features. The compliance testing focused on integrating the DuVoice DV2000 VS6 with Avaya Aura® Communication Manager and Aura® Session Manager.

2. General Test Approach and Test Results

The general test approach was to manually place intra-switch calls and inbound trunk calls that were ultimately answered by the DuVoice DV2000 VS6. Depending on the type of call, the user had the option to leave a voicemail message, retrieve a voicemail message, schedule a wake-up call, or transfer to another extension. All inbound calls to the DuVoice DV2000 VS6 hunt group on Communication Manager were routed to Session Manager, which were then answered by DV2000 VS6 with the automated attendant greeting. Internal calls that were unanswered covered to the DV2000 VS6 hunt group. DV2000 VS6 would answer these calls with the voice mailbox greeting of the subscriber. Lastly, internal calls placed to DV2000 VS6 directly were answered by the DV2000 VS6 with the voicemail menu of the originating extension with an option to retrieve messages. For serviceability testing, the DV2000 VS6 and Communication Manager were each restarted separately.

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 testing included feature and serviceability testing.
The feature testing focused on exercising the core features of the DV2000 VS6 to validate the integration interface to Session Manager using SIP Trunks. This included the automated attendant, voicemail, wakeup call and performing guest check-in and checkout using the Hospitality Tester and InnerDesk functionality. The serviceability testing introduced failure scenarios to verify operation of the DuVoice DV2000 VS6 after failure recovery.

2.2. Test Result

Interoperability testing of the sample configuration was completed with successful results for DuVoice DV2000 VS6. The following observations were noted:

- Newly created SIP guest mailboxes will not have guest name updated for the first time the room is checked in for example, if guest check in room 200, the guest name will not be updated on mailbox 200, the name still show default name.
- If the user name of a SIP deskphone is manually changed on Session Manager or via the Communication Manager SAT, DuVoice DV2000 VS6 will not aware of this name change. Therefore, there will be name mismatch between the deskphone and DuVoice DV2000 VS6 system.
- Guest move to another vacant room will not move DND setup to the new room.
- Guest move to another vacant room will move Wakeup call setup to the new room but the Wakeup call will also still exist for the old room. DuVoice will fix it in future release.

2.3. Support

Technical support on DuVoice DV2000 VS6 can be obtained by phone or email as follows:

- **Phone:** (425) 250-2393
- **Email:** support@duvoice.com

3. Reference Configuration

Figure 1 illustrates the test configuration used during compliance testing and consisted of the following:

- Avaya Aura® Communication Manager with Avaya G450 Media Gateway
- Avaya Aura® Media Server
- Avaya Aura® Session Manager
- Avaya Aura® System Manager
- DuVoice DV2000 VS6 running on Windows 10 Pro

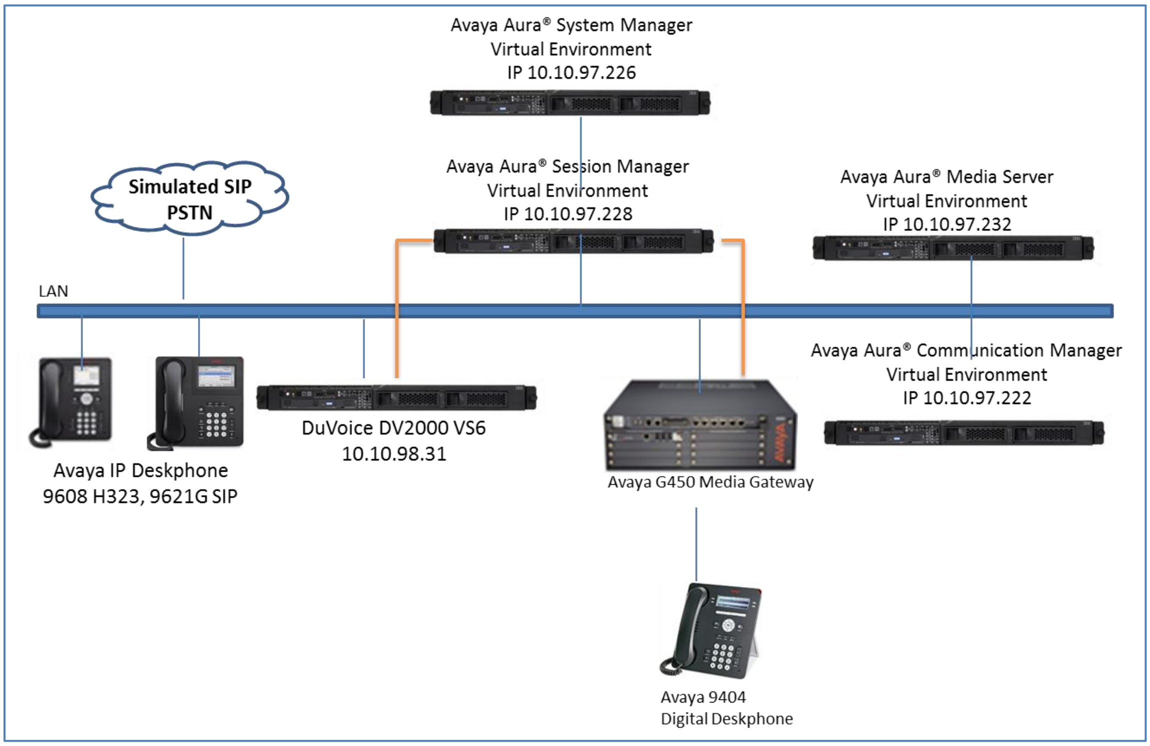


Figure 1: Reference Configuration

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 with Avaya G450 Media Gateway	7.0 SP2
Avaya Aura® Media Server	7.7 SP2 (v.7.7.0.281)
Avaya Aura® Session Manager in Virtual Environment	7.0.1 SP2
Avaya Aura® System Manager in Virtual Environment	7.0.1.1
DuVoice DV2000 VS6 running on Windows 10 Pro in Virtual Environment	6.0.0.45

5. Configure Avaya Aura® Communication Manager

This section provides steps for configuring Communication Manager. All configuration for Communication Manager is done through System Access Terminal (SAT). It is assumed that the SIP trunk between Communication Manager and Session Manager is already in place and will not be mentioned in this Application Notes.

5.1. Administer IP Network Region

Use the **change ip-network-region *n*** command to configure a network region, where ***n*** is an existing network region.

Configure this network region as follows:

- Set **Location** to **1.**
- Set **Codec Set** to **1.**
- Set **Intra-region IP-IP Direct Audio** to **yes.**
- Set **Inter-region IP-IP Direct Audio** to **yes.**
- Enter **Authoritative Domain**, e.g. bvwdev.com.

```
change ip-network-region    1                               Page   1 of  20
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: bvwdev.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS
 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
```

5.2. Administer IP Codec Set

Use the **change ip-codec-set *n*** command to configure IP codec set, where ***n*** is an existing codec set number.

Configure this codec set as follows. On **Page 1**, set **Audio Codec 1** to **G.711MU.**

```
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:
```

5.3. Administer IP Node Names

Use the **change node-names ip** command to add an entry for Session Manager. For compliance testing, **SM-VM** and **10.10.97.228** entry was added.

```
display node-names ip
                                  IP NODE NAMES
    Name               IP Address
GW-G450              10.10.97.223
SM-VM                10.10.97.228
procr                10.10.97.222
```

5.4. Administer SIP Signaling Group

Use the **add signaling-group *n*** command to add a new signaling group, where ***n*** is an available signaling group number.

Configure this signaling group as follows:

- Set **Group Type** to **sip.**
- Set **Near-end Node Name** to **procr.**
- Set **Far-end Node Name** to the configured Session Manager.
- Set **Far-end Network Region** to the configured region in **Section 5.1**, i.e. 1.
- Enter a **Far-end Domain,** e.g. bvwdev.com.

```
add signaling-group 1                                       Page   1 of   2
                                SIGNALING GROUP

 Group Number: 1              Group Type: sip
  IMS Enabled? n        Transport Method: tls
        Q-SIP? 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: 1
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? 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
```

Note: Signaling Group, Trunk Group and Route Pattern for simulated PSTN calls and inter-site calls over ISDN/PRI and SIP were pre-configured and are not shown in this document.

5.5. Administer SIP Trunk Group

Use the **add trunk-group *n*** command to add a trunk group, where ***n*** is an available trunk group number.

Configure this trunk group as follows, on **Page 1**:

- Set **Group Type** to **sip.**
- Enter a **Group Name**, e.g. SM.
- Enter a valid **TAC**, e.g. *001.
- Set **Service Type** to **tie.**
- Enter **Signaling Group** value to the signaling group configured in **Section 5.4**, i.e. 1.
- Enter a desired number in **Number of Member** field.

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

Group Number: 1                    Group Type: sip            CDR Reports: y
  Group Name: SM-VM                       COR: 1        TN: 1        TAC: *001
   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: 1
                                                     Number of Members: 25
```

On **Page 3**:

- Set **Numbering Format** to private.

```
add trunk-group 1                                              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
```

5.6. Administer Route Pattern

Use the **change route-pattern *n*** command to configure a route pattern, where ***n*** is an available route pattern.

Configure this route pattern as follows:

- Enter a name in the **Pattern Name** field.
- For preference 1, set **Grp No** to the trunk group configured in **Section 5.5**, i.e. 1.
- For preference 1, set **FRL** to **0.**

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1      Pattern Name: To SM on VM
    SCCAN? n    Secure SIP? n     Used for SIP stations? n

    Grp FRL NPA Pfx Hop Toll No.  Inserted                            DCS/ IXC
    No          Mrk Lmt List Del  Digits                              QSIG
                             Dgts                                     Intw
 1: 1    0                    0                                        n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM Sub  Numbering LAR
    0 1 2 M 4 W     Request                                 Dgts Format
 1: y y y y y n  n            rest                               lev1-pvt  none
 2: y y y y y n  n            rest                                         none
```

5.7. Administer Hunt Group

Use the **add hunt-group *n*** command to configure a hunt group, where ***n*** is an available hunt group number.

Configure the hunt group as follows:

- Type a descriptive name in **Group Name** field.
- Type in a available extension number for **Group Extension.**

```
add hunt-group 4                                                Page   1 of  60
                                  HUNT GROUP

            Group Number: 4                             ACD? n
              Group Name: DuVoice Voicemail           Queue? n
         Group Extension: 56604                      Vector? n
              Group Type: ucd-mia              Coverage Path:
                      TN: 1        Night Service Destination:
                     COR: 1                 MM Early Answer? n
           Security Code:             Local Agent Preference? n
 ISDN/SIP Caller Display:
```

5.8. Administer Coverage Path

Use the **add coverage path *n*** command to add a coverage path, where ***n*** is available coverage path number.

Configure the coverage path as follows:

- Under **COVERAGE POINTS**, set **Point1** to the hunt group that was configured in the previous section. e.g., h4, where h stands for hunt group and 4 is the hunt group number.

```
add coverage path 4                                        Page   1 of   1
                                 COVERAGE PATH

                   Coverage Path Number: 4
     Cvg Enabled for VDN Route-To Party? n        Hunt after Coverage? n
                       Next Path Number:          Linkage

COVERAGE CRITERIA
    Station/Group Status    Inside Call     Outside Call
            Active?             n               n
              Busy?             y               y
       Don't Answer?            y               y        Number of Rings: 3
               All?             n               n
 DND/SAC/Goto Cover?            y               y
   Holiday Coverage?            n               n


COVERAGE POINTS
    Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h4             Rng: 3  Point2:
  Point3:                        Po\
]]
```

5.9. Administer Private Numbering

Use the **change private-numbering 1** command to define the calling party number to send to Session Manager.

Configure private numbering as follows:

- Add entries for trunk group configured in **Section 5.5.**

Note: For compliance testing, 5-digit extensions starting with “56” routed over trunk group 1 results in a 5-digit calling party number being sent over the trunk.

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

Ext Ext            Trk        Private          Total
Len Code           Grp(s)     Prefix           Len
 5  56             1                           5     Total Administered: 1
                                                        Maximum Entries: 540
```

5.10. Administer AAR Analysis

Use the **change aar analysis *n*** command to configure routing for hunt group extension number ***n***. For compliance testing, hunt group extension 56604 was used for routing calls to DV2000 VS6.

- Set **Dialed String** to hunt group extension, e.g. 566.
- Set **Min** and **Max** to 5 for 5-digit extensions.
- Set **Route Pattern** to pattern configured in **Section 5.6**, i.e. 1.
- Set **Call Type** to **aar.**

Note: During compliance test, dialed string 56 has call type set to lev0 to remove extra "+" in dialed string when DuVoice DV2000 VS6 (56604) try to send a notify message to SIP deskphone 56202 to turn on MWI, the extra "+" is added causing SM cannot route Notify message to appropriated phone

```
change aar analysis 25099                                       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
    53                    5    5     1         aar          n
    54                    5    5     1         aar          n
    56                    5    5     1         lev0         n
    566                   5    5     1         aar          n
```

5.11. Administer Stations

It is assumed that stations for guests or staff are already configured on Communication Manager. Please see document listed in Reference **Section 10** for how to create stations on Communication Manager. Below is the screenshot of a station used during the compliance test.

Detail of guest station 56203 with **Coverage Path** to DuVoice Server, which was created in **Section 5.8**. In this example, the call used coverage path 4 if there was no answer to forward to DuVoice DV2000 VS6.

```
change station 56203                                            Page   1 of   6
                                     STATION

Extension: 56203                        Lock Messages? n               BCC: 0
     Type: 9641SIPCC                    Security Code: *                TN: 1
     Port: S00052                     Coverage Path 1: 4               COR: 1
     Name: SIP203, Lab                Coverage Path 2:                 COS: 1
                                      Hunt-to Station:
STATION OPTIONS
                                          Time of Day Lock Table:
              Loss Group: 19
                                                Message Lamp Ext: 56203

        Display Language: english                 Button Modules: 0

          Survivable COR: internal
   Survivable Trunk Dest? y                         IP SoftPhone? y
```

```
                                         IP Video Softphone? n
                        Short/Prefixed Registration Allowed: default
```

On **Page 2** of station 56203, **MWI Served User Type** is set to sip-adjunct.

```
change station 56203                                          Page   2 of   6
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe
          LWC Activation? y                         Coverage Msg Retrieval? y
                                                               Auto Answer: none
             CDR Privacy? n                                 Data Restriction? n
                                                Idle Appearance Preference? n
 Per Button Ring Control? n                   Bridged Idle Line Preference? n
   Bridged Call Alerting? n                       Restrict Last Appearance? y
  Active Station Ringing: single

        H.320 Conversion? n          Per Station CPN - Send Calling Number?


    MWI Served User Type: sip-adjunct

                                                 Coverage After Forwarding? s

 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? y
  Emergency Location Ext: 56203         Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? n
```

Below is detail of H323 staff station used during compliance test, with Coverage Path to DuVoice DV2000 VS6 as configured in **Section 5.8**.

```
change station 56101                                            Page   1 of   5
                                     STATION

Extension: 56101                         Lock Messages? n               BCC: 0
     Type: 9608                          Security Code: *                TN: 1
     Port: S00000                      Coverage Path 1: 4               COR: 1
     Name: StationNameOneOOne          Coverage Path 2:                 COS: 1
                                       Hunt-to Station:               Tests? y
STATION OPTIONS
                                           Time of Day Lock Table:
              Loss Group: 19        Personalized Ringing Pattern: 1
                                                 Message Lamp Ext: 56101
            Speakerphone: 2-way               Mute Button Enabled? y
        Display Language: english                  Button Modules: 0
 Survivable GK Node Name:
          Survivable COR: internal              Media Complex Ext:
   Survivable Trunk Dest? y                          IP SoftPhone? y

                                              IP Video Softphone? n
                        Short/Prefixed Registration Allowed: default

                                             Customizable Labels? y
```

On **Page 2** of station, **MWI Served User Type** is set to sip-adjunct.

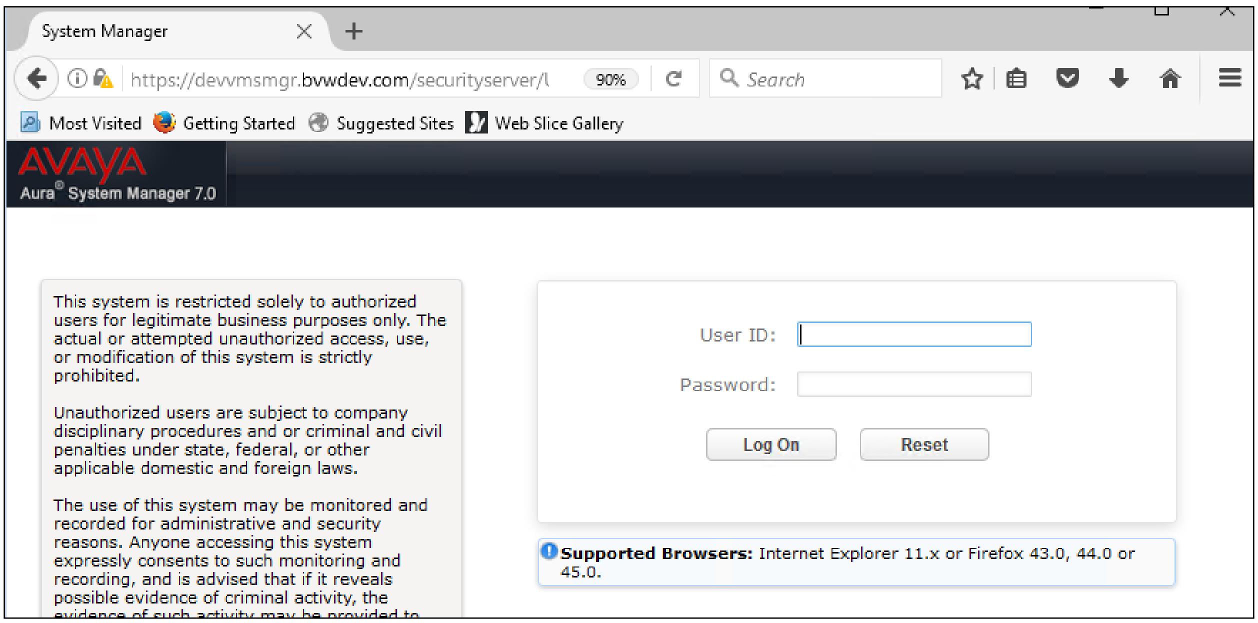
```
change station 56101                                        Page   2 of   5
                                     STATION
FEATURE OPTIONS
           LWC Reception: spe             Auto Select Any Idle Appearance? n
          LWC Activation? y                       Coverage Msg Retrieval? y
  LWC Log External Calls? n                                  Auto Answer: 
none
             CDR Privacy? n                            Data Restriction? n
   Redirect Notification? y                  Idle Appearance Preference? n
 Per Button Ring Control? n                Bridged Idle Line Preference? n
   Bridged Call Alerting? n                    Restrict Last Appearance? y
  Active Station Ringing: single
                                                        EMU Login Allowed? n
        H.320 Conversion? n          Per Station CPN - Send Calling Number?
       Service Link Mode: as-needed                     EC500 State: enabled
         Multimedia Mode: enhanced                 Audible Message Waiting? n
    MWI Served User Type: sip-adjunct           Display Client Redirection? n
                                               Select Last Used Appearance? n
                                                 Coverage After Forwarding? s
                                                   Multimedia Early Answer? n
 Remote Softphone Emergency Calls: as-on-local Direct IP-IP Audio Connections? 
y
  Emergency Location Ext: 56302         Always Use? n IP Audio Hairpinning? n
     Precedence Call Waiting? n
```

On **Page 4**, enter DuVoice DV2000 VS6 number 56604 in **voice-mail** field. On the deskphone, if user presses Message button, deskphone will automatically make a call to DuVoice DV2000 VS6 Messaging number, in this case it is 56604:

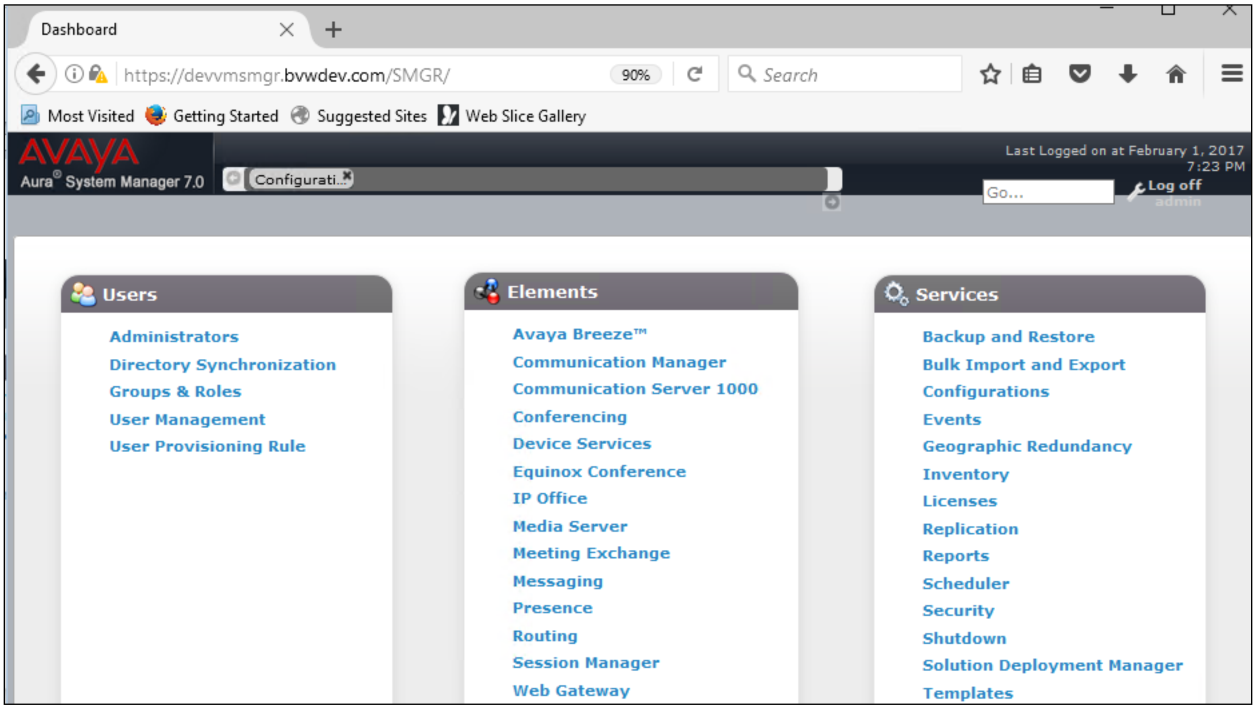
```
change station 56101                                   Page   4 of   5
                                     STATION
 SITE DATA
       Room:                                          Headset? n
       Jack:                                          Speaker? n
      Cable:                                         Mounting: d
      Floor: 2                                   Cord Length: 0
   Building: 1                                      Set Color:

ABBREVIATED DIALING
     List1:                    List2:                    List3:

BUTTON ASSIGNMENTS
 1: call-appr                           5: manual-in          Grp:
 2: call-appr                           6: after-call         Grp:
 3: auto-in           Grp:              7:
 4: aux-work    RC:   Grp:              8:

    voice-mail 56604
```

6. Configure Avaya Aura® Session Manager

Configuration of Avaya Aura® Session Manager is performed via System Manager. It is assumed that Session Manager, System Manager and Communication Manager are already setup and operational and are outside the scope of these App Notes. This section only describes steps needed to configure DuVoice DV2000 VS6 to work with Communication Manager and Session Manager. On the System Manager Web administration interface enter https://<ip-address>/SMGR URL in a web browser, where <ip-address> is the IP address of System Manager. Log in using appropriate credentials.



Once logged in, the following screen is displayed.

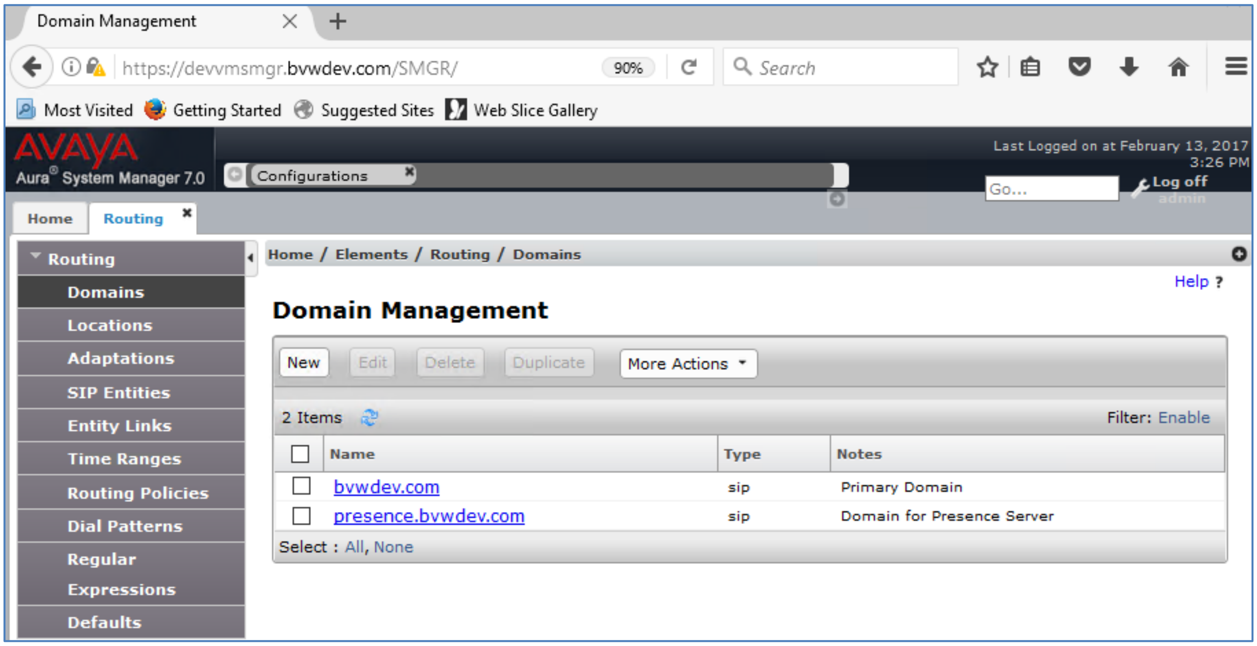


6.1. Administer SIP Domain

Navigate to **Home → Elements → Routing → Domains**, click on **New** button (not shown) and configure as follows:

- In **Name** field type in a domain (authoritative domain used in **Section 5.3**) i.e. bvwdev.com.
- Set **Type** to **sip.**

Click **Commit** to save changes.



6.2. Add Location

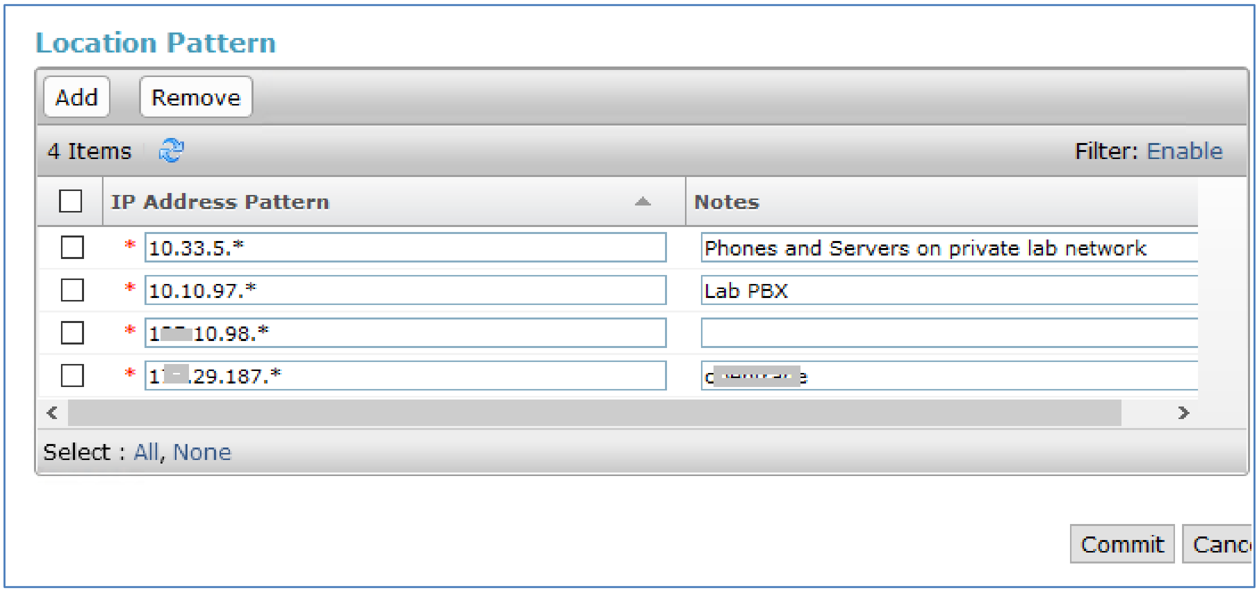
Navigate to **Home → Elements → Routing → Location**, click on **New** button (not shown) and configure as follows:

Under **General**: Type in a descriptive **Name.**

Under **Location Pattern** click on **New** (not shown):

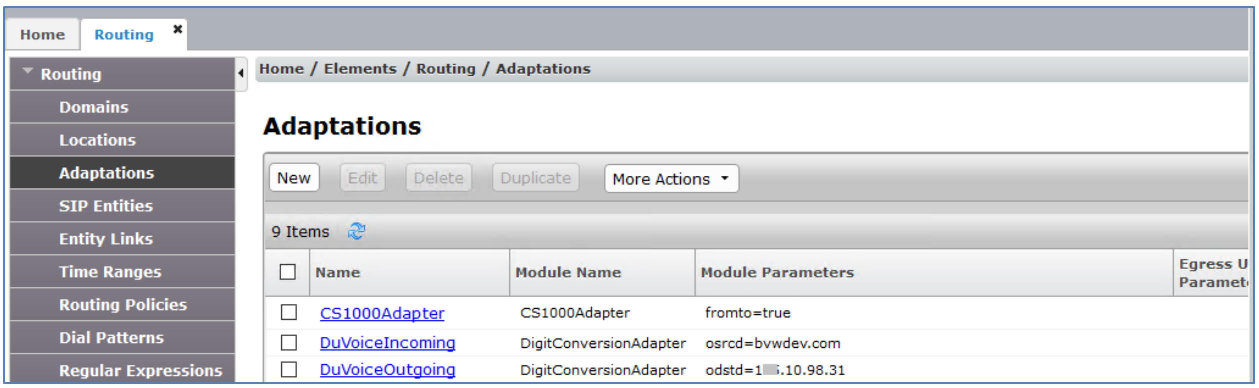
- Type in an **IP Address Pattern** of Session Manager, e.g.10.10.97.*

Click **Commit** to save changes. Screen shot shown on next page.



6.3. Administer Adaptation

Avaya Aura® Session Manager can be configured with adaptation modules that can modify SIP messages before or after routing decisions have been made. A generic adaptation module **DigitConversionAdapter** supports digit conversion of telephone numbers in specific headers of SIP messages. Other adaptation modules are built on this generic, and can modify other headers to permit interoperability with third party SIP products. To view or change adaptations, select **Routing → Adaptations**. Click on the checkbox corresponding to the name of an adaptation and **Edit** to edit an existing adaptation, or the **New** button to add an adaptation. Click the **Commit** button after changes are completed. The following screen shows a portion of the list of adaptations in the sample configuration. Adaptations were used for DuVoice DV2000 VS6, an incoming adaptation (for incoming calls from DuVoice DV2000 VS6) and an outgoing adaptation (for outgoing calls to DuVoice DV2000 VS6).

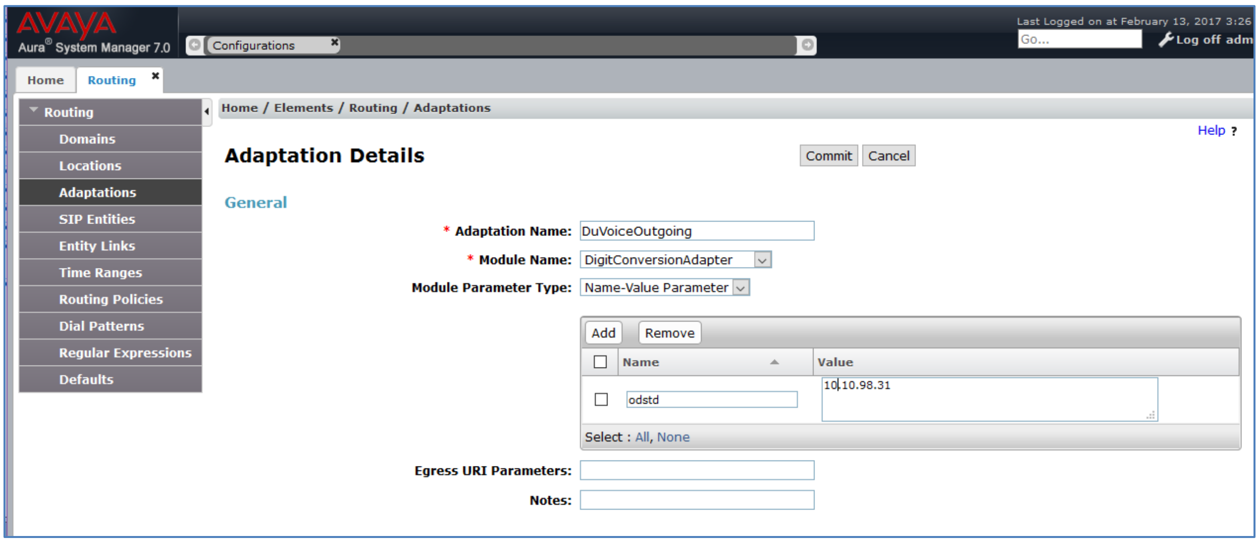


The adaptations named **DuVoice Incoming** and **DuVoice Outgoing** were configured and used in the compliance test.

Settings for the **DuVoice Outgoing** Adaptation**:**
In the **General** section, enter the following values. Use default values for all remaining fields:

- **Adaptation Name:** Enter a descriptive name for the adaptation.
- **Module Name:** Select **DigitConversionAdapter**.
- **Module Parameter Type:** Set to Name-Value Parameter. Next, enter **odstd** for the **Name** parameter and the IP address of the DuVoice DV2000 VS6 server for the **Value** parameter, e.g. 10.10.98.31.

Click **Commit** to save. The **DuVoice Outgoing** adaptation shown below will later be assigned to the **DuVoice** SIP Entity. This adaptation uses the **DigitConversionAdapter**.



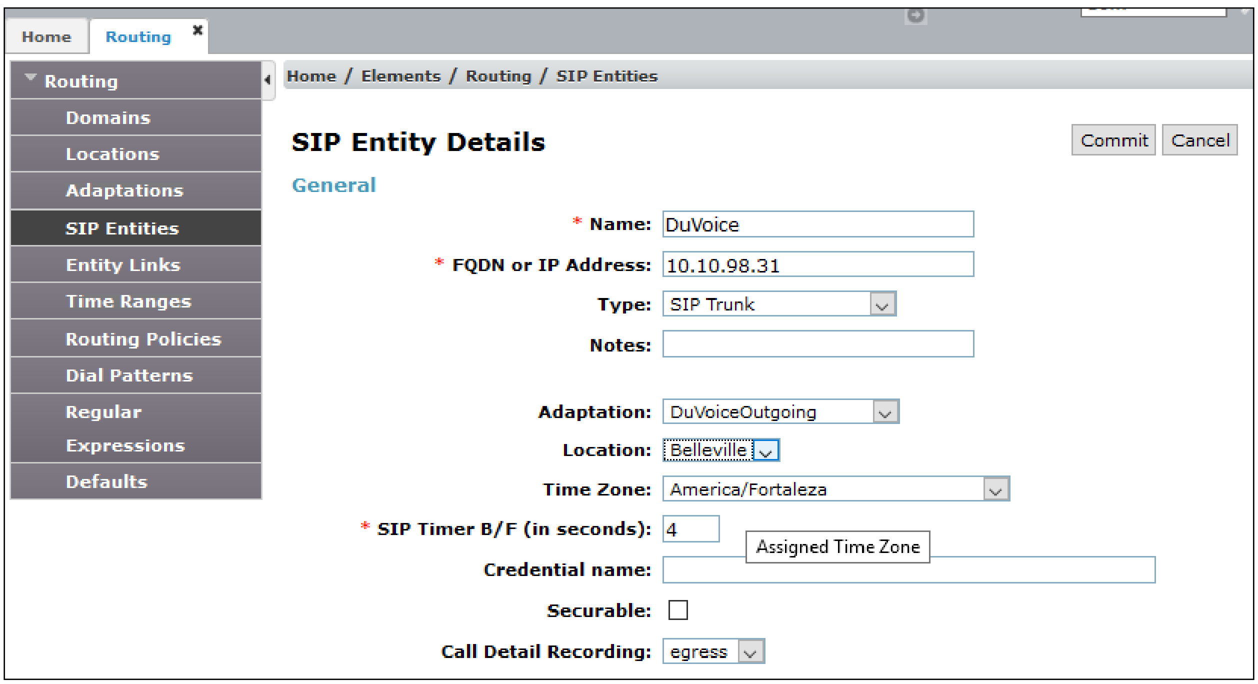
6.4. Add SIP Entity – DuVoice

Add Communication Manager as a SIP Entity. Navigate to **Home → Elements → Routing → SIP Entities,** click on **New** (no shown) and configure as follows:

- Type in a descriptive name in **Name** field.
- Type in the IP address or FQDN of DuVoice DV2000 VS6 in **FQDN or IP Address** field.
- Set **Type** to **SIP Trunk.**
- Set **Location** to the appropriated location.

Click **Commit** to save changes.

Note: It is assumed that SIP Entity for Session Manager has been already configured.

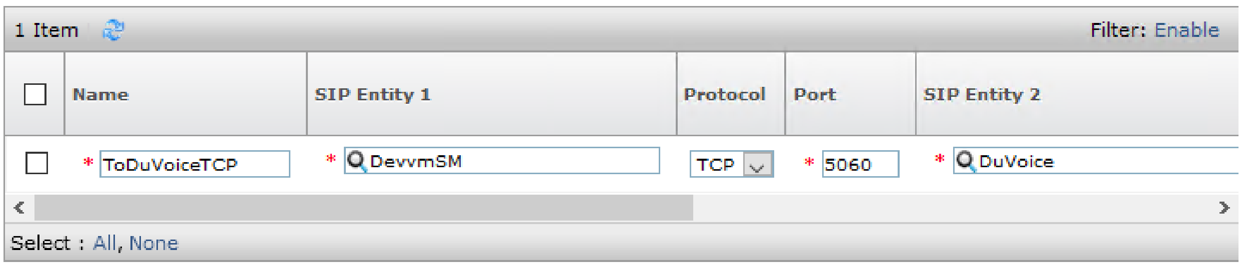


6.5. Add Entity Link – DuVoice

Navigate to **Home → Elements → Routing → Entity Links**, click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field.
- Set **SIP Entity 1** to the name of Session Manager SIP Entity
- Set **SIP Entity 2** to DuVoice DV2000 VS6 SIP Entity configured in **Section 6.5.**
- Set **Protocol** to **TCP.**

Click **Commit** to save changes.

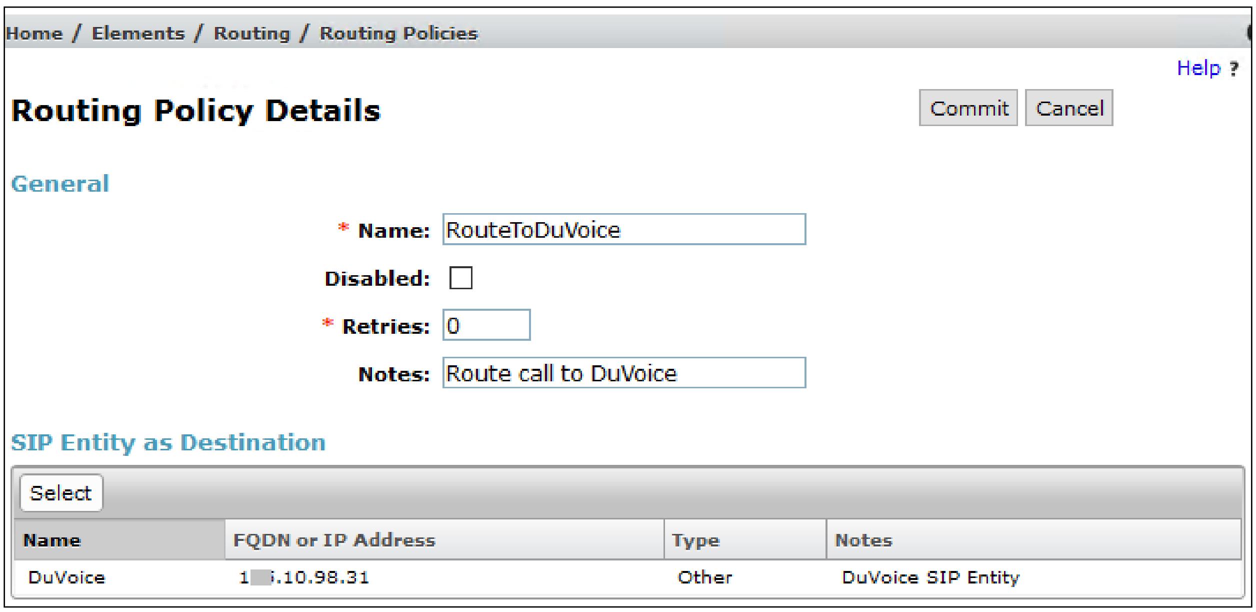


6.6. Add Routing Policy – DuVoice

Navigate to **Home → Elements → Routing → Routing Policies**, click on **New** (not shown) and configure as follows:

- Type in a descriptive name in **Name** field.
- Under **SIP Entity as Destination**, click on **Select** (not shown):
  - Select DuVoice DV2000 VS6 SIP entity added in **Section 6.5.**
- Under **Time of Day**, click on **Add** (not shown):
  - Select time range added in previous step.

Click **Commit** to save changes.



6.7. Add Dial Patterns – DuVoice

Navigate to **Home → Elements → Routing → Dial Patterns**, click on **New** (not shown) and configure as follows:

Under **General:**

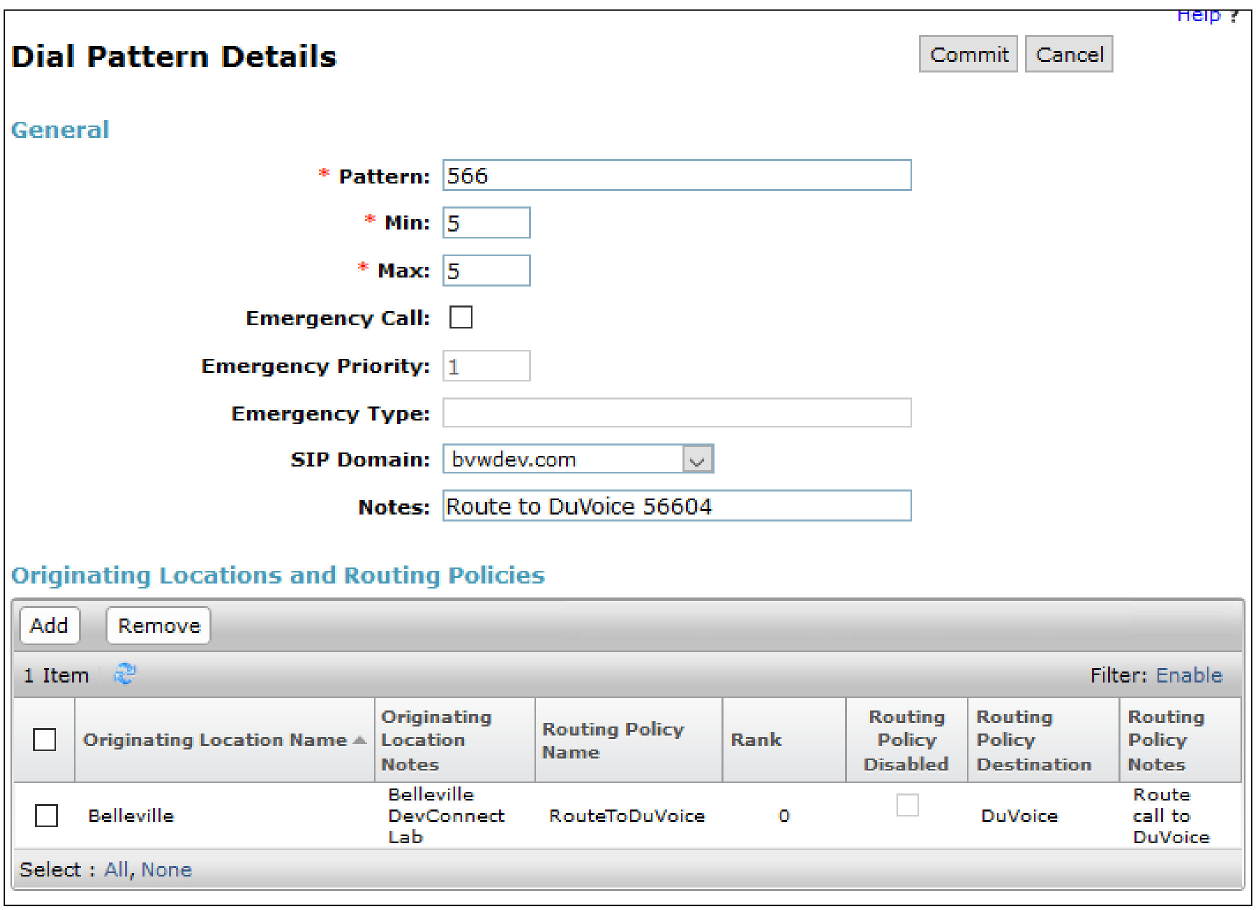
- Set **Pattern** to prefix of dialed number.
- Set **Min** to minimum length of dialed number.
- Set **Max** to maximum length of dialed number.
- Set **SIP Domain** to the appropriate value (e.g., bvwdev.com).

Under **Originating Locations and Routing Policies:**

- Click **Add** and select originating location and DuVoice DV2000 VS6 routing policy as configured in **Section 6.9.**

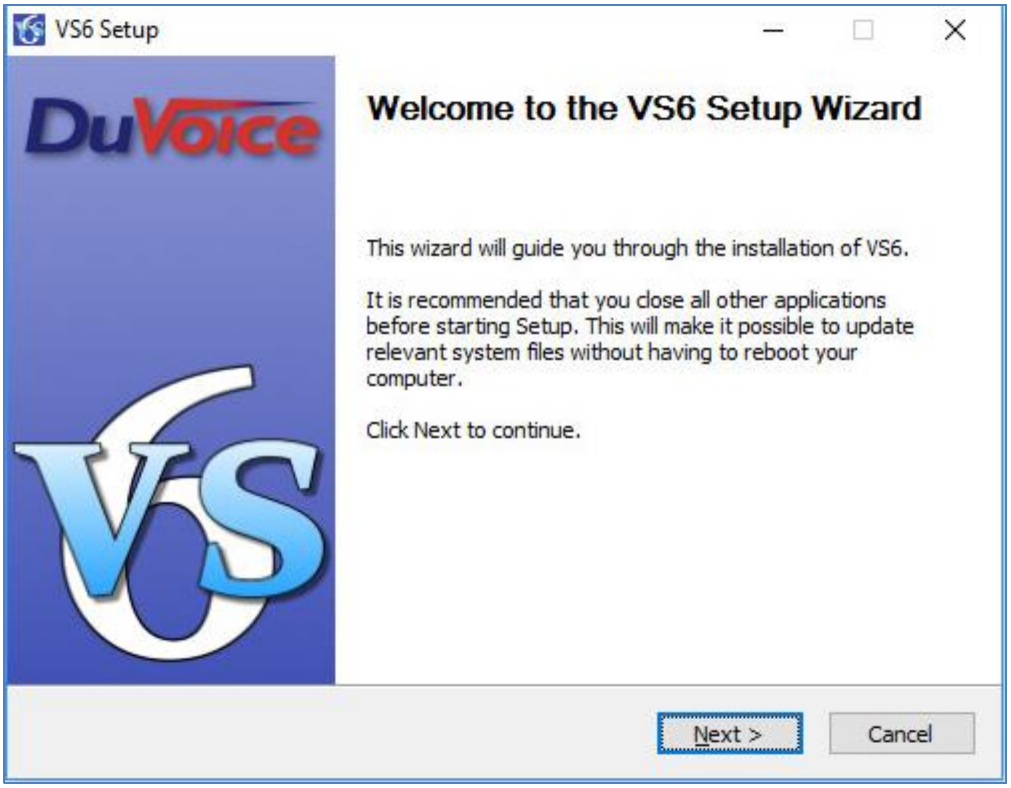
Click **Commit** to save changes.

Note: For Compliance testing, dialed number of 56604 was used to route calls to DuVoice. Thus, **Pattern** was set to 566 and **Min** and **Max** values were set to 5.

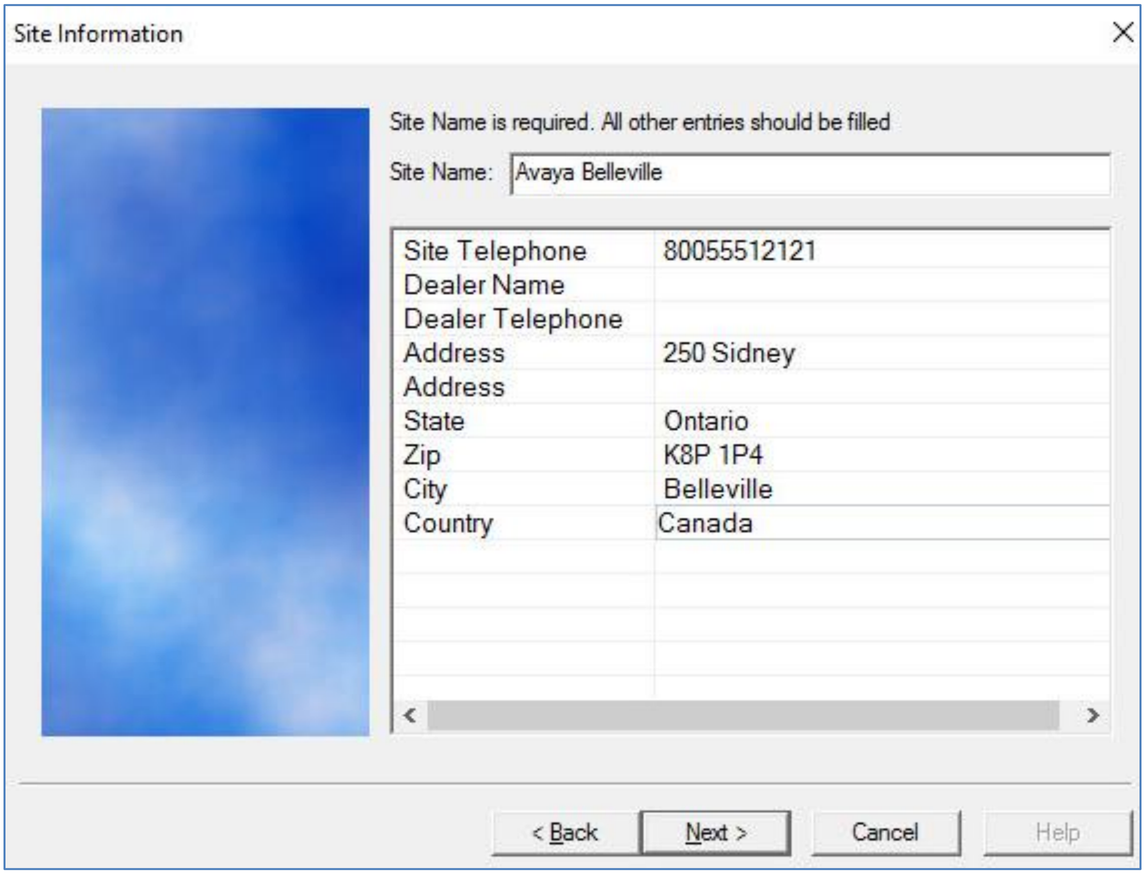


7. Configure DuVoice

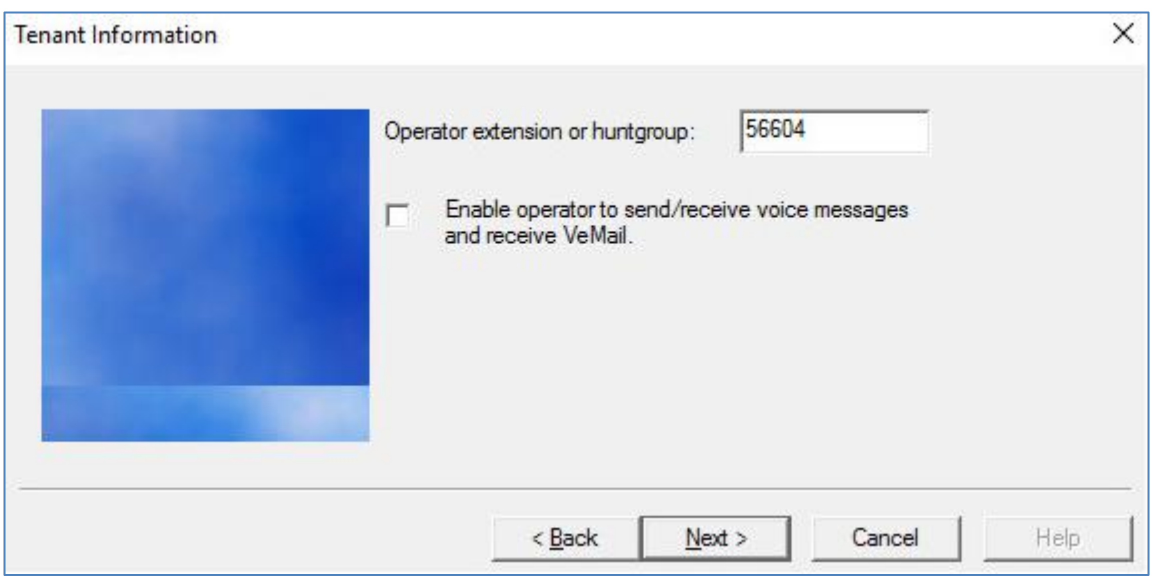
During compliance testing, DuVoice DV2000 VS6 was installed on a Windows 7 Enterprise desktop PC. To configure SIP connectivity to Session Manager, locate the SETUP.exe file for DuVoice DV2000 VS6 and open it. SETUP.exe can be found in the installation directory for DuVoice. On the **Wizard Start** window, select **Next.**



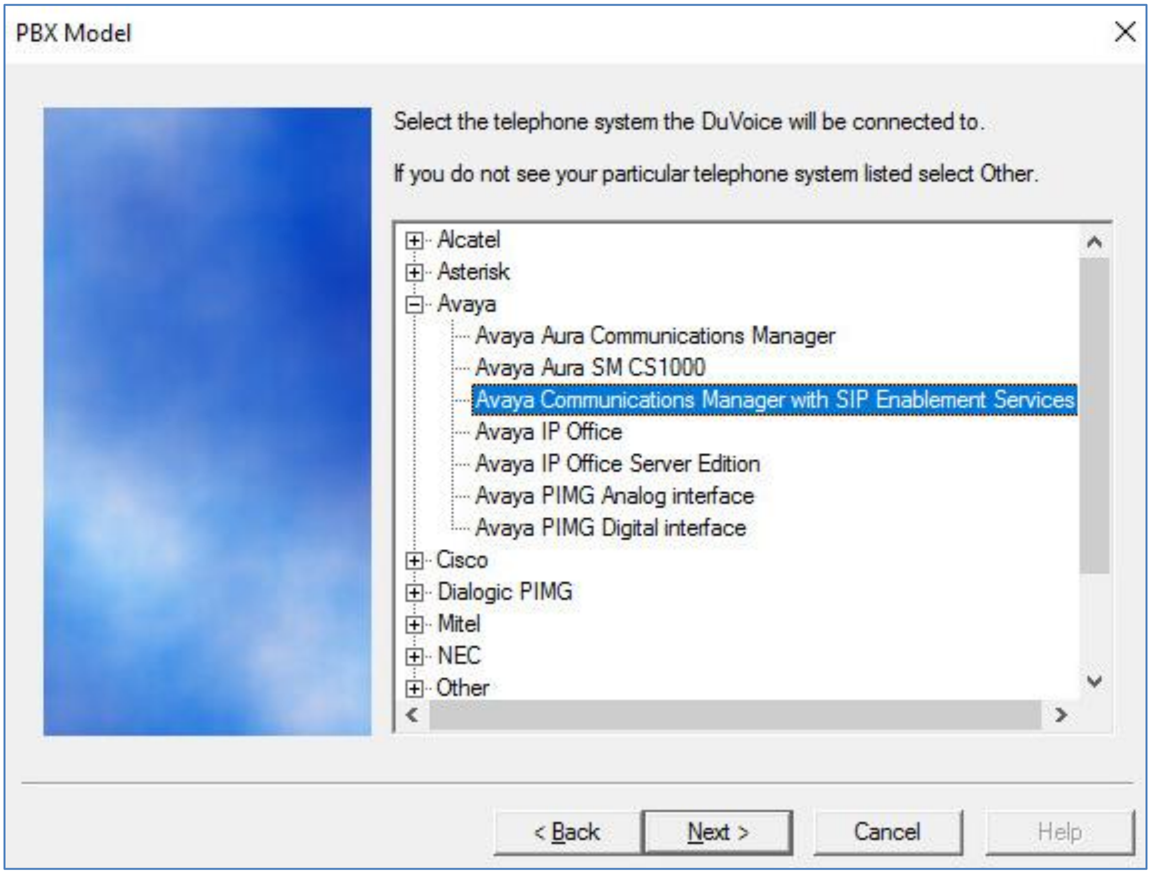
On the **Site Information** window, fill in the fields marked with * and click **Next**.



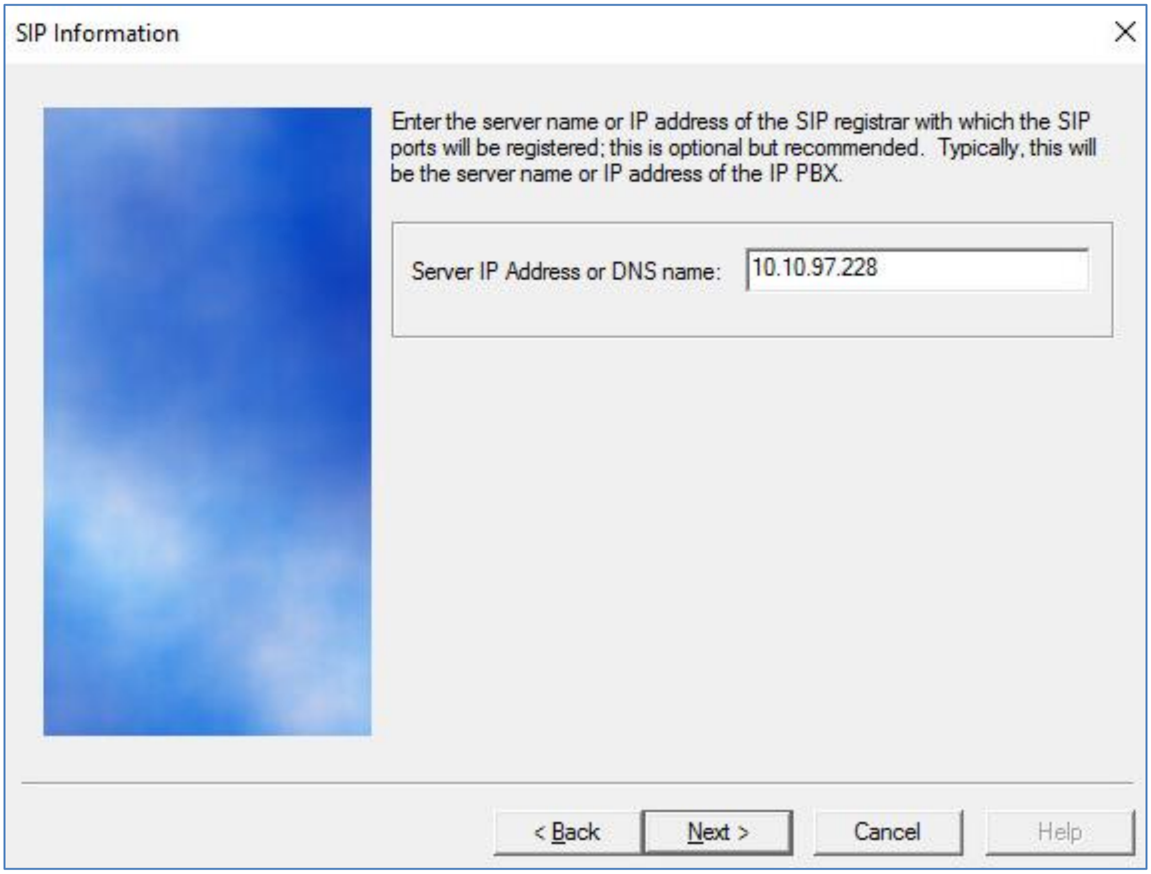
Enter DV2000 VS6 hunt group number in **Tenant Information** windows.



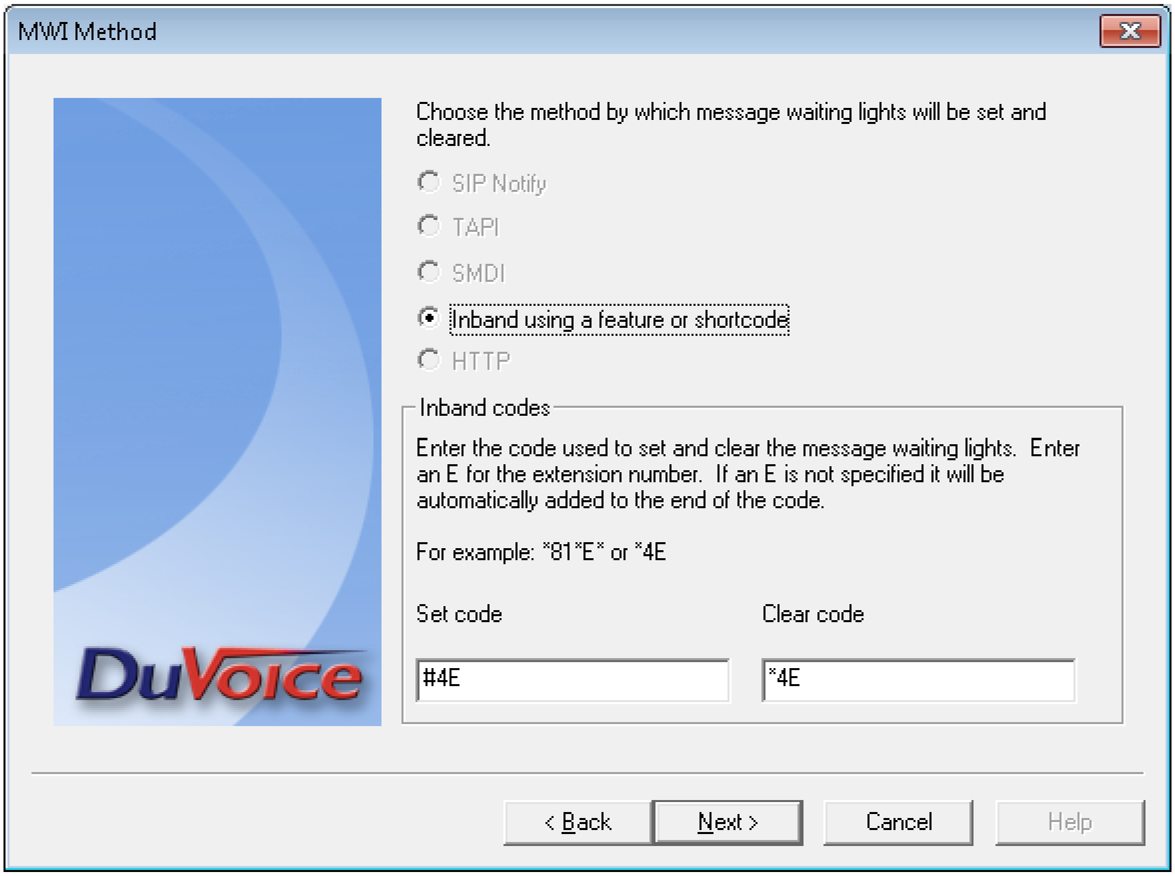
On the **PBX Model** window, select **Avaya → Avaya Communication Manager with SIP Enablement Services** and click **Next**.



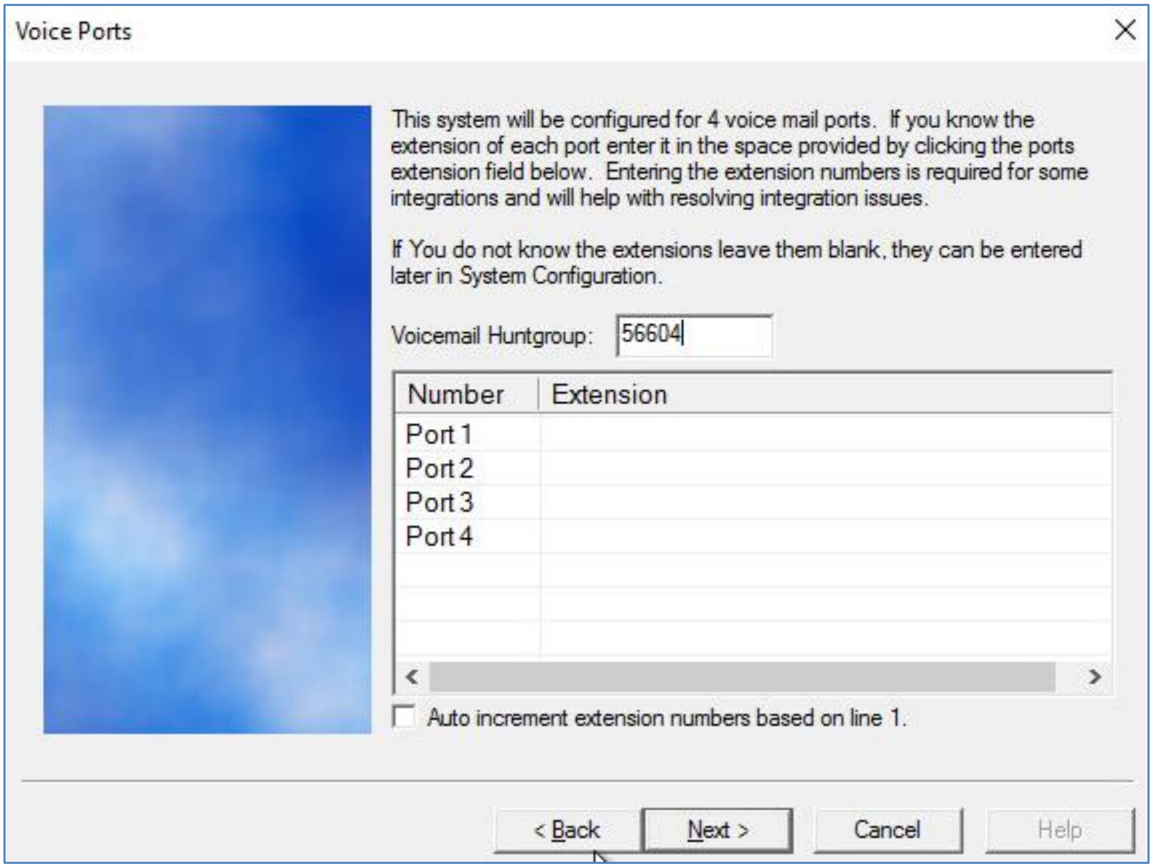
On the **SIP Information** window, type in the Session Manager IP Address in **Server IP Address or DNS Name** field and click **Next**.



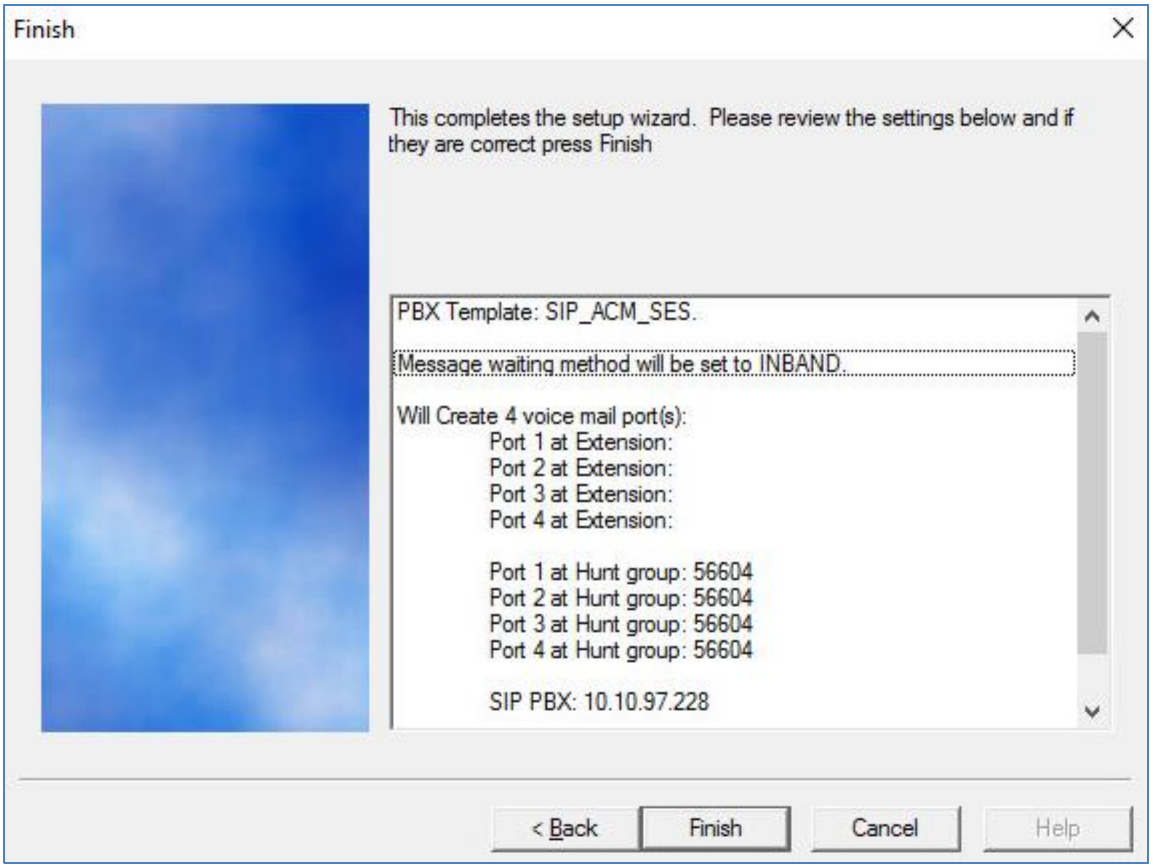
On the **MWI Method** window, accept the default values and click **Next**. Please note that MWI method will be changed to SIP in **Section 7.2**.



On the **Voice Ports** window, type in the Hunt Group that was configured in Communication Manager in **Voicemail Huntgroup** field and click **Next**.

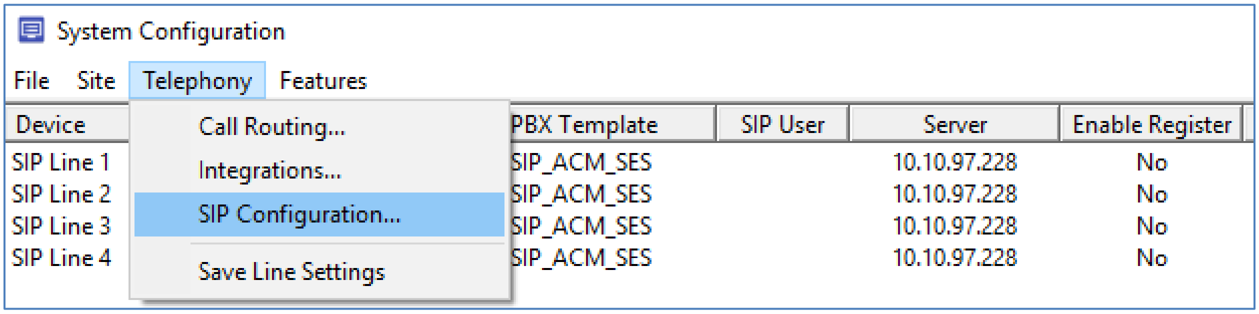


The final screen shows the configuration, click **Finish.**



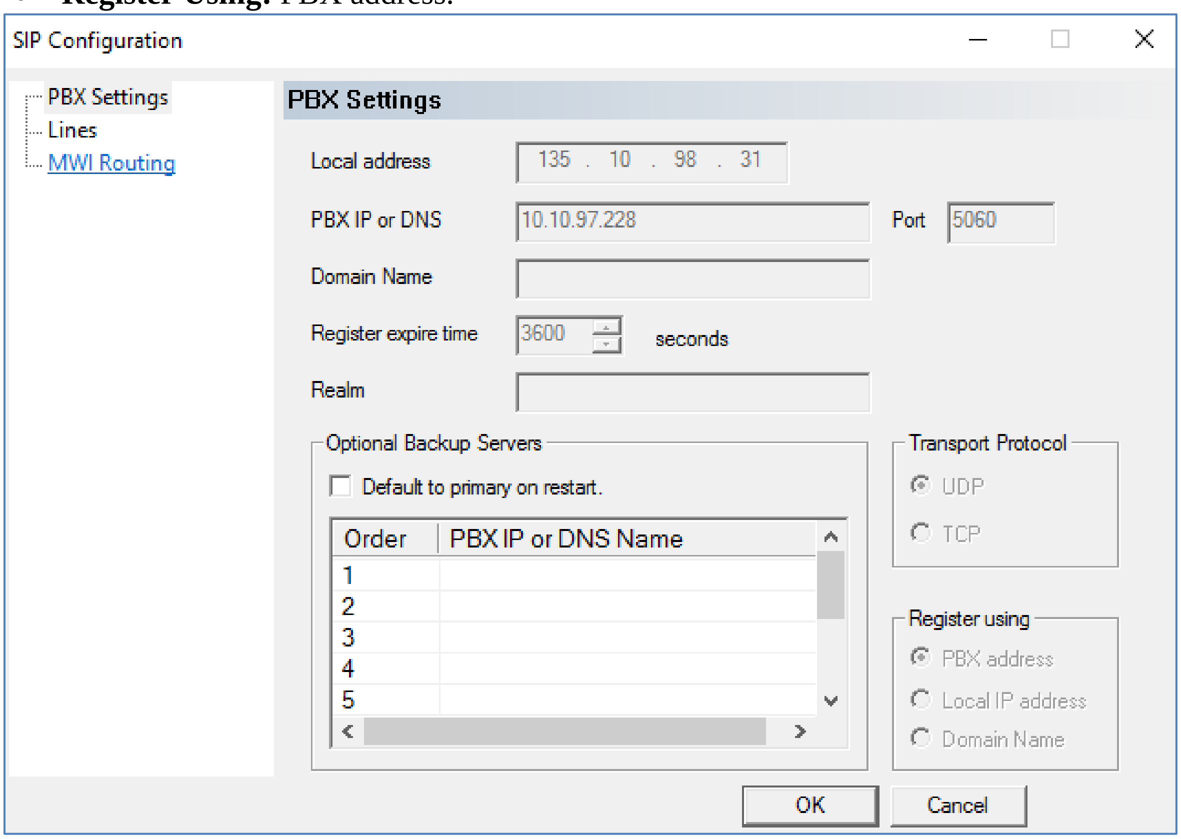
7.1. Configure SIP Configuration

Open the DV2000 VS6 system configuration program located on the desktop or the start under program file VS6. Once open select **Telephony →SIP Configuration** from top menu



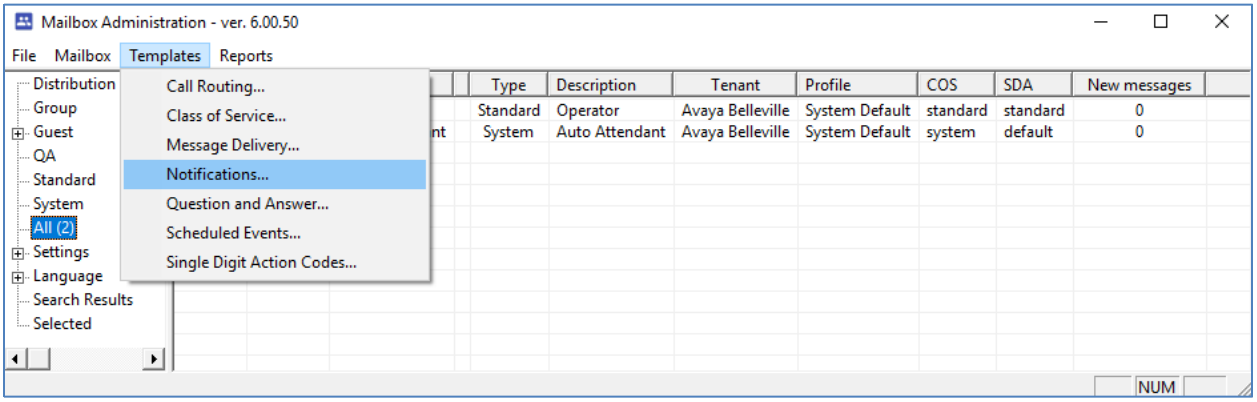
From the SIP configuration tab please confirm the following settings.

- **Local IPaddress:** DV2000 VS6e IP address.
- **PBX or DNS:** enter the SM IP address and select port 5060.
- **Domain Name:** leave blank (default).
- **Register expire time:** using default value 3600.
- **Realm:** enter the SM Sip domain found in **Section 6.1** of this document, Enter the Session Managers sip domain name.
- **Transport protocol:** UDP.
- **Register Using:** PBX address.

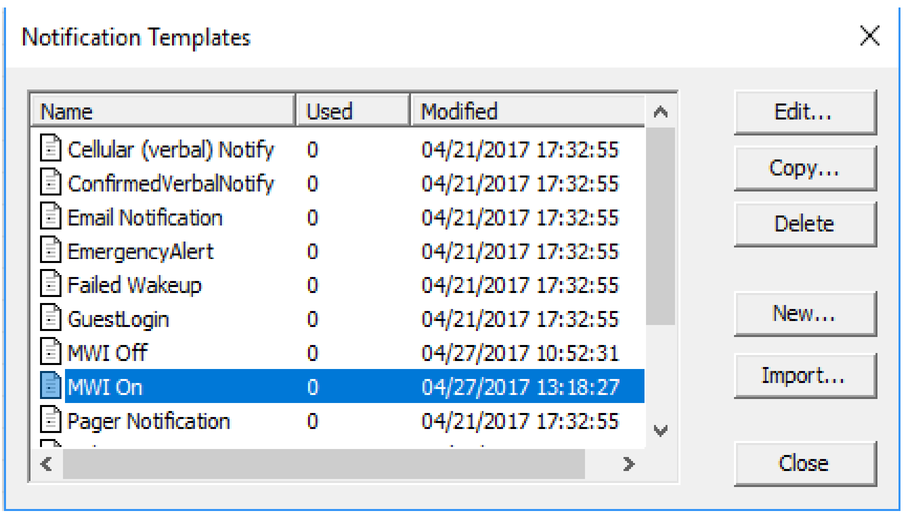


7.2. Configure SIP MWI

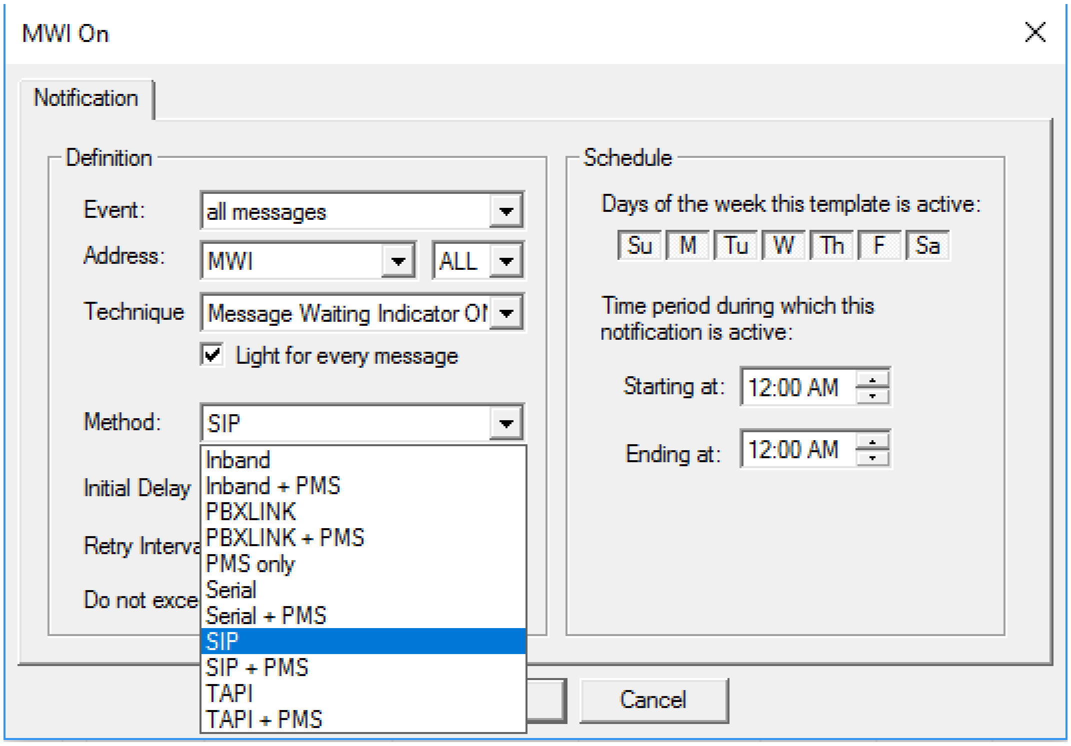
In the mailbox **Administration** select **Templates → Notifications…** to change MWI method to SIP.



In **Notification Templates**, select **MWI On** template and click on **Edit** button:

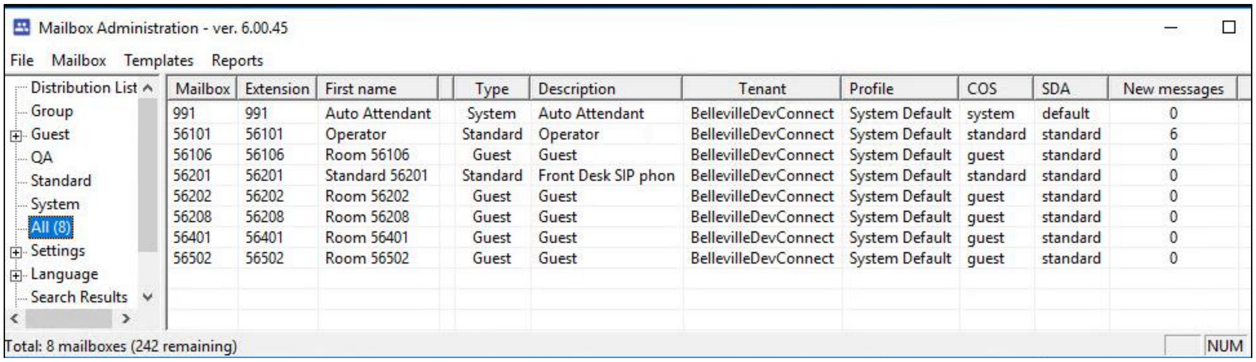


In the **MWI On** template window, in **Method** drop down list select SIP. Click **OK** to save changes.

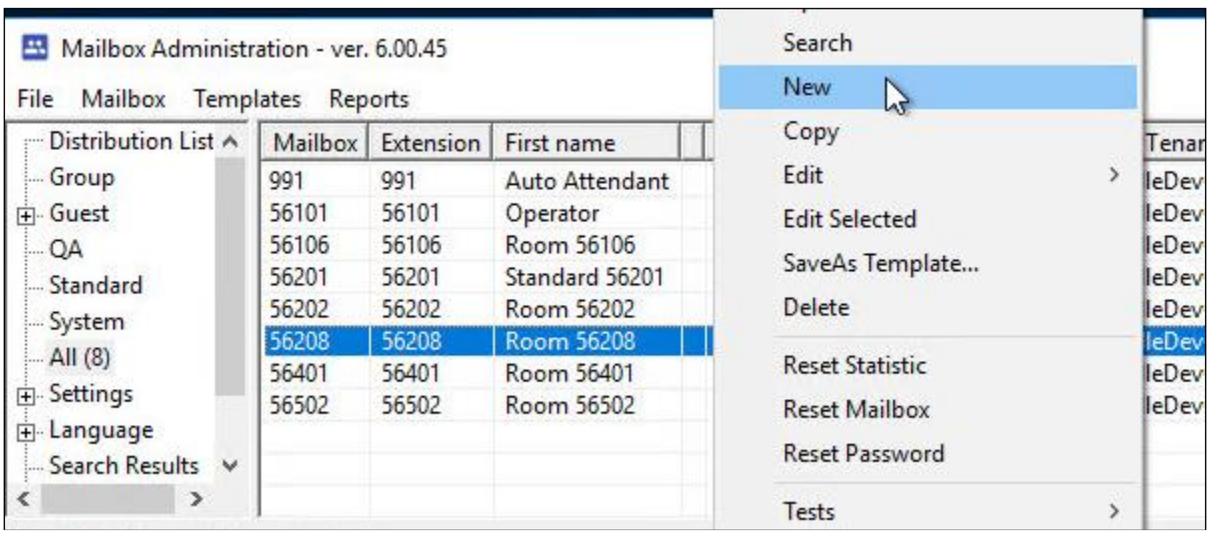


7.3. Configure Mailbox

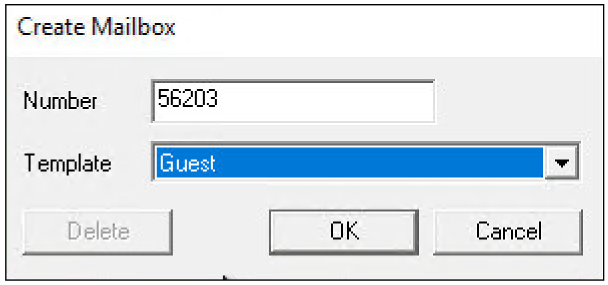
To configure mail boxes for guests, open **Mailbox Administration**, and select **All** in the left pane. A shortcut icon for **Mailbox Administration** can be found on desktop PC.



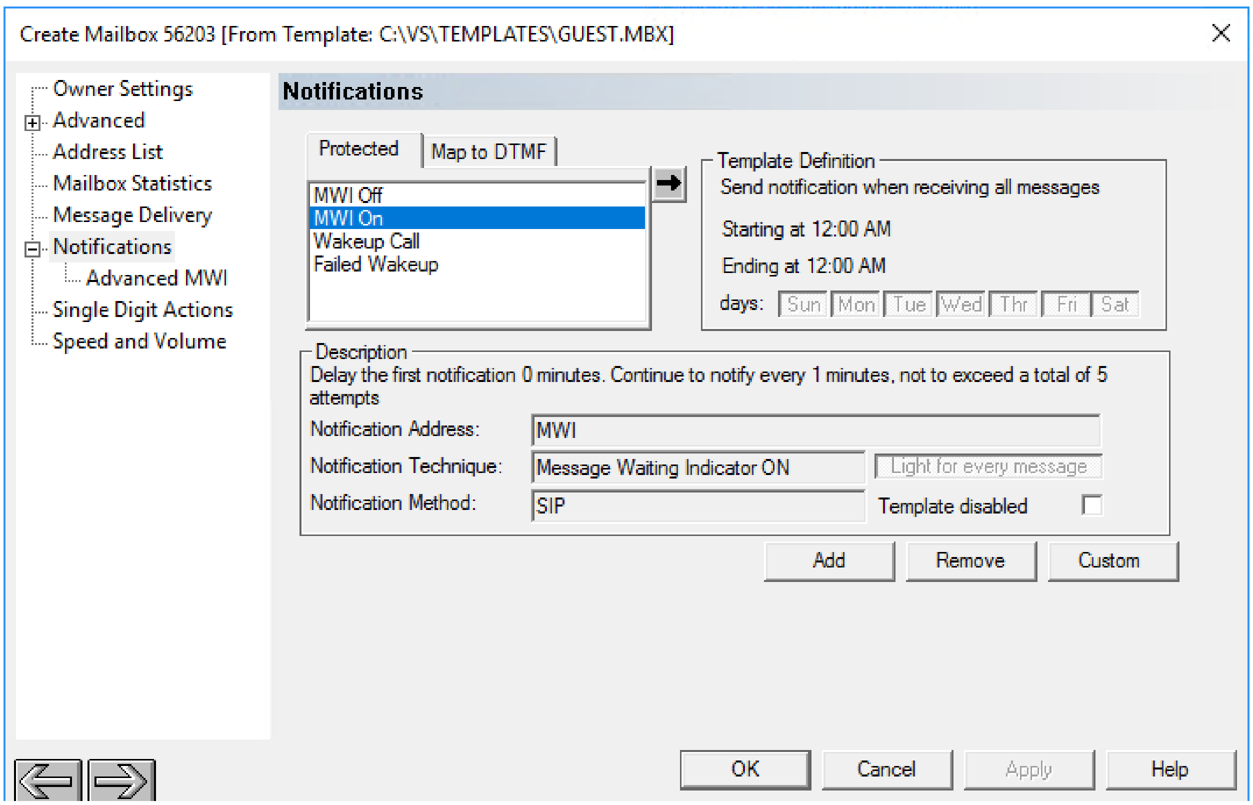
Add a new mail box, right-click on the right pane and select **New.**



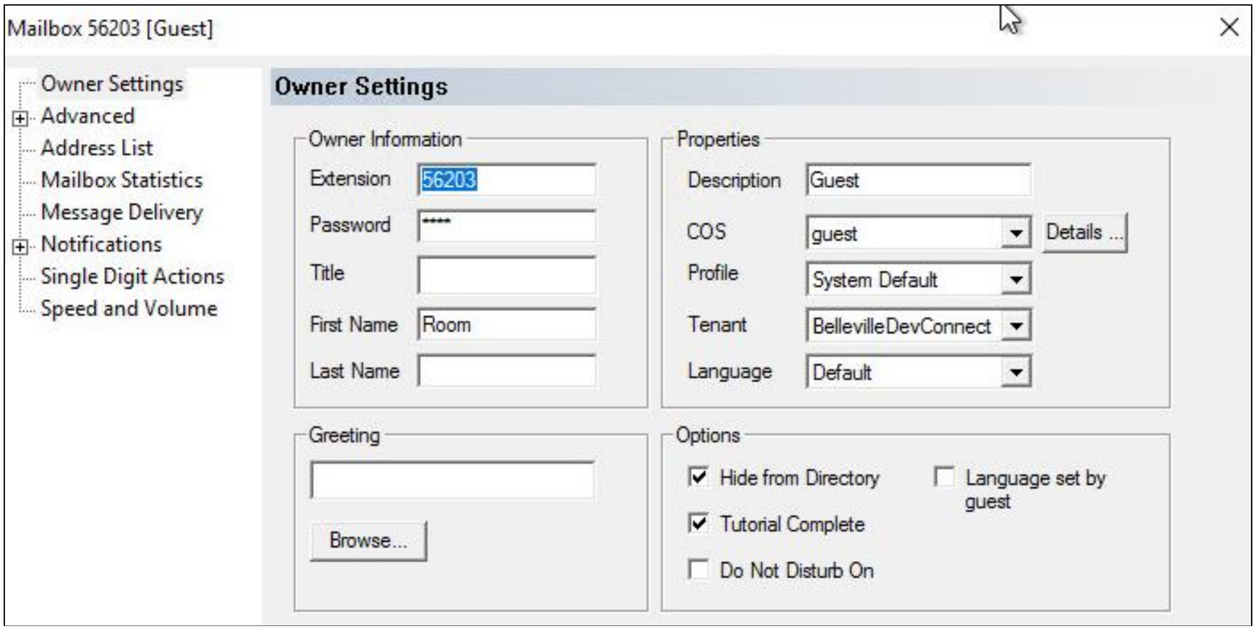
On the **Create Mailbox** window, enter the station extension and select **Guest** for **Template**, and click **OK.**



Select **Notification** and add **MWI On** in the mailbox **Notifications** as shown below:



On the next window, accept default values and click **OK.**



8. Verification Steps

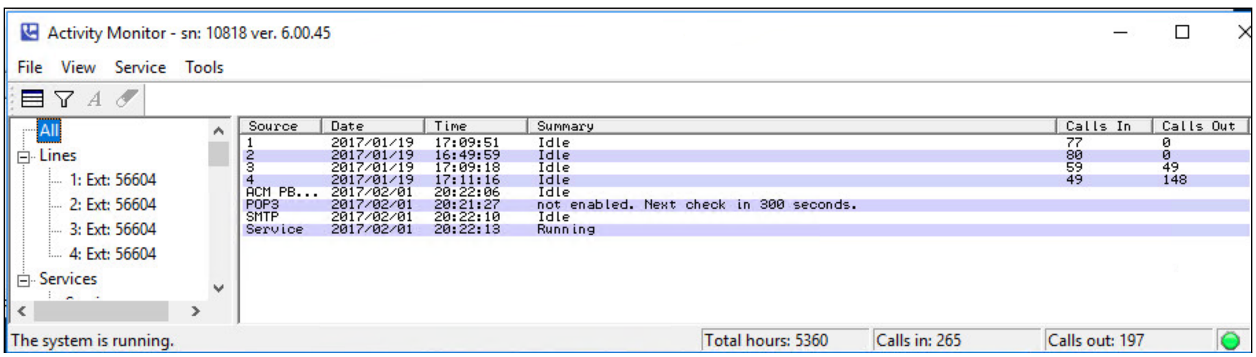
This section describes verification steps that may be used to verify SIP connectivity between DuVoice DV2000 VS6 and Session Manager.

8.1. Avaya Aura® Session Manager

On the System Manager, navigate to **Home → Element → Session Manager → System Station → SIP Entity Monitoring** (not shown). Verify the **Conn. Status** and **Reason Code** are **Up** and **200 OK**.

8.2. Verify DuVoice Activity

Open **Activity Monitor** application on DuVoice DV2000 VS6 desktop PC. Verify all the lines connection to DuVoice DV2000 VS6 hunt group number are Idle as displayed in below screenshot.



9. Conclusion

These Application Notes describe the procedures required to configure DuVoice DV2000 VS6 to interoperate with Avaya Aura® Session Manager and Avaya Aura® Communication Manager to support the network shown in **Figure 1**. DuVoice DV2000 VS6 passed compliance testing.

10. Additional References

Product documentation for Avaya products may be found at http://support.avaya.com.

[1] *Administering Avaya Aura® Communication Manager*, Release 7.0.1 03-300509 Issue 2.1 August 2016.

[2] *Administering Avaya Aura® Session Manager*, Release 7.0.1 Issue 2 May 2016.

Product documentation for DuVoice DV2000 VS6 may be obtained directly from DuVoice.

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.