Avaya Solution & Interoperability Test Lab

Application Notes for DuVoice DV2000 with Avaya Communication Server 1000 Release 7.6 and Avaya Aura® Session Manager 6.3 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for the DuVoice DV2000 Hospitality Voice Messaging System to operate with Avaya SIP enabled enterprise solution. The Avaya SIP enabled enterprise solution consists of Avaya Communication Server 1000, Avaya Aura® Session Manager, and various Avaya endpoints. In the compliance testing SIP trunks were used in between the DuVoice DV2000 Messaging System and Avaya Aura® Session Manager. DuVoice DV2000 uses rlogin through ELAN to access Avaya Communication Server 1000 to provide Property Management System features such as check in/out, room clean status, do not disturb, guest name change, and move room.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for the DuVoice DV2000 to operate with Avaya SIP enabled enterprise solution. The Avaya SIP enabled enterprise solution consists of Avaya Aura® Communication Manager, Avaya Aura® Session Manager, and various Avaya endpoints.

DuVoice DV2000 is a hospitality application that provides voicemail, automated attendant, wake-up call features. And DV2000 provides Property Management System (PMS) features such as check in/out, room clean status, do not disturb, guest name change, and move room.

In the compliance testing SIP trunks were used in between the DuVoice DV2000 server and Avaya Aura® Session Manager, Avaya Communication Server 1000, the DuVoice server with a physical connection to the Local Area Network (LAN).

For the voicemail coverage scenarios, voicemail messages were recorded and saved on the DuVoice server. Standard SIP messaging was used to activate/deactivate the MWI, to transfer the call via automated attendant or to schedule wakeup calls when requested manually by the guests.

InnDesk is a Web based used by the hotel staff to manage wakeup calls. InnDesk was used to schedule wakeup call, to view failed wakeup call. Not all capabilities of InnDesk were tested, only capabilities related to wake up services.

Hospitality Tester is Window base application, used to check in/out room, update guest name, move room, set/clear DND. The Hospitality features are enabled by a PMS data link to Avaya Communication Server 1000. The data link used between Avaya CS1000 and DV2000 is Rlogin via ELAN of Communication Server 1000.

Please note that DuVoice DV2000 will be referred as DV2000 for rest of the document.

2. General Test Approach and Test Results

Feature functionality testing was performed manually. Inbound calls were made to the Avaya IP Telephones (i.e. the guest telephones) over PRI and SIP trunks, as well as from other local extensions (analog, digital, and IP Telephone). A Hospitality Tester was used to launch changes to telephone message waiting lamps and phone privileges during room check in / checkout / move requests, receive room status updates, and activate/deactivate DND.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute a full product performance or feature testing performed by third party vendors, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a third party solution.

2.1. Interoperability Compliance Testing

To verify SIP trunk interoperability, and PMS interface the following areas were tested for compliance:

Automated Attendant

- Incoming trunk calls to DV2000 Voice Messaging System answered by Auto Attendant
- Incoming trunk calls to DV2000 Voice Messaging System answered by Auto Attendant, originated from a PSTN extensions
- Transfers to Staff Extensions
- Transfers to Guest Extensions
- Remote Disconnects
- Invalid Options

Voice Mail

- Incoming trunk calls to DuVoice Voice Messaging System for voicemails. Verifying message waiting indicator (light on/off) on different types of end-point (Analog, UNSTim, Digital and SIP phones).
- Guest to Guest Voice Messaging
- Staff Voice Messaging
- Voicemail retrieval
- Voicemail retrieval from a simulated PSTN extension
- Call Blocking

Wake-up call

- Schedule wake-up calls from guest extensions
- Schedule wake-up calls from InnDesk
- Wake-up calls retries
- Wake-up call failed coverage (routes to front desk after expiration of 4 retries)

PMS

- Check in/out with guest name.
- Verify MWI light
- Verify Controlled Class of Service On/Off
- Room change
- Guest info update
- DND On/Off
- Update room status.

2.2. Test Results

All executed test cases were completed successfully. Here is a list of observation:

1. Make an incoming trunk call, unplug the cable for 30 or 60 second, the call will not be disconnected.
2. Perform feature move the guest whom has the new message in their mailbox. The new message is successfully moved to the new mailbox but there is no MWI light lit on the phone of the new room.
3. If a guest requests their phone to have Do Not Disturb ON, they will not able to receive the wakeup call, as DV2000 will received a busy signal when trying to make a call to guest.

2.3. Support

Avaya: For technical support on the Avaya products described in these Application Notes visit http://support.avaya.com.

DuVoice: For technical support on DuVoice products visit the online support site at http://www.duvoice.com/

3. Reference Configuration

Figure 1 below illustrates the test configuration diagram that has an Rlogin via ELAN IP for PMS connected from DV2000 server to ELAN of CS1000 Call server. And the test configuration simulates an enterprise site with Avaya SIP-enabled enterprise solution connected to the DV2000 server via the Local Area Network (LAN).

The transport protocol between the Avaya Aura® Session Manager and the DuVoice Server is SIP over UDP. The transport protocol between Avaya Aura® Session Manager and Avaya Aura® Communication Server 1000 across the enterprise IP network is SIP over UDP.

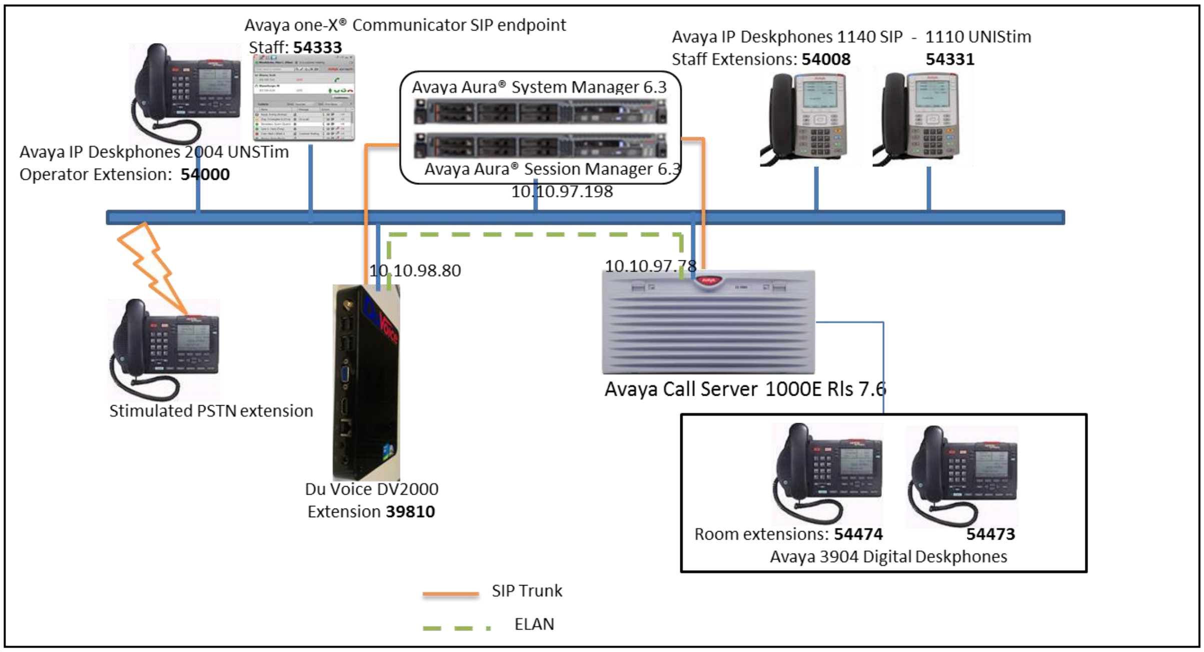


Figure 1: Test Configuration Diagram

4. Equipment and Software Validated

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

Equipment	Software
Avaya S8800 Server running Avaya Aura® Session Manager Server	6.3 (Build No 6.3.2.0.632023)
Avaya S8800 Server running Avaya Aura® System Manager Server	6.3 (Build No: 6.3.0.8.5682-6.3.8.1627)
Avaya Communication Server 1000E/CPPM	Avaya Communication Server Release 7.6 Q+ Deplist 1 (created: 2012-09-20) and Service Update 1 (Created: Sept 19, 2012)
Avaya IP SIP Phone 1140	4.3
Avaya Digital phone 3904	N/A
Avaya Analog M8003	N/A
Avaya IP Unistim Phone 2004, 1110	0604DCN
Avaya one-X® Communicator for CS1000	6.1
DuVoice NaNo Server DV2000	5.20.026

5. Configure Avaya Communication Server 1000

This document assumes that the CS1000 system used for the compliance test was already installed and configured. This section just provides necessary procedure to configure for CS1000 to work with DV2000. For more detail on how to administer the CS1000 system, please refer to **Section 10**.

Please note that Avaya Communication Server 1000 will be refered as CS1000 for rest of this document.

5.1. Configure Property Management System Interface (PMSI)

The Property Management System Interface is an optional software package that allows the CS1000 system to interface directly with a customer-provided Property Management System (PMS) through Rlogin via Embedded LAN (ELAN). This provides an effective means of information between the PMS and the CS1000 system.

This section provides the procedure how to check the software package and to configure the Property Management System Interface on the CS1000. Log in the CS1000 Call Server and execute the following overlay (LD) commands.

1. Use overlay LD 22 to check all necessary software packages that are required for the PMS feature on the CS1000.

Prompt	Response	Comment
REQ	**PRT**	Request: Print
TYPE	**PKG**	Type: package
DNDI	**9**	Do Not Disturb Individual package
DNDG	**16**	Do Not Disturb Group package
MWC	**46**	Message Waiting Center package
CCOS	**81**	Controlled Class of Service package
BGD	**99**	Background Terminal package
RMS	**100**	Room Status package
MR	**101**	Mange Registration package
AWU	**102**	Automatic Wake UP package
PMSI	**103**	Property Management Service Interface

2. Use overlay LD 17 to create a TTY port number for a PTY connection on the CS1000. This PTY port was used for DuVoice DV2000 to connect to the Call Server via ELAN.

Prompt	Response	Comment
REQ	CHG	Request: Change
TYPE	ADAN	Action Device and Number
ADAN	NEW TTY 7	Add a new TTY port
CTYP	PTY	Card type: Pseudo TTY
DNUM	7	Device number for I/O port
PORT	**7**	Port number
FLOW	NO	Flow control capability
USER	BGD PMS	Output message type

3. Use overlay (LD) 17 to enable the PMS interface in the CS1000 system.

Prompt	Response	Comment
REQ	CHG	Request
TYPE	PARM	System Parameters
PMSI	YES	Modify properties management system interface
MANU	PMS1	PMS interface
PMCR	20	Number of call registers used for PMSI
PORT	**7**	Port number
XTMR	2	PMS acknowledgment time
XNUM	1	Number of retransmissions per message
PMIN	YES	Minor alarm when link is not responding
PTMR	0	Polling time for PMSI

4. Use overlay (LD) 15 to enable the Controlled Class of Service (CCOS) feature in the customer data block.

Prompt	Response	Comment
REQ	CHG	Request change
TYPE	CCS	Controlled class of service
CUST	0	Customer
CCRS	**UNR**	Restricted Service
ECC1	**FRE**	Enhance Level 2
ECC2	**UNR**	Enhance Level 2

5. Use overlay (LD) 15 to enable Automatic Wake Up feature in the customer data block. Note that RAN routes 16, 17, and 18 below were used just for example and they need to be defined in LD 16 before it can be used in the Automatic Wake Up feature.

Prompt	Response	Comment
REQ	CHG	Request change
TYPE	AWU	Type of data block: Automatic wake up
CUST	0	Customer 0
AWU	**YES**	Automatic wake up
RANF	16	Music route
RAN1	17	Primary RAN route
RAN2	18	Secondary RAN route

6. User overlay (LD) 15 to enable Do Not Disturb feature in the customer data block.

Prompt	Response	Comment
REQ	CHG	Request change
TYPE	FTR	Features and options
CUST	0	Customer 0
DNDL	**YES**	Do not disturb lamb

7. Use overlay (LD) 15 to enable Message Waiting Indicator feature in the customer data block (CDB).

Prompt	Response	Comment
REQ	CHG	Request change
TYPE	FTR	Features and options
CUST	0	Customer 0
OPT	**MCI**	Options: Message center included

8. Use overlay (LD) 10 and 11 to administer analog, digital and IP phone.

Prompt	Response	Comment
REQ	CHG	Request change
TYPE	1165	Type of set
CUST	0	Customer ID
ECHG	YES	Easy change
ITEM	**CLS CCSA MWA**	Class of service
ITEM	**KEY 1 RMK**	Room status key
ITEM	**KEY 2 WUK**	Wakeup key

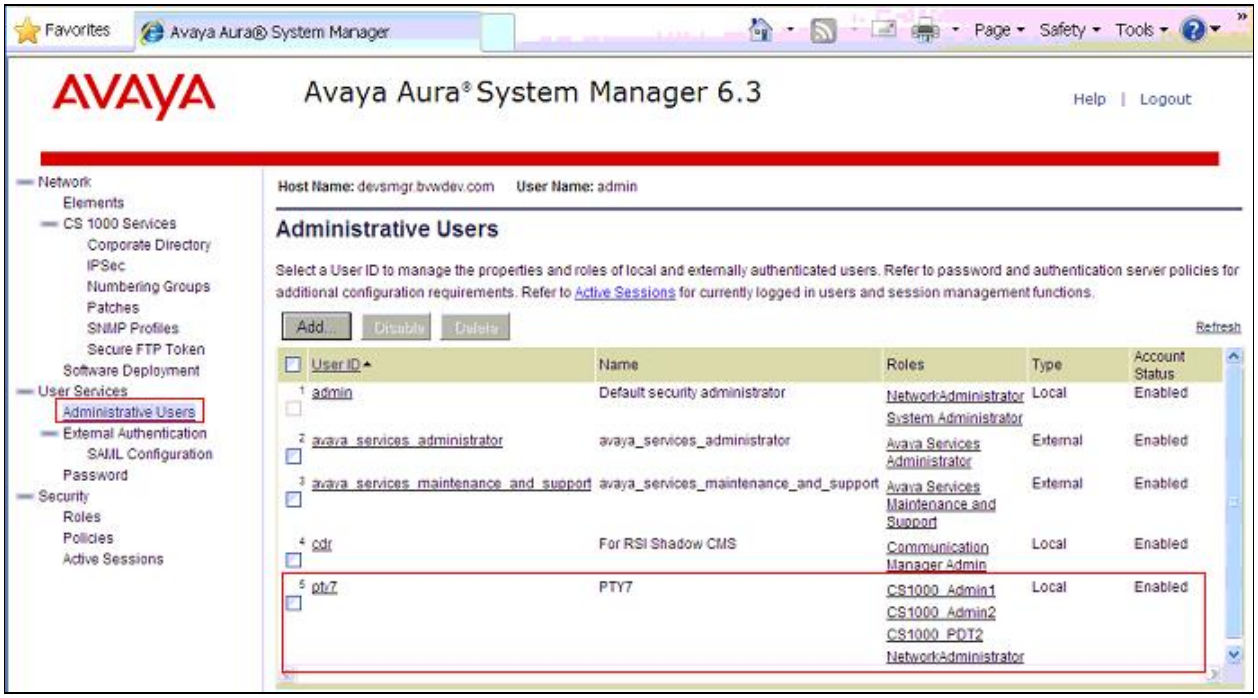
With definitions for class of services:
- **CCSA**: Controlled Class of service Allowed.
- **MWA**: Messaging Waiting Indicator Allowed.

5.2. Configure Username in Unified Communications Management (UCM)

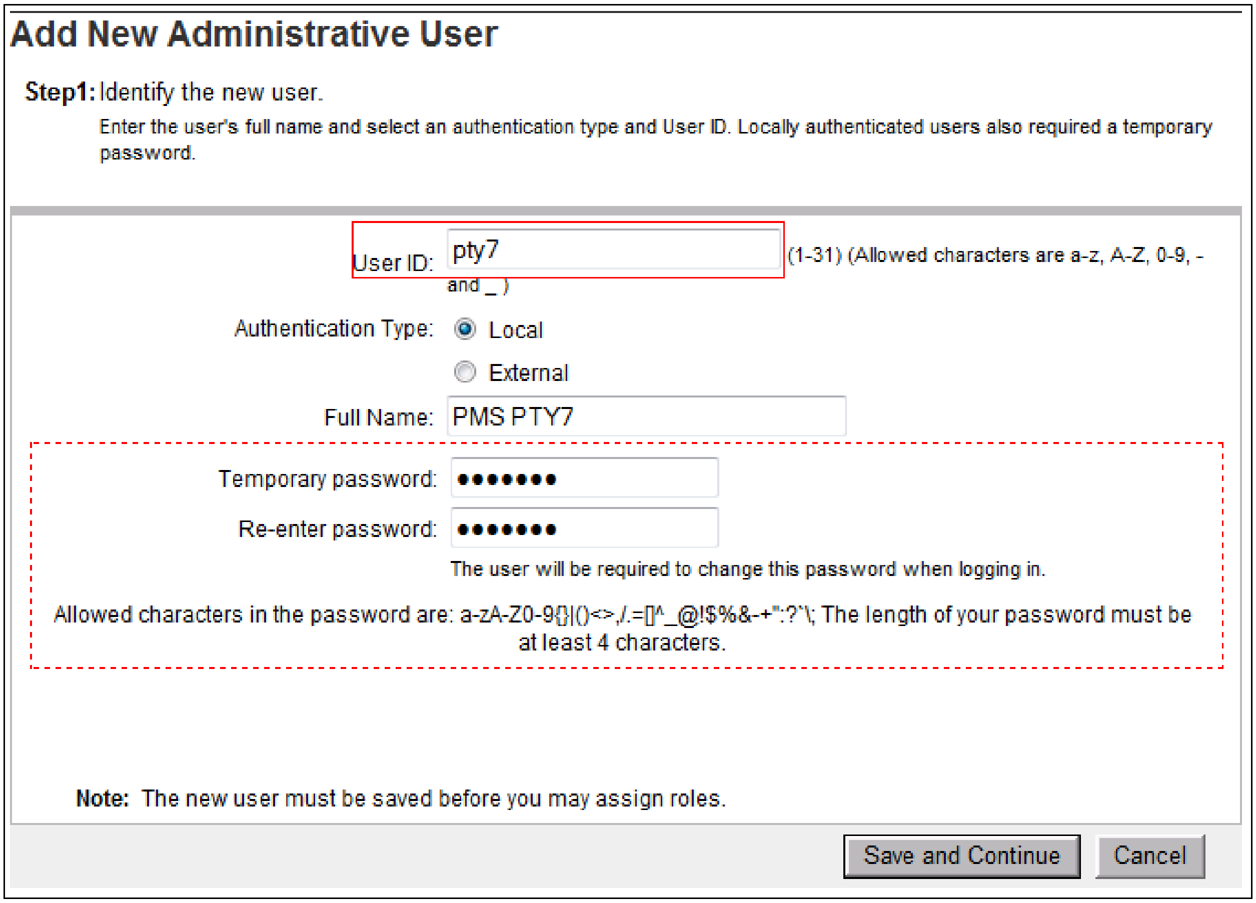
In order to integrate DV2000 logs in to the Call server via Rlogin with the dedicated PTY port 7 above they must use a dedicated username created in the Unified Communications Management (UCM). This special username has to be named like **pty7** which is matched with port 7 in the PTY port above.

Log in to the UCM by using administrator privilege; enter the user name **admin** in the **User ID** field and the password in to the **Password** field. Click **Log In** button.

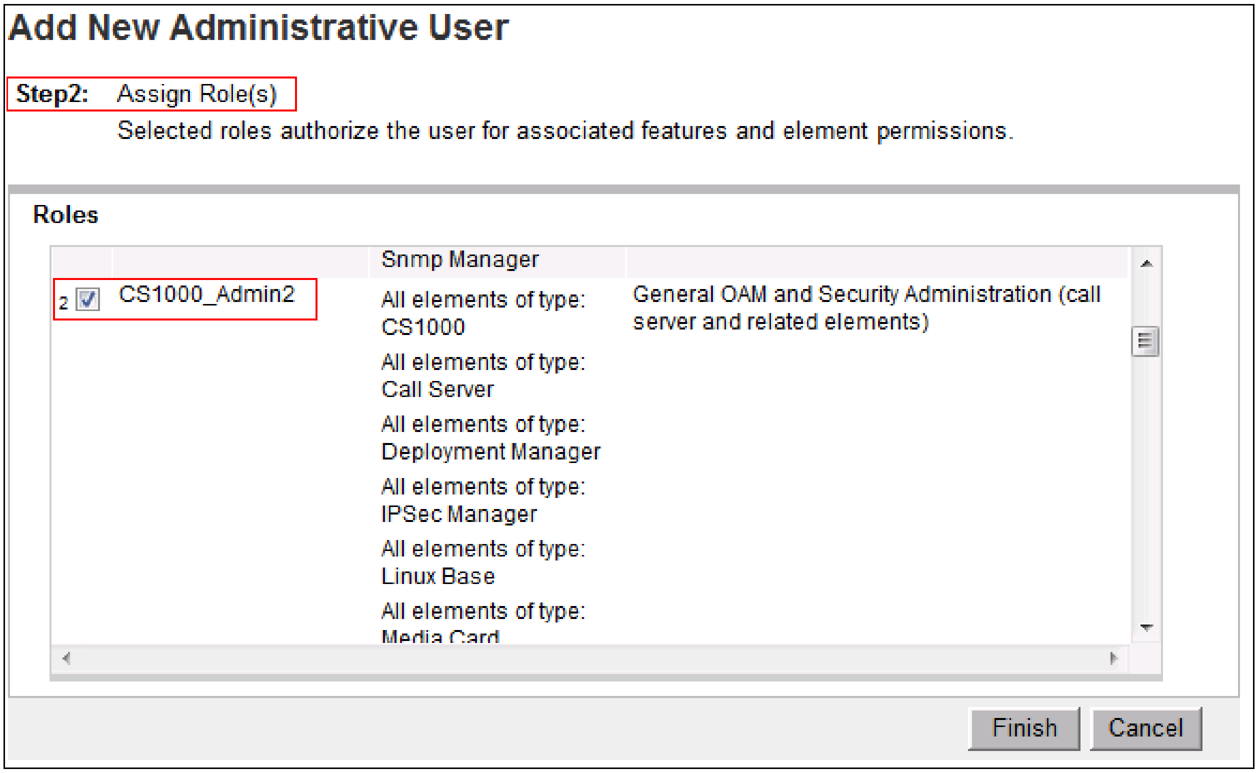
The Avaya Unified Communications Management homepage is displayed as per the screen shot below. Click on the **Administrative Users** in the left navigation pane. Below screenshot shown user **pty7** had been created. Continue to next section for detail step on how to create new user.



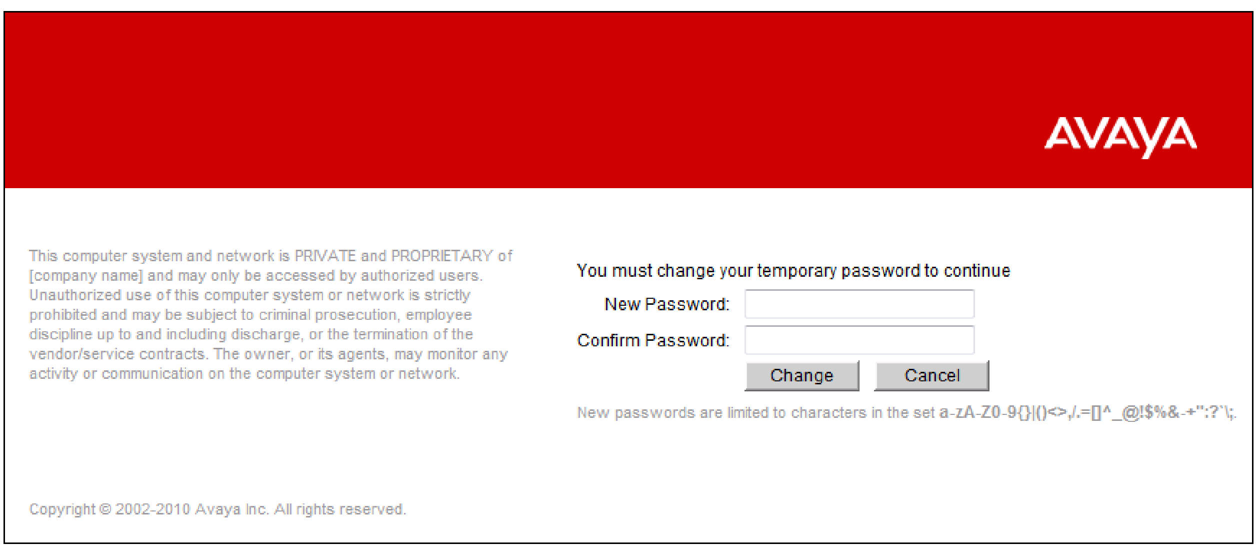
The **Administrative Users** page is displayed in the right. Click on **Add** button to add a new user name (not shown). The **Add New Administrative User** page is displayed. Enter **pty7** in the **User ID** field and select **Local** radio option. Enter a descriptive name in the **Full Name** field and a password in the **Temporary password** and **Re-enter password** fields. Click on **Save and Continue** button to go to next page.



In the **Step2: Assign Role(s)** page, assign **CS1000_Admin2** and **Network Administrator** roles to this user as shown below. Click on **Finish** button to save and complete.



The temporary password of the new **pty7** user must be changed before it can be used to Rlogin to CS1000 Call server. To change the temporary password, launch the UCM webpage and use the **pty7** username and its temporary password to log in. Enter a new password in both **New Password** and **Confirm Password** fields and click on the **Change** button to change it to new one.



5.3. Configure Avaya Communication Server 1000 for DuVoice Messaging system

This section describes the procedure for setting up CS1000E. The steps include setting up

- Node properties.
- Route, Route List Block (RLB) and Distant Steering Code (DSC).
- Endpoints/Telephones.

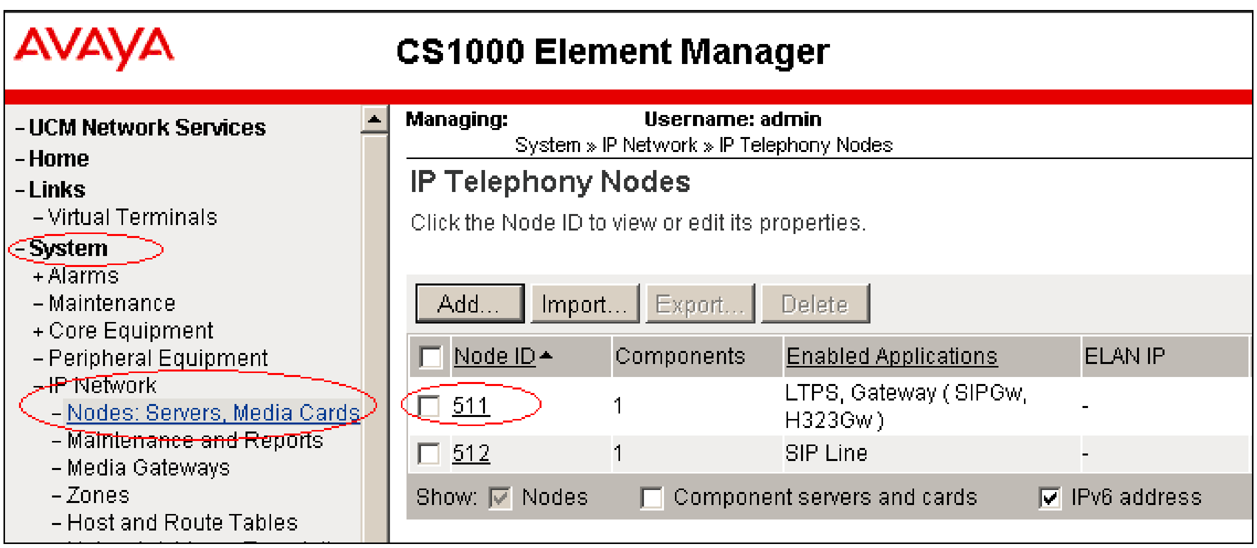
The values used in this guide may be unique to the example shown. User will have to use values unique to their site, where this solution is being deployed e.g. site's IP address, extension numbers, etc. CS1000E configurations are performed through Unified Communications Manager (UCM), Element Manager (EM) and Command Line Interface (CLI) via a telnet session to the Call Server.

It may not be necessary to create all the items above when creating a connection to Session Manager since some of these items would have already been defined as part of the initial Avaya Aura® Session Manager and Avaya Communication Server 1000 installation. This includes items such as certain SIP domains, Node, Route, Route List Block and Distant Steering Code. However, each item should be reviewed to verify the configuration.

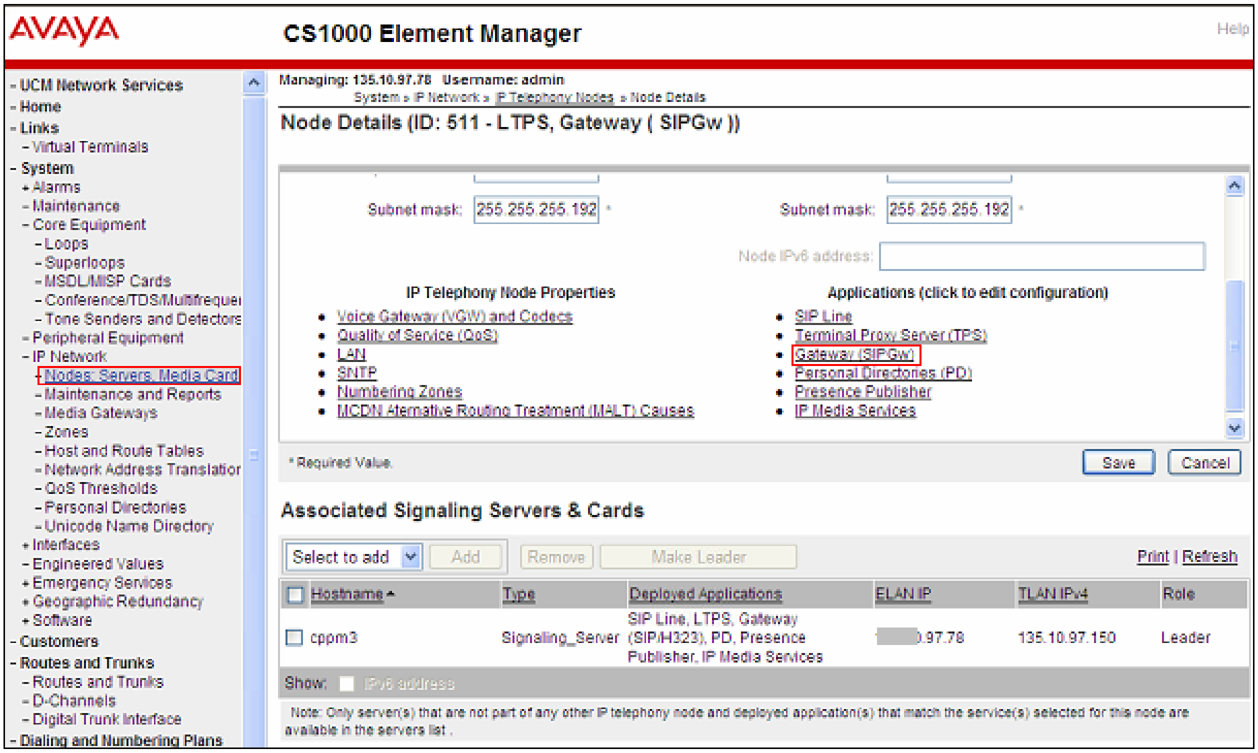
5.3.1. Node IP (SIP Gateway) Configuration

This section only describes the configuration of the SIP Gateway application running on the CS1000E signaling server. In the solution test, Node ID **511** is configured, that has the SIP Gateway application enabled on it. For additional information on Nodes configuration refer to **Section 10.**

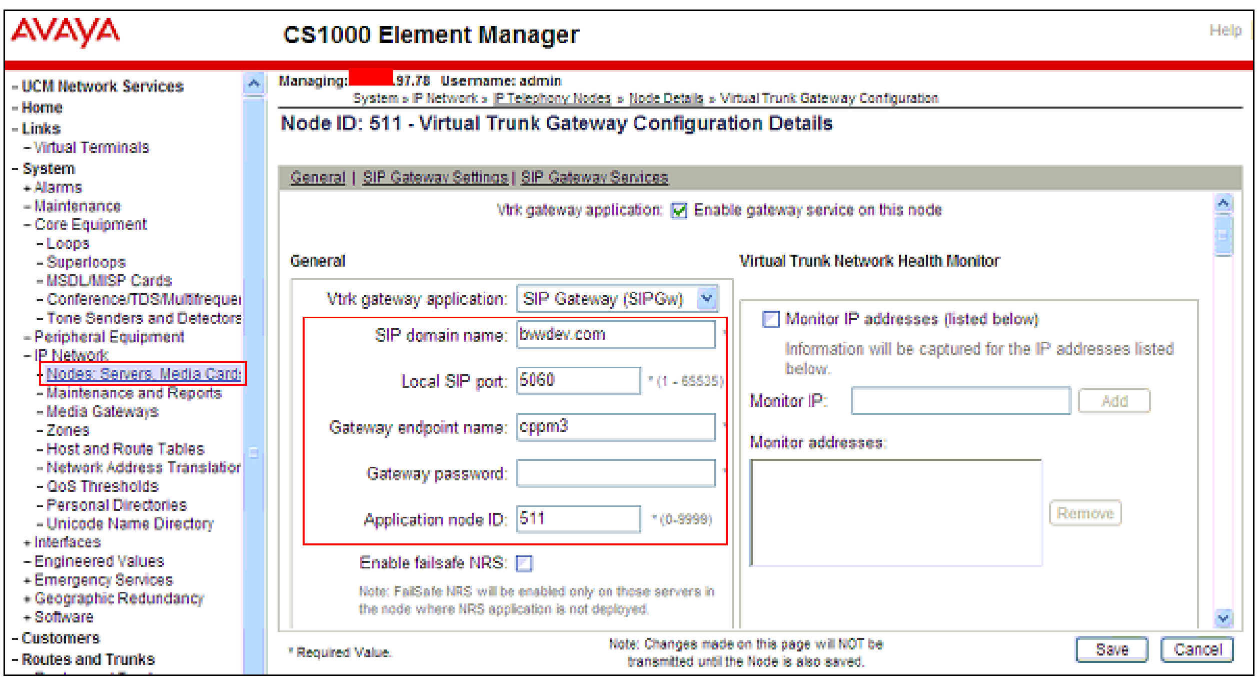
To configure the SIP Gateway from EM, navigate to **System →IP Network →Nodes: Servers, Media Cards** and click on the **Node ID 511** as shown below.



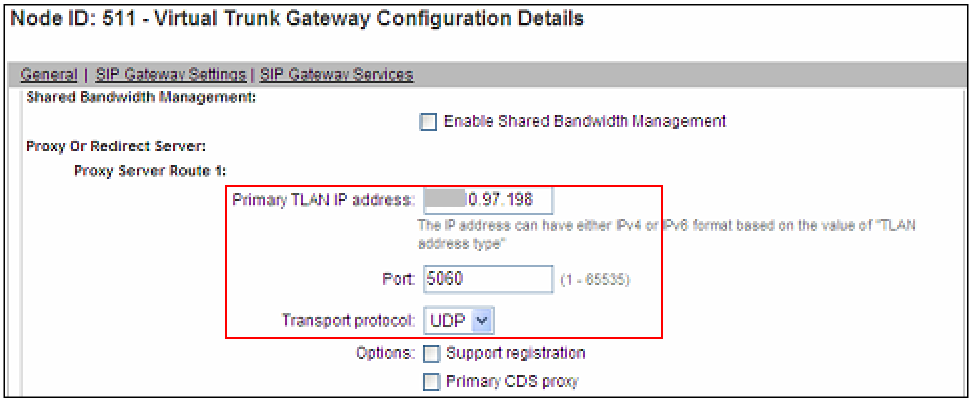
Click on the link **Gateway (SIPGw)** link as shown below.



In the General section enter the **SIP domain name** as **bvwdev.com**, **Local SIP port** as **5060**, **Gateway endpoint name** as **cppm3** and **Application node ID** as **511**.



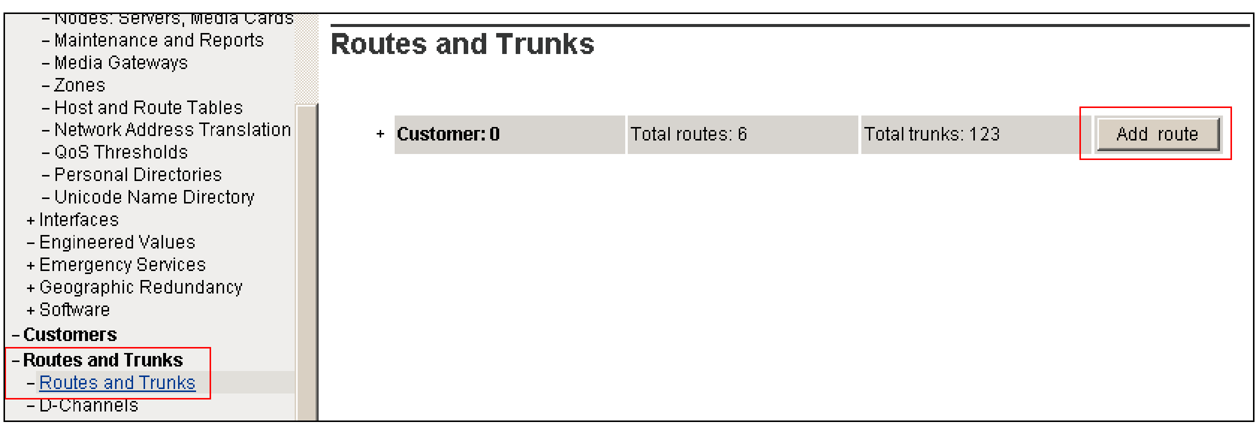
In Proxy Server Route 1, verify **Primary TLAN IP address**, which is the **IP address of the Session Manager**. Rest of the fields is left at default.



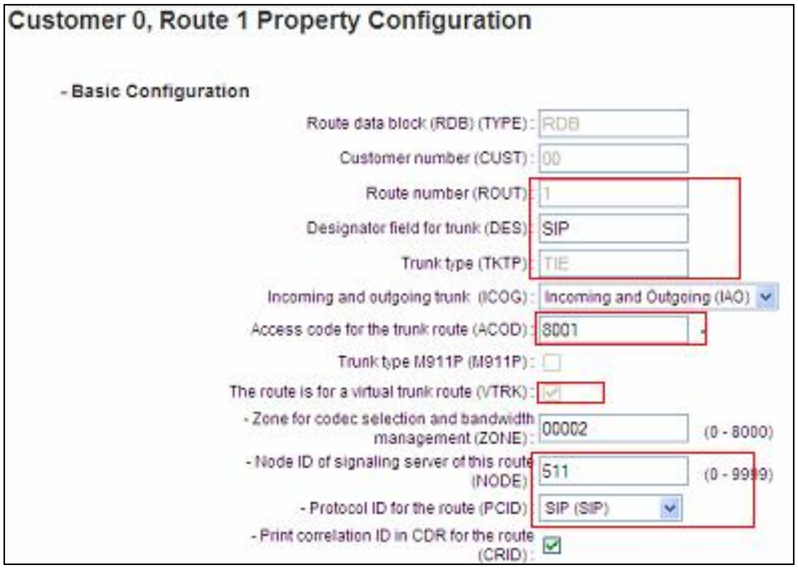
In the **SIP URI Map** verify the following information: **UDP** field is configured as **udp.** The rest of the fields are left as default.

5.3.2. Route, RLB and DSC Configuration

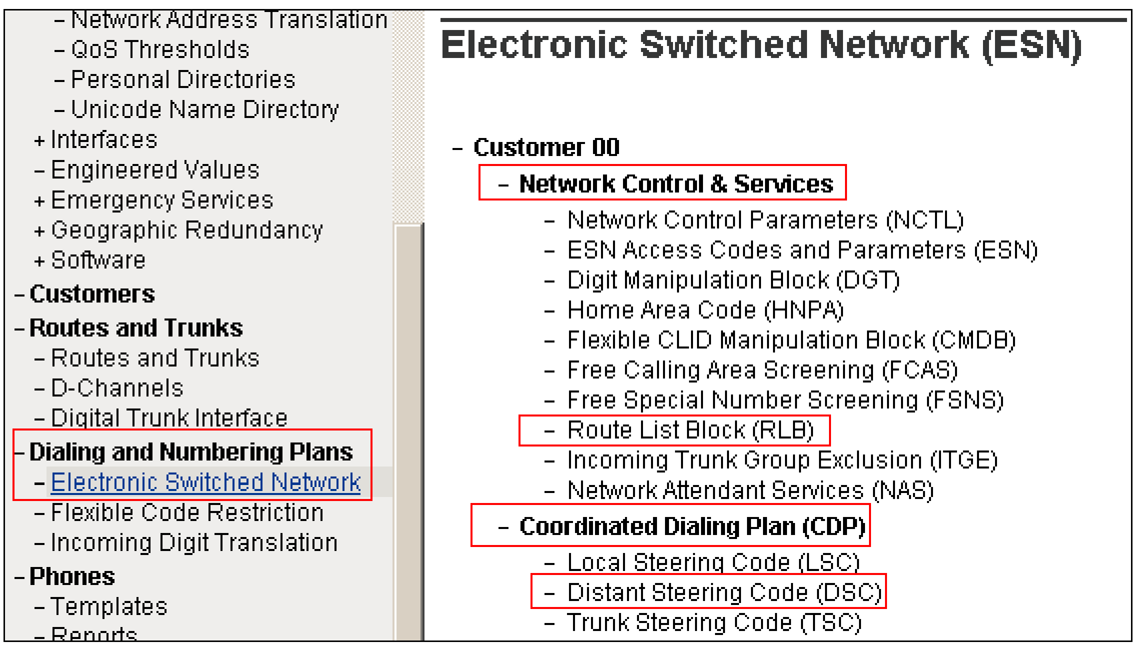
This section explains the steps to configure a routing entry that will access the Office-LinX server from the CS1000E using the RLB and DSC values. After logging into the UCM, click on the EM link of the respective CS1000E (Not Shown). In the EM navigate to **Routes and Trunks → Routes and Trunks.** Click on **Add route.**



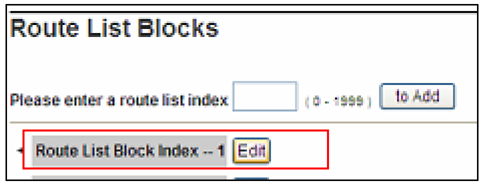
Below is the configuration of the **Route 1** used during the compliance test. The values that are circled in red are to be configured by the user. The values shown are examples used during the solution testing.



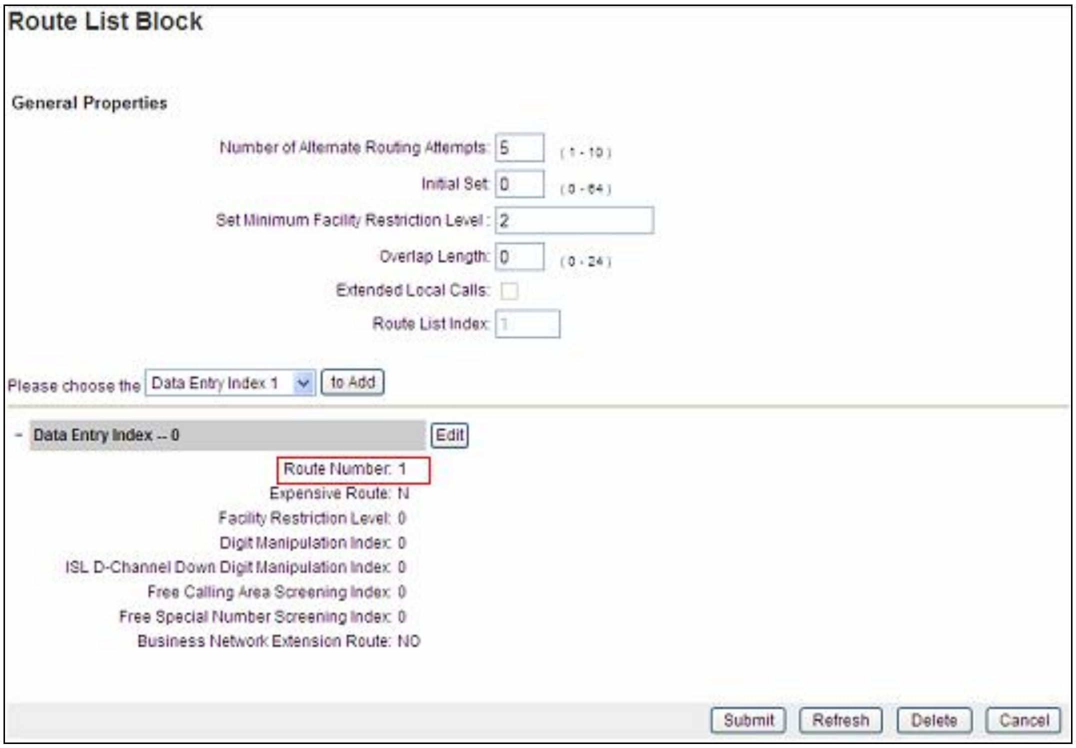
To configure the RLB using EM navigate to **Dialing and Numbering Plans →Electronic Switched Network →Network Control & Services →Route List Block (RLB)**.



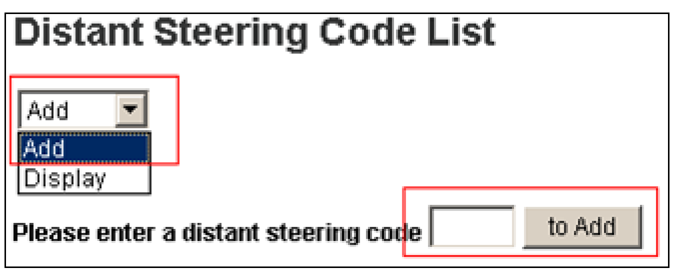
Enter the value of the route list index and click on **to Add** button to continue the configuration as shown below. During the solution testing the value of **1** was added.



The **Route Number 1** being selected to the RLB created. Route **1** is selected since it was the route number assigned while adding a route. Below is detail of RLB 1

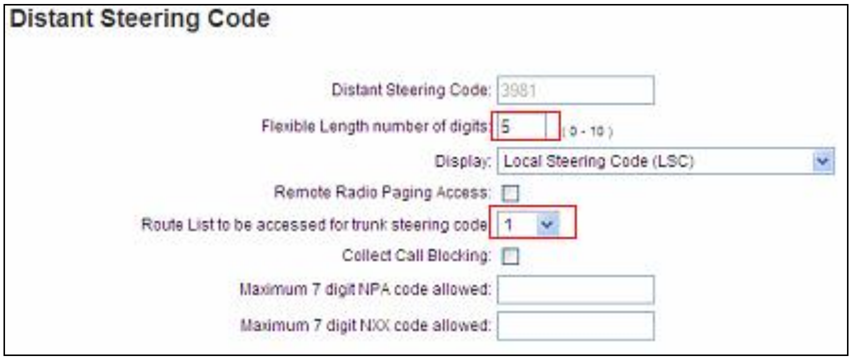


To configure the DSC using EM navigate to **Dialing and Numbering Plans → Electronic Switched Network → Coordinated Dialing Plan (CDP) → Distant Steering Code (DSC)**. In the Distant Steering Code List page, select **Add** from the drop down list as shown below.



Enter the value of the DSC and click on the **to Add** button (Not Shown). As shown below 53 was added during the solution testing. The value **3981** was configured since the pilot DN of the DV2000 was **39810**.

Flexible Length number of digits indentifies length of the directory number (DN). During solution testing value of **5** was configured.
Route List to be accessed for trunk steering code is selected as **1** from the drop down list. This value is selected based on the RLB created in above step.



For additional information on Route, RLB and DSC configuration, refer to **Section 10** of these Application Notes.

5.3.3. Endpoint/Telephone Configuration

This section explains the provisioning of an endpoint/telephone for Guest or Staff that was configured for the solution testing. Endpoint/Telephone can be configured using the CLI of the CS1000E from overlay LD 11/20. Refer to **Section 10** for further information regarding add/configuration of endpoints/telephones.

Below are values that are shown in red are to be configured by the user. The **FDN** and **HUNT** value of **39810** was used during the solution testing as the pilot DN of the DV2000.

```
Ld 11
REQ: prt
TYPE: 1165
TN   096 0 00 17
FDN  39810
…
CLS  UNR FBA WTA LPR MTD FNA HTA TDD HFA CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSA SWD LNA CNDA
     CFTD SFA MRD DDV CNID CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDA CFXA ARHD CLTD ASCD
     CPFA CPTA ABDD CFHD FICD NAID BUZZ AGRD MOAD
     UDI RCC HBTD AHD IPND  DDGA NAMA MIND PRSD NRWD NRCD NROD
RCO  0
HUNT 39810
…
KEY  00 SCR 54312 0     MARP
        CPND
          CPND_LANG ROMAN
            NAME DN 54312
            XPLN 13
         DISPLAY_FMT FIRST,LAST
```

6. Configure Session Manager

This section provides the procedures for configuring Session Manager as provisioned in the reference configuration. Session Manager is comprised of two functional components: the Session Manager server and the System Manager server. All SIP call provisioning for Session Manager is performed through the System Manager Web interface and is then downloaded into Session Manager.

The following sections assume that Session Manager and System Manager have been installed and that network connectivity exists between the two platforms.

In this section, the following topics are discussed:

- SIP Domains
- Locations: Logical/physical location that can be occupied by SIP Entities.
- SIP Entities corresponding to Communication Server 1000 and Session Manager.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policy, which control call routing between the SIP Entities.
- Dial Patterns, which govern to which SIP Entity a call is routed.

It may not be necessary to create all the items above since some of these items would have already been defined as part of the initial Session Manager installation. This includes items such as certain SIP domains, locations, SIP entities and Session Manager itself. However, each item should be reviewed to verify the configuration.

6.1. Configure SIP Domain

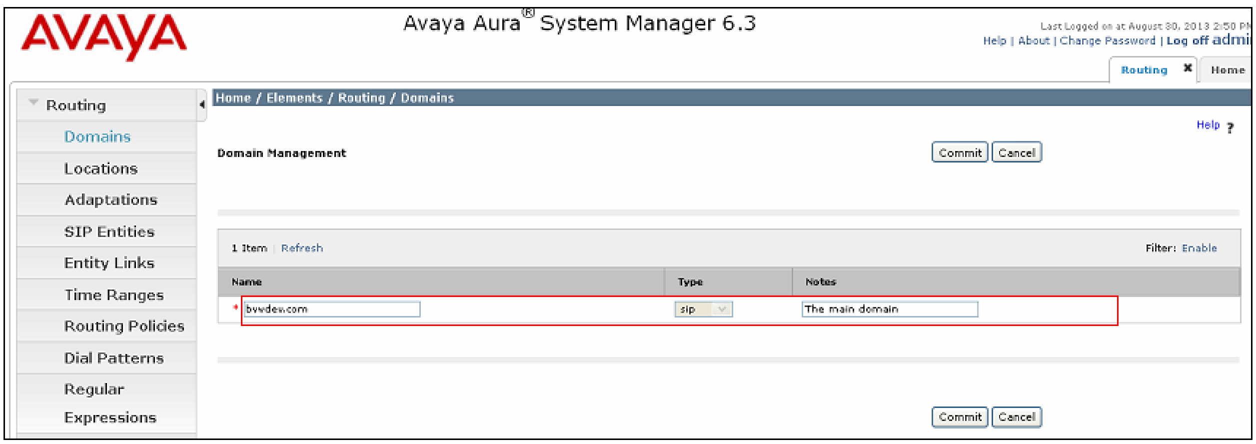
Launch a web browser, enter "**https://<IP address of System Manager>/SMGR**" in the URL, and log in with the appropriate credentials.

Create a SIP domain for each domain for which Avaya Aura® Session Manager will need to be aware in order to route calls. For the compliance test, this includes the enterprise domain.

Add a domain, navigate to **Routing → Domains**, and click on the **New** button (not shown) to create a new SIP Domain. Enter the following values and use default values for remaining fields:

- **Name**: Enter the Authoritative Domain Name, which is **bvwdev.com**.
- **Type** : Select **SIP**

Click **Commit** to save. The following screen shows the Domains page used during the compliance test.



6.2. Configure Locations

Locations are used to identify logical and/or physical locations where SIP Entities reside, for purposes of bandwidth management or location-based routing. Navigate to **Routing → Locations**, and click on the **New** button (not shown) to create a new SIP endpoint location.

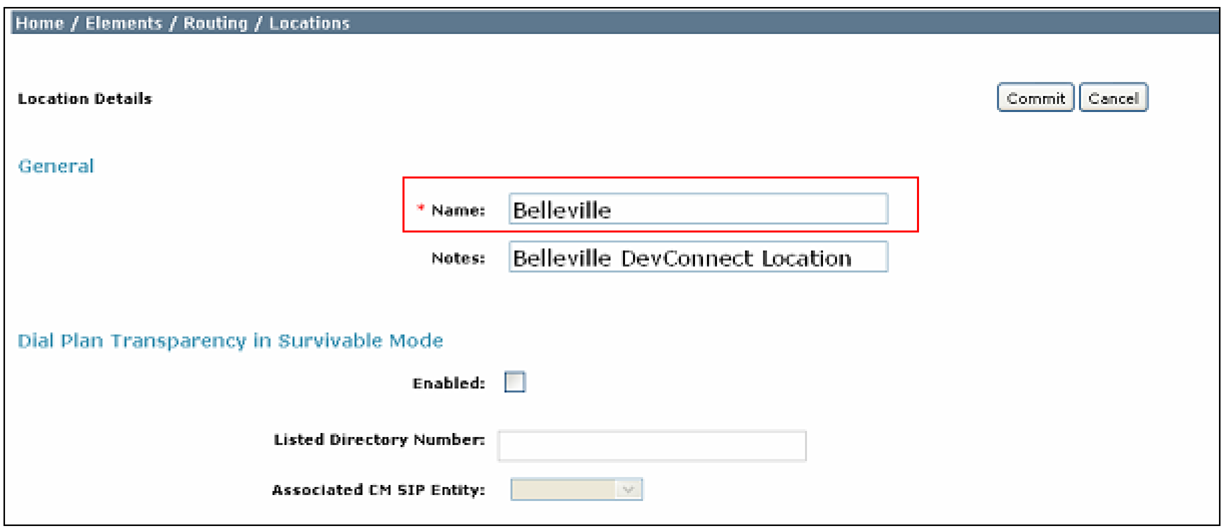
In General section, enter the following values and use default values for remaining fields.

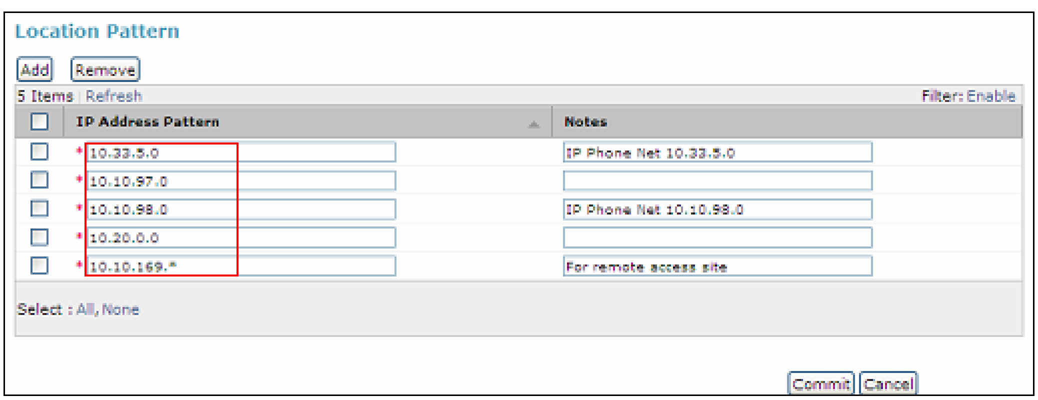
- Enter a descriptive Location name in the **Name** field.
- Enter a description in the **Notes** field if desired.

In Location Pattern section, click **Add** and enter the following values:

- **IP address Pattern**: Enter the IP Pattern to identify the location.
- **Notes**: Enter a description in the **Notes** field if desired.

The following screen shows the Locations page used during the compliance test. Click on the **Commit** button.





6.3. Configure Adaptation module

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. The adaptations named **CS1000** and **DuVoice Outgoing** Adaptations were configured and used in the compliance test.

6.3.1. Settings for DuVoice Outgoing Adaptation:

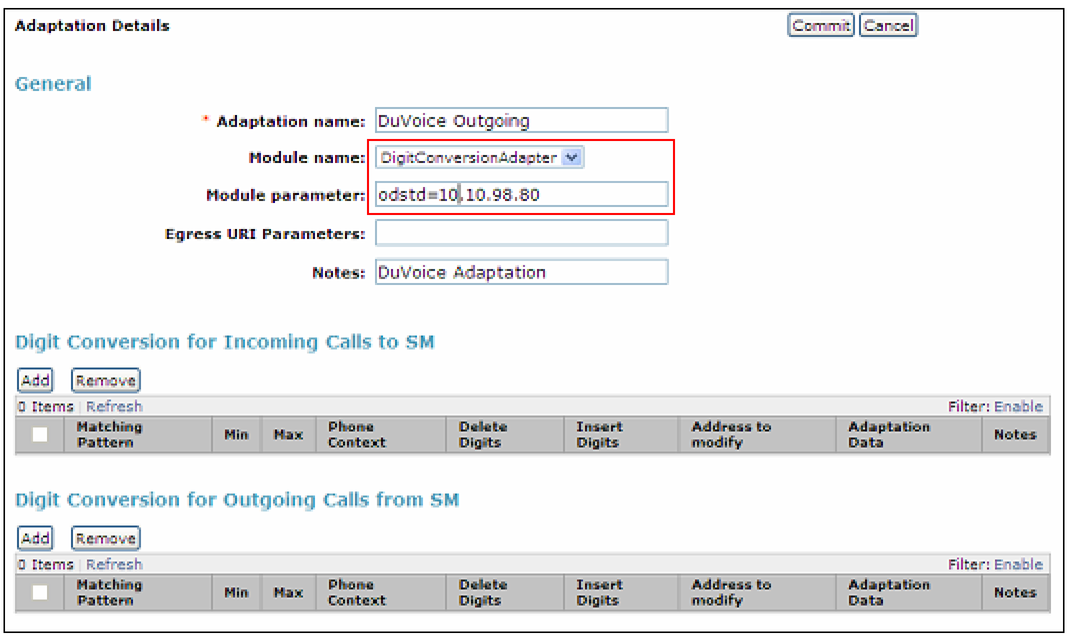
In the **General** section, enter the following values. Use default values for all remaining fields:

- **Adaptation Name:** Enter a descriptive name for the adaptation.
- **Module Name:** Enter **DigitConversionAdapter**.
- **Module parameter:** Enter **odstd=x** where **x** is the IP address of the DuVoice server.

The **odstd=10.10.98.80** module parameter enables the outbound destination domain to be overwritten with the IP address of the DuVoice server. For example, for outbound calls from Avaya to DuVoice, the Request-URI will contain IP address **10.10.98.80** as expected by DuVoice.

Click **Commit** to save.

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



6.3.2. Settings for CS1000 Adaptation:

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

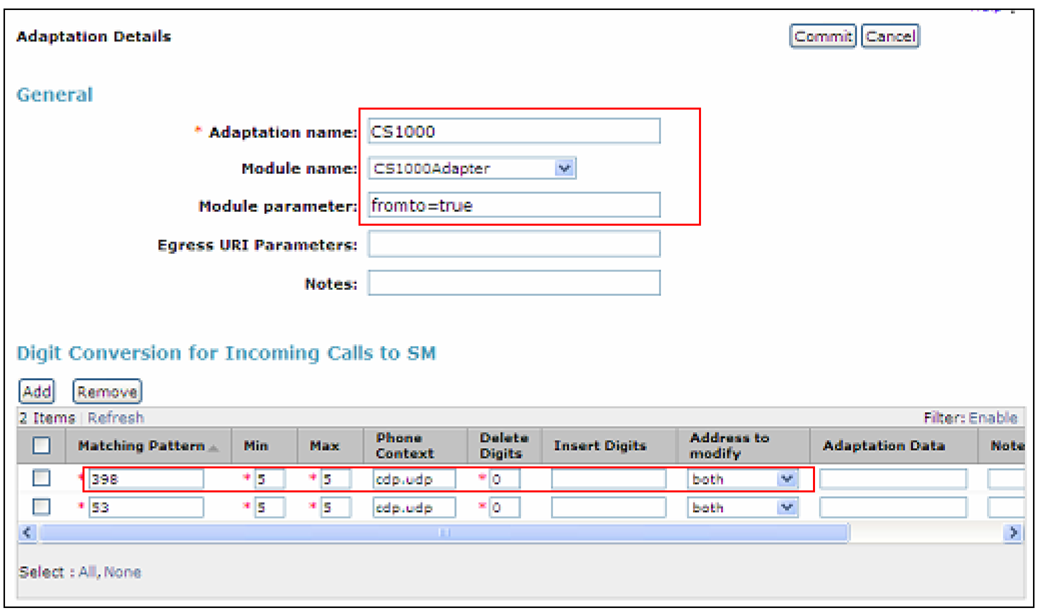
- **Adaptation Name:** Enter a descriptive name for the adaptation, example **CS1000**.
- **Module Name:** Select CS1000Adaptor.
- **Module parameter:** Enter **fromto=true**, adaptation will modify From and To headers of the message.

In Digit Conversion for Incoming calls to SM, add item for DV2000 pilot number, as following:

- **Matching Pattern:** Enter a matching pattern, **398**.
- **Min:** Enter **5.**
- **Max:** Enter **5.**
- **Phone Context**: **cdp.udp**
- **Delete Digits**: Enter **0**
- **Address to modify**: Select **both.**

Click **Commit** to save.

The **CS1000** adaptation shown below will later be assigned to the CS1000 SIP Entity. This adaptation uses the **CS1000Adapter**.



6.4. Configure SIP Entities

A SIP Entity must be added for Session Manager and for each network component that has a SIP trunk provisioned to Session Manager. During the compliance test, the following SIP Entities were configured:

- Session Manager itself.
- Communication Server 1000
- DuVoice DV2000

Navigate to **Routing → SIP Entities**, and click on the **New** button (not shown) to create a new SIP entity. Provide the following information:

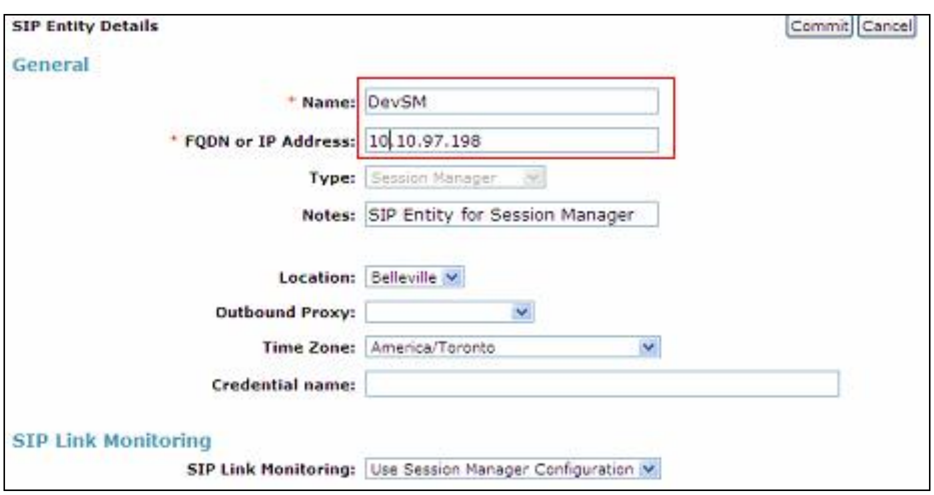
Enter the following values and use default values for remaining fields.

- Enter a descriptive name in the **Name** field.
- Enter IP address of SIP Entity that is used for SIP signaling in the **FQDN or IP Address** field. Enter IP address of Communication Server, Session Manager, or DV2000.
- From the **Type** drop down menu select a type that best matches the SIP Entity. For Communication Server, select **Other**. For Session Manager, select **Session Manager**. For DuVoice DV2000, select **Other**.
- Select Adaptation for Communication Server and DV2000 entities.
- Enter a description in the **Notes** field if desired.
- Select the appropriate time zone.
- Accept the other default values.

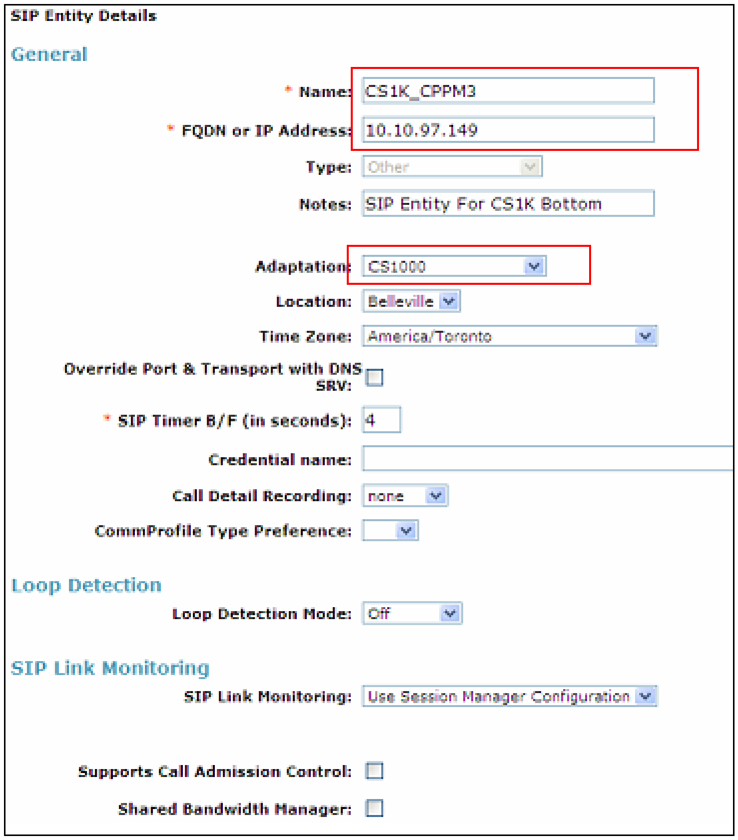
Click on the **Commit** button to save configuration for each SIP Entity.

The following screens show the SIP Entities page used during the compliance test.

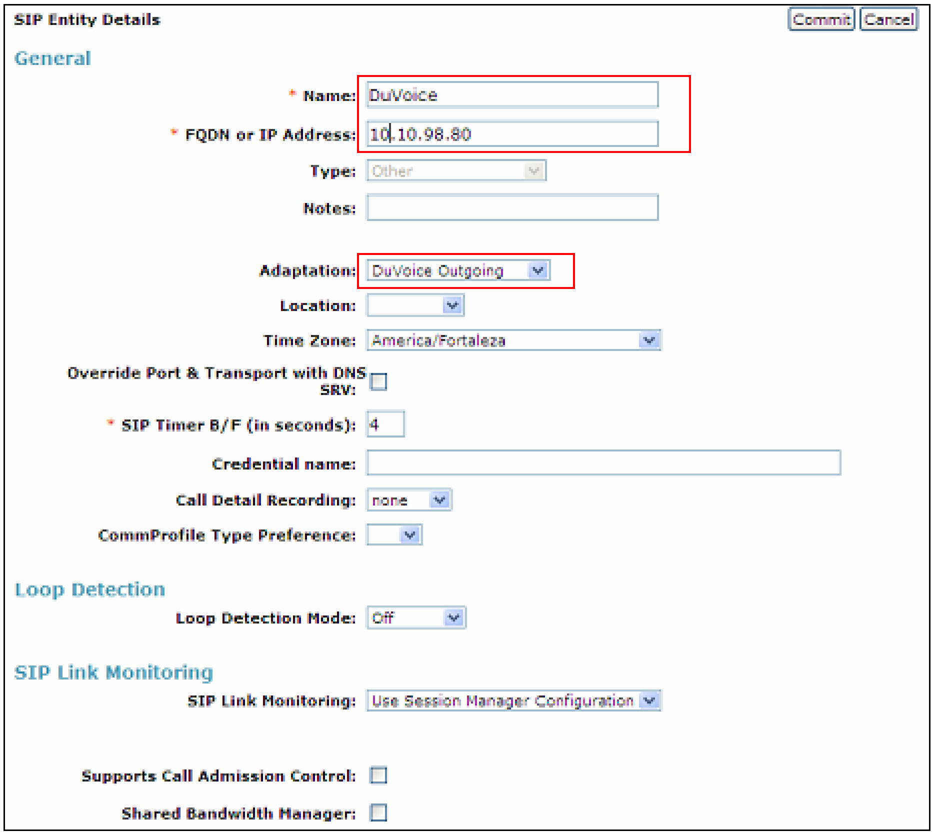
Session Manager SIP Entity:



Communication Server SIP Entity with Adaptation CS1000:



DuVoice DV2000 SIP Entity with Adaptation DuVoice Outgoing:

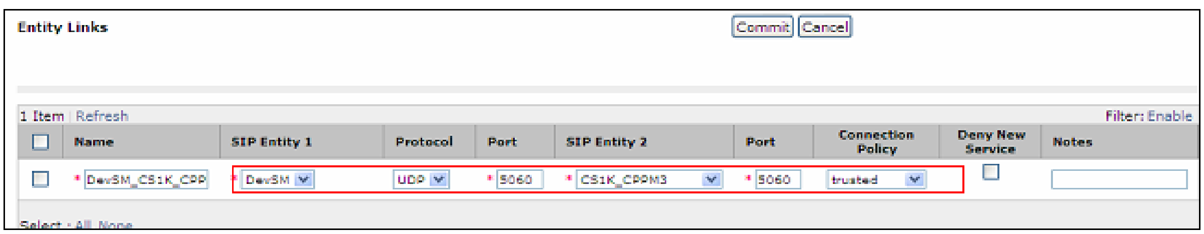


6.5. Configure Entity Links

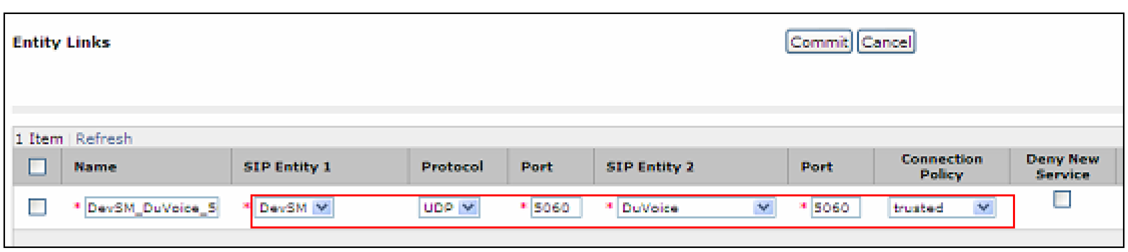
Entity Links define the connections between the SIP Entities and Session Manager. In the compliance test, the 2 entities links are defined: one to Communication Manager (Avaya G450 with S8300D Server) and one to Messaging. Add an entity link, navigate to **Routing → Entity Links**, and click on the **New** button (not shown) to create a new entity link. Provide the following information:

- Enter a descriptive name in the **Name** field.
- In the **SIP Entity 1** drop down menu, select the Session Manager SIP Entity.
- In the **Protocol** drop down menu, select the protocol to be used.
- In the **Port** field, enter the port to be used, UDP or TCP – 5060
- In the **SIP Entity 2** drop down menu, select an entity for desired entity.
- In the **Port** field, enter the port to be used (e.g. **5060**).
- Check the **Trusted** box.
- Enter a description in the **Notes** field if desired.

Click on the **Commit** button to save each Entity Link definition. The following screen shows an Entity Links page used during the compliance test between Session manager and Communication Server 1000.



Repeat the steps to define Entity Links between Session Manager and DV2000.



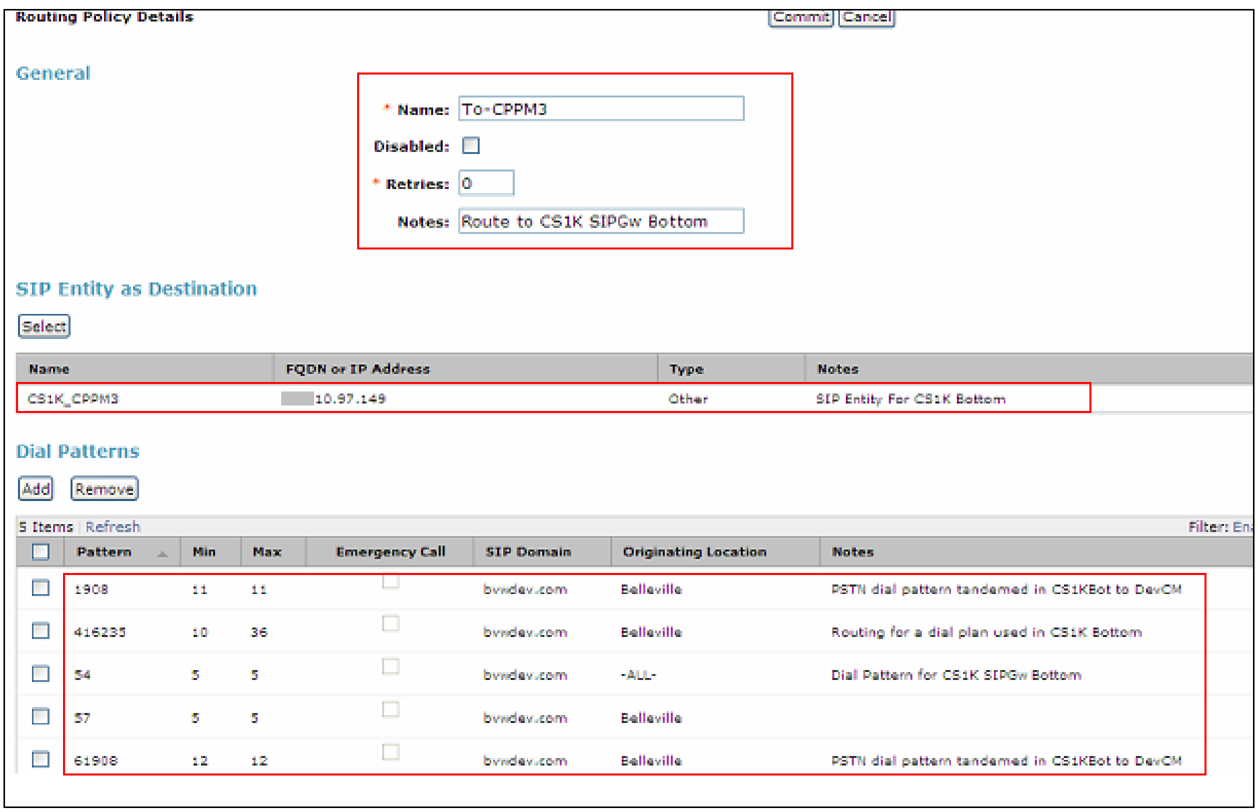
6.6. Configure Routing Policies

Routing policies describe the conditions under which calls will be routed to the SIP Entities. Two routing policies must be added: one for Avaya Aura® Communication Manager and one for Messaging. To add a routing policy, navigate to **Routing →Routing Policies** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). The following screen is displayed. Fill in the following: In the **General** section, enter the following values. Use default values for all remaining fields:

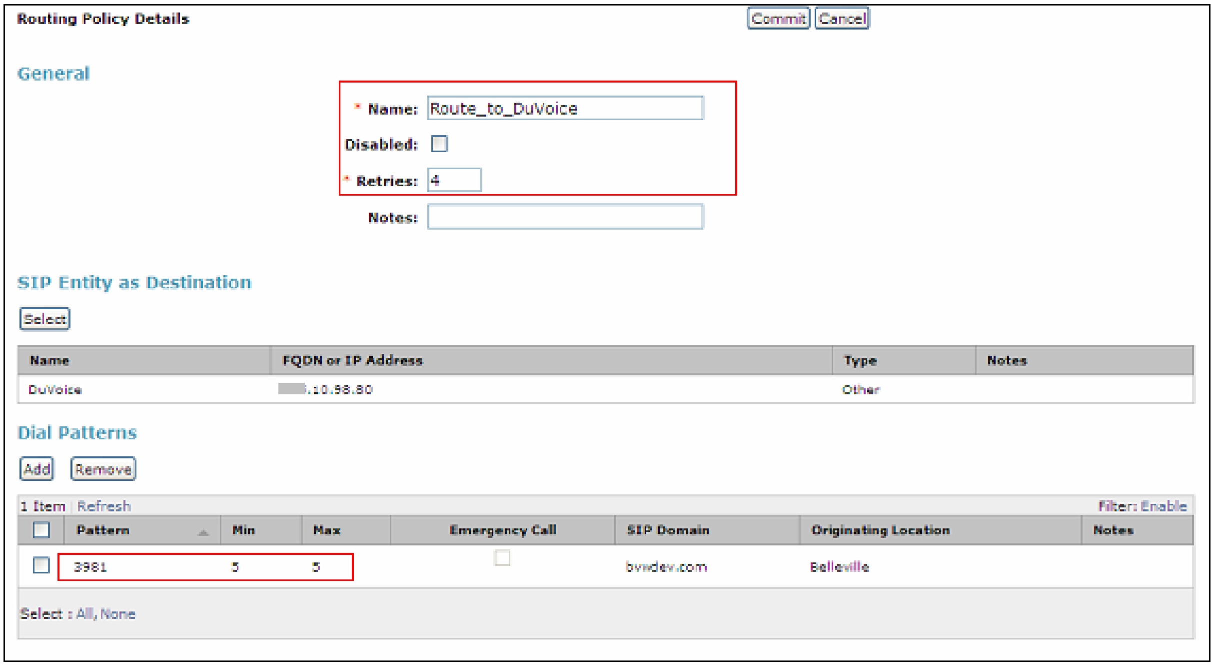
- **Name:** Enter a descriptive name.
- **Notes:** Add a brief description (optional).

In the **SIP Entity as Destination** section, click **Select**. The **SIP Entity List** page opens (not shown). Select the appropriate SIP entity to which this routing policy applies and click **Select**. The selected SIP entity displays on the **Routing Policy Details** page as shown below. Use default values for the remaining fields. Click **Commit** to save. The following screens show the routing policy for Avaya Aura® Communication Manager.

The following screen shows the Routing Policy used to Communication Server 1000 Communication Manager.



Repeat the steps to define routing policies to DV2000.



6.7. Configure Dial Patterns

Dial Patterns define digit strings to be matched for inbound and outbound calls. In addition, the domain in the request URI is also examined. In the compliance test, the following dial patterns are defined from Session Manager.

- 54xxx – SIP endpoints in Avaya CS1000
- 39810 –DV2000 Pilot Number.

To add a Dial Pattern, select **Routing → Dial Patterns,** and click on the **New** button (not shown) on the right. During the compliance test, 5 digit dial plan was utilized. Provide the following information:

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

- **Pattern:** Enter a dial string that will be matched against the Request-URI of the call.
- **Min:** Enter a minimum length used in the match criteria.
- **Max:** Enter a maximum length used in the match criteria.
- **SIP Domain:** Enter the destination domain used in the match criteria.
- **Notes:** Add a brief description (optional).

In the **Originating Locations and Routing Policies** section, click **Add**. From the **Originating Locations and Routing Policy List** that appears (not shown), select the appropriate originating location for use in the match criteria. Lastly, select the routing policy from the list that will be used to route calls that match the specified criteria. Click **Select**.

Default values can be used for the remaining fields. Click **Commit** to save. See screenshot in **Section 6.6** for detail of dial pattern for each SIP entity.

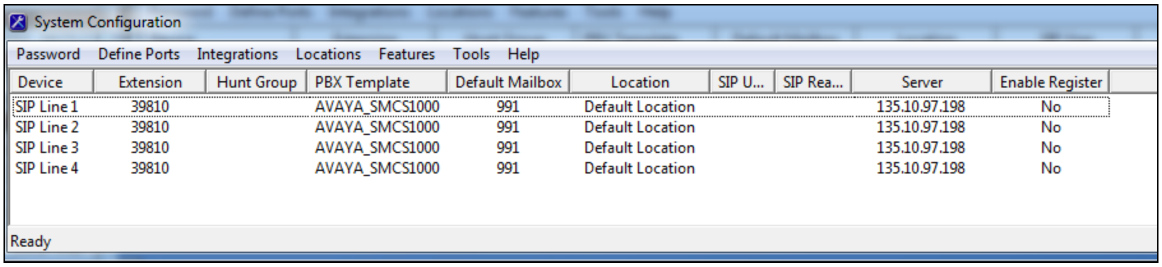
7. Configure DuVoice DV2000 Application

This section describes details the essential portion of the DuVoice DV2000 configuration to interoperate with Avaya Session Manager and Avaya Communication Server 1000. These Application Notes assume that the DuVoice DV2000 has already been properly installed by DuVoice services personnel.

At the time of taking the screenshot all setup has been in place. This section will capture the detail of the configuration had been in place on DV20000 for review.

7.1. Administer PMS Pass-through Connectors

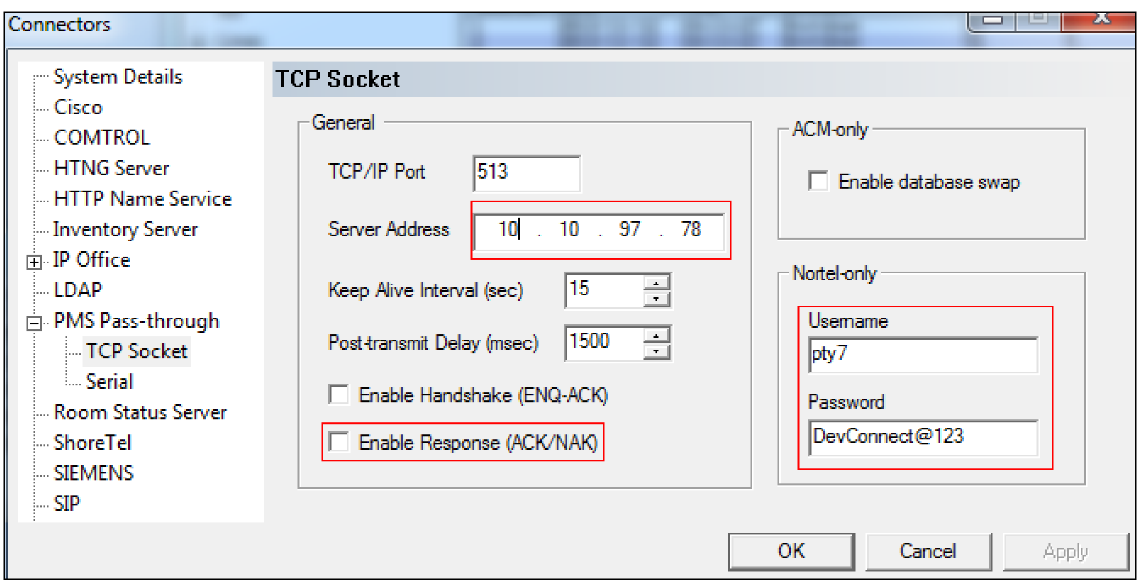
From the DuVoice server, select **Start → All Programs → DuVoice → System Configuration**. Below is the **System Configuration** window.



Open PMS Pass-through by select menu **Features → Connectors.** In the **Connectors** window, and select **TCP Socket**, in TCP Socket enter the information of rlogin that create in **Section 5.1** as following:

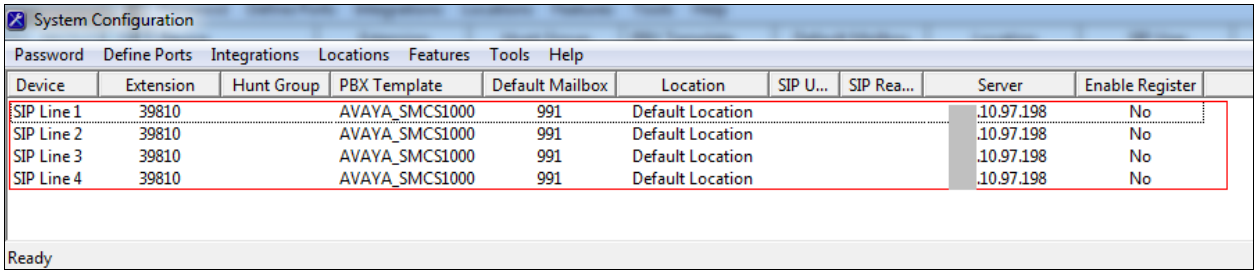
- **TCP/IP Port:** Default port is 513.
- **Server Address:** Enter the ELAN IP address of Communication Server 1000
- **Enable Handshake:** Do not check this option.
- **Enable Response:** **Make sure this option is uncheck.**
- **User name:** Enter user **pty7**
- **Password:** Enter password of pty7 user. In compliance test the password is **DevConnect@123.**

Below is screenshot of **TCP Socket** detail.



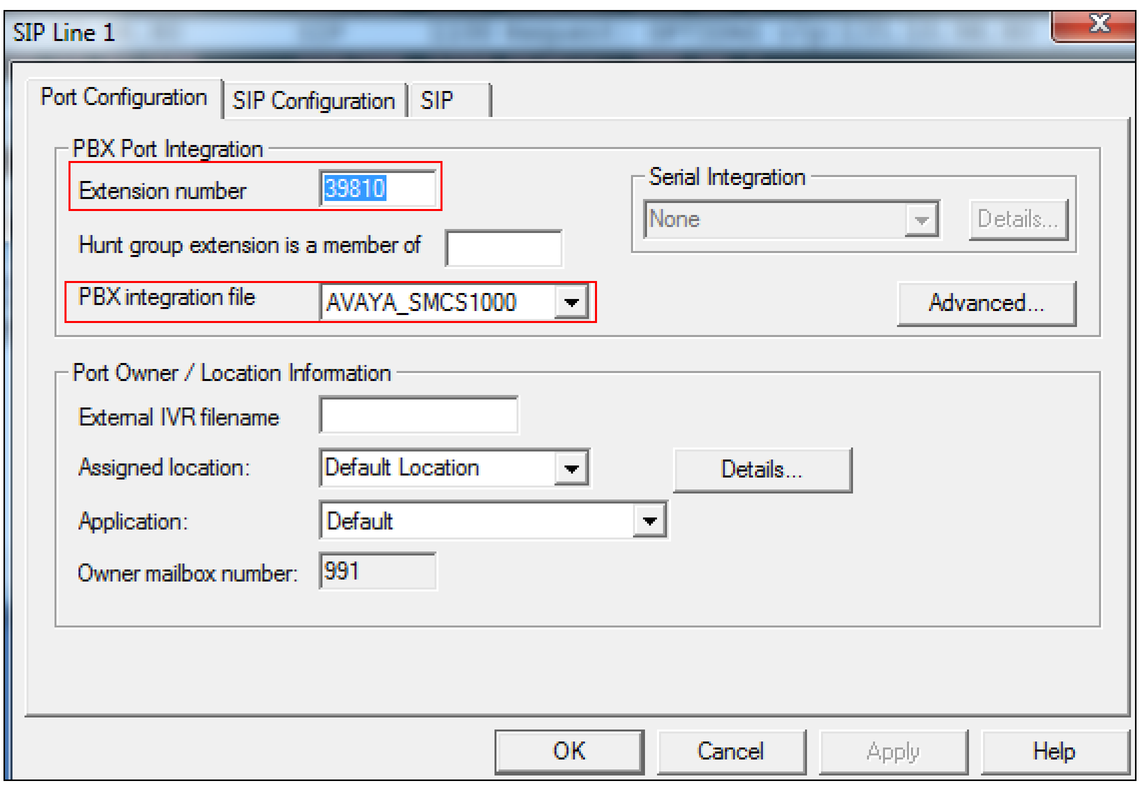
7.2. Verify the System Configuration and SIP Line Setting

Select **Start →All Programs→DuVoice→System Configuration**. The **System Configuration** screen is displayed:



Double click on **SIP Line 1**. Under the **Port Configuration** tab, verify the following values. Use default values for all remaining fields:

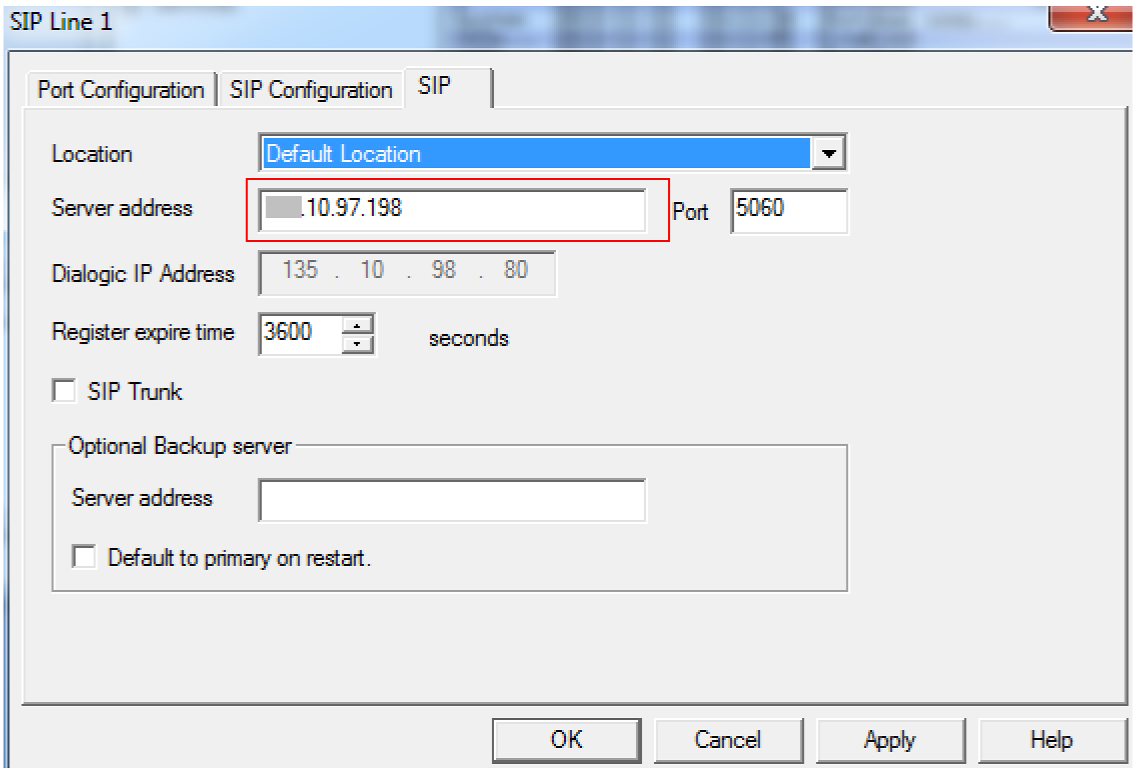
- **Extension number:** Verify that the extension number is set to the DuVoice pilot number, during compliance test, extension 39810 is used as pilot number for DV2000.
- **PBX integration file:** Verify that the PBX integration file is set to **AVAYA_SMCS1000**.



Under the **SIP** tab verify SIP setting:

- **Server address:** Verify that the Registrar address is set to the **IP address of Session Manager**.
- **Port:** **5060**
- **Dialogic IP Address:** Verify it set to **IP address of DuVoice** device**.**
- **SIP trunk:** Verify that SIP Trunk is not checked.

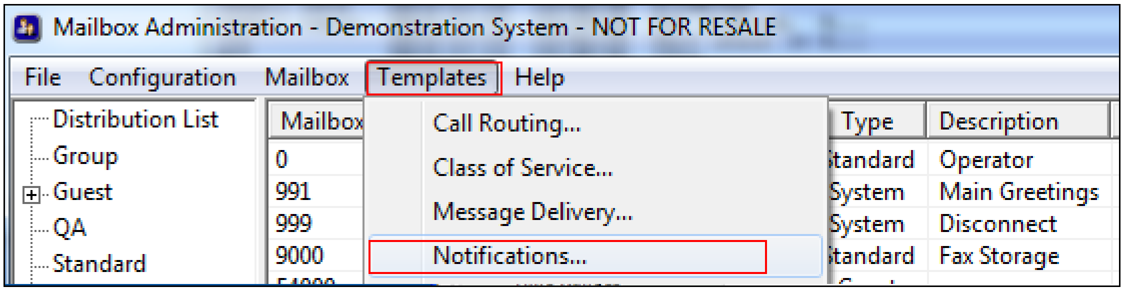
Leave other fields as default.



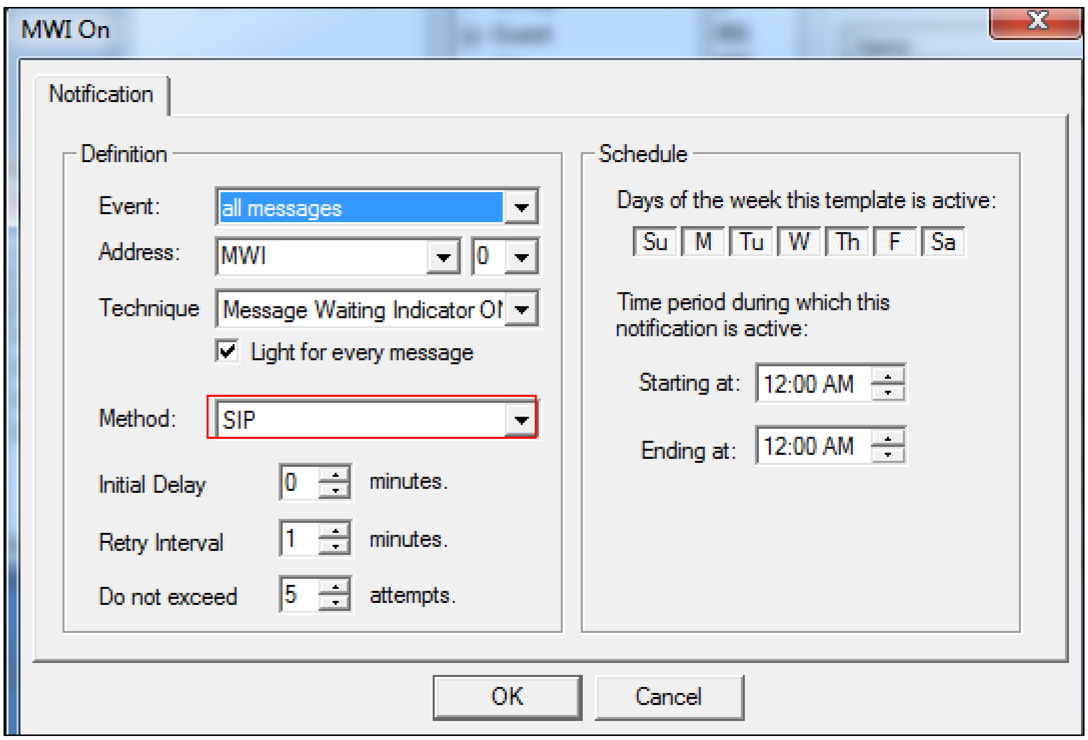
Repeat this section for each remaining Voice Port Number **2-4** for the sample configuration.

7.3. Administer MWI

From the DuVoice server, select **Start →All Programs→Mailbox Administration**, the **Mailbox Administration** screen is displayed. Select **Templates→Notifications….**



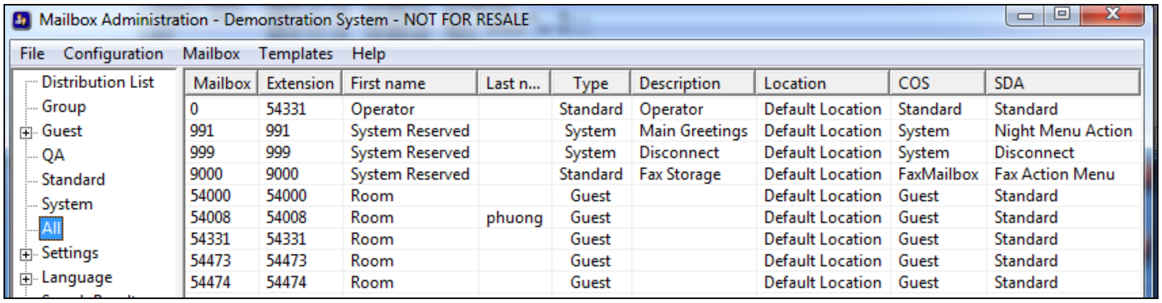
In the **Notifications** window, double click on **MWI On. In The MWI On** window, verify SIP method is selected as shown below.



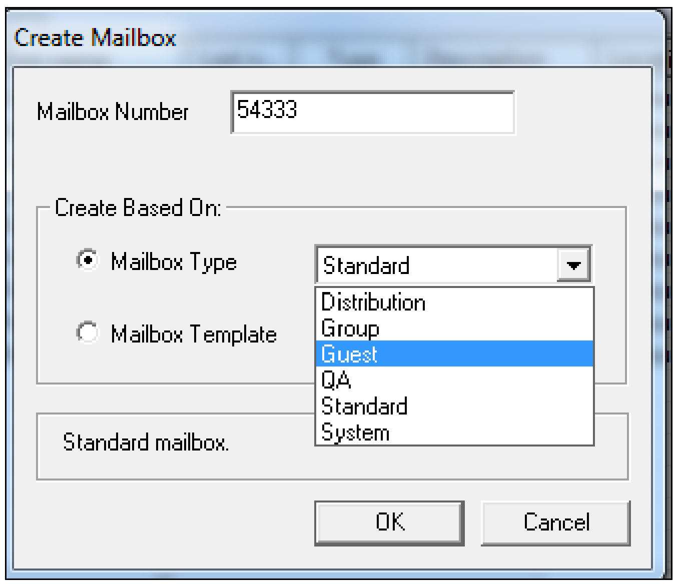
Perform the same for **MWI Off** notification.

7.4. Administer Mailboxes

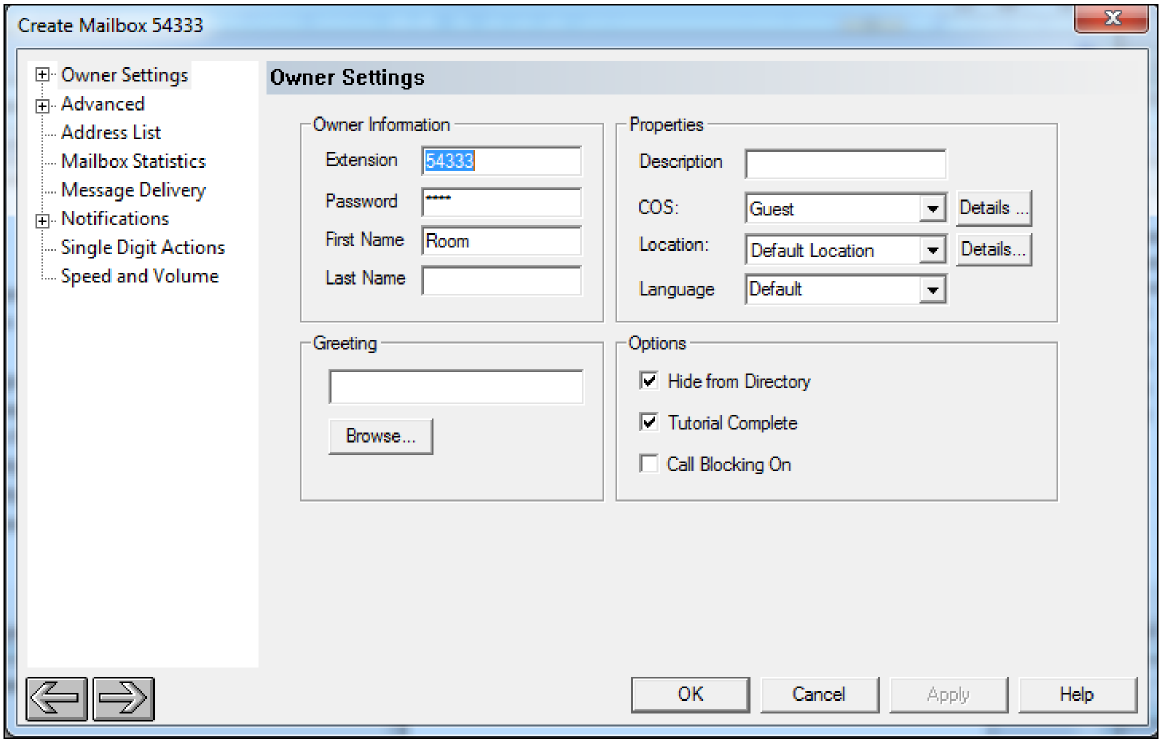
From the DuVoice server, select **Start →All Programs→Mailbox Administration**.
The **Mailbox Administration** screen is displayed. Below is the Mailbox window with the list of Guest mailbox used during compliance test:



To add new mailbox, right click on the Mailbox window, the **Create Mailbox** screen is displayed next. For **Mailbox Number**, enter the first voicemail user extension, in this case "54333" was created. For **Mailbox Type**, select "Guest" for guest users and "Standard" for front desk and staff users.



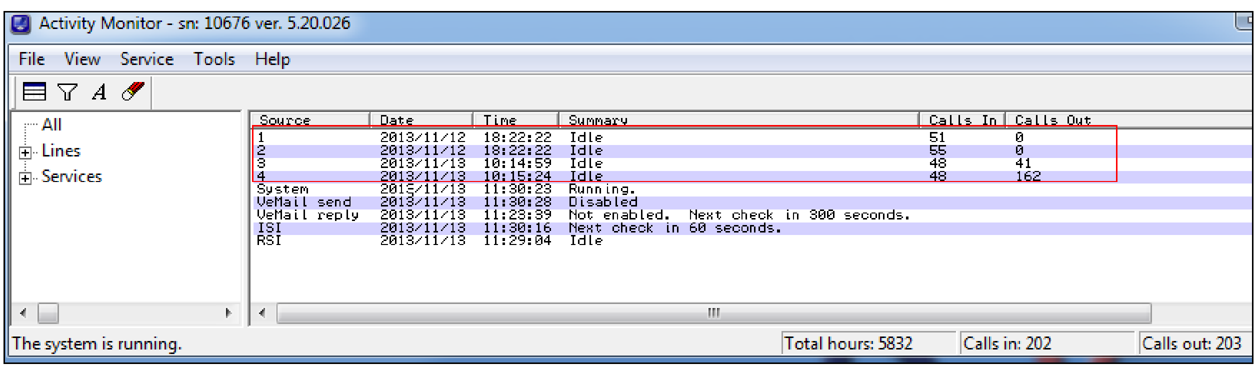
The **Mailbox 54333** screen is displayed next. Enter desired values for **Password**, **First Name**, and **Last Name**, and retain the default values in the remaining fields.



Repeat this section for all voicemail users.

7.5. Verify Port Activity

From the DuVoice server, select **Start → All Programs → Activity Monitor**. The **Activity Monitor** screen is displayed. Verify that all configured ports are **IDLE** and ready to accept calls.



8. Verification Steps

8.1. Verify Property Management System Interface

The following steps might be used to verify the connection between Avaya CS1000 switch and DuVoice DV2000.

To verify the DV2000 can successfully Rlogin to ELAN of Communication Server 1000 with the "**pty7**" user, in CS1000 use overlay **LD 37** to print all status of TTY ports. The TTY port 7 should be shown as "**ENBL**".

```
>ld 37
IOD000
.stat
TTY  0 : DSBL
TTY  1 : DSBL
TTY  2 : DSBL
TTY  3 : DSBL
TTY  4 : ENBL
TTY  5 : DSBL
TTY  6 : DSBL   DES: TTY1
TTY  7 : ENBL   DES: pms
TTY  8 : DSBL ( MGC  4 0 )  DES: CDR
TTY  9 : DSBL   DES: pty9
TTY  10 : ENBL   DES: PTY10
```

To verify the ELAN connection for PMS, from the Hospitality Tester do a room check-in with guest name for an extension and verify that the CPND name is updated on this extension. The screen below shows the DuVoice DV2000 terminal console with the check-in command sent to the switch.

```
ACMSRVR
IG | PID | TID | Date-Time | Filter | Data
.. 2... 4... 2013/1... ERROR  RLoginPort::open(1025) Unable to connect to server [10.10.97.78:513:pty7]; Unable...
.. 2... 4... 2013/1... ERROR  Failed initialization [10.10.97.78:513]; err = 7
.. 2... 4... 2013/1... FFF... Settings: [10.10.97.78:513:pty7, NORTEL]
.. 2... 4... 2013/1... ERROR  RLoginPort::open(1025) Unable to connect to server [10.10.97.78:513:pty7]; Unable...
.. 2... 4... 2013/1... ERROR  Failed initialization [10.10.97.78:513]; err = 7
.. 2... 4... 2013/1... FFF... Settings: [10.10.97.78:513:pty7, NORTEL]
.. 2... 4... 2013/1... STATUS Standard Access: 0 of 0
.. 2... 4... 2013/1... STATUS Administrator Access: 0 of 5
.. 2... 4... 2013/1... STATUS Standard Access: 0 of 0
.. 2... 4... 2013/1... STATUS Administrator Access: 0 of 5
.. 2... 4... 2013/1... ERROR  RLoginPort::open(1025) Unable to connect to server [10.10.97.78:513:pty7]; Unable...
.. 2... 4... 2013/1... ERROR  Failed initialization [10.10.97.78:513]; err = 7
.. 2... 1... 2013/1... STATUS Standard Access: 0 of 0
.. 2... 1... 2013/1... STATUS Administrator Access: 0 of 5
.. 2... 1... 2013/1... STATUS Standard Access: 0 of 0
.. 2... 1... 2013/1... STATUS Administrator Access: 0 of 5
.. 2... 4... 2013/1... FFF... Settings: [10.10.97.78:513:pty7, NORTEL]
.. 4... 4... 2013/1... STATUS Built on Oct 16 2013 Version: 5.20.027
.. 4... 3... 2013/1... FFF... Settings: [10.10.97.78:513:pty7, NORTEL]
.. 4... 3... 2013/1... ERROR  RLoginPort::open(1025) Unable to connect to server [10.10.97.78:513:pty7]; Unable...
.. 4... 3... 2013/1... ERROR  Failed initialization [10.10.97.78:513]; err = 7
.. 4... 3... 2013/1... FFF... Settings: [10.10.97.78:513:pty7, NORTEL]
.. 4... 0... 2013/1... STATUS Built on Oct 16 2013 Version: 5.20.027
.. 4... 2... 2013/1... FFF... Settings: [135.10.97.78:513:pty7, NORTEL]
.. 4... 2... 2013/1... STATUS Connected to [135.10.97.78] on port [513]
.. 4... 4... 2013/1... STATUS Standard Access: 0 of 0
.. 4... 4... 2013/1... STATUS Administrator Access: 0 of 5
.. 4... 4... 2013/1... STATUS Standard Access: 0 of 0
.. 4... 4... 2013/1... STATUS Administrator Access: 0 of 5
.. 4... 2... 2013/1... STATUS Port 513; TX [<02>SE CP  54000 "1111, Room" CH IN<03><08>]
.. 4... 2... 2013/1... STATUS RLoginPort::rxPacket(1148) RX: [<02>SE CP  54000 "1111, Room" CH IN<03><08>]
.. 4... 4... 2013/1... STATUS Standard Access: 0 of 0
.. 4... 4... 2013/1... STATUS Administrator Access: 0 of 5
.. 4... 4... 2013/1... STATUS Standard Access: 0 of 0
.. 4... 4... 2013/1... STATUS Administrator Access: 0 of 5
.. 4... 2... 2013/1... STATUS Port 513; TX [<02>SE CP  54000 "Phuong, Nguyen" CH IN<03>2]
.. 4... 2... 2013/1... STATUS RLoginPort::rxPacket(1148) RX: [<02>SE CP  54000 "Phuong, Nguyen" CH IN<03>2]
```

Verify Automated Attendant features:

Place an incoming trunk call to the DV2000 pilot number, when asked enter a valid guest extension (defined in the DuVoice server) to be transferred to. Verify that the transfer takes place, ring back and speech path in both directions.

Verify Voice Mail features:

Place a call to reach a guest, do not answer the call. Verify the caller hears the system greeting, leave a voice message. Verify the MWI is turned on at the guest telephone. Make a call from the guest extension to the hunt group pilot number, Verify the greeting is played and that the message can be retrieved. Verify the MWI is turned off.

Verify Wakeup call feature:

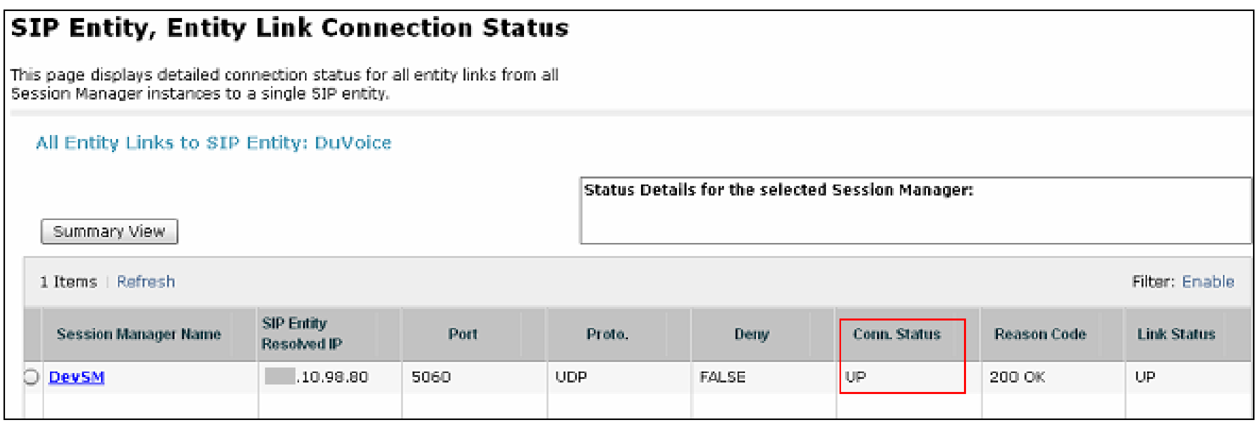
From a guest extension call the DV2000 pilot number to schedule a wakeup call. Verify that the wakeup call takes place at the scheduled time.

8.2. Verify SIP Entity Links

Navigate to **Elements → Session Manager → System Status → SIP Entity Monitoring** (not shown) to view more detailed status information for one of the SIP Entity Links.

Select the SIP Entity for DevACEsrv from the **All Monitored SIP Entities** table (not shown) to open the **SIP Entity, Entity Link Connection Status** page.

In the **All Entity Links to SIP Entity: DuVoice** table, verify the **Conn. Status** for the link is "**Up**" as shown below.



9. Conclusion

These Application Notes describe the procedures for configuring DuVoice DV2000 to interoperate with Session Manager and Communication Server 1000. All interoperability compliance test cases executed against such a configuration were completed successfully.

10. Additional References

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

[1] Hospitality Features Fundamentals, Release 7.0, Issue 04.01, Date June 2010.
[3] Software Input Output Reference — Administration Avaya Communication Server 1000, Release 7.6, Issue 04.02, Date Apr 04, 2013.

Product documentation for DuVoice DV2000 products may be found at

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.