



Avaya Solution & Interoperability Test Lab

**Application Notes for Configuring DuVoice DV2000
with Avaya Aura® Session Manager and Avaya Aura®
Communication Manager – Draft 0.1**

Abstract

These Application Notes contains interoperability instructions for configuring DuVoice DV2000 with Avaya Aura® Session Manager and Avaya Aura® Communication Manager. Compliance testing was conducted to verify the interoperability.

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 with Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and various Avaya endpoints. In the compliance testing SIP trunks were used in between the DuVoice DV2000 Voice Messaging System and Avaya Aura® Session Manager.

DuVoice DV2000 is a hospitality application that provides voicemail, automated attendant, and wake-up call features. The compliance testing focused on integrating the DuVoice DV2000 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. Depending on the type of call, the user then 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 were routed by Communication Manager to the DuVoice DV2000 hunt group via Session Manager, which were answered by the DV2000 with the automated attendant greeting. Internal calls that were unanswered were covered to the DV2000 hunt group. The DV2000 would answer these calls with the voice mailbox greeting of the subscriber extension. Lastly, internal calls placed to the DV2000 directly were answered by the DV2000 with the voicemail menu of the originating extension with an option to retrieve messages. For serviceability testing, the DV2000 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 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 Room Status Monitor functionality. The serviceability testing introduced failure scenarios to verify operation of the DuVoice DV2000 after failure recovery.

2.2. Test Results

All executed test cases were passed.

2.3. Support

3. Reference Configuration

Test configuration used during compliance testing consisted of following:

- Avaya G450 Media Gateway with Avaya 8300D Media Server running Avaya Aura® Communication Manager
- Avaya Aura® Session Manager
- Avaya Aura® System Manager
- DuVoice DV2000 running on a Windows 7 Enterprise

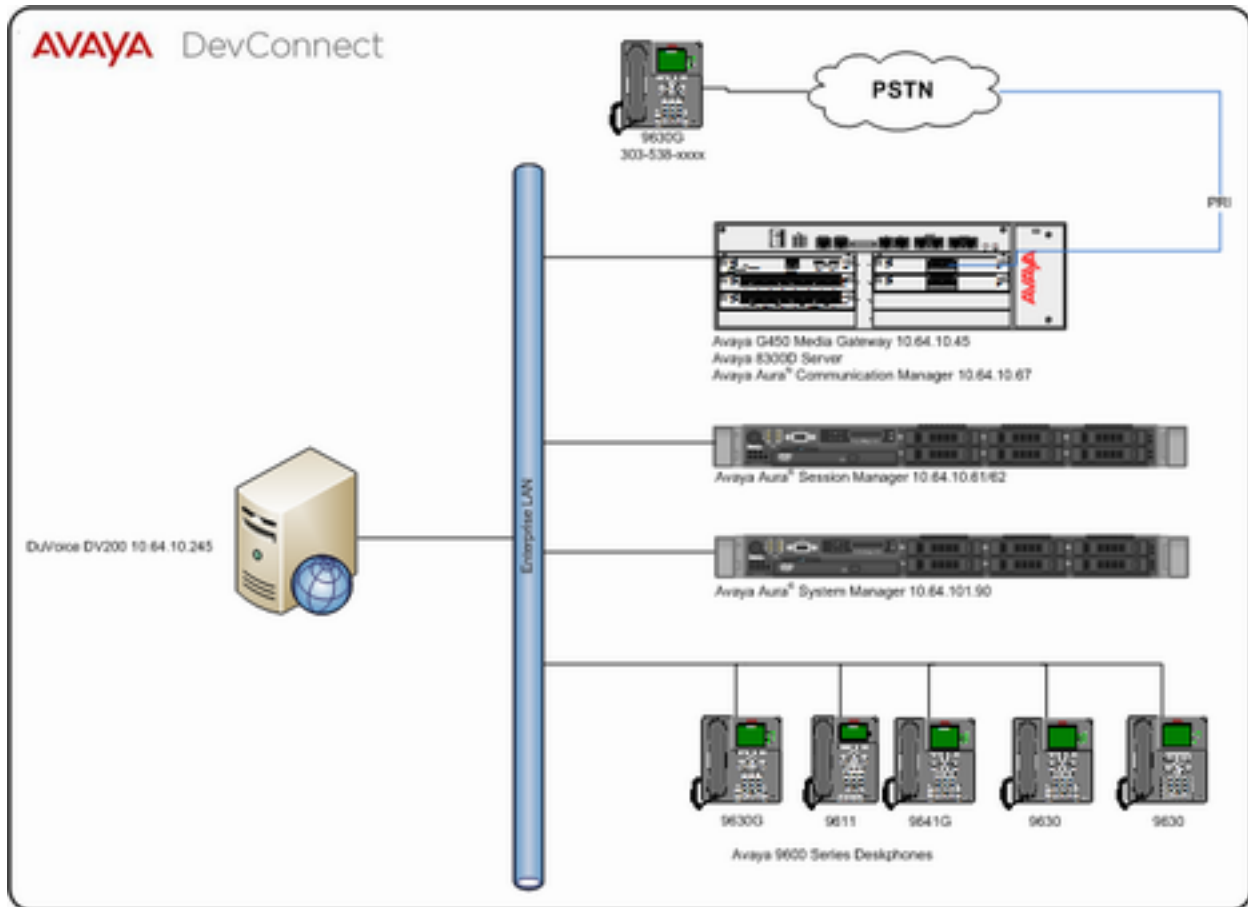


Figure 1: Reference Configuration

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 running on Avaya S8300D Server	6.2 SP5
Avaya Aura® Session Manager	6.3 SP5
Avaya Aura® System Manager	6.3 SP4
DuVoice DV2000 running on Windows Server 2008 R2	?

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).

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 and **Authoritative Domain**, e.g. avaya.com

```
change ip-network-region 1                                     Page 1 of 20
                                                           IP NETWORK REGION
Region: 1
Location: 1          Authoritative Domain: avaya.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
H.323 IP ENDPOINTS          AUDIO RESOURCE RESERVATION PARAMETERS
      H.323 Link Bounce Recovery? y          RSVP Enabled? n
      Idle Traffic Interval (sec): 20
      Keep-Alive Interval (sec): 5
      Keep-Alive Count: 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:
3:
4:
5:
6:
7:

Media Encryption
1:
2:
3:
```

5.3. Administer IP Node Names

Use the **change node-names ip** command to add an entry for Session Manager. For compliance testing, **sm** and **10.64.10.61** entry was added.

```
change node-names ip                                     Page 1 of 2

                               IP NODE NAMES

Name      IP Address
default   0.0.0.0
msgsrvr   10.64.10.67
procr     10.64.10.67
procr6    ::
sm        10.64.10.61
```

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 in **Section 5.4**, i.e. **sm**
- Set **Far-end Network region** to the configured region in **Section 5.2**, i.e. **1**

- Enter a **Far-end Domain**, e.g. avaya.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
Near-end Listen Port: 5061        Far-end Listen Port: 5061
                Far-end Network Region: 1

Far-end Domain: avaya.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 for 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 5.5**, 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                                     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 **Number 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:

- Type a name in **Pattern Name** field
- For line 1, set **Grp No** to the trunk group configured in **Section 5.6**, i.e. 1
- For line 1, set **FRL** to **0**

```
change route-pattern 1                                     Page 1 of 3
      Pattern Number: 1   Pattern Name: Voice
      SCCAN? n           Secure SIP? n
      Grp FRL NPA Pfx Hop Toll No.  Inserted           DCS/ IXC
      No           Mrk Lmt List Del  Digits           QSIG
                                                    Intw
1: 1      0
2:
```

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 6                                         Page 1 of 60
      HUNT GROUP
      Group Number: 6
      Group Name: DuVoice Voicemail
      Group Extension: 25099
      Group Type: ucd-mia
      TN: 1
      COR: 1
      Security Code:
      ISDN/SIP Caller Display:
      ACD? n
      Queue? n
      Vector? n
      Coverage Path:
      Night Service Destination:
      MM Early Answer? n
      Local Agent Preference? n
```

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**, for **Point1** type in the hunt group that was configured in previous section. e.g., h6, where h stands for hunt group and 6 is the hunt group number.

```

add coverage path 6                                     Page 1 of 1
                                COVERAGE PATH
                                Coverage Path Number: 6
                                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: 2
    All?                   n              n
  DND/SAC/Goto Cover?    y              y
  Holiday Coverage?      n              n

COVERAGE POINTS
  Terminate to Coverage Pts. with Bridged Appearances? n
  Point1: h6              Rng:          Point2:
  Point3:                  Point4:
  Point5:                  Point6:
  
```

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 5.6**

Note: For compliance testing, 5-digit hunt group extension 25099 routed over trunk groups 1 resulted in a 5-digit calling party number.

```

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

Ext  Ext      Trk      Private      Total
Len  Code      Grp(s)     Prefix       Len
  5  25099      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 25099 was used for routing calls to DV2000.

- Set **Dialed String** to hunt group extension, e.g. 25099
- Set **Min** and **Max** to 5 for 5 digit extensions
- Set **Route Pattern** to pattern configured in **Section 5.7**, i.e. 1
- Set **Call Type** to **aar**

Note: An entry to dial plan will need to be added for extension range used in this step.

```
change aar analysis 25099                                     Page 1 of 2
AAR DIGIT ANALYSIS TABLE
Location: all                                               Percent Full: 2
```

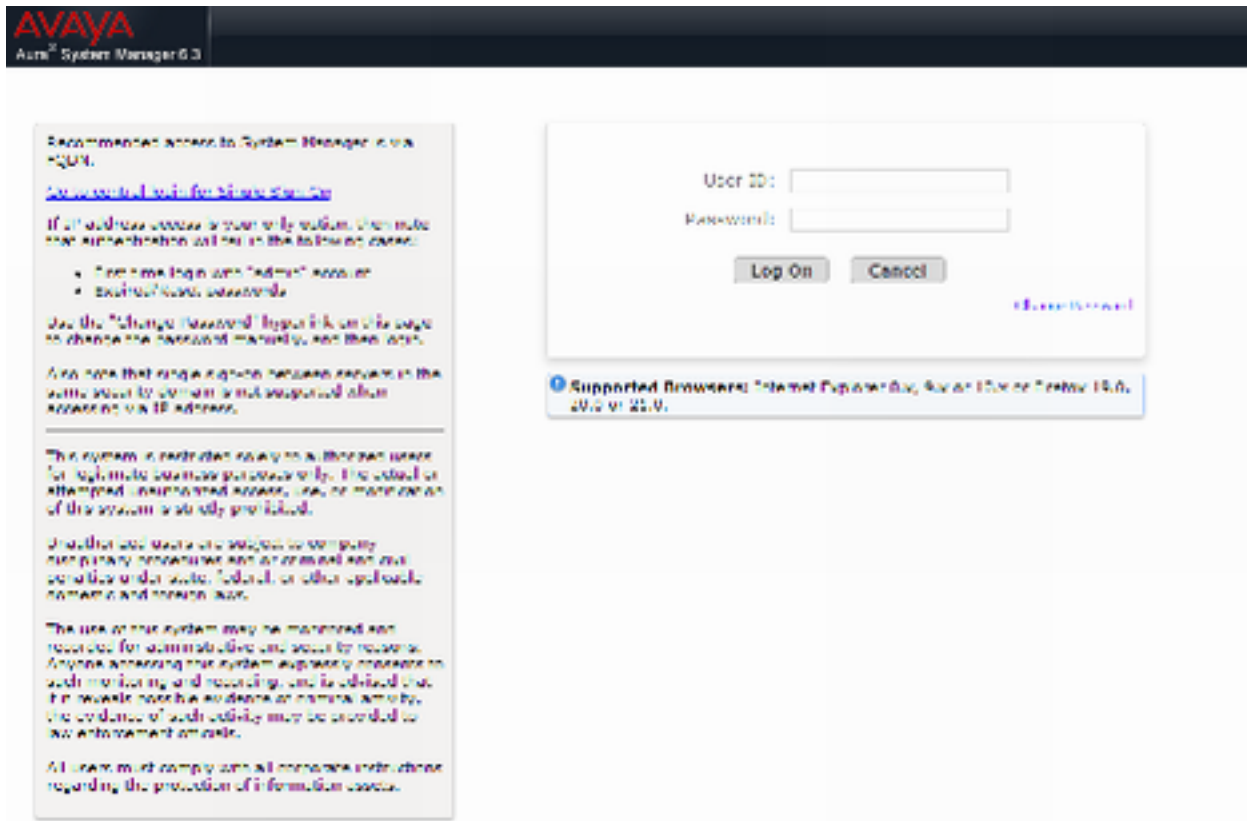
Dialed String	Total		Route Pattern	Call Type	Node Num	ANI Reqd
	Min	Max				
25099	5	5	1	aar	n	
252	5	5	2	aar	n	
257	5	5	10	aar	n	
258	5	5	10	aar	n	
25990	5	5	13	aar	n	
25999	5	5	98	aar	n	
26	5	5	10	aar	n	
27	5	5	21	aar	n	
275	5	5	10	aar	n	

5.11. Administer Stations

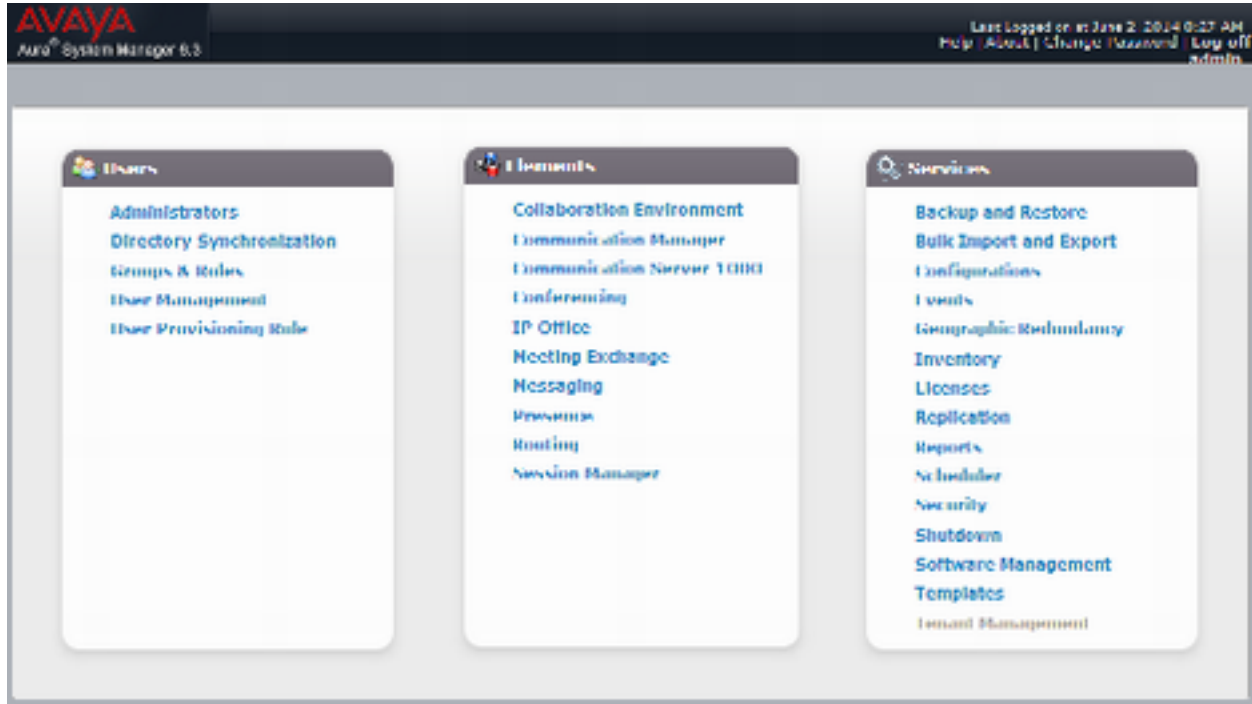
Administration of Avaya Stations/Extensions in Communication Manager and Session Manager is not shown in this document. Please refer to document [1] and/or [2] in reference section of this document.

6. Configure Avaya Aura® Session Manager

Configuration of Avaya Aura® Session Manager is performed via Avaya Aura® System Manager. Access the System Manager Administration web interface by entering <https://<ip-address>/SMGR> URL in a web browser, where <ip-address> is the IP address of System Manager.



Log in using appropriate credentials.

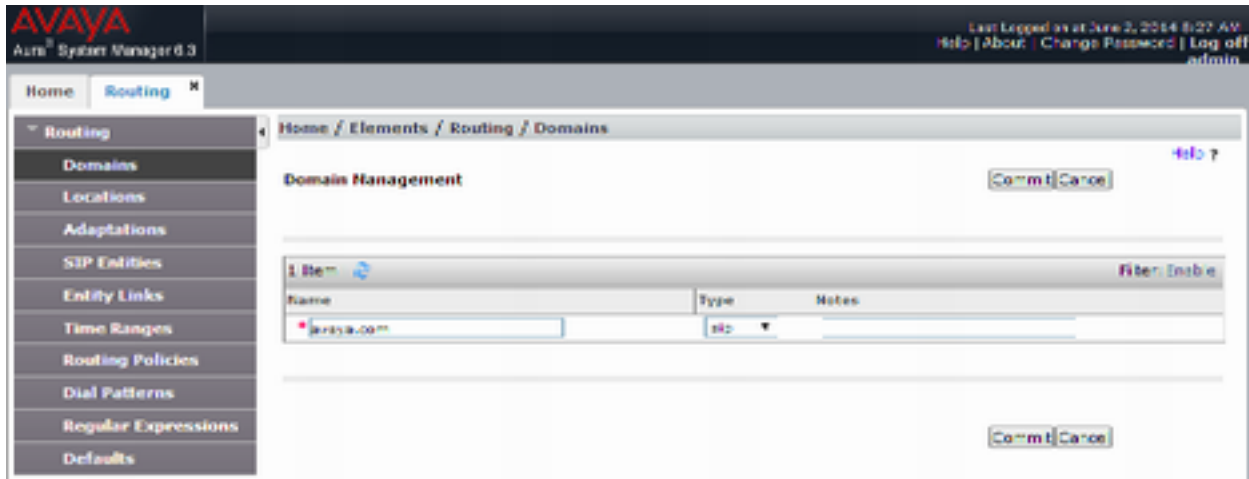


6.1. Add SIP Domain

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**) i.e. avaya.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**, e.g. 10.64.10.*

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

Home / Devices / Routing / LocalBase

Location Details Commit Cancel

General

Room: **Test Room 1**
 Name:

Dual Plan Transparency in Summable Mode

Initial Download Method:

Associated ONSCP Entry:

Overall Managed Bandwidth

Managed Bandwidth Unit: **kb/sec**

Total Bandwidth:

FullMedia Bandwidth:

Associated Control Plane Bandwidth

Per-Call Bandwidth Parameters

Maximum FullMedia Bandwidth (kb/sec) kb/sec

Maximum FullMedia Bandwidth (kb/sec) kb/sec

Maximum FullMedia Bandwidth kb/sec

Default Audio Bandwidth: kb/sec

Alarm Threshold

Overall Alarm Threshold: %

FullMedia Alarm Threshold: %

Latency before Overall Alarm Trigger: seconds

Latency before FullMedia Alarm Trigger: seconds

Location Pattern

Pattern	Total
IP Address Pattern	
* 0.0.0.0	
* 0.0.0.0	

Select: All, None

6.3. Add SIP Entity – Communication Manager

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 Communication Manager in **FQDN or IP Address** field.
- Set **Type** to **CM**
- Set **Location** to the location configured in **Section 6.2**

Click **Commit** to save changes.

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

SIP Entity Details

Commit Cancel

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

Loop Detection

Loop Detection Mode:

SIP Link Monitoring

SIP Link Monitoring:

Supports Call Admission Control:

Shared Bandwidth Manager:

Primary Session Manager Bandwidth Association:

Backup Session Manager Bandwidth Association:

6.4. Add Entity Link – Communication Manager

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 Communication Manager SIP Entity configured in **Section 6.3**

Click **Commit** to save changes.

Name	SIP Entity 1	Protocol	Port	SIP Entity 2	Port	Connection Policy	Notes
CM	SM_Public	TLS	5061	Communication Manager	5061	Trusted	

6.5. 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 in **FQDN or IP Address** field.
- Set **Type** to **SIP Trunk**
- Set **Location** to the location configured in **Section 6.2**

Click **Commit** to save changes.

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

SIP Entity Details Commit Cancel

General

* Name:

* FQDN or IP Address:

Type:

Notes:

Adaptation:

Location:

Time Zone:

* SIP Timer B/F (in seconds):

Credential name:

Call Detail Recording:

Loop Detection

Loop Detection Mode:

SIP Link Monitoring

SIP Link Monitoring:

6.6. 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 SIP Entity configured in **Section 6.5**
- Set **Protocol** to **UDP**

Click **Commit** to save changes.

Entity Links Commit Cancel

1 Item Filter: Enable									
<input type="checkbox"/>	Name	SIP Entity 1	Protocol	Port	SIP Entity 2	DNS Override	Port	Connect Policy	
<input type="checkbox"/>	* asm-tr1_dv2000-tr	* asm-tr1	TCP	* 5060	* dv2000-tr1	<input type="checkbox"/>	* 5060	trusted	

Select : All, None

6.7. Add Time Ranges

Navigate to **Home** → **Elements** → **Routing** → **Time Ranges**, click on **New** (now shown) and configure as follows:

- Type in a descriptive name in **Name** field

Click **Commit** to save changes.

Time Ranges Commit Cancel

1 Item | Refresh Filter: Enable

Name	Mo	Tu	We	Th	Fr	Sa	Su	Start Time	End Time	Notes
* TimeRange1	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	* 00:00	* 23:59	

6.8. Add Routing Policy – Communication Manager

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 Communication Manager SIP entity added in **Section 6.3**
- Under **Time of Day**, click on **Add** (not shown):
 - Select time range added in previous step

Click **Commit** to save changes.

Routing Policy Details Commit Cancel

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
cm-tr1	10.64.10.67	CM	Avaya Aura® Communication Manager - Test Room 1

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	TimeRange	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	00:00	23:59	

Select : All, None

6.9. 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 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.

Routing Policy Details
Commit Cancel

General

* Name:

Disabled:

* Retries:

Notes:

SIP Entity as Destination

Select

Name	FQDN or IP Address	Type	Notes
dv2000-tr1	10.64.10.245	SIP Trunk	DuVoice DV2000

Time of Day

Add Remove View Gaps/Overlaps

1 Item Filter: Enable

Ranking	Name	Mon	Tue	Wed	Thu	Fri	Sat	Sun	Start Time	End Time	Notes
0	TimeRange	☑	☑	☑	☑	☑	☑	☑	00:00	23:59	

Select : All, None

6.10. Add Dial Patterns – Communication Manager

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 **Domain** to domain configured on **Section 6.1**

Under **Originating Locations and Routing Policies**:

- Click **Add** and select originating location and Communication Manager routing policy as configured in **Section 6.8**

Click **Commit** to save changes.

Note: For Compliance testing, dialed number of 25xxx were used to route calls to Communication Manager. Thus, pattern, min and max values were all set to 5.

Dial Pattern Details Commit Cancel

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

1 Item Filter: 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"/>	-ALL-		cm-tr1		<input type="checkbox"/>	cm-tr1	

Select : All, None

6.11. 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 **Domain** to **-All-**

Under **Originating Locations and Routing Policies**:

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

Click **Commit** to save changes.

Note: For Compliance testing, dialed number of 25099 were used to route calls to DuVoice. Thus, pattern, min and max values were all set to 5.

Dial Pattern Details Commit Cancel

General

* **Pattern:**

* **Min:**

* **Max:**

Emergency Call:

Emergency Priority:

Emergency Type:

SIP Domain:

Notes:

Originating Locations and Routing Policies

3 Items Filter: Enable

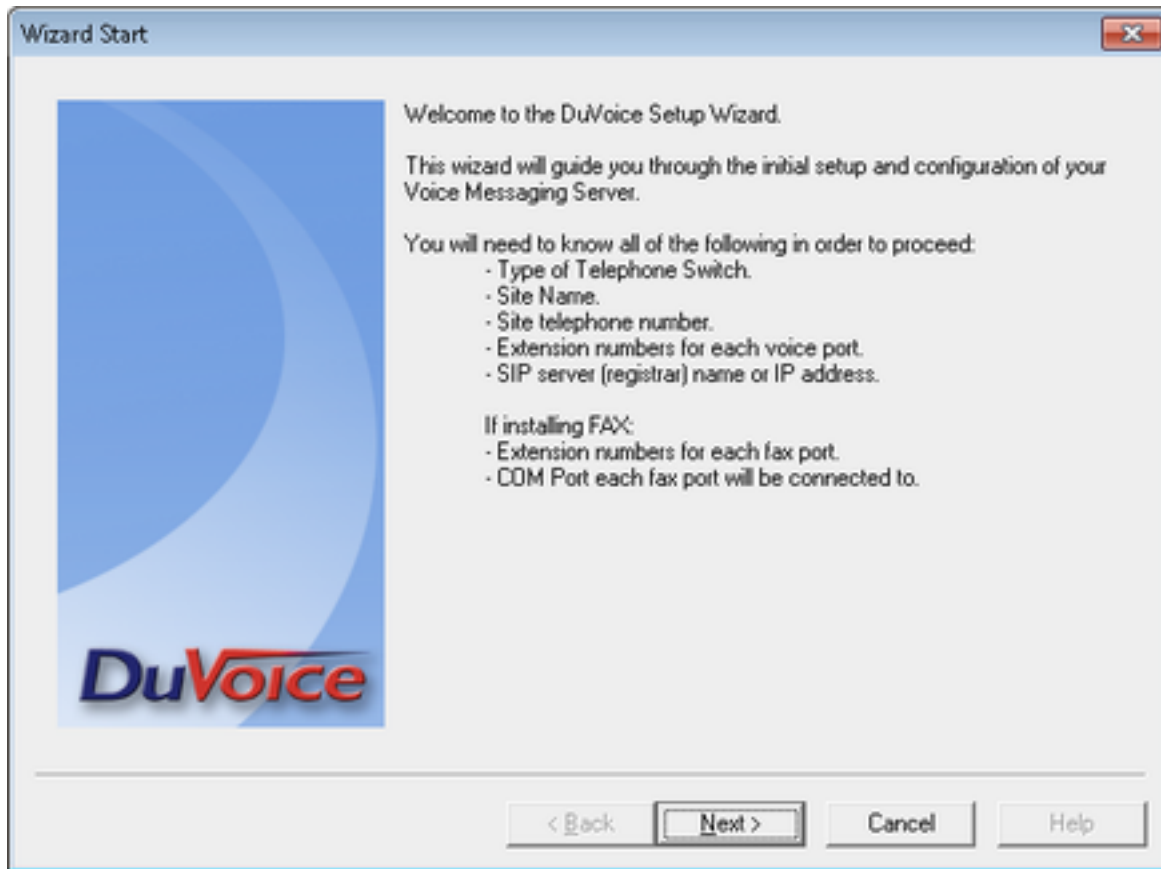
<input type="checkbox"/>	Originating Location Name	Originating Location Notes	Routing Policy Name	Rank	Routing Policy Disabled	Routing Policy Destination	Routing Policy Notes
<input type="checkbox"/>	Test Room 1		dv2000-tr1	0	<input type="checkbox"/>	dv2000-tr1	
<input type="checkbox"/>	Test Room 2		dv2000-tr1	0	<input type="checkbox"/>	dv2000-tr1	
<input type="checkbox"/>	Test Room 3		dv2000-tr1	0	<input type="checkbox"/>	dv2000-tr1	

Select : All, None

7. Configure DuVoice

To configure SIP connectivity to Session Manager, locate the SETUP.exe file for DuVoice DV2000 and open it.

On the **Wizard Start** window select **Next**



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

Site Information

Enter the required site information. The optional information should be filled in if you are using any type of FAX application.

Site Information

Name: *

Site telephone number: () - *

Fax telephone number: () -

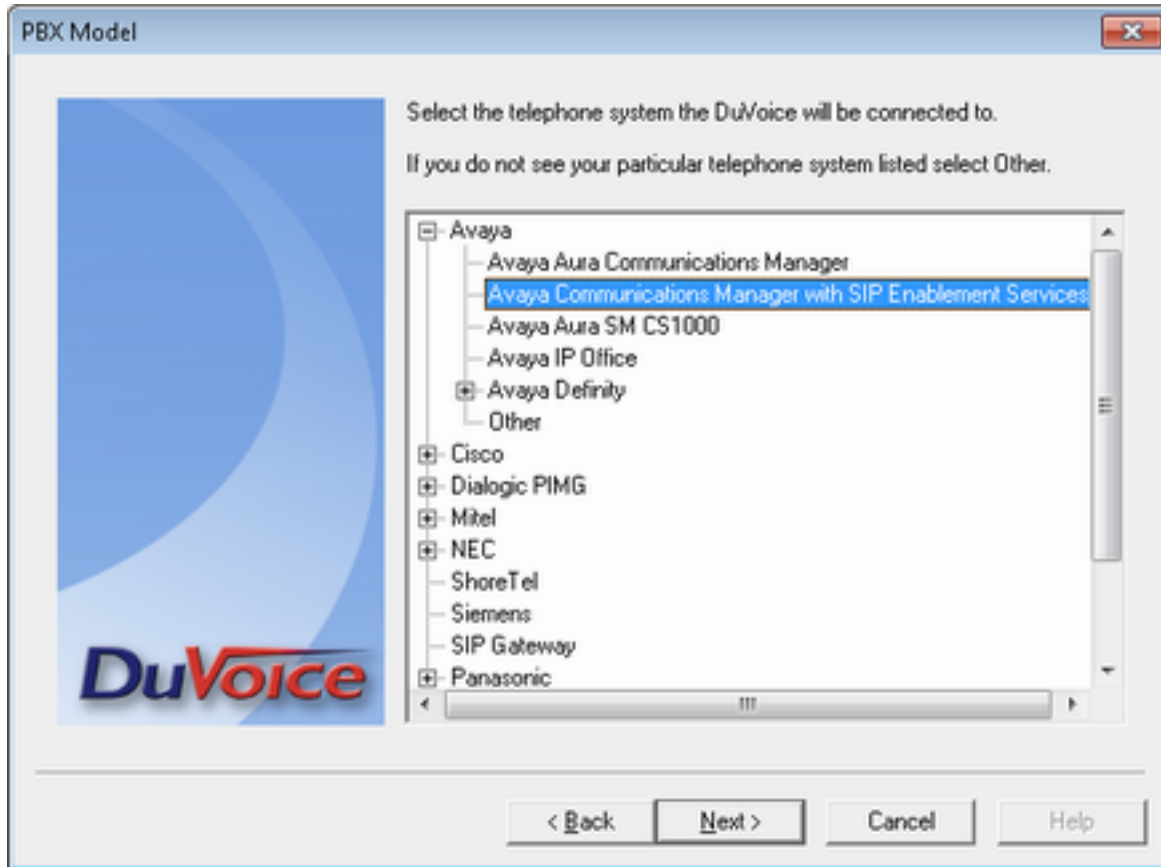
Address: * City: *

State or Province: * Zip or Postal Code: *

* Items required.

< Back **Next >** Cancel Help

On the **PBX Model** window, select **Avaya** → **Avaya Communication Manager with SIP Enablement Services** 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 a later section.

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.

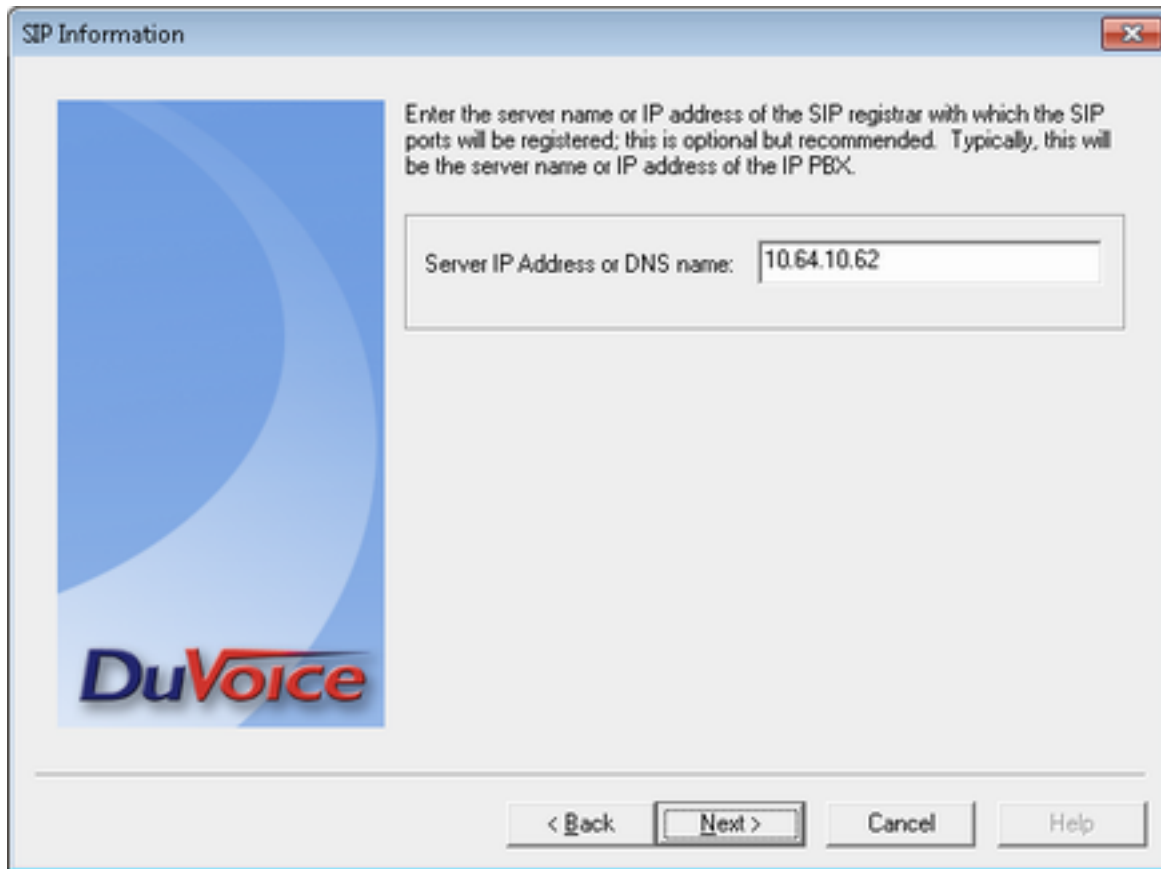
For example: *81*E* or *4E

Set code Clear code

#4E *4E

< Back Next > Cancel Help

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

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:

Auto increment extension numbers based on line 1.

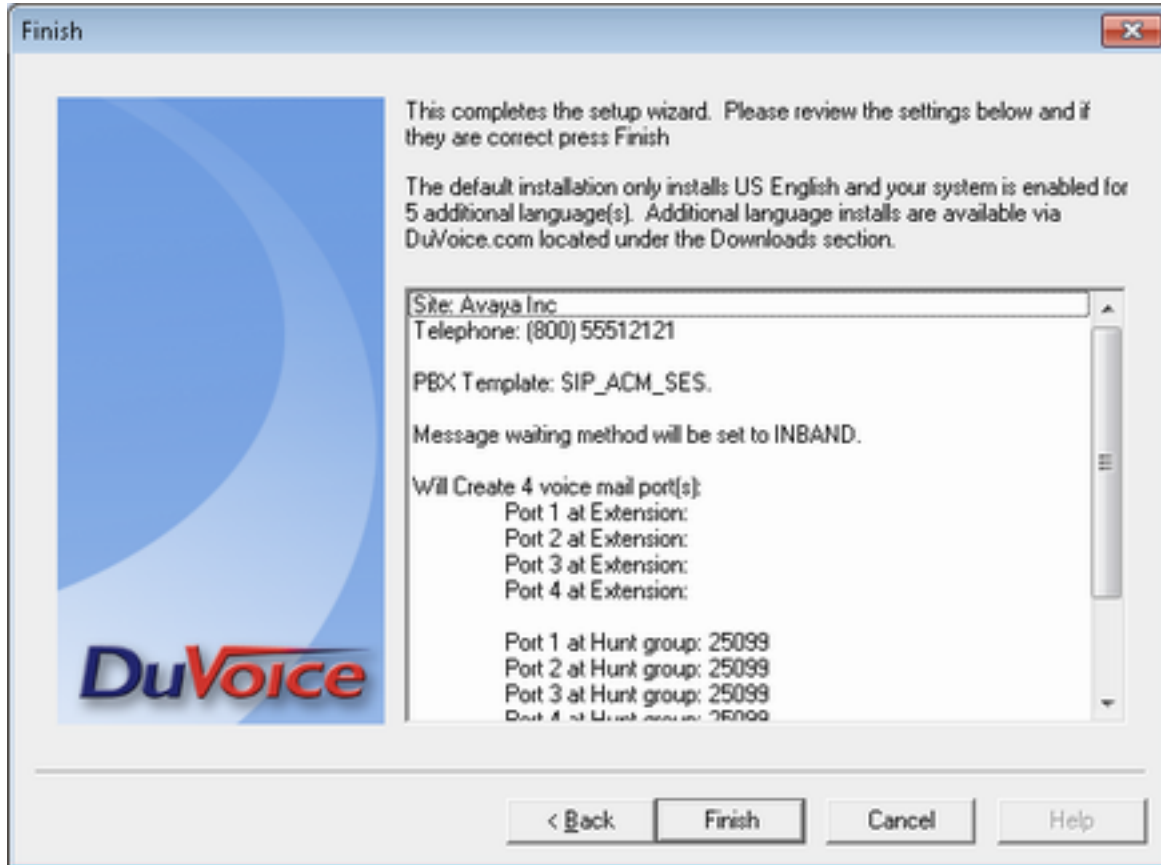
Number	Extension
Port 1	
Port 2	
Port 3	
Port 4	

< Back Next > Cancel Help

Update:
Enter the huntgroup number in the extension field for each line.

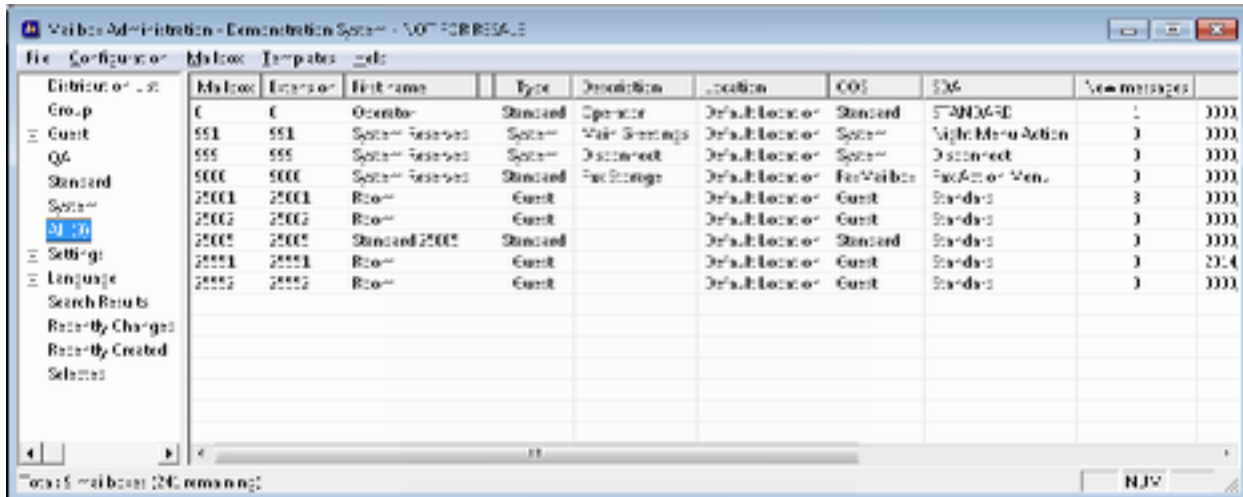
If this is not done, on some systems the PBX will give an error about the identity being wrong when callouts are attempted.

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

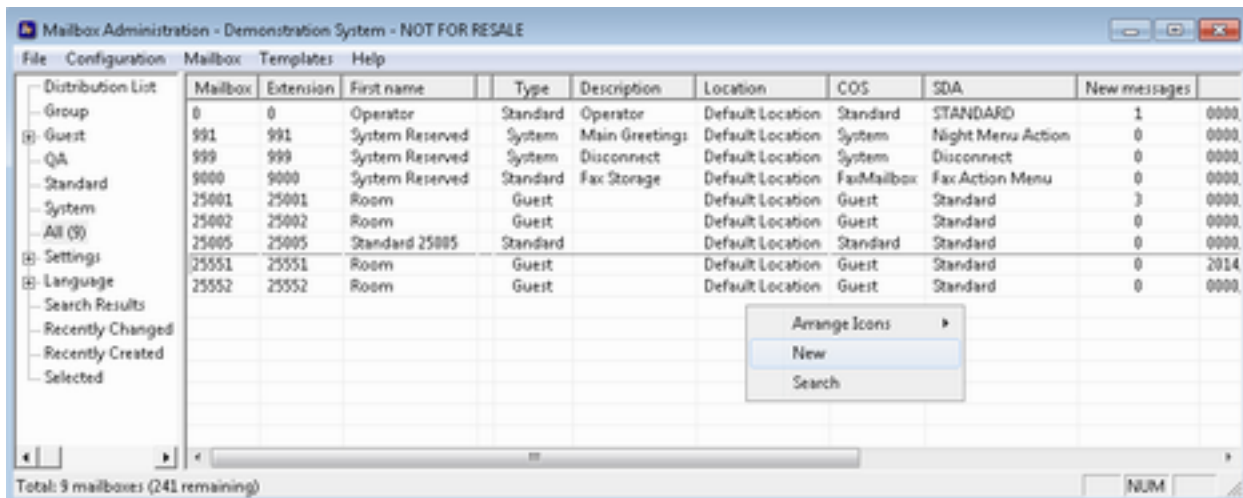


7.1. Configure MailBox

To configure mail boxes for guests, open Mailbox Administration, and select **All** in the left pane.



To add a mail box, right click on the right pane and select **New**.



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

Create Mailbox

Mailbox Number: 25001

Create Based On:

- Mailbox Type: Guest
- Mailbox Template: AudioText

Guest mailbox.

OK Cancel

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

Create Mailbox 25004

Owner Settings

Owner Information

Extension: 25001

Password: ****

First Name: Room

Last Name:

Properties

Description:

COS: Guest

Location: Default Location

Language: Default

Greeting

Browse...

Options

- Hide from Directory
- Tutorial Complete
- Call Blocking On
- Language set by guest

OK Cancel Apply Help

8. Verification Steps

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

8.1. Avaya Aura® Session Manager

On the System Manager, navigate to **Home → Element → Session Manager → System Station → SIP Entity Monitoring**.

Verify the **Conn. Status** and **Reason Code** are **Up** and **200 OK**.

SIP Entity, Entity Link Connection Status

This page displays detailed connection status for all entity links from all Session Manager instances to a single SIP entity.

All Entity Links to SIP Entity: **dv2000-tr1**

Status Details for the selected Session Manager:

Summary View

1 Items | Refresh

Filter: Enable

	Session Manager	SIP Entity Resolved IP	Port	Proto.	Deny	Conn. Status	Reason Code	Link Status
<input type="radio"/>	asm-tr1	10.64.10.24	5060	UDP	FALSE	UP	200 OK	UP

9. Conclusion

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

10. Additional References

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

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