



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

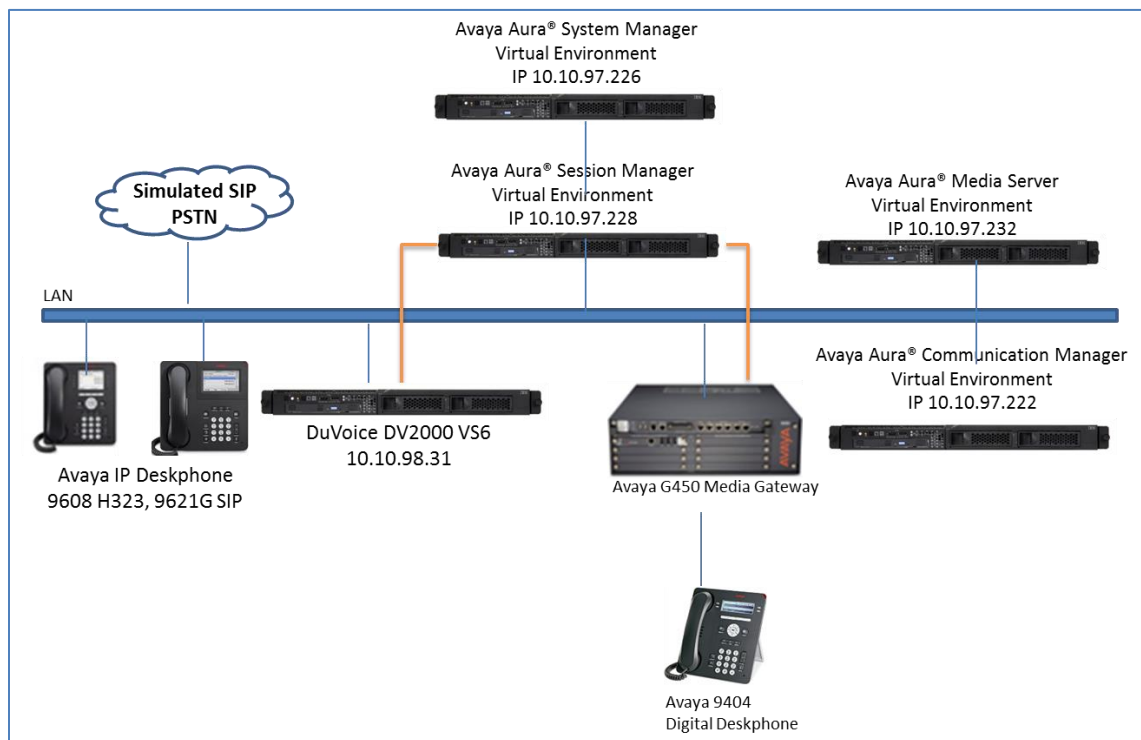


Figure 1: Reference Configuration

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
Codec Set: 1		IP Codec Set		
Audio Codec	Silence Suppression	Frames Per Pkt	Packet 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. bvwddev.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: bvwddev.com		
Incoming Dialog Loopbacks: eliminate	Bypass If IP Threshold Exceeded? n	
DTMF over IP: rtp-payload	RFC 3389 Comfort Noise? n	
Session Establishment Timer(min): 3	Direct IP-IP Audio Connections? y	
Enable Layer 3 Test? y	IP Audio Hairpinning? n	
H.323 Station Outgoing Direct Media? n	Initial IP-IP Direct Media? n	
	Alternate Route Timer(sec): 6	

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	Night Service:	
Dial Access? n	Auth Code? n		
Queue Length: 0	Member Assignment Method: auto		
Service Type: tie	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 String	Total Min Max	Route Pattern	Call Type	Node Num	ANI Req'd	
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			
Loss Group: 19		Time of Day Lock Table:	
		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	7:	
4: aux-work	8:	
RC:	Grp:	
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.

System Manager

https://devvmmgr.bvwdev.com/securityserver/ 90%

AVAYA
Aura System Manager 7.0

This system is restricted solely to authorized users for legitimate business purposes only. The actual or attempted unauthorized access, use, or modification of this system is strictly prohibited.

Unauthorized users are subject to company disciplinary procedures and or criminal and civil penalties under state, federal, or other applicable domestic and foreign laws.

The use of this system may be monitored and recorded for administrative and security reasons. Anyone accessing this system expressly consents to such monitoring and recording, and is advised that if it reveals possible evidence of criminal activity, the evidence of such activity may be provided to

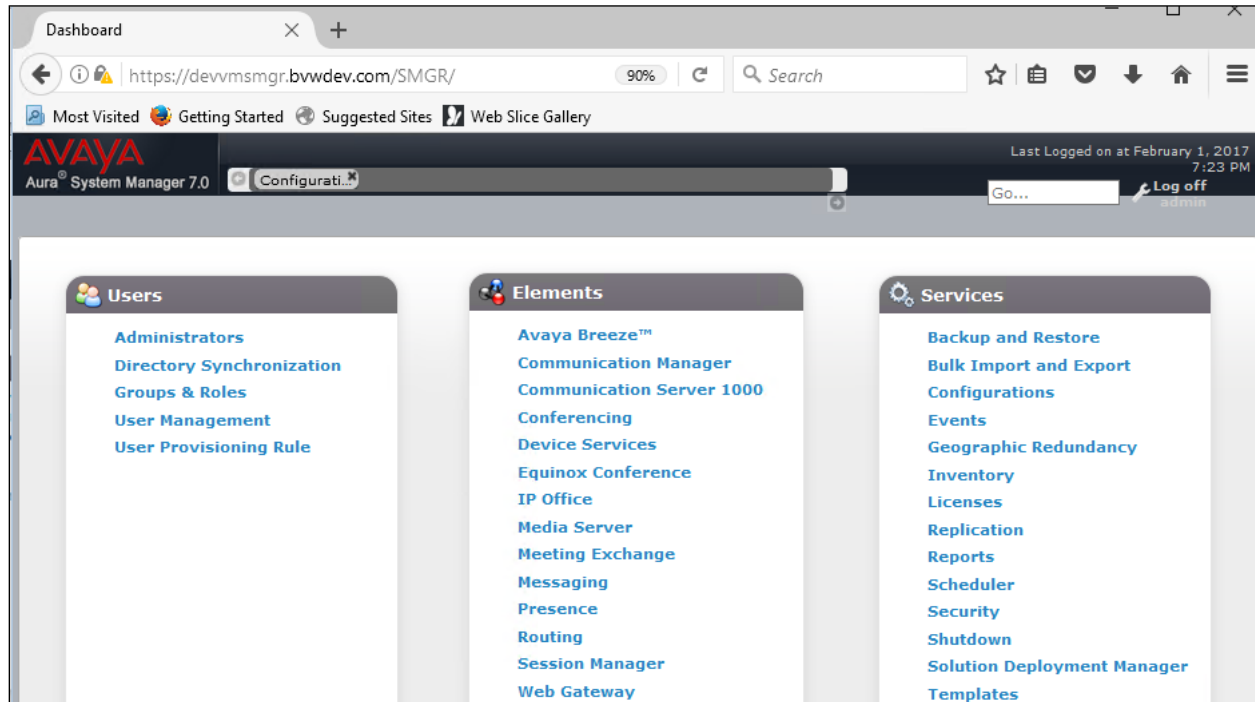
User ID:

Password:

Log On Reset

Supported Browsers: Internet Explorer 11.x or Firefox 43.0, 44.0 or 45.0.

Once logged in, the following screen is displayed.

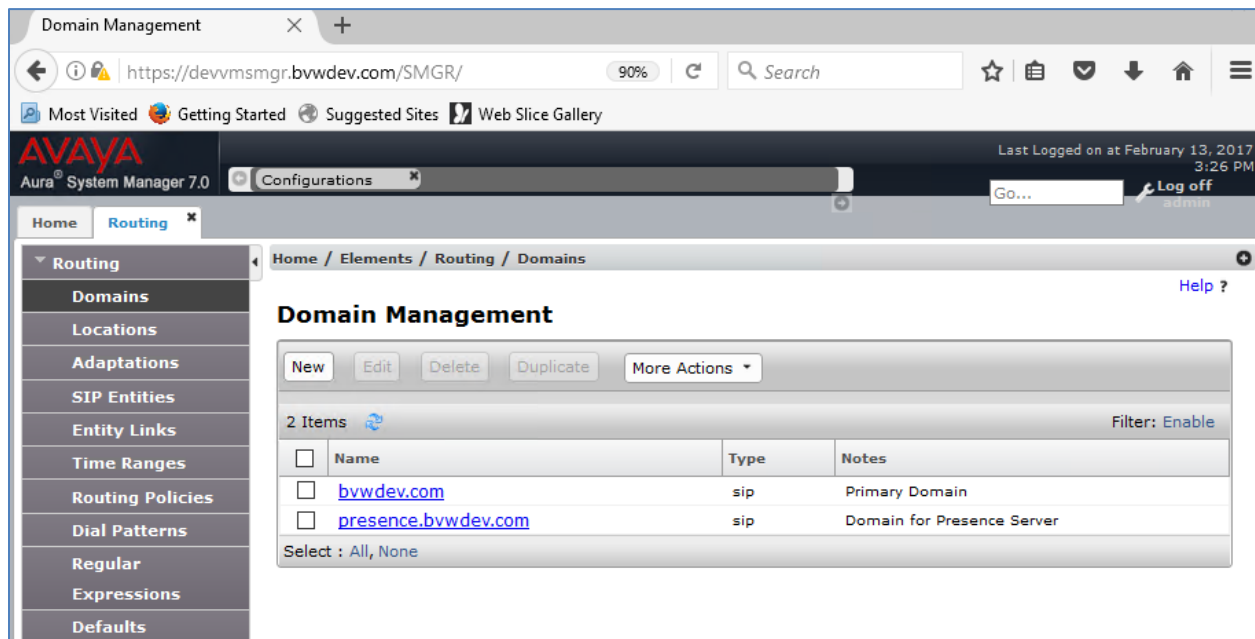


6.1. Administer SIP Domain

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. **bvwddev.com**.
- Set **Type** to **sip**.

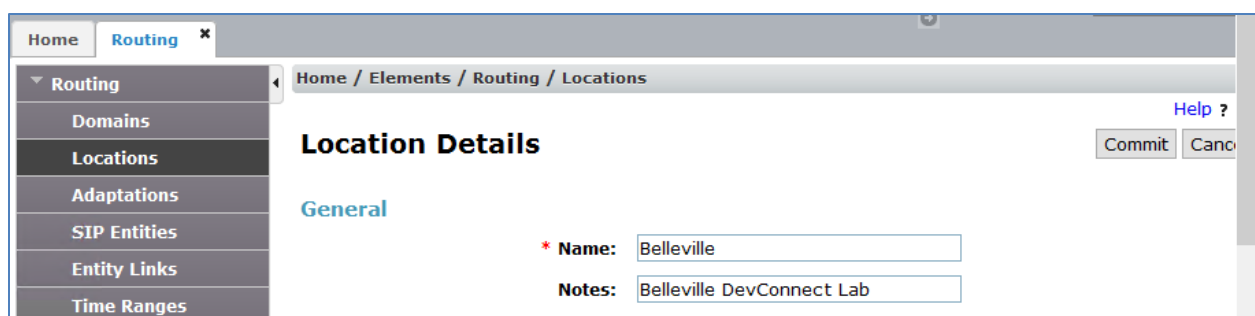
Click **Commit** to save changes.



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.

Location Pattern

AddRemove

4 Items Filter: [Enable](#)

<input type="checkbox"/>	IP Address Pattern ▲	Notes
<input type="checkbox"/>	* 10.33.5.*	Phones and Servers on private lab network
<input type="checkbox"/>	* 10.10.97.*	Lab PBX
<input type="checkbox"/>	* 10.10.98.*	
<input type="checkbox"/>	* 10.29.187.*	

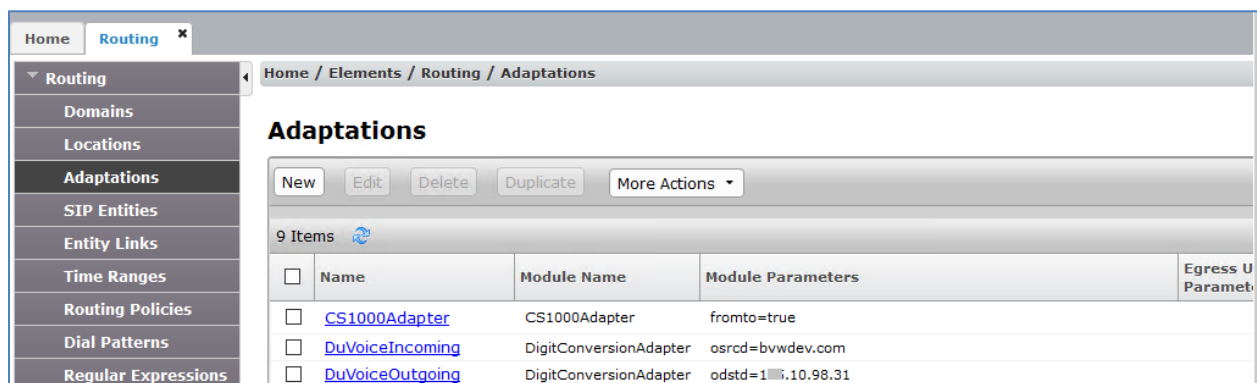
< >

Select : [All](#), [None](#)

CommitCancel

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).



	Name	Module Name	Module Parameters	Egress U Parameters
<input type="checkbox"/>	CS1000Adapter	CS1000Adapter	fromto=true	
<input type="checkbox"/>	DuVoiceIncoming	DigitConversionAdapter	osrcd=bvwdev.com	
<input type="checkbox"/>	DuVoiceOutgoing	DigitConversionAdapter	odstd=10.10.98.31	

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**.

The screenshot shows the Avaya Aura System Manager 7.0 interface. The left sidebar contains a navigation menu with options: Home, Routing, Domains, Locations, Adaptations, SIP Entities, Entity Links, Time Ranges, Routing Policies, Dial Patterns, Regular Expressions, and Defaults. The main content area is titled 'Adaptation Details' and shows the 'General' tab. The 'Adaptation Name' field is set to 'DuVoiceOutgoing'. The 'Module Name' dropdown is set to 'DigitConversionAdapter'. The 'Module Parameter Type' dropdown is set to 'Name-Value Parameter'. Below this, there is a table with two columns: 'Name' and 'Value'. The first row has 'odstd' in the 'Name' column and '10.10.98.31' in the 'Value' column. There are 'Add' and 'Remove' buttons above the table. Below the table, there is a 'Select' dropdown set to 'All, None'. At the bottom, there are fields for 'Egress URI Parameters' and 'Notes'. The top right of the interface shows the user is logged in as 'off adm' and the last login was on February 13, 2017 at 3:26.

Name	Value
odstd	10.10.98.31

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.

The screenshot shows the 'SIP Entity Details' configuration page for an entity named 'DuVoice'. The page is part of a web application with a navigation menu on the left and a breadcrumb trail at the top: 'Home / Elements / Routing / SIP Entities'. The 'SIP Entities' menu item is selected. The configuration is under the 'General' tab. Fields include: 'Name' (DuVoice), 'FQDN or IP Address' (10.10.98.31), 'Type' (SIP Trunk), 'Notes' (empty), 'Adaptation' (DuVoiceOutgoing), 'Location' (Belleville), 'Time Zone' (America/Fortaleza), 'SIP Timer B/F (in seconds)' (4), 'Credential name' (empty), 'Securable' (unchecked), and 'Call Detail Recording' (egress). There are 'Commit' and 'Cancel' buttons at the top right.






Field	Value
Name	DuVoice
FQDN or IP Address	10.10.98.31
Type	SIP Trunk
Notes	
Adaptation	DuVoiceOutgoing
Location	Belleville
Time Zone	America/Fortaleza
SIP Timer B/F (in seconds)	4
Credential name	
Securable	<input type="checkbox"/>
Call Detail Recording	egress

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.

1 Item 					
	Name	SIP Entity 1	Protocol	Port	SIP Entity 2
<input type="checkbox"/>	* ToDuVoiceTCP	*  DevvmSM	TCP 	* 5060	*  DuVoice
					
Select : All, None					

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.

Home / Elements / Routing / Routing Policies

Routing Policy Details [Help ?](#)

General

* **Name:**

Disabled: ☐

* **Retries:**

Notes:

SIP Entity as Destination

Name	FQDN or IP Address	Type	Notes
DuVoice	10.10.98.31	Other	DuVoice SIP Entity

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.

Dial Pattern Details

CommitCancel

General

* Pattern: 566

* Min: 5

* Max: 5

Emergency Call: ☐

Emergency Priority: 1

Emergency Type:

SIP Domain: bvwdev.com

Notes: Route to DuVoice 56604

Originating Locations and Routing Policies

AddRemove

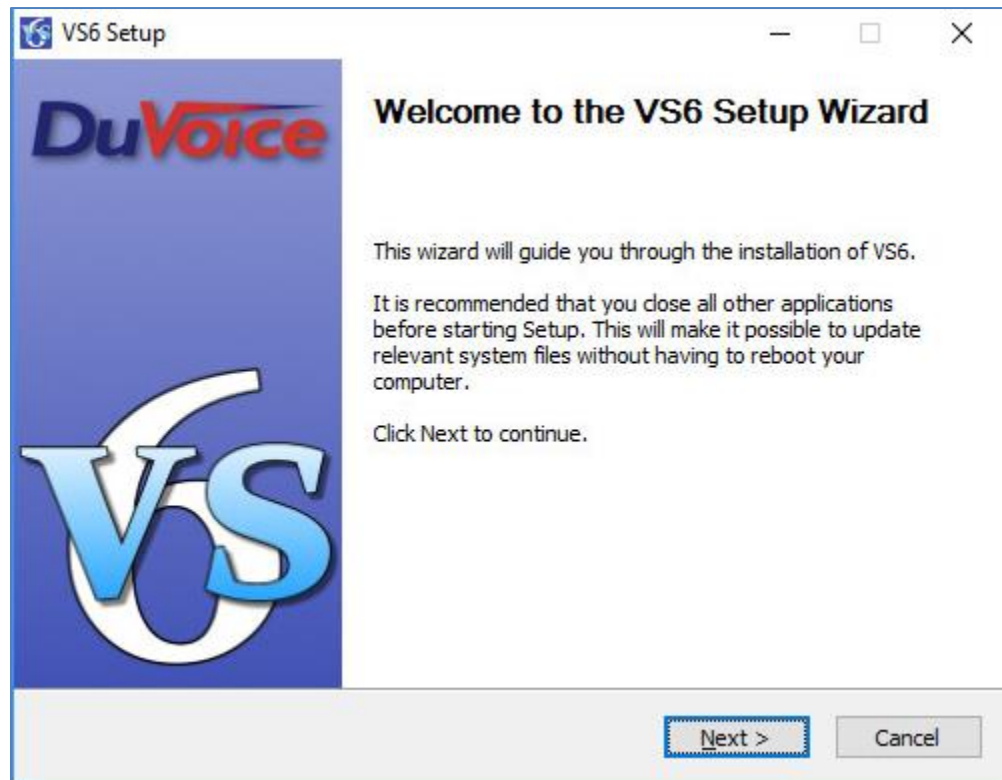
1 ItemFilter: Enable

<input type="checkbox"/>	Originating Location Name ▲	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Belleville	Belleville DevConnect Lab	RouteToDuVoice	0	<input type="checkbox"/>	DuVoice	Route call to DuVoice

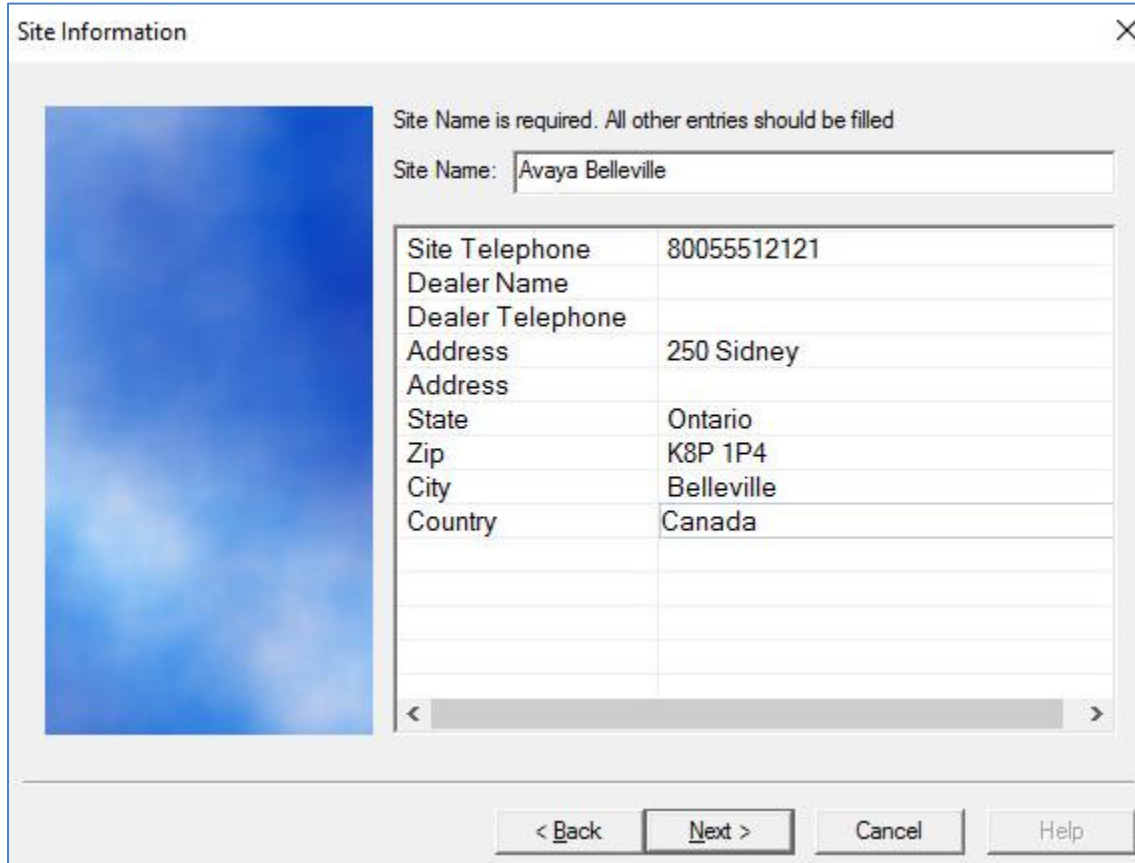
Select : All, None

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**.

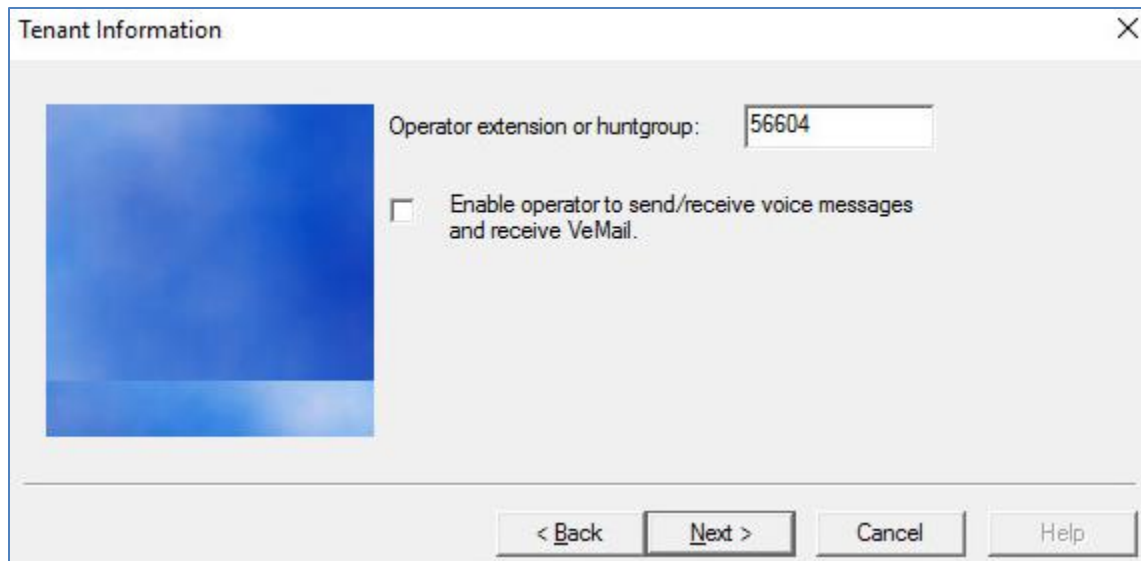


The 'Site Information' window contains a blue placeholder image on the left. On the right, a message states 'Site Name is required. All other entries should be filled'. Below this, the 'Site Name' field is populated with 'Avaya Belleville'. A table-like form contains the following information:

Site Telephone	80055512121
Dealer Name	
Dealer Telephone	
Address	250 Sidney
Address	
State	Ontario
Zip	K8P 1P4
City	Belleville
Country	Canada

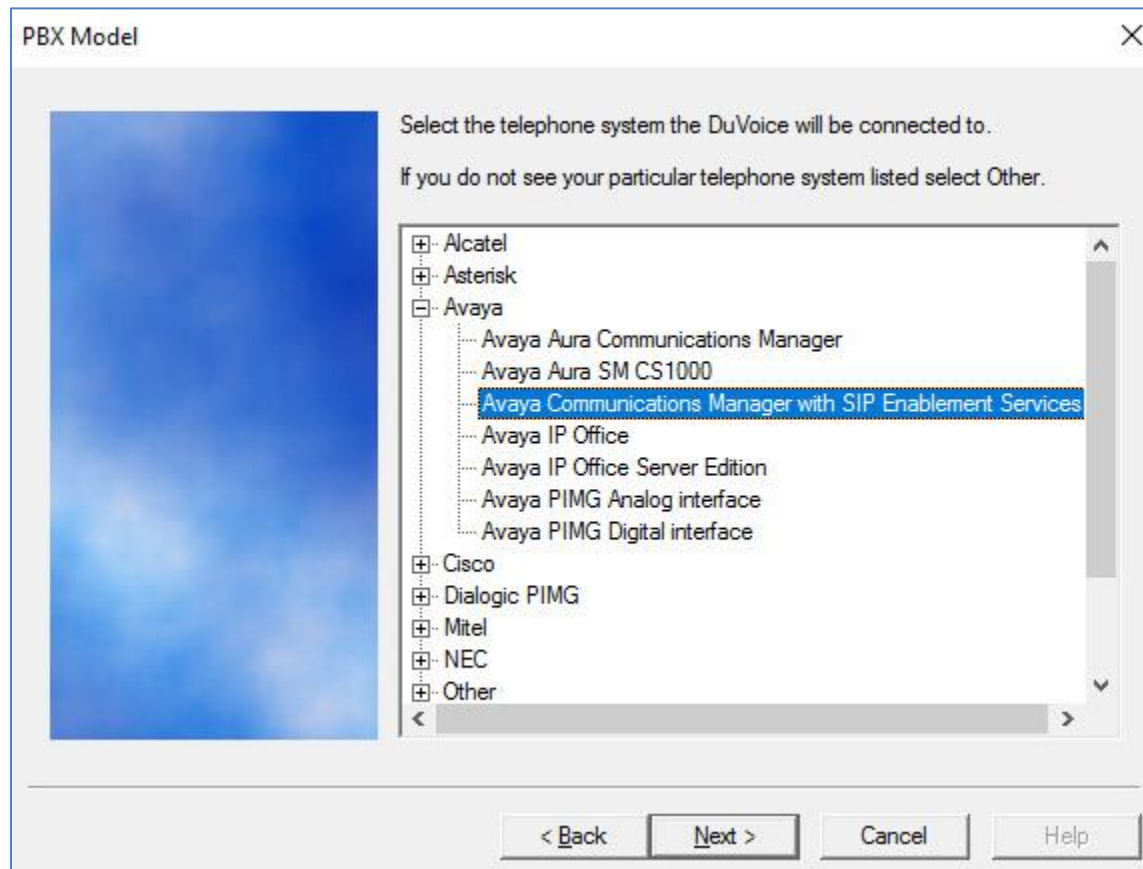
At the bottom are buttons for '< Back', 'Next >', 'Cancel', and 'Help'.

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

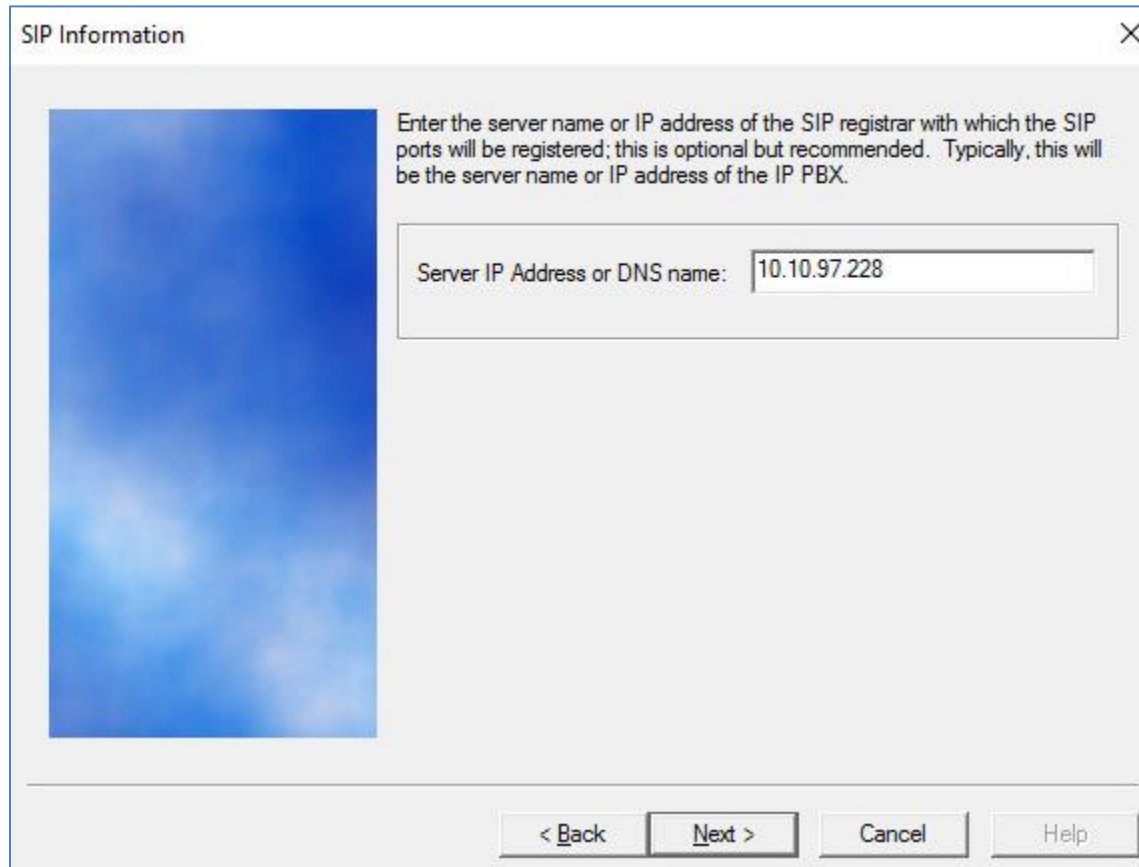


The 'Tenant Information' window features a blue placeholder image on the left. On the right, the 'Operator extension or huntgroup:' field is filled with '56604'. Below this is an unchecked checkbox with the text 'Enable operator to send/receive voice messages and receive VeMail.'. The bottom of the window includes buttons for '< Back', 'Next >', 'Cancel', and 'Help'.

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**.



The screenshot shows a window titled "SIP Information" with a close button (X) in the top right corner. On the left side of the window is a large blue rectangular area. To the right of this area, there is instructional text: "Enter the server name or IP address of the SIP registrar with which the SIP ports will be registered; this is optional but recommended. Typically, this will be the server name or IP address of the IP PBX." Below this text is a text input field labeled "Server IP Address or DNS name:" containing the IP address "10.10.97.228". At the bottom of the window, there are four buttons: "< Back", "Next >", "Cancel", and "Help".

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**.

MWI Method

Choose the method by which message waiting lights will be set and cleared.

- ☐ SIP Notify
- ☐ TAPI
- ☐ SMDI
- ☒ Inband using a feature or shortcode
- ☐ HTTP

Inband codes

Enter the code used to set and clear the message waiting lights. Enter an E for the extension number. If an E is not specified it will be automatically added to the end of the code.

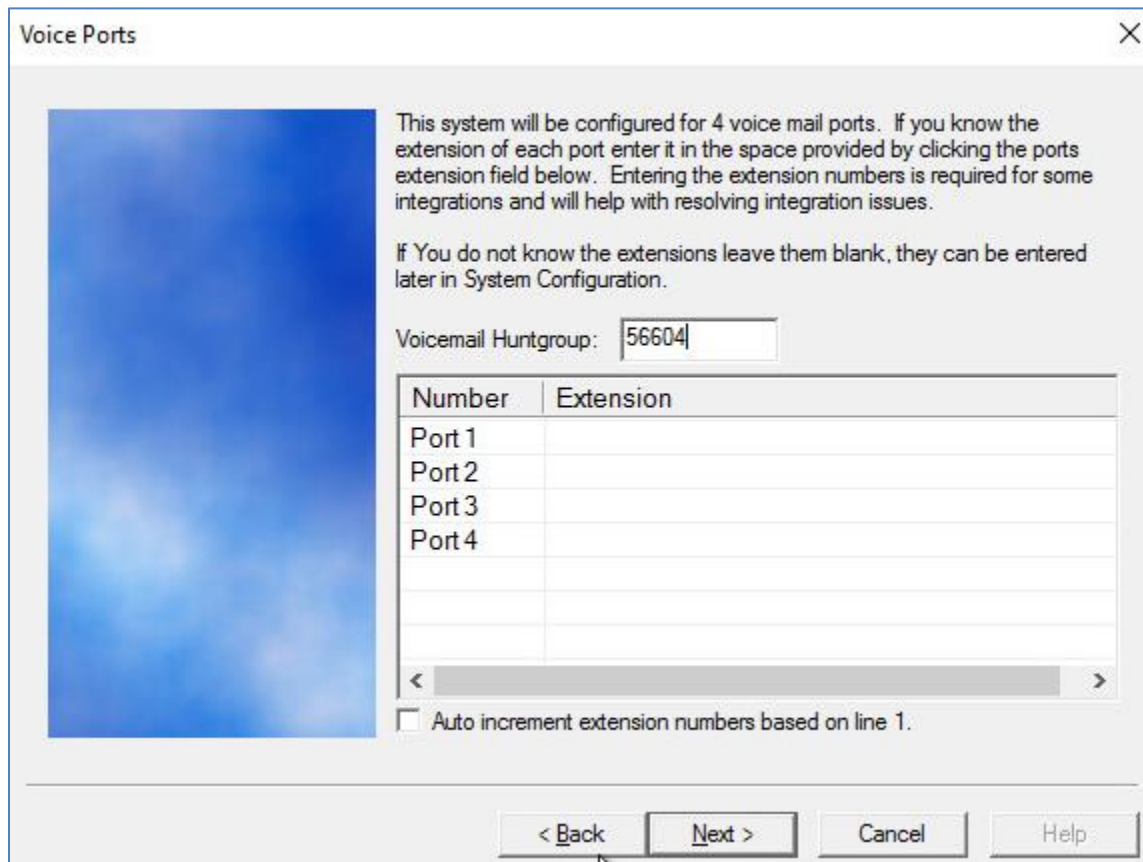
For example: *81*E* or *4E

Set code: #4E

Clear code: *4E

< Back Next > Cancel Help

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



Voice Ports

This system will be configured for 4 voice mail ports. If you know the extension of each port enter it in the space provided by clicking the ports extension field below. Entering the extension numbers is required for some integrations and will help with resolving integration issues.

If You do not know the extensions leave them blank, they can be entered later in System Configuration.

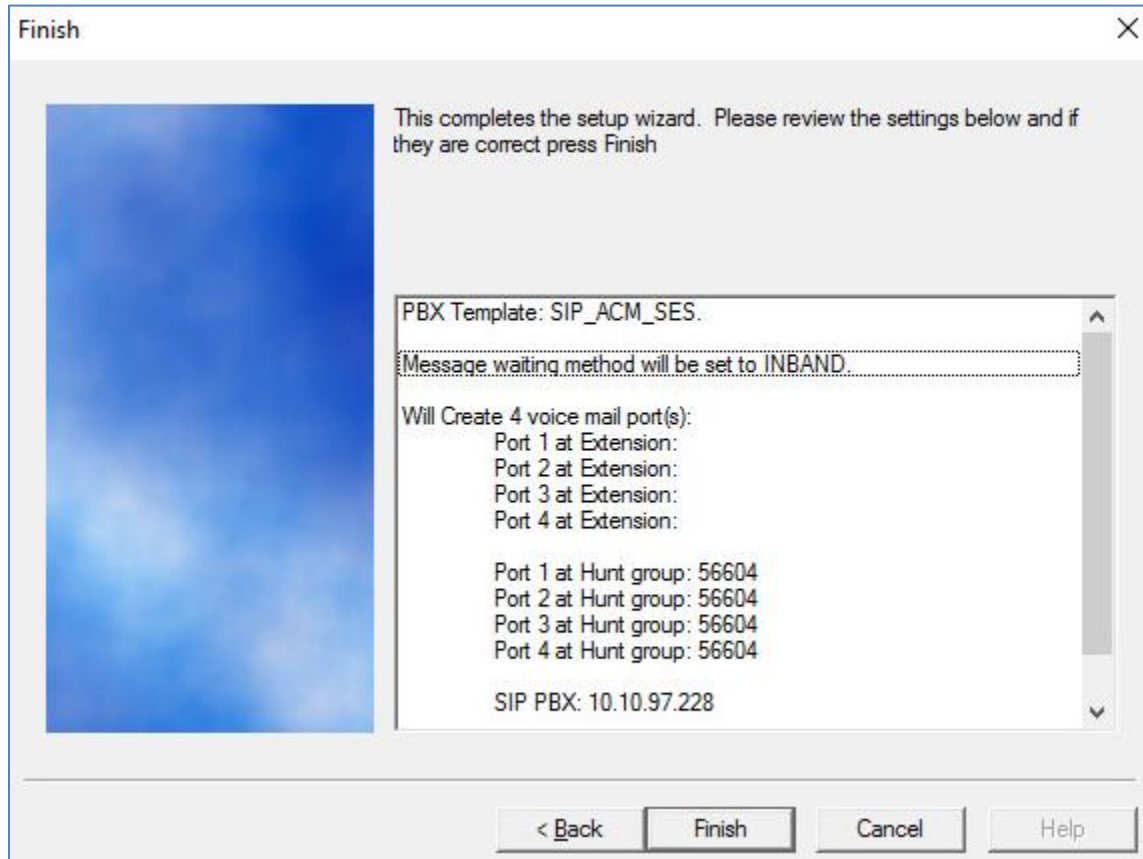
Voicemail Huntgroup: 56604

Number	Extension
Port 1	
Port 2	
Port 3	
Port 4	

☐ Auto increment extension numbers based on line 1.

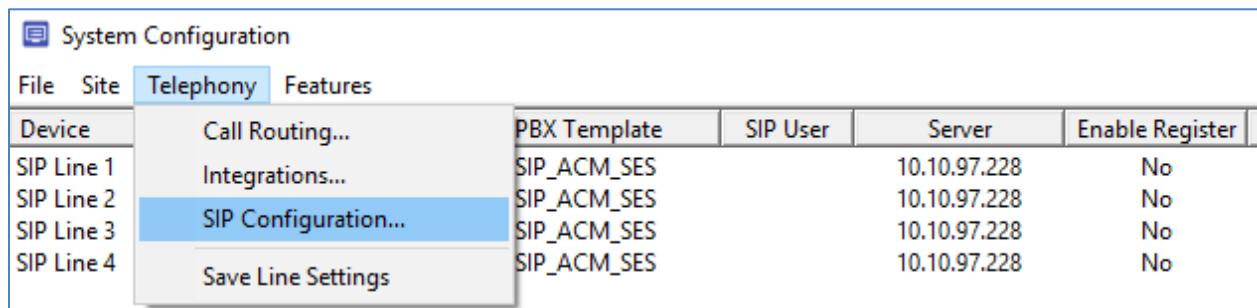
< Back Next > Cancel Help

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



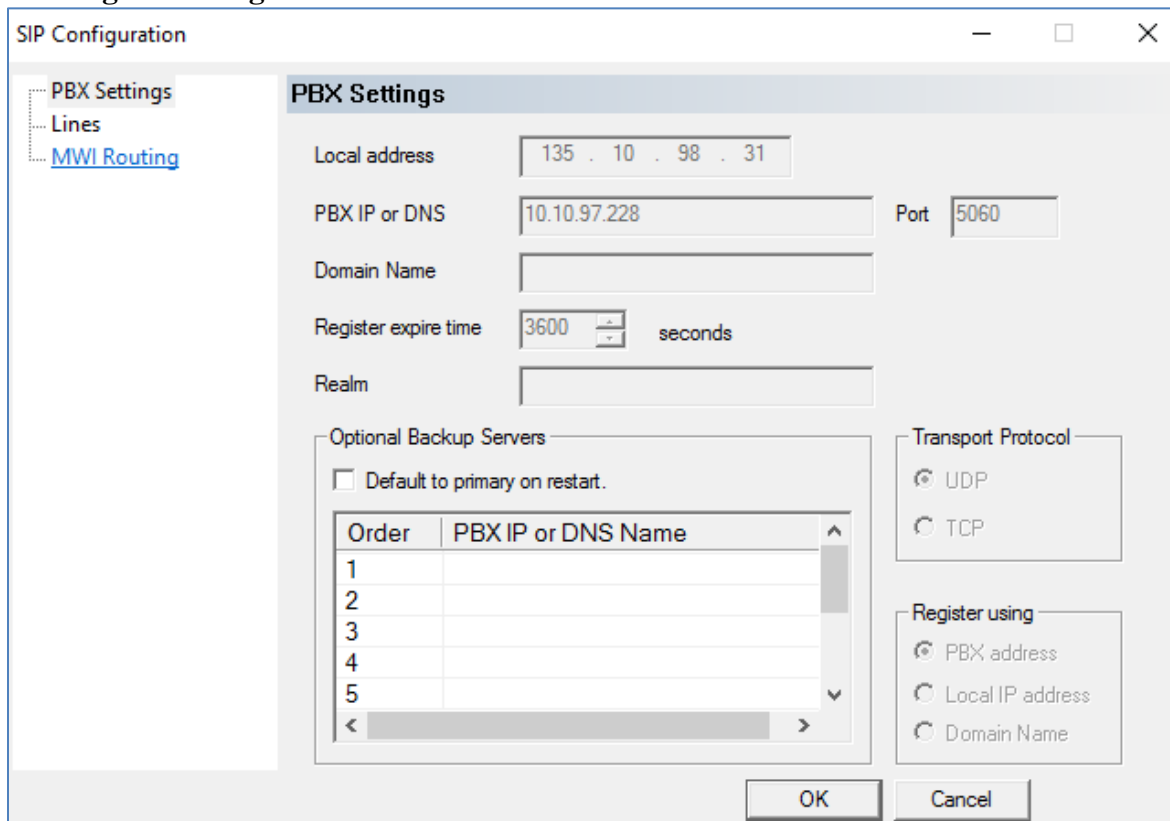
7.1. Configure SIP Configuration

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



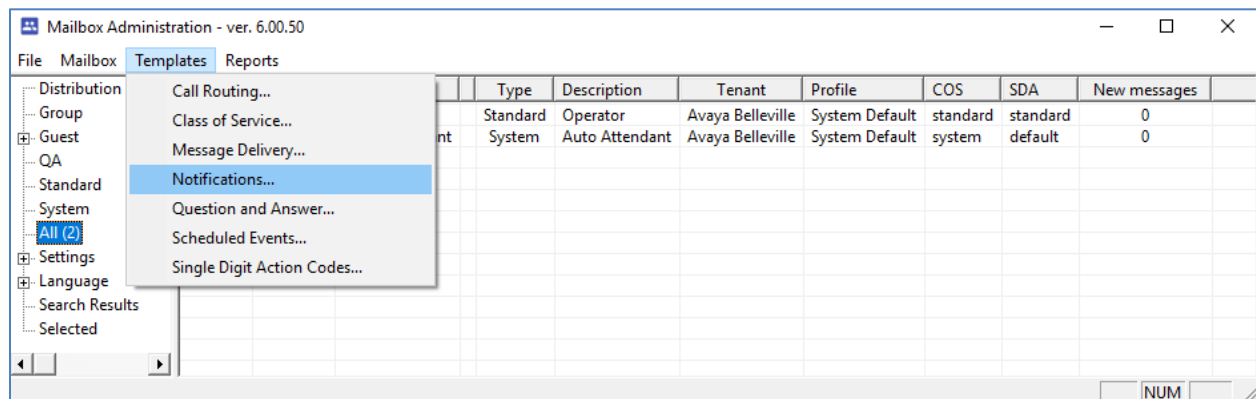
From the SIP configuration tab please confirm the following settings.

- **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.

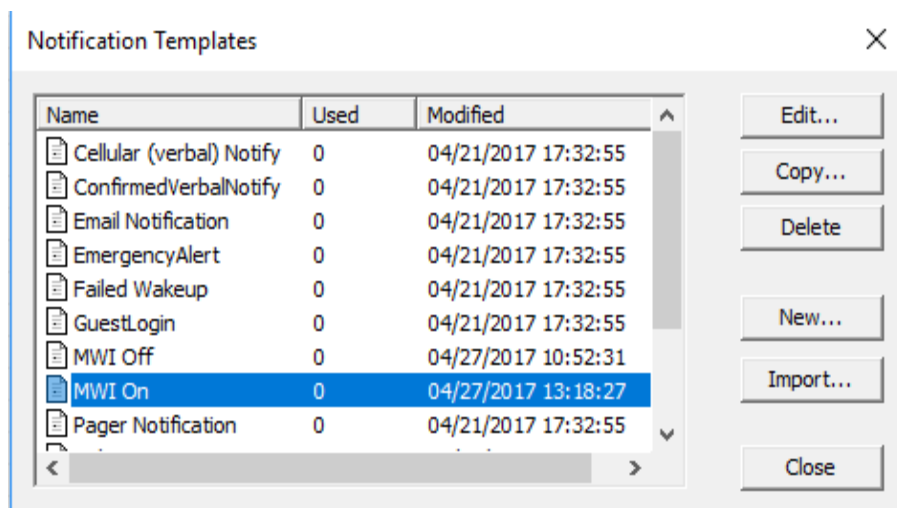


7.2. Configure SIP MWI

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:

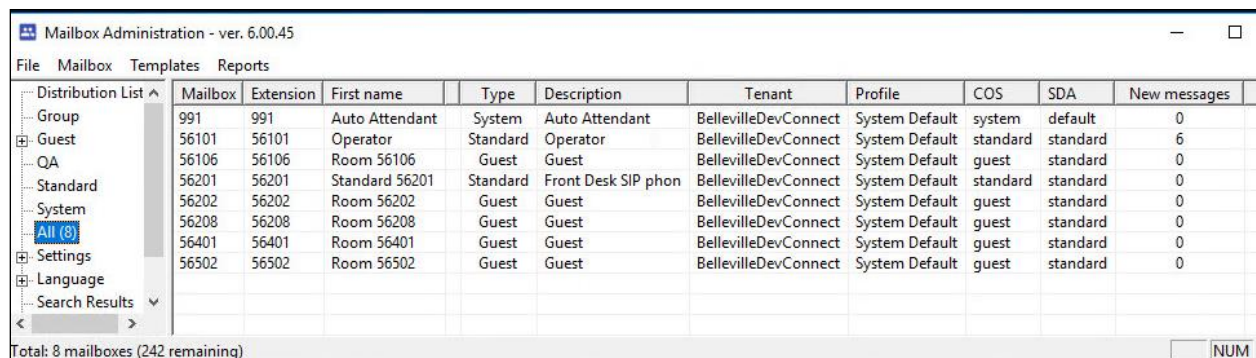


In the **MWI On** template window, in **Method** drop down list select SIP. Click **OK** to save changes.

The screenshot shows the 'MWI On' template window. The 'Definition' tab is active. The 'Event' is set to 'all messages', 'Address' is 'MWI', and 'Technique' is 'Message Waiting Indicator On'. The 'Method' dropdown menu is open, showing options: Inband, Inband + PMS, PBXLINK, PBXLINK + PMS, PMS only, Serial, Serial + PMS, SIP (highlighted), SIP + PMS, TAPI, and TAPI + PMS. The 'Schedule' tab is also visible, showing 'Days of the week this template is active' (Su, M, Tu, W, Th, F, Sa) and 'Time period during which this notification is active' (Starting at: 12:00 AM, Ending at: 12:00 AM). A 'Cancel' button is at the bottom right.

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.



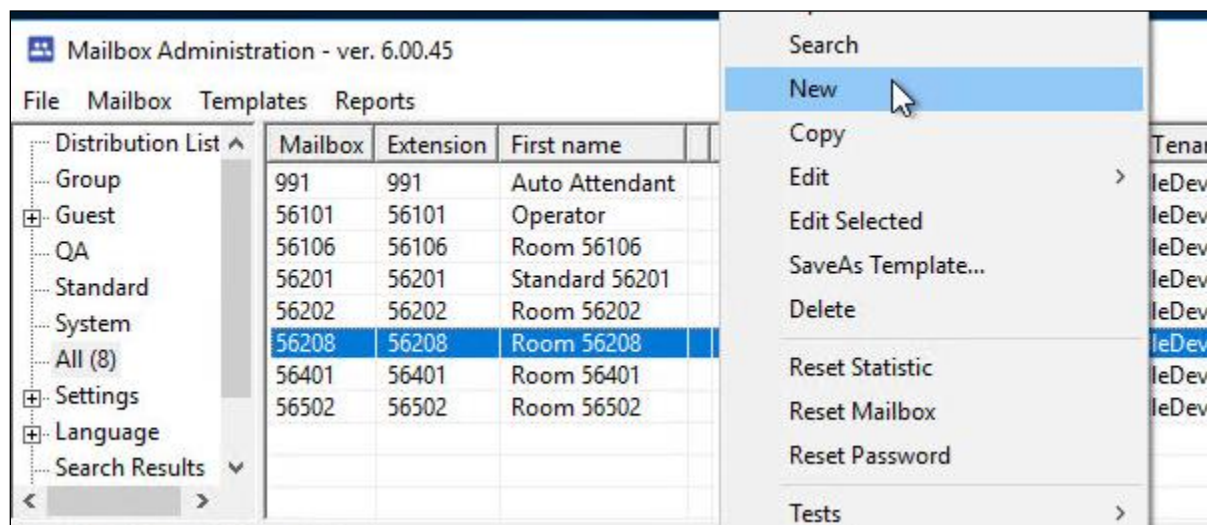
Mailbox Administration - ver. 6.00.45

File Mailbox Templates Reports

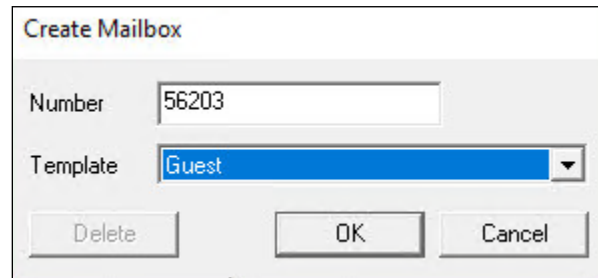
	Mailbox	Extension	First name	Type	Description	Tenant	Profile	COS	SDA	New messages
Group	991	991	Auto Attendant	System	Auto Attendant	BellevilleDevConnect	System Default	system	default	0
Guest	56101	56101	Operator	Standard	Operator	BellevilleDevConnect	System Default	standard	standard	6
QA	56106	56106	Room 56106	Guest	Guest	BellevilleDevConnect	System Default	guest	standard	0
Standard	56201	56201	Standard 56201	Standard	Front Desk SIP phon	BellevilleDevConnect	System Default	standard	standard	0
System	56202	56202	Room 56202	Guest	Guest	BellevilleDevConnect	System Default	guest	standard	0
All (8)	56208	56208	Room 56208	Guest	Guest	BellevilleDevConnect	System Default	guest	standard	0
Settings	56401	56401	Room 56401	Guest	Guest	BellevilleDevConnect	System Default	guest	standard	0
Language	56502	56502	Room 56502	Guest	Guest	BellevilleDevConnect	System Default	guest	standard	0
Search Results										

Total: 8 mailboxes (242 remaining) NUM

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

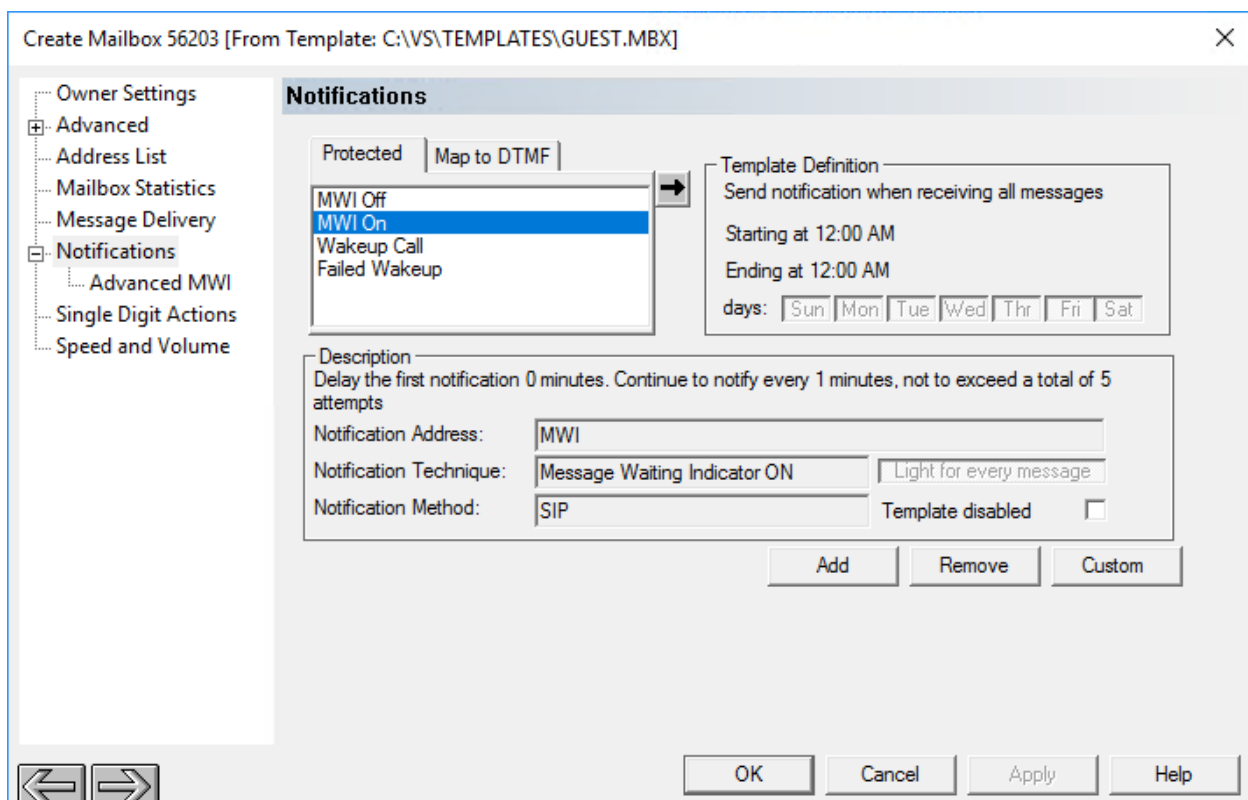


On the **Create Mailbox** window, enter the station extension and select **Guest** for **Template**, and click **OK**.



The 'Create Mailbox' dialog box is shown. It has a title bar 'Create Mailbox'. Inside, there is a 'Number' text box containing '56203' and a 'Template' dropdown menu with 'Guest' selected. At the bottom are three buttons: 'Delete', 'OK', and 'Cancel'.

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



The 'Create Mailbox 56203 [From Template: C:\VS\TEMPLATES\GUEST.MBX]' window is shown. It has a title bar with a close button. On the left is a tree view with the following items: 'Owner Settings', 'Advanced', 'Address List', 'Mailbox Statistics', 'Message Delivery', 'Notifications' (selected), 'Advanced MWI', 'Single Digit Actions', and 'Speed and Volume'. The main area is titled 'Notifications' and contains two tabs: 'Protected' and 'Map to DTMF'. Under 'Protected', there is a list box with 'MWI Off', 'MWI On' (selected), 'Wakeup Call', and 'Failed Wakeup'. To the right of this list is a 'Template Definition' box with the text 'Send notification when receiving all messages', 'Starting at 12:00 AM', 'Ending at 12:00 AM', and a 'days' row with buttons for 'Sun', 'Mon', 'Tue', 'Wed', 'Thr', 'Fri', and 'Sat'. Below the list box is a 'Description' box with the text 'Delay the first notification 0 minutes. Continue to notify every 1 minutes, not to exceed a total of 5 attempts'. Below the description are three text boxes: 'Notification Address' (containing 'MWI'), 'Notification Technique' (containing 'Message Waiting Indicator ON'), and 'Notification Method' (containing 'SIP'). To the right of the 'Notification Technique' box is a checkbox labeled 'Light for every message'. To the right of the 'Notification Method' box is a checkbox labeled 'Template disabled'. At the bottom of the main area are three buttons: 'Add', 'Remove', and 'Custom'. At the bottom of the window are four buttons: 'OK', 'Cancel', 'Apply', and 'Help'.

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

The screenshot shows a configuration window titled "Mailbox 56203 [Guest]". On the left is a tree view with the following items: "Owner Settings" (selected), "Advanced", "Address List", "Mailbox Statistics", "Message Delivery", "Notifications", "Single Digit Actions", and "Speed and Volume". The main area is titled "Owner Settings" and contains four sections:

- Owner Information:** Fields for Extension (56203), Password (masked with four asterisks), Title, First Name (Room), and Last Name.
- Properties:** Fields for Description (Guest), COS (guest), Profile (System Default), Tenant (BellevilleDevConnect), and Language (Default). A "Details ..." button is next to the COS field.
- Greeting:** A text input field and a "Browse..." button.
- Options:** Checkboxes for "Hide from Directory" (checked), "Tutorial Complete" (checked), and "Do Not Disturb On" (unchecked). A checkbox for "Language set by guest" is unchecked.

8. Verification Steps

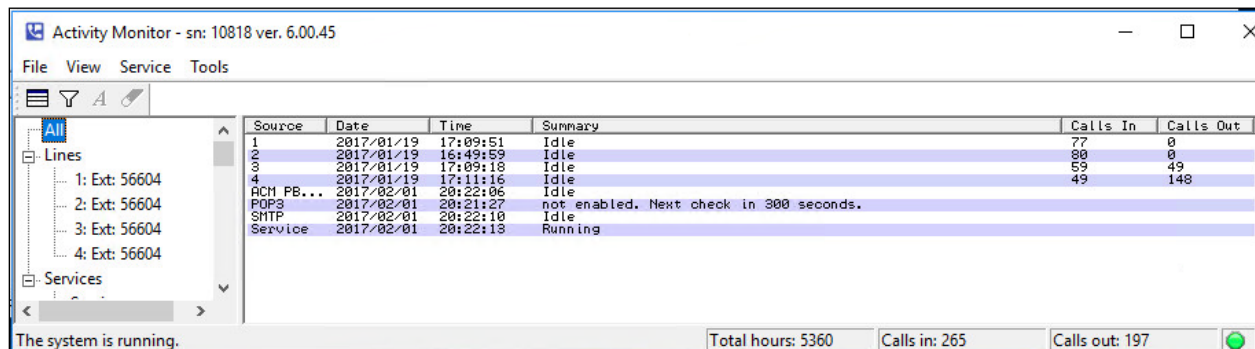
This section describes verification steps that may be used to verify SIP connectivity between DuVoice DV2000 VS6 and Session Manager.

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.



The screenshot shows the 'Activity Monitor' application window. The title bar reads 'Activity Monitor - sn: 10818 ver. 6.00.45'. The menu bar includes 'File', 'View', 'Service', and 'Tools'. On the left, there is a tree view with 'Lines' and 'Services' expanded. Under 'Lines', four entries are listed: '1: Ext: 56604', '2: Ext: 56604', '3: Ext: 56604', and '4: Ext: 56604'. Under 'Services', 'POP3' and 'SMTP' are listed. The main pane displays a table with columns: 'Source', 'Date', 'Time', 'Summary', 'Calls In', and 'Calls Out'. The table contains several rows of data, including 'Idle' status for various sources and a 'Running' status for 'Service'. At the bottom, a status bar indicates 'The system is running.' and provides summary statistics: 'Total hours: 5360', 'Calls in: 265', and 'Calls out: 197'.

Source	Date	Time	Summary	Calls In	Calls Out
1	2017/01/19	17:09:51	Idle	77	0
2	2017/01/19	16:49:59	Idle	80	0
3	2017/01/19	17:09:18	Idle	59	49
4	2017/01/19	17:11:16	Idle	49	148
ACM PB...	2017/02/01	20:22:06	Idle		
POP3	2017/02/01	20:21:27	not enabled. Next check in 300 seconds.		
SMTP	2017/02/01	20:22:10	Idle		
Service	2017/02/01	20:22:13	Running		

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.

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