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Application Notes for Configuring Vocal IP Networx SIP Trunk service with Avaya Communication Server 1000 Release 7.6, Avaya Aura® Session Manager Release 6.3 and Avaya Session Border Controller for Enterprise Release 6.2 – Issue 1.0

Abstract

These Application Notes describe procedure for configuration Vocal IP Networx SIP Trunk service with Avaya Communication Server 1000 Release 7.6, Avaya Aura® Session Manager Release 6.3, and Avaya Session Border Controller for Enterprise Release 6.2.

The test was performed to verify SIP trunk features including basic calls, call forward (all calls, busy, no answer), call transfer (blind and consult), conference, and voice mail. Calls were placed to and from the PSTN with various Avaya endpoints.

Vocal IP Networx SIP Trunk service provides PSTN access via SIP trunks between enterprise and Vocal IP Networx's network as an alternative to legacy analog or digital trunks. This approach generally results in lower cost for the enterprise.

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.

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1. Introduction

These Application Notes illustrate a sample configuration using Avaya Communication Server 1000 Release 7.6, Avaya Aura® Session Manager Release 6.3, Avaya Session Border Controller for Enterprise (Avaya SBCE) Release 6.2 with Vocal IP Networx SIP Trunk service. Vocal IP Networx SIP Trunk service provides PSTN access via SIP Trunks between the enterprise and Vocal IP Networx's network as an alternative to legacy analog or digital trunks.

2. General Test Approach and Test Results

Communication Server 1000 was connected to Avaya SBCE via SIP Trunks to Session Manager. Avaya SBCE was connected to Vocal IP Networx's network via SIP trunks. Various call types were made from Communication Server 1000 to Vocal IP Networx and vice versa to verify interoperability.

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

2.1. Interoperability Compliance Testing

Compliance testing scenarios for the configuration described in these Application Notes included the following:

- General call processing between Communication Server 1000 and Vocal IP Networx SIP Trunk service, including the following:
    - Codec/ptime (G.711 u-law/20ms, G.711 a-law/20ms and G.729/20ms), no VAD.
    - Hold/Resume on both ends.
    - Calling Line Identification Display (CLID).
    - Ring-back tone.
    - Speech (audio) path.
    - Dialing plan support (local, long distance, international, outbound toll-free, inbound toll-free, 411, and 911).
    - Advanced features (Call on Mute, Call Park, and Call Waiting).
    - Abandoned Call.
- Call redirection verification: all supported methods (blind transfer, consultative transfer, call forward, and conference) including CLID. Call redirection is performed from both ends.
- Response to SIP OPTIONS queries.
- Fax G.711 Pass Through
- Inbound and outbound long hold time call stability.
- Caller number/ID presentation.
- Privacy requests (i.e., caller anonymity) and Caller ID restriction for inbound and outbound calls.

- DTMF (RFC2833) in both directions
- SIP Transport UDP, port 5060
- Voice Mail Server Call Pilot (hosted on Avaya system)
- Use SIP UPDATE on Blind Transfer.
- Use SIP DIVERSION HEADER on Call Redirection.

The following assumptions were made for the compliance tested configuration:
1. Communication Server 1000 R7.6 software with latest patches.
2. Vocal IP Networx service provides support to setup, configure and troubleshoot on carrier switch during testing execution.

During testing, the following activities were made to each test scenario:
1. Calls were checked for the correct call progress tones and cadences.
2. During the ringing state the ring back tone and destination ringing were checked.
3. Calls were checked in both hands-free and handset mode due to internal Avaya requirement.
4. Calls were checked for speech path in both directions using spoken words to ensure clarity of speech.
5. The display(s) of the sets/clients involved were checked for consistent and expected CLID and redirection information both prior to answer and after call establishment.
6. The speech path and messaging system were observed for timely and quality End to End tone audio path generation and application responses.
7. The call server maintenance terminal window was open during the test cases execution for the monitoring of BUG(s), ERROR and AUD messages (See **Session 5.1.2**).
8. Speech path was checked before and after calls were put on/off hold from each end.
9. Calls were checked to ensure that all resources such as Virtual trunks, TDM trunks, Sets and VGWs (Voice Gateways) were released when calls were ended (See **Session 9.2** - SIP Trunk monitoring (LD 32)).

2.2. Test Results

The objectives outlined in **Section 2.1** were verified. All the applicable test cases were executed. However, the following observations were noted during the compliance testing:

1. If the Avaya Communication Server 1000 phone holds/resumes an outbound call, the dialed digits are no longer displayed. This is a Communication Server 1000 known issue.
2. Vocal IP Networx service does not support registration and authentication.

2.3. Support

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

For technical support on the Vocal IP Networx service, please contact customer service or visit http://www.vocalipx.com/index.php/vocal-voice/vocal-sip

3. Reference Configuration

Figure 1 illustrates the test configuration used during the compliance test between Communication Server 1000 and Vocal IP Networx service. For confidentiality and privacy purposes, actual public IP addresses used in this testing have been masked and replaced with fictitious IP addresses throughout the document.
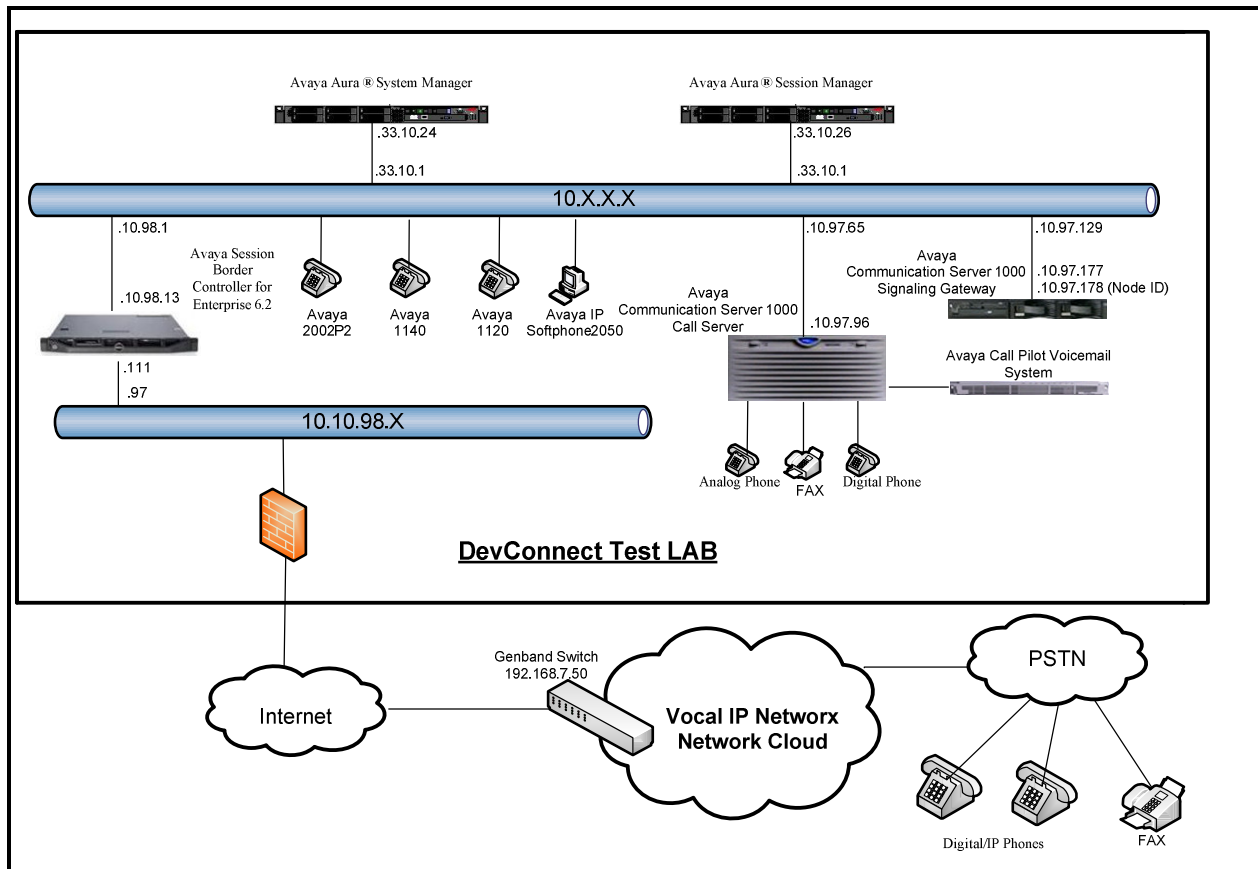

Figure 1- Network diagram for Avaya and Vocal IP Networx SIP Trunk Service

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4. Equipment and Software Validated

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

Avaya systems:

Equipment/Software	Release/Version
Avaya Communication Server 1000 (CPPM)	Call Server: 765 P + Signaling Server: 7.65.16 GA SIP Line Server: 7.65.16 GA
Avaya Call Pilot C201i	Call Pilot Voice Mail Manager: 05.00.41.143
Avaya S8800 Server	Avaya Aura® Session Manager R6.3.0 – 6.3.0.0.630002 - 6.3.4.634012
Avaya S8800 Server	Avaya Aura ®System Manager R6.3.4 – FP3 6.3.0.8.5682 – 6.3.8.2631
Avaya Session Border Controller for Enterprise	6.2.0 Q48

Avaya UNIStim Phones:
    2002 p2
    1140
    1120
    2007
	
	 0604DCO
	 0625C8Q
	 0624C8Q
	 0621C8L

Avaya 3904 Digital Phone	N/A
Avaya IP Softphone 2050	3.1
Analog Symphony 2000	N/A
HP Office jet 4500 Fax	N/A

Vocal IP Networx service:

System	Software
Genband	5.2

Additional patch lineup for the configuration listed as follows:

Call Server: 7.65 P+ GA plus latest DEPLIST – CPL_7.6_1.zip (X2107.65P)
Signaling Server: 7.65.16 GA plus latest DEPLIST – SP_7.6_1.ntl

5. Configure Avaya Communication Server 1000

These Application Notes use the Incoming Digit Translation feature to receive calls, the Numbering Plan Area Code (NPA), and Special Number (SPN) features to route calls from the Communication Server 1000 to the PSTN, via SIP trunks to Vocal IP Networx.

These application notes assume that the basic Communications Server 1000 configuration has already been administered. For further information on Communications Server 1000, please consult the references in **Section 11.**

The procedures below describe the configuration details for configuring the Communication Server 1000.

5.1. Log into Communication Server 1000 System

5.1.1. Log into System Manager and Element Manager (EM)

Open an instance of a web browser and connect to the System Manager using the following address: https://<System Manager IP address>/SMGR/. Log in using an appropriate User ID and Password (not shown). Select **Elements → Communication Server 1000**
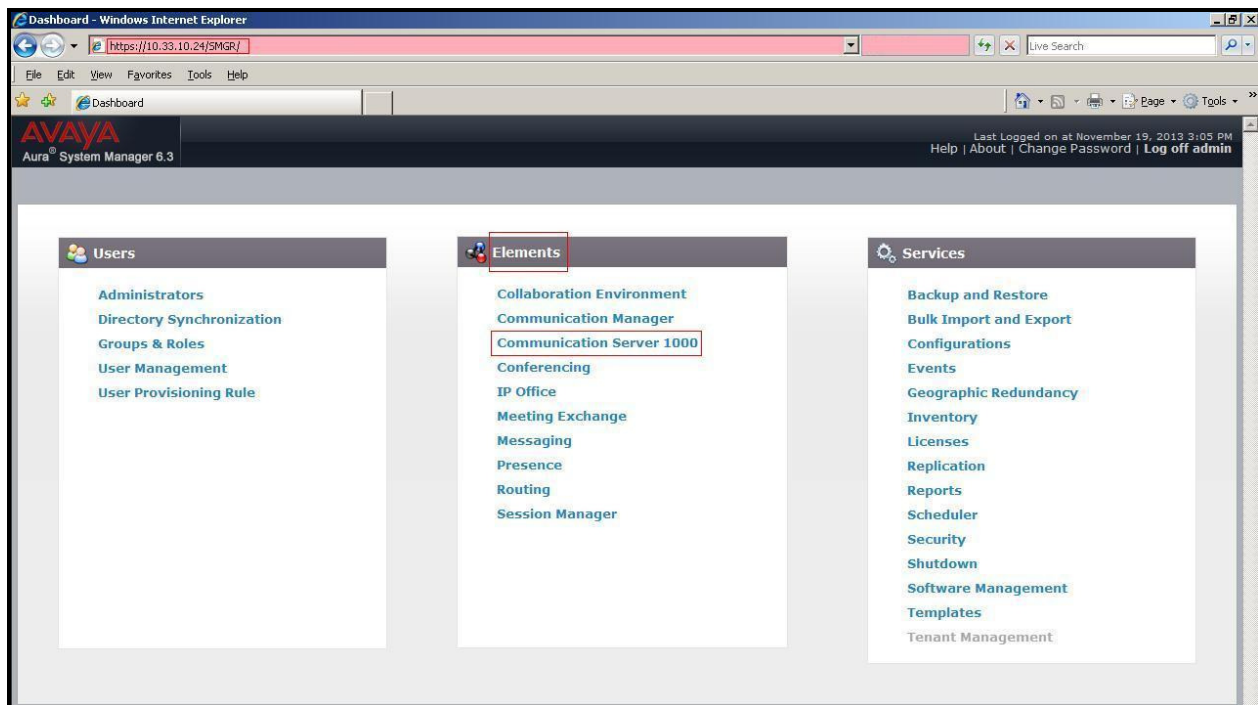
Figure 2 –System Manager Home Screen

The **Avaya Communication Server 1000 Management** screen is displayed. Click on the **Element Name** of the Communication Server 1000 Element as highlighted in red box as below:
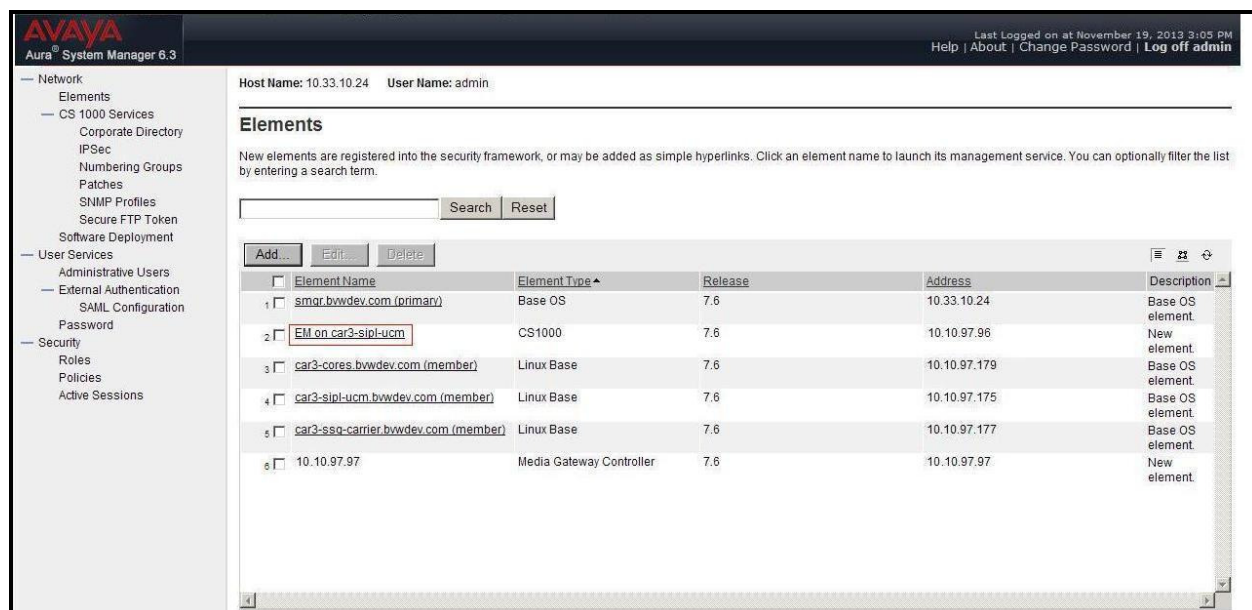

Figure 3 – Communication Server 1000 Management

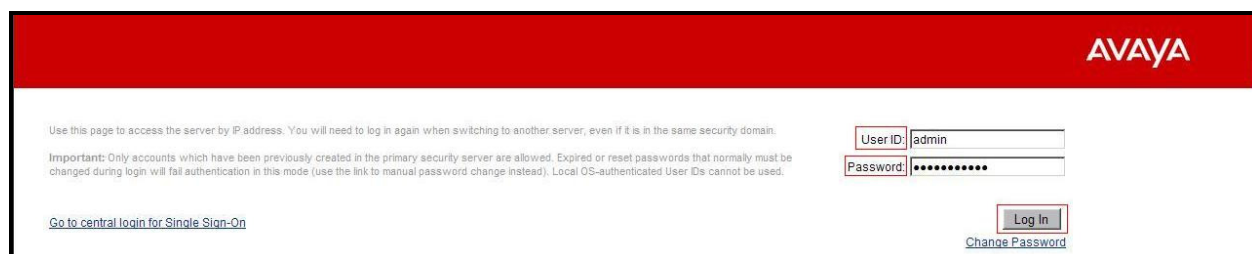Log into the Communication Server 1000 using an appropriate **User ID** and **Password**.

Figure 4 – Communication Server 1000 Log In Screen

The Communication Server 1000 Element Manager **System Overview** page is displayed as shown in **Figure 5**.

> IP Address: 10.10.97.96
> Type: Avaya Communication Server 1000E CPPM Linux
> Version: 4121
> Release: 765 P +

Figure 5 – Element Manager System Overview

5.1.2. Log into the Call Server by using the Overlay Command Line Interface (CLI)

Using Putty, SSH to the IP address of the Communication Server 1000 Signaling Server using an account with administrator credentials.

Run the command **cslogin** and log in with the appropriate user account and password. Sample output is shown below.

login as: **< --- enter an account with administrator credentials**

The software and data stored on this system are the property of, or licensed to, Avaya Inc and are lawfully available only to authorized users for approved purposes. Unauthorized access to any software or data on this system is strictly prohibited and punishable under appropriate laws. If you are not an authorized user then do not try to login. This system may be monitored for operational purposes at any time.

admin@10.10.97.177's password: **<----enter the password**
Last login: Thu Nov 21 11:20:18 2013 from 10.10.98.78
[admin@car3-ssg-carrier ~]$ **cslogin**

SEC054 A device has connected to, or disconnected from, a pseudo tty without authenticating
>login

USERID? **< --- enter the user account**

```
PASS?  <----enter the password
.
TTY #08 LOGGED IN ADMIN 11:39 21/11/2013
The software and data stored on this system are the property of, or licensed to, Avaya Inc and are
lawfully available only to authorized users for approved purposes. Unauthorized access
to any software or data on this system is strictly prohibited and punishable under appropriate
laws. If you are not an authorized user then log out immediately. This system may be monitored
for operational purposes at any time.

>
```

Note: Leave this screen on for monitoring of BUG(s), ERROR and AUD messages.

5.2. Administer an IP Telephony Node

This section describes how to configure an IP Telephony Node on Communication Server 1000.

5.2.1. Obtain Node IP address

These application notes assume that the basic Communication Server 1000 configuration has already been administered and that a Node has already been created. This section describes the steps for configuring a Node (Node ID 3000) in Communication Server 1000 IP network to work with Vocal IP Networx SIP Trunk service. For further information on Communications Server 1000, please consult the references in **Section 11**.

Select **System → IP Network → Nodes: Servers, Media Cards** and then click on the **Node ID** as shown in **Figure 6**.

Figure 6 – IP Telephony Nodes

The **Node Details** screen is displayed in **Figure 7** with the IP address of the Communication Server 1000 node. **Call server IP address: 10.10.97.96**. The **Node IPv4 address 10.10.97.178** is a virtual address which corresponds to the **TLAN IP address 10.10.97.177** of the Signaling Server/SIP Signaling Gateway. The SIP Signaling Gateway uses this Node IP address to communicate with other components to process SIP calls.

Figure 7 –Node Details 1

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The **Node Details** screen is displayed in **Figure 8** with the IP Telephony Node Properties and Applications.

Figure 8 –Node Details 2

5.2.2. Administer Terminal Proxy Server (TPS)

Continuing from **Section 5.2.1**, on the **Node Details** page, select the **Terminal Proxy Server** (**TPS**) link as shown in **Figure 8**. Check the **UNIStim Line Terminal Proxy Server** checkbox to enable proxy service on this node and then click the **Save** button as shown in **Figure 9**.

Figure 9 – TPS Configuration Details

5.2.3. Administer Quality of Service (QoS)

Continuing from **Section 5.2.1**, on the **Node Details** page, select the **Quality of Service (QoS)** link as shown in **Figure 8**. The default Diffserv values are as shown in **Figure 10**. Click on the **Save** button.

Figure 10 – QoS Configuration Details

5.2.4. Synchronize New Configuration

Continuing from **Section 5.2.3**, return to the **Node Details** page (**Figure 7**) and click on the **Save** button. The **Node Saved** screen is displayed. Click on **Transfer Now** (not shown). The **Synchronize Configuration Files (Node ID <3000>)** screen is displayed (not shown). Check the **Signaling Server** checkbox and click on **Start Sync** (not shown). When the synchronization completes, check the **Signaling Server** checkbox and click on the **Restart Applications** (not shown).

5.3. Administer Voice Codec

5.3.1. Enable Voice Codec G.711, G.729

Select **IP Network → Nodes: Servers, Media Cards** from the left pane, and on the **IP Telephony Nodes** screen displayed (not shown), select the **Node ID** of the Communication Server 1000 system. The **Node Details** screen is displayed, (see **Section 5.2.1** for more details). On the **Node Details** page shown in **Figure 8**, click on **Voice Gateway (VGW) and Codecs**. Vocal IP Networx supports **G.711/time 20ms** and **G.729/time 20ms** with **Voice Activity Detection (VAD)** checkbox unchecked. Click on the **Save** button.

Figure 11 – Voice Gateway and Codec Configuration Details

Synchronize the new configuration (please refer to **Section 5.2.4**).

5.3.2. Enable Voice Codec on Media Gateways

From the left menu of the Element Manager page in **Figure 11**, select **IP Network** → **Media Gateways**. The Media Gateways page will appear (not shown). Click on the **MGC** which is located on the right of the page. In the following screen, scroll down to select the **Codec G.711** and **Codec G.729A** and uncheck **VAD** as shown in **Figure 12**. Scroll down to the bottom of the page and click on the **Save** button (not shown).

Figure 12 – Media Gateways Configuration Details

5.4. Zones and Bandwidth Management

This section describes the steps to create two zones: zone 10 for the VGW and IP sets, and zone 255 for the SIP Trunk.

5.4.1. Create a Zone for IP Phones (Zone 10)

The following figures show how to configure a zone for VGW and IP sets for bandwidth management purposes. The bandwidth strategy can be adjusted to preference.
Select **IP Network → Zones** configuration from the left pane (not shown), click on **Bandwidth Zones** as shown in **Figure 13**.
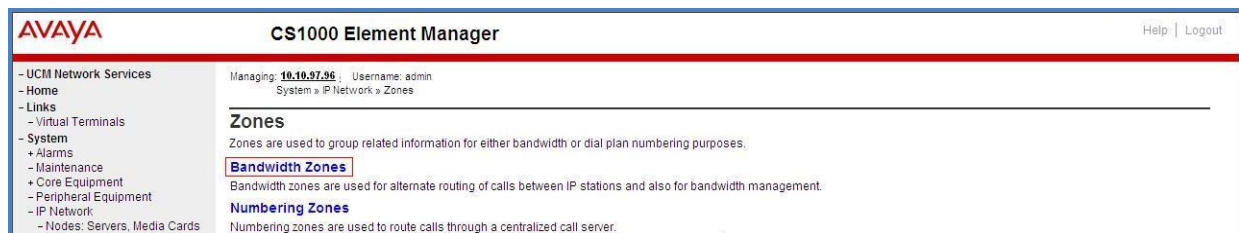
Figure 13 – Zones Page

The **Bandwidth Zones** screen is displayed as shown in **Figure 14**. Click **Add** to create new zone for IP Phones.
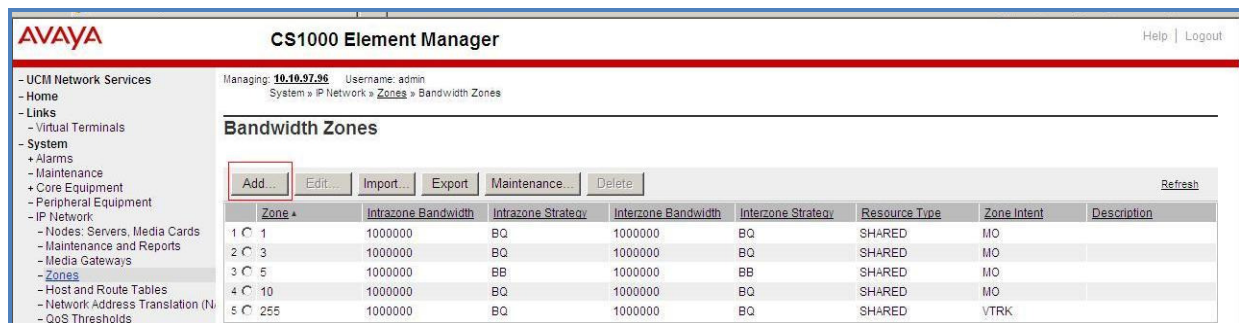
Figure 14 – Bandwidth Zones

Select and input the values as shown below (in the red boxes) in **Figure 15,** and click on the **Submit** button.

- • **INTRA_BW: 1000000**
- • **INTRA_STGY**: Set codec for local calls. Select **Best Quality (BQ)** to use G.711 as the first priority codec for negotiation.
- • **INTER_BW: 1000000**
- • **INTER_STGY**: Set codec for the calls over trunk. Select **Best Quality (BQ)** to use G.711 as the first priority codec for negotiation.
- • **Zone Intent** (**(ZBRN)**): Select **MO (MO)** for IP phones, and VGW.

Figure 15 –Bandwidth Management Configuration Details – IP phone

5.4.2. Create a Zone for Virtual SIP Trunk (Zone 255)

Follow the steps described in **Section 5.4.1** to create a zone for the virtual SIP trunk. The difference is in the **Zone Intent (ZBRN)** field. Select **VTRK** for virtual trunk as shown in **Figure 16** and then click on the **Submit** button.

Figure 16 –Bandwidth Management Configuration Details –virtual SIP trunk

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5.5. Administer SIP Trunk Gateway

This section describes the steps for establishing a SIP connection between the SIP Signaling Gateway and Session Manager.

5.5.1. Integrated Services Digital Network (ISDN)

Select **Customers** in the left pane (not shown). The **Customers** screen is displayed. Click on the link associated with the appropriate customer, in this case **00**. The system can support more than one customer with different network settings and options. The **Customer 00 Edit** page will appear (not shown). Select the **Feature Packages** option from **Customer 00 Edit** page.
The screen is updated with a listing of available **Feature Packages** (not all features are shown in **Figure 17** below). Select **Integrated Services Digital Network** to edit the parameters shown below. Check the **Integrated Services Digital Network** option, and retain the default values for all remaining fields. Scroll down to the bottom of the screen, and click on the **Save** button (not shown).

Figure 17 –Customer – ISDN Configuration

5.5.2. Administer SIP Trunk Gateway to Avaya Aura® Session Manager

Select **IP Network → Nodes: Servers, Media Cards** configuration from the left pane. In the **IP Telephony Nodes** screen displayed (not shown), select the **Node ID** of the Communication Server 1000 system. The **Node Details** screen is displayed as shown in **Figure 8**, **Section 5.2.1**.

On the **Node Details** screen, select **Gateway (SIPGw)**. Under the **General** tab of the **Virtual Trunk Gateway Configuration Details** screen, enter the following values (highlighted in red boxes) for the specified fields, and retain the default values for the remaining fields as shown in **Figure 18**. The **SIP domain name** and **Local SIP port** should be matched in the configuration of Session Manager (in **Sections 6.2 and 6.5**).

Figure 18 – Virtual Trunk Gateway Configuration Details

Click on the **SIP Gateway Settings** tab, under **Proxy or Redirect Server**, and enter the following values (highlighted in red boxes) for the specified fields, retaining the default values for the remaining fields as shown in **Figure 19**. Enter the IP address of Session Manager in the **Primary TLAN IP address** field (This IP address is defined in **Session 6.5.1**). Enter **Port**: **5060** and **Transport protocol**: **UDP**. Uncheck **Support registration** checkbox.

Figure 19 – Virtual Trunk Gateway Configuration Details

On the same page as shown in **Figure 19**, scroll down to the **SIP URI Map** section.
Under the **Public E.164 domain names**, enter the following:
- **National**: leave this SIP URI field blank
- **Subscriber**: leave this SIP URI field blank
- **Special Number**: leave this SIP URI field blank
- **Unknown**: leave this SIP URI field blank

Under the **Private domain names**, enter the following:
- **UDP**: leave this SIP URI field blank
- **CDP**: leave this SIP URI field blank
- **Special Number**: leave this SIP URI field blank
- **Vacant number**: leave this SIP URI field blank
- **Unknown**: leave this SIP URI field blank

The remaining fields can be left at their default values as shown in **Figure 20**. Then click on the **Save** button.

Figure 20 – Virtual Trunk Gateway Configuration Details

Synchronize the new configuration (please refer to **Section 5.2.4**).

5.5.3. Administer Virtual D-Channel

Select **Routes and Trunks** → **D-Channels** (not shown) from the left pane to display the **D-Channels** screen. In the **Choose a D-Channel Number** field, select an available D-channel from the drop-down list and type **DCH** as shown in **Figure 21**. Click the **to Add** button.

Figure 21 – D-Channels

The **D-Channels 100 Property Configuration** screen is displayed next, as shown in **Figure 22**. Enter the following values for the specified fields, and retain the default values for the remaining fields.

- **D channel Card Type:** D-Channel is over IP (DCIP)
- **Designator:** A descriptive name
- **User:** Integrated Services Signaling Link Dedicated (ISLD)
- **Interface type for D-channel:** Meridian Meridian1 (SL1)
- **Meridian 1 node type:** Slave to the controller (USR)
- **Release ID of the switch at the far end:** 25

Click on **Advanced options (ADVOPT)**. Check on the **Network Attendant Service Allowed** checkbox as shown in **Figure 22**. Other fields are left as default.

Figure 22 – D-Channels Configuration Details

Click on the **Basic Options (BSCOPT)** and click on the **Edit** button on the **Remote Capabilities** field as shown in **Figures 23**.

Figure 23 – D-Channel Configuration Details

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The **Remote Capabilities Configuration** page appears as shown in **Figures 24**. Check on the **ND2** and the **MWI** checkboxes.

Figure 24 – Remote Capabilities Configuration Details

Click on the **Return – Remote Capabilities** button (not shown).

Click on the **Submit** button (not shown).

5.5.4. Administer Virtual Super-Loop

Select **System → Core Equipment → Superloops** from the left pane to display the **Superloops** screen. If the Superloop does not exist, please click the **Add** button to create a new one as shown in **Figure 25**. In this example, Superloop 4, 96, 100, and 124 have been added and are being used.
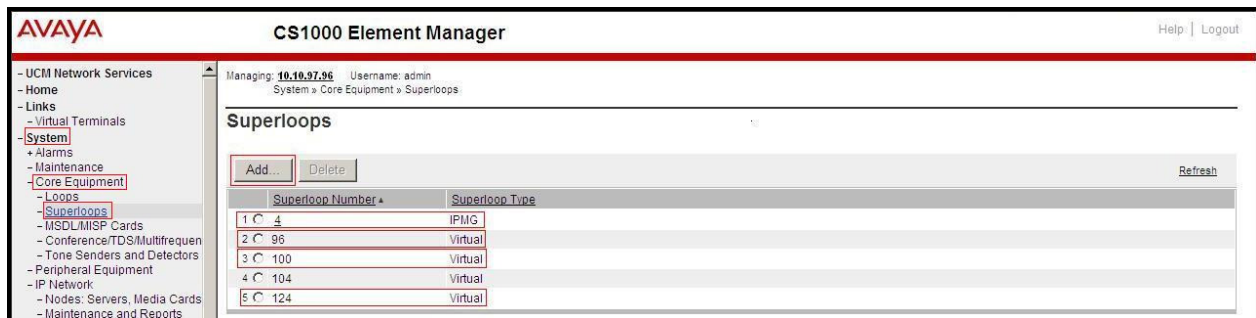
Figure 25 – Administer Virtual Super-Loop Page

5.5.5. Administer Virtual SIP Routes

Select **Routes and Trunks → Routes and Trunks** (not shown) from the left pane to display the **Routes and Trunks** screen. In this example, **Customer 0** is being used. Click on the **Add route** button as shown in **Figure 26**.
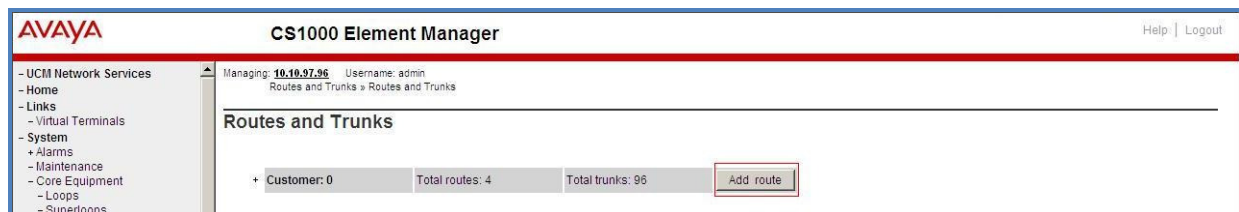
Figure 26 – Add route

The **Customer 0**, New **Route Configuration** screen is displayed next (not shown). The **Basic Configuration** section is displayed to put the following values for the specific fields, and retain the default values for the remaining fields. The screenshot of Basic Configuration section of existing route 100 is displayed to edit as shown in **Figures 27**.

- **Route number** (ROUT): Select an available route number (example: route **100**).
- **Designator field for trunk** (DES): A descriptive text (**100**).
- **Trunk type** (TKTP): TIE trunk data block (**TIE**)
- **Incoming and outgoing trunk** (ICOG): Incoming and Outgoing (**IAO**)
- **Access code for the trunk route** (ACOD): An available access code (example: **8100**).
- Check the **The route is for a virtual trunk route** (VTRK) field, to enable four additional fields to appear.

- For the **Zone for codec selection and bandwidth management** (ZONE) field, enter **255** (created in **Section 5.4.2**). Note: The Zone value is filled out as 255, but after it is added, the screen is displayed with prefix 00.
- For the **Node ID of signaling server of this route** (NODE) field, enter the node number **3000** (created in **Section 5.2.1**).
- Select **SIP (SIP)** from the drop-down list for the **Protocol ID for the route (PCID)** field.
- Check the **Integrated Services Digital Network option** (ISDN) checkbox to enable additional fields to appear. Scrolling down to the bottom of the screen, enter the following values for the specified fields, and retain the default values for the remaining fields.
  - **Mode of operation** (MODE): Select Route uses ISDN Signalling Link (**ISLD**)
  - **D channel number** (DCH): Enter **100** (created in **Section 5.5.3**)
  - **Network calling name allowed** (NCNA): Check the field.
  - **Network call redirection** (NCRD): Check the field.
  - **Insert ESN access code** (INAC)**:** Check the field.

Figure 27 – Route Configuration Details

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Click on **Basic Route Options**, check the **North American toll scheme (NATL)** and **Incoming DID digit conversion on this route (IDC)** checkboxes. Enter **1** for both **Day IDC tree number** and **Night IDC tree number** as shown in **Figure 28**.
Click on the **Submit** button.

Figure 28 – Route Configuration Details

5.5.6. Administer Virtual Trunks

Select **Routes and Trunks** → **Route and Trunks** (not shown). The Route list is now updated with the newly added routes. In the example, the Route 100 was being added. Click on the **Add trunk** button as shown in **Figure 29**.

Figure 29 – Routes and Trunks Page

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The **Customer 0, Route 100, Trunk 1 Property Configuration** screen is displayed. Enter the following values for the specified fields and retain the default values for the remaining fields. The Media Security (sRTP) needs to be disabled at the trunk level by editing the **Class of Service** (CLS) at the bottom of the basic trunk configuration page. Click on the **Edit** button as shown in **Figure 30**.

Note: The Multiple trunk input number (MTINPUT) field may be used to add multiple trunks in a single operation, or repeat the operation for each trunk. In the sample configuration, 32 trunks were created.

- **Trunk data block**: IP Trunk (**IPTI**)
- **Terminal Number**: Available terminal number (Superloop 100 created in **Section 5.5.4**)
- **Designator field for trunk**: A descriptive text
- **Extended Trunk**: Virtual trunk (**VTRK**)
- **Member number**: Current route number and starting member
- **Card Density**: **8D**
- **Start arrangement Incoming**: Immediate (**IMM**)
- **Start arrangement Outgoing**: Immediate (**IMM**)
- **Trunk group access restriction**: Desired trunk group access restriction level
- **Channel ID for this trunk**: An available starting channel ID

Figure 30 – New Trunk Configuration Details

For **Media Security**, select **Media Security Never (MSNV)**. Enter the values for the specified fields as shown in **Figure 31**. Scroll down to the bottom of the screen and click **Return Class of Service** and click on the **Save** button (shown in **Figure 30**).

Figure 31 – Class of Service Configuration Details Page

5.5.7. Administer Calling Line Identification Entries

Select **Customers → 00 → ISDN and ESN Networking** on the left pane. Click on **Calling Line Identification Entries** as shown in **Figure 32**.

Figure 32 – ISDN and ESN Networking

Click on **Add** as shown in **Figure 33**.

Figure 33 – Calling Line Identification Entries

The add entry **0** screen is displayed to put the following values for the specified fields and retain the default values for the remaining fields. The Edit Calling Line Identification of existing entry 0 is displayed as shown in **Figure 34**:

- **National Code**: leave it blank.
- **Local Code**: input prefix digits assigned by Vocal IP Networx, in this case it is 6 digits – **917267**. This **Local Code** will be used for call display purpose for Call Type = Unknown.
- **Home Location Code**: input prefix digits assigned by Vocal IP Networx, in this case it is 6 digits - **917267**. This **Home Location Code** will be used for call display purpose for Call Type = National (NPA).
- **Local Steering Code**: input prefix digits assigned by Vocal IP Networx, in this case it is 6 digits - **917267**. This **Local Steering Code** will be used for call display purpose for Call Type = Local Subscriber (NXX).
- **Use DN as DID**: **YES**.
- **Calling Party Name Display**: Uncheck for **Roman characters**.

Click on the **Save** button as shown in **Figure 34**

Figure 34 – Edit Calling Line Identification 0

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5.5.8. Enable External Trunk to Trunk Transfer

This section shows how to enable the External Trunk to Trunk Transfer feature, which is a mandatory configuration to make call transfer and conference work properly over a SIP trunk.

Log in Call Server Overlay CLI (please refer to **Section 5.1.2** for more details).
Allow External Trunk to Trunk Transfer for Customer Data Block by using **LD 15**.

```
>ld 15
CDB000
MEM AVAIL: (U/P): 33600126    USED U P: 8345621 954062    TOT: 45579868
DISK SPACE NEEDED: 1722 KBYTES
REQ: chg
TYPE: net

TYPE NET_DATA
CUST 0
OPT
…
TRNX YES (←Enable transfer feature)
EXTT YES (← Enable external trunk to trunk Transfer )
…
```

5.6. Administer Dialing Plans

5.6.1. Define ESN Access Codes and Parameters (ESN)

Select **Dialing and Numbering Plans** → **Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (**ESN**) screen as shown in **Figure 35**.
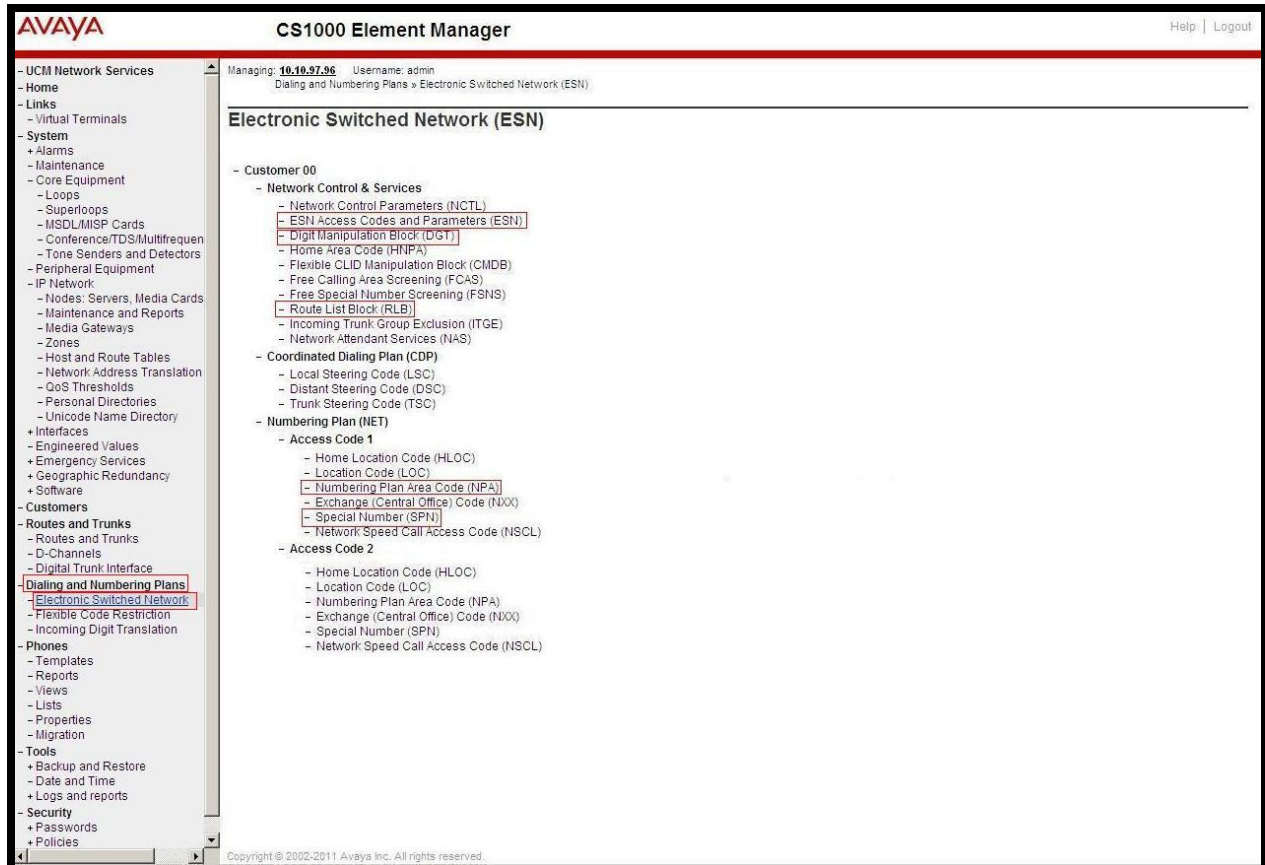
Figure 35 –ESN Configuration Details

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Select **ESN Access Codes and Basic Parameters** to define **NARS/BARS Access Code 1** as shown in **Figure 36**.

Click the **Submit** button (not shown).

Figure 36 – ESN Access Codes and Basic Parameters

5.6.2. Associate NPA and SPN call to ESN Access Code 1

Log in Call Server CLI (please refer to **Section 5.1.2** for more details), change Customer Net Data block by using **LD 15**.

```
>ld 15
CDB000
MEM AVAIL: (U/P): 35600086    USED U P: 8325631 954152    TOT: 44879869
DISK SPACE NEEDED: 1722 KBYTES
REQ: chg
TYPE: net

TYPE NET_DATA
CUST 0
OPT
AC2 xNPA xSPN    →  (Set NPA, SPN not to associate to ESN Access Code 2)
FNP
CLID
…
```

Verify Customer Net Data block by using **LD 21**.

```
>ld 21
PT1000

REQ: prt
TYPE: net
TYPE NET_DATA
CUST 0

TYPE NET_DATA
CUST 00
OPT RTA
AC1 INTL NPA SPN NXX LOC ------  > (NPA, SPN are associated to ESN Access Code 1)
AC2
FNP YES
…
```

5.6.3. Digit Manipulation Block Index (DMI)

The following steps show how to add DMI for the outbound call. There is an index, which was added to the Digit Manipulation Block List (14).

Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen as shown in **Figure 35**. Select **Digit Manipulation Block** (DGT). The **Digit Manipulation Block List** is displayed as shown in **Figure 37**. In the **Please choose the** field, , select an available **Digit Manipulation Block Index** from the drop-down list, and click on the **to Add** button.

Figure 37 – Add a DMI

The DMI_14 screen will open. In this testing, it is not supposed to delete any number of leading digits, therefore enter **0** for the **Number of leading digits to be deleted** field and select **NPA (Numbering Plan Area)** for the **Call Type to be used by the manipulated digits** and then click on **Submit** button as shown in **Figure 38**.

Figure 38 – DMI_14 Configuration Details

5.6.4. Route List Block (RLB) (RLB 14)

This session shows how to add a RLB associated with the DMI created in **Section 5.6.3**. Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen as shown in **Figure 35**. Select **Route List Block** (RLB).

Enter an available value in the textbox for the **Please enter a route list index** (in this case 14) and click on the **to Add** button as shown in **Figure 39**. The screen shown in **Figure 40** will open.

Figure 39 – Add a Route List Block.

Enter the following values for the specified fields, and retain the default values for the remaining fields (**Figure 40**). Scroll down to the bottom of the screen, and click on the **Submit** button (not shown).

- **Digit Manipulation Index**: 14 (created in **Section 5.6.3**)
- **Incoming CLID Table**: 0 (created in **Section 5.5.7**)
- **Route number** 100 (created in **Section 5.5.5**)

Figure 40 – RLB_14 Route List Block Configuration Details

5.6.5. Inbound Call – Incoming Digit Translation Configuration

This section describes the configuration steps required in order to receive calls from PSTN via the Vocal IP Networx SIP Trunk service.

Select **Dialing and Numbering Plans → Incoming Digit Translation** (not shown) from the left pane to display the **Incoming Digit Translation** screen. Click on the **Edit IDC** button as shown in **Figure 41**.
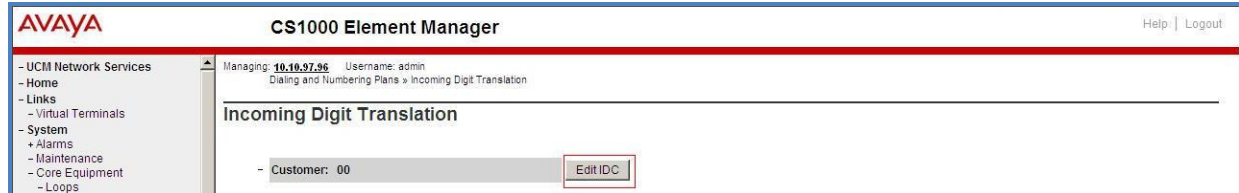
Figure 41 – Incoming Digit Translation

Click on the **New DCNO** to create the digit translation mechanism. In this example, Digit Conversion Tree Number 1 has been created as shown in **Figure 42**.

Figure 42 – Incoming Digit Conversion Property

Detail configuration of the Digit Conversion Tree Configuration is shown in **Figure 43**. The **Incoming Digits** can be added to map to the Converted Digits which would be the associated Communication Server 1000 system phone DN. This **DCNO** has been assigned to route 100 as shown in **Figure 27**.

In the following configuration, the incoming call from PSTN with DID with prefix 917267 will be translated to the associated DN with 4 digits. For voicemail testing purpose, configure the DID number 9172675528 to translate to 1700 (not shown).

Figure 43 – Digit Conversion Tree

5.6.6. Outbound Call - Special Number Configuration

There are special numbers which have been configured to be used for this testing such as: 0, 1800, 411, 911 and so on.

Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen as show in **Figure 35**. Select **Special Number** (SPN). Enter a SPN number and then click on **to Add** button. **Figure 44** shows all the special numbers used for this testing.

Figure 44 – Add a SPN

5.6.7. Outbound Call - Numbering Plan Area (NPA)

This section describes the creation of NPA used in this test configuration.

Select **Dialing and Numbering Plans → Electronic Switched Network** from the left pane to display the **Electronic Switched Network** (ESN) screen (not shown). Select **Numbering Plan Area Code** (NPA) as shown in **Figure 35**. Enter the area code desired in the textbox and click on the **to Add** button. The 1212, 1613, 1972, and 613 area codes were used in this configuration as shown in **Figure 45**.

Figure 45 – Numbering Plan Area Code List

5.7. Administer a Phone

This section describes the creation of Communication Server 1000 clients used in this configuration.

5.7.1. Phone creation

Refer to **Section 5.5.4** to create a Virtual Superloop - **96** used for IP phone. Refer to **Section 5.4.1** to create a bandwidth zone - **10** for IP phone. Log in to the Call Server Command Line Interface (please refer to **Section 5.1.2** for more detail). Create an IP phone by using LD 11 as shown below:

```
REQ: prt
TYPE: 2002p2
TN   96 0 0 2
DATE
PAGE
DES
MODEL_NAME
EMULATED
DES  2002P2 < --- Describe information for IP Phone
TN   96 0 00 02  VIRTUAL < --- Set Terminal Number for IP Phone
TYPE 2002P2
CDEN 8D
CTYP XDLC
CUST 0
NUID
NHTN
CFG_ZONE 00010 < --- Set bandwidth zone for IP phone
CUR_ZONE 00010
MRT
ERL  12345
ECL  0
FDN
TGAR 0
LDN  NO
NCOS 7
SGRP 0
RNPG 0
SCI  0
SSU
LNRS 16
XLST
SCPW
SFLT NO
CAC_MFC 0
CLS  UNR FBA WTA LPR MTD FNA HTA TDD CRPD
     MWA LMPN RMMD SMWD AAD IMD XHD IRD NID OLD VCE DRG1
     POD SLKD CCSD SWD LNA CNDA
     CFTD SFA MRD DDV CNIA CDCA MSID DAPA BFED RCBD
     ICDD CDMD LLCN MCTD CLBD AUTU
     GPUD DPUD DNDD 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
    DRDD EXR0
    USMD USRD ULAD CCBD RTDD RBDD RBHD PGND OCBD FLXD FTTC DNDY DNO3 MCBN
    FDSD NOVD VOLA VOUD CDMR PRED RECD MCDD T87D SBMD
    MSNV FRA  PKCH MWTD DVLD CROD ELCD
CPND_LANG ENG
HUNT
PLEV 02
PUID
UPWD
DANI NO
AST
IAPG 0
AACS NO
ITNA NO
DGRP
MLWU_LANG 0
MLNG ENG
DNDR 0
KEY  00 SCR 5528 0     MARP < --- Set the position of DN 5528 to display on key 0 of the phone
    CPND
     CPND_LANG ROMAN
      NAME Vocal1 < --- Set name to display
      XPLN 13
      DISPLAY_FMT FIRST,LAST
   01
<Text removed for brevity>
```

5.7.2. Enable Privacy for the Phone

This section shows how to enable Privacy for a phone by changing its class of service (CLS) and this feature cannot be enabled or disabled from the phone. By modifying the configuration of the phone created in **Section 5.7.1**, the display of the outbound call will be changed appropriately.

To hide the display number, set **CLS** (Class of Service) to **DDGD**. Communication Server 1000 will include "Privacy:id" in the SIP message header before sending it to Vocal IP Networx.

```
>ld 11
REQ: chg
TYPE: 2002p2
TN   96 0 0 2
ECHG yes
ITEM CLS DDGD
…
```

To allow the display number, set **CLS** to **DDGA**. Communication Server 1000 will not send the Privacy header to Vocal IP Networx.

```
>ld 11
REQ: chg
TYPE: 2002p2
TN   96 0 0 2
ECHG yes
ITEM CLS DDGA
…
```

5.7.3. Enable Call Forward for Phone

This section shows how to configure the Call Forward feature at the system and phone level.

Select **Customer → 00 → Call Redirection**. The Call Redirection page is shown in **Figure 46**.
- **Total redirection count limit**: **0** (unlimited)
- **Call Forward: Originating**
- **Number of normal ring cycle for CFNA: 3**
- Click **Save** to save the configuration.

Figure 46 – Call Redirection

To enable **Call Forward All Call** (**CFAC**) for a phone over a trunk, use **LD 11**. Change its **CLS** to **CFXA**, and **SFA,** then program the forward number on the phone set. The following is the configuration of a phone that has CFAC enabled with forwarding number 616139675205.

```
ld 11
REQ: chg
TYPE: 2007
TN   96 0 0 4

ECHG yes
ITEM CLS CFXA SFA
ITEM key 19 CFW 16 616139675205
```

To enable **Call Forward Busy (CFB)** for phone over trunk by using **LD 11**, change its **CLS** to **FBA**, **HTA**, and **SFA,** then program the forward number as **HUNT** and **FDN**. Following is the configuration of a phone has **CFB** enabled with forward number is 616139675205.

```
ld 11
REQ: chg
TYPE: 2007
TN   96 0 0 4
ECHG yes
ITEM CLS FBA HTA SFA
ITEM HUNT 616139675205
ITEM FDN 616139675205
```

To enable **Call Forward No Answer (CFNA)** for a phone over a trunk by using **LD 11**, change its **CLS** to **FNA,** and **SFA,** then program the forward number as **HUNT** and **FDN**. Following is the configuration of a phone that has CFNA enabled with forward number 616139675205.

```
ld 11
REQ: chg
TYPE: 2007
TN   96 0 0 4
ECHG yes
ITEM CLS FNA SFA
ITEM HUNT 616139675205
ITEM FDN 616139675205
```

5.7.4. Enable Call Waiting for Phone

This section shows how to configure the Call Waiting feature at the phone level.

Log in to the Call Server CLI (please refer to **Section 5.1.2** for more details), configure Call Waiting feature for phone by using **LD 11** to change **CLS** to **HTD**, and **SWA** and adding a **CWT** key.

```
ld 11
REQ: chg
TYPE: 2002p2
TN   96 0 0 2
ECHG yes
ITEM CLS HTD SWA
ITEM key 2 CWT
```

6. Configure Avaya Aura® Session Manager

This section provides the procedures for configuring Session Manager. The procedures include configuring the following items:

- SIP Domain.
- Logical/physical Location that can be occupied by SIP Entities.
- Adaptation module to perform dial plan manipulation.
- SIP Entities corresponding to Communication Server 1000, Avaya SBCE and Session Manager.
- Entity Links, which define the SIP trunk parameters used by Session Manager when routing calls to/from SIP Entities.
- Routing Policies, which define route destinations and control call routing between the SIP Entities.
- Dial Patterns, which specify dialed digits and govern which Routing Policy is used to service a call.

It may not be necessary to create all the items above when configuring a connection to the service provider 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. Avaya Aura® System Manager Login and Navigation

Session Manager configuration is accomplished by accessing the browser-based GUI of System Manager, using the URL "https://<ip-address>/SMGR", where "<ip-address>" is the IP address of System Manager. At the **System Manager Log On** screen, enter an appropriate **User ID** and **Password** and press the **Log On** button (not shown). The initial screen shown below is then displayed.
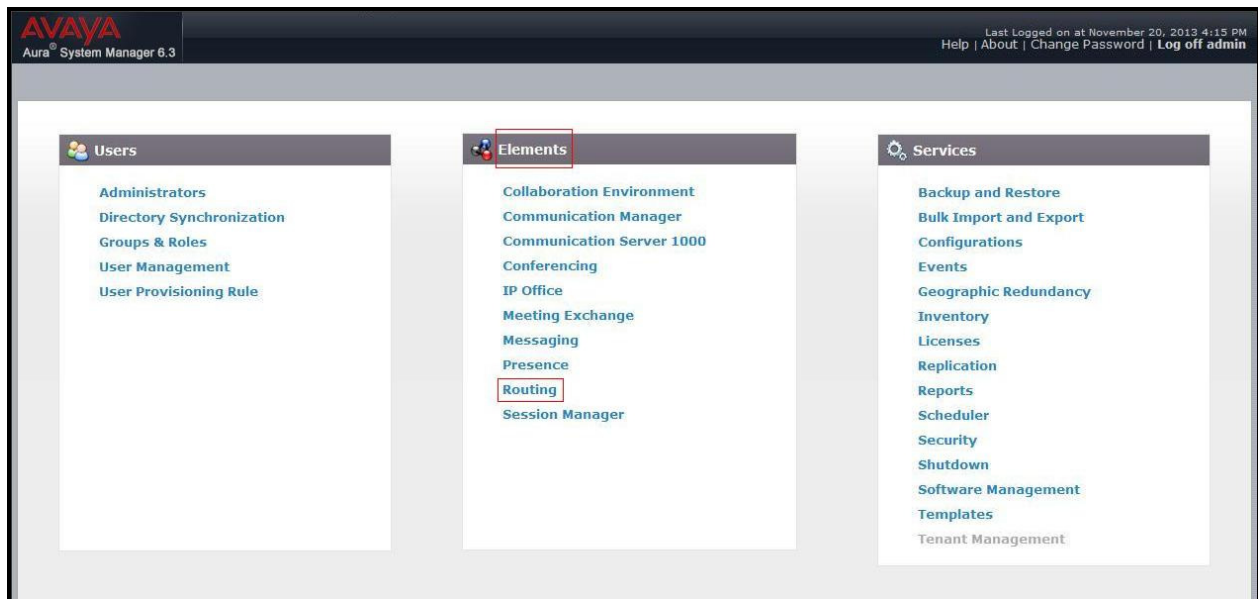
Figure 47 – System Manager Home Screen

Most of the configuration items are performed in the Routing Element. Click on **Routing** in the **Elements** column to bring up the **Introduction to Network Routing Policy** screen.

The navigation tree displayed in the left pane will be referenced in subsequent sections to navigate to items requiring configuration.
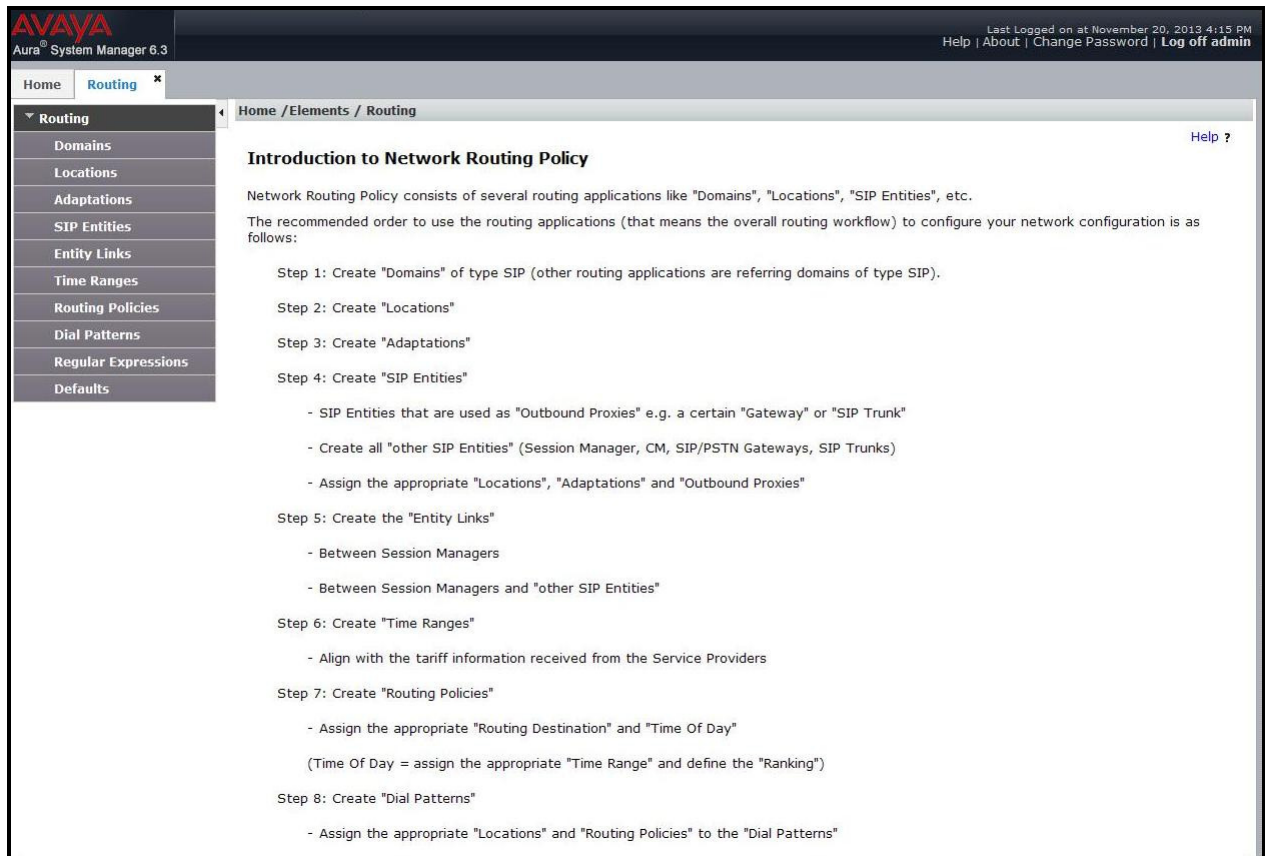
Figure 48 – Network Routing Policy

6.2. Specify SIP Domain

Create a SIP Domain for each domain of which Session Manager will need to be aware in order to route calls. For the compliance test, this includes the enterprise domain **bvwdev7.com**. Navigate to **Routing → Domains** in the left-hand navigation pane and click the **New** button in the right pane. In the new right pane that appears (not shown), fill in the following:

- **Name:** Enter the domain name.
- **Type:** Select **sip** from the pull-down menu.
- **Notes:** Add a brief description (optional).

Click **Commit** (not shown) to save.

The screen below shows the existing entry for the enterprise domain.
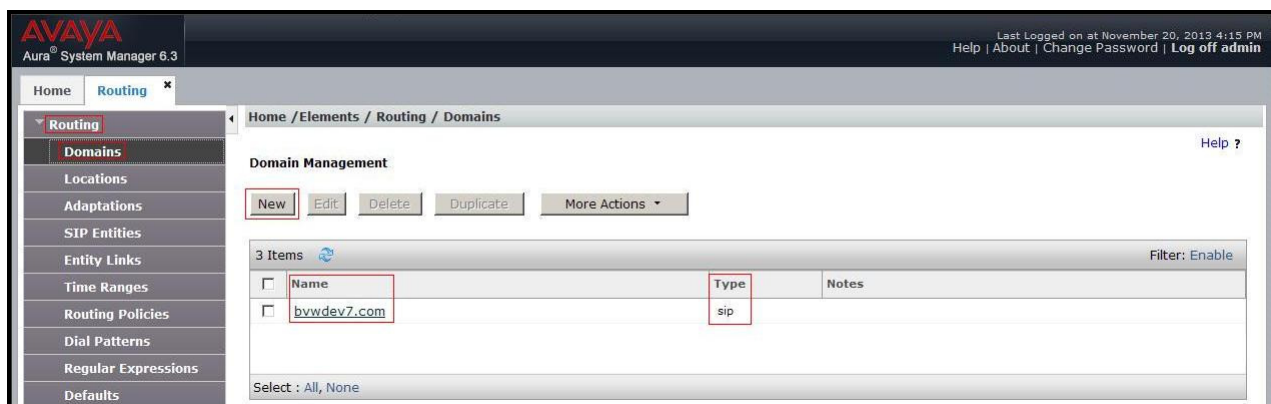
Figure 49 – Domain Management

6.3. Add Location

Locations can be used to identify logical and/or physical locations where SIP Entities reside for purposes of bandwidth management and call admission control. A single Location was defined for the enterprise even though multiple subnets were used. The screens below show the addition of the Location named **Belleville**, which includes all equipment in the enterprise including Communication Server 1000, Session Manager and Avaya SBCE.

To add a Location, navigate to **Routing → Locations** in the left-hand navigation pane and click the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

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

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Figure 50 – Location Configuration

In the **Location Pattern** section, click **Add** to enter IP Address patterns. The following patterns were used in testing:

- **IP Address Pattern:** 10.33.*, 10.10.97.*, 10.10.98.*

Figure 51 – IP Ranges Configuration

Click **Commit** to save.

Note that call bandwidth management parameters should be set per customer requirement.

6.4. Configure Adaptations

Adaptation is configured to format the History Info on Communication Server 1000 to be compatible with Avaya History Info form. In order to add a new adaptation, select **Routing →** **Adaptations**. Click the **New** button to add an adaptation. Enter an appropriate **Adaptation Name** to identify the adaptation. Select **CS1000Adapter** from the **Module Name** drop-down menu. Select **Name-Value Parameter** from the **Module Parameter Type** drop-down menu. Click **Add** button to add **Name** as **fromto** and **Value** as **true**. Click the **Commit** button after changes are completed.

Figure 52 - CS1000 Adaptation

Adaptation is configured to convert the History Info to Diversion Header and to remove MINE. In order to add a new adaptation, select **Routing → Adaptations**. Click the **New** button to add an adaptation. Enter an appropriate **Adaptation Name** to identify the adaptation. Select **DiversionTypeAdapter** from the **Module Name** drop-down menu. Select **Name-Value Parameter** from the **Module Parameter Type** drop-down menu. Click **Add** button to add **Name** as **MINE** and **Value** as **no**. Click the **Commit** button after changes are completed.

Figure 53 – Diversion Header Adaptation

6.5. Add SIP Entities

A SIP Entity must be added for Session Manager and for each SIP telephony system connected to Session Manager which includes Communication Server 1000 and Avaya SBCE. Navigate to **Routing → SIP Entities** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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

- **Name:** Enter a descriptive name.
- **FQDN or IP Address:** Enter the FQDN or IP address of the SIP Entity that is used for SIP signaling.
- **Type:** Select **Session Manager** for Session Manager, **Other** for Communications Server 1000 and the Avaya SBCE.
- **Adaptation:** This field is only present if **Type** is not set to **Session Manager**. If applicable, select the appropriate Adaptation module that will be applied to the SIP Entity being created.
- **Location:** Select the Location that applies to the SIP Entity being created. For the compliance test, all components were located in Location **Belleville**.
- **Time Zone:** Select the time zone for the Location above.

In this configuration, there are three SIP Entities.
- Session Manager SIP Entity
- Communication Manager 1000 SIP Entity
- Session Border Controller for Enterprise SIP Entity

6.5.1. Configure Session Manager SIP Entity

The following screen shows the addition of the Session Manager SIP Entity named **SM63**. The IP address of Session Manager's signaling interface is entered for **FQDN or IP Address 10.33.10.26**. The **Location** field is set to **Belleville**. Select **Time Zone** as **America/Toronto**.

Figure 54 – Session Manager SIP Entity

To define the ports used by Session Manager, scroll down to the **Port** section of the **SIP Entity Details** screen. This section is only present for the **Session Manager** SIP Entity.

In the **Port** section, click **Add** and enter the following values. Use default values for all remaining fields:

- **Port:**   Port number on which Session Manager listens for SIP requests.
- **Protocol:**   Transport protocol to be used with this port.
- **Default Domain:**   The default domain associated with this port. For the compliance test, this was the enterprise SIP Domain.

Defaults can be used for the remaining fields. Click **Commit** button (not shown) to save.

The compliance test used port **5060** with **UDP** for connecting to Communication Server 1000 and Avaya SBCE.

Other entries defined for other projects as shown in the screen were not used.

Figure 55 – Session Manager SIP Entity Port

6.5.2. Configure Communication Server 1000 SIP Entity

The following screen shows the addition of the Communication Server 1000 SIP Entity named **car3-ssg-carrier**. In order for Session Manager to send SIP service provider traffic on a separate Entity Link to Communication Server 1000, it is necessary to create a separate SIP Entity for Communication Server 1000, in addition to the one created at Session Manager installation, for use with all other SIP traffic within the enterprise. The **FQDN or IP Address** field is set to the IP address of Communication Server 1000 signaling Node **10.10.97.178**. Select **Type** as **Other**. Select **Adaptation** as **CS1K76_Adaptation** (in **Section 6.4**). The **Location** field is set to **Belleville** which is the location that includes the subnet where Communication Server 1000 resides. Select **Time Zone** as **America/Toronto**.

Figure 56 – Communication Server 1000 SIP Entity

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6.5.3. Configure Avaya SBCE SIP Entity

The following screen shows the addition of Avaya SBCE SIP entity named **SBCE**. The **FQDN or IP Address** field is set to the IP address of the SBC's private network interface **10.10.98.13**. Select **Type** as **Other**. Select **Adaptation** as **Diversion-Type-Remove-MINE** (in **Section 6.4**). The **Location** field is set to **Belleville** which includes the subnet where the Avaya SBCE resides. Select **Time Zone** as **America/Toronto**.

Figure 57 – Avaya SBCE SIP Entity

6.6. Add Entity Links

A SIP trunk between Session Manager and a telephony system is described by an Entity Link. Two Entity Links were created: one to Communication Server 1000 and one to the Avaya SBCE.

To add an Entity Link, navigate to **Routing → Entity Links** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

- **Name:**   Enter a descriptive name.
- **SIP Entity 1:** Select the Session Manager being used.
- **Protocol:**   Select the transport protocol used for this link.
- **Port:**   Port number on which Session Manager will receive SIP requests from the far-end.
- **SIP Entity 2:** Select the name of the other system as defined in **Section 6.5**.
- **Port:**   Port number on which the other system receives SIP requests from the Session Manager.
- **Trusted:**   Check this box. Note: If this box is not checked, calls from the associated SIP Entity specified in **Section 6.5** will be denied.

Click **Commit** to save.

The following screens illustrate the Entity Link to the Communication Server 1000. The protocol and ports defined here must match the values used for the Communication Server 1000 signaling in **Section 5.5.2**.

Figure 58 – Communication Server 1000 Entity Link

The following screens illustrate the Entity Links to the Avaya SBCE. The protocol and ports defined here must match the values used for the Avaya SBCE mentioned in **Section 7.2.7**

Figure 59 – Avaya SBCE Entity Link

6.7. Configure Time Ranges

Time Ranges are configured for time-based-routing. In order to add Time Ranges, select
Routing → **Time Ranges** and then click **New** button. The Routing Policies shown subsequently
will use the **24/7** range since time-based routing was not the focus of these Application Notes.
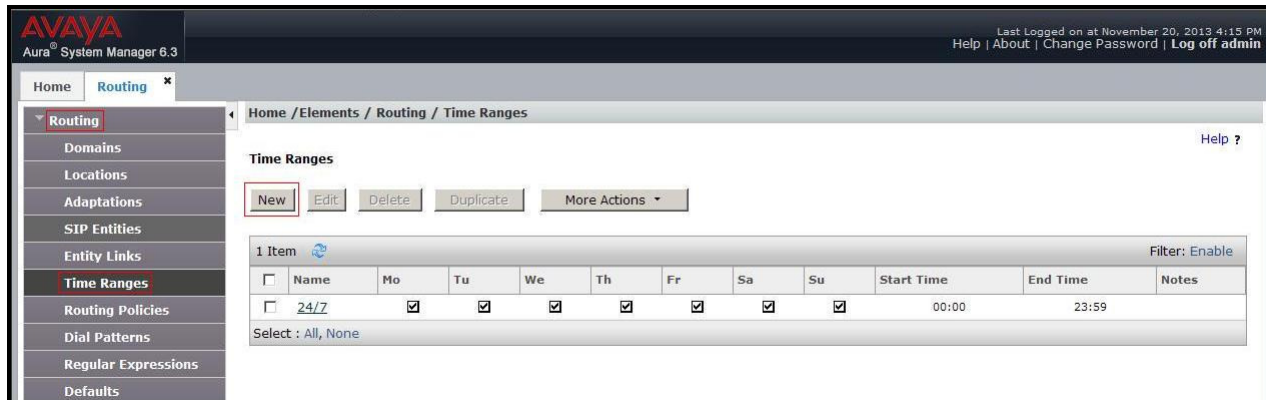
Figure 60 – Time Ranges

6.8. Add Routing Policies

Routing Policies describe the conditions under which calls will be routed to the SIP Entities
specified in **Section 6.5**. Two Routing Policies must be added: one for the Communication
Server 1000 and one for the Avaya SBCE. 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). In the new right pane that appears (shown below), 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 remaining fields. Click **Commit** to save.

The following screen shows the **Routing Policy Details** for the policy named **To car3-ssg-
carrier** associated with incoming PSTN calls from Vocal IP Networx to the Communication
Server 1000. Observe the **SIP Entity as Destination** is the entity named **car3-ssg-carrier**.

Figure 61 – Routing to Communication Server 1000

The following screen shows the **Routing Policy Details** for the policy named **To_Vocal.** This is associated with outgoing calls from the Communication Server 1000 to the PSTN via Vocal IP Networx, through the Avaya SBCE. Observe the **SIP Entity as Destination** is the entity named **SBCE**.

Figure 62 – Routing to Vocal IP Networx

6.9. Add Dial Patterns

Dial Patterns are used to route calls through Session Manager. For the compliance test, Dial Patterns were configured to route calls from the Communication Server 1000 to Vocal IP Networx SIP Trunk service and vice versa. Dial Patterns define which Route Policy will be selected as route destination for a particular call based on the dialed digits, destination Domain and originating Location.

To add a Dial Pattern, navigate to **Routing → Dial Patterns** in the left-hand navigation pane and click on the **New** button in the right pane (not shown). In the new right pane that appears (shown below), fill in the following:

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 all calls that match the specified criteria. Click **Select**.

Default values can be used for the remaining fields. Click **Commit** to save.

Two examples of the Dial Patterns used for the compliance test are shown below, one for outbound calls from the enterprise to the PSTN and one for inbound calls from the PSTN to the enterprise. Other Dial Patterns (e.g., 1800 Toll free call, 411, etc.) were similarly defined.

The first example shows that outbound 11-digit dialed numbers that begin with **1613** and have a destination SIP Domain of **bvwdev7.com** uses the Routing Policy Name **To_Vocal** as defined in **Section 6.8**.

Figure 63 – Dial Pattern_1613

Note that the above Dial Pattern did not restrict outbound calls to specific US area codes. In real deployments, appropriate restriction can be exercised per customer business policies.

The second example shows that inbound 10-digit numbers that start with **917** uses Routing Policy Name **To car3-ssg-carrier** as defined in **Section 6.8**. This Dial Pattern matches the DID numbers assigned to the enterprise by Vocal IP Networx.

Figure 64 – Dial Pattern_917

The following screen illustrates a list of dial patterns used for inbound and outbound calls between the enterprise and the PSTN.

Figure 65 – Dial Pattern List

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7. Configure Session Border Controller for Enterprise

This section describes the configuration of the Avaya SBCE necessary for interoperability with the Session Manager and Vocal IP Networx SIP Trunk service.

Avaya elements reside on the Private side and the Vocal IP Networx SIP Trunk service reside on the Public side of the network, as illustrated in **Figure 1**,

Note: The following section assumes that Avaya SBCE has been installed and that network connectivity exists between the systems. For more information on Avaya SBCE, see **Section 11** of these Application Notes.

7.1. Log into the Avaya SBCE

Access the web interface by typing "**https://x.x.x.x/sbc/**" (where x.x.x.x is the management IP of the Avaya SBCE).

Enter the **Username** and **Password**.

Figure 66 - Avaya SBCE Login

7.2. Global Profiles

When selected, Global Profiles allows for configuration of parameters across all Avaya SBCE appliances.

7.2.1. Configure Server Interworking - Avaya site

Server Interworking allows one to configure and manage various SIP call server-specific capabilities such as call hold, 180 handling, etc.

From the menu on the left-hand side, select **Global Profiles** → **Server Interworking** → **Add**
- Enter Profile name**: SM63**
- All options on the **General** Tab can be left at default.

On the **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** Tabs: All options can be left at default. Click **Finish** (not shown).

The following screen is shown that Session Manager server interworking (named: **SM63**) was added.

Figure 67 - Server Interworking – Avaya site

7.2.2. Configure Server Interworking – Vocal IP Networx site

From the menu on the left-hand side, select **Global Profiles** → **Server Interworking** → **Add**
- Enter Profile name**: Vocal**
- All options on the **General** Tab can be left at default.

On the **Timers**, **URI Manipulation**, **Header Manipulation** and **Advanced** Tabs: All options can be left at default. Click **Finish** (not shown).

The following screen is shown that Vocal IP Networx server interworking (named: **Vocal**) was added.
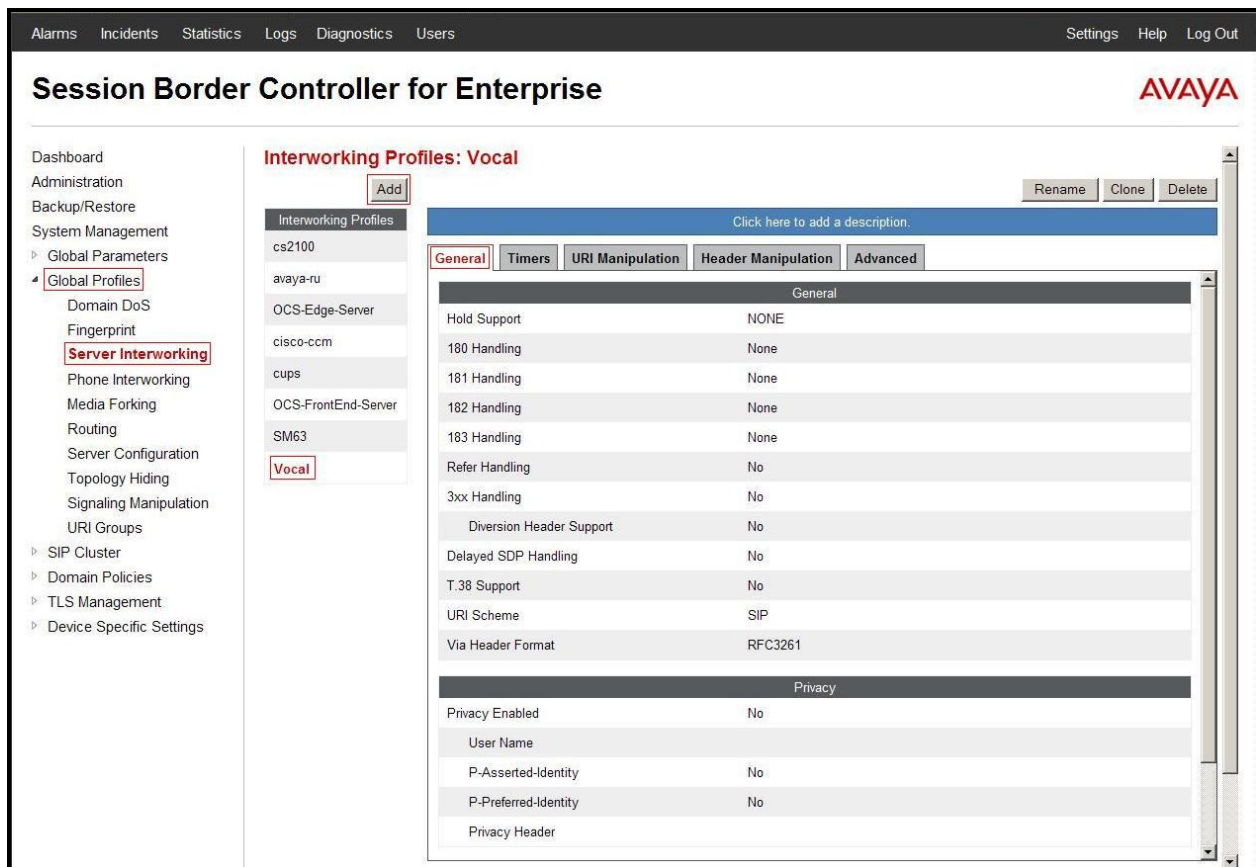
Figure 68 - Server Interworking – Vocal IP Networx site

7.2.3. Configure URI Groups

The URI Group feature allows administrator to create any number of logical URI groups that are comprised of individual SIP subscribers located in that particular domain or group.
The following URI Group configuration is used for this specific testing in DevConnect Lab environment. The URI-Group named **Vocal** was used to match the "From" and "To" headers in a SIP call dialog received from both Enterprise and Vocal IP Networx SIP Trunk service. If there is a match, the Avaya SBCE will apply the appropriate Routing profiles (see **Section 7.2.4, 7.2.5**), Server Flows (see **Section 7.4.4**), and Session Flow (see **section 7.4.5**) to route incoming and outgoing calls to the right destinations. In production environment, there is not a requirement to define this URI.

From the menu on the left-hand side, select **Global Profiles → URI Groups**. Select **Add.**
- Enter Group Name: **Vocal**.
- Edit the URI Type: **Regular Expression** (not shown).
- **Add** URI: **.*10\.10\.98\.111** (Avaya SBCE public interface IP address), **.*10\.10\.98\.13** (Avaya SBCE internal interface IP address), **.*10\.10\.33\.26** (Session Manager IP address), **.*192\.168\.7\.50** (Vocal IP Networx SIP Signaling server IP address**), .*nonymous\.invalid** (Anonymous URI), **.*bvwdev7\.com** (Enterprise domain).
- Click **Finish** (not shown))
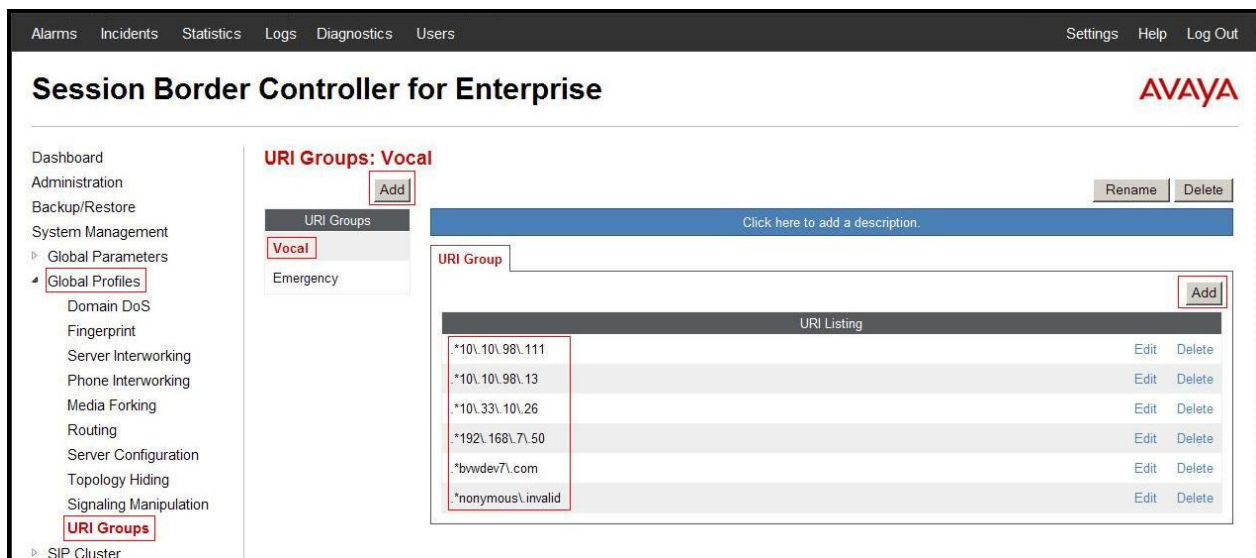
Figure 69 - URI Group

7.2.4. Configure Routing – Avaya site

Routing profiles define a specific set of packet routing criteria that are used in conjunction with other types of domain policies to identify a particular call flow and thereby ascertain which security features will be applied to those packets. Parameters defined by Routing Profiles include packet transport settings, name server addresses and resolution methods, next hop routing information, and packet transport types.

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From the menu on the left-hand side, select **Global Profiles → Routing → Add**
Enter Profile Name: **Vocal_To_SM63**

- **URI Group**: **Vocal**
- **Next Hop Server 1**: **10.33.10.26:5060** (Session Manager IP address)
- Check **Routing Priority based on Next Hop Server** (not shown)
- **Outgoing Transport**: **UDP** (not shown)
- Click **Finish** (not shown).
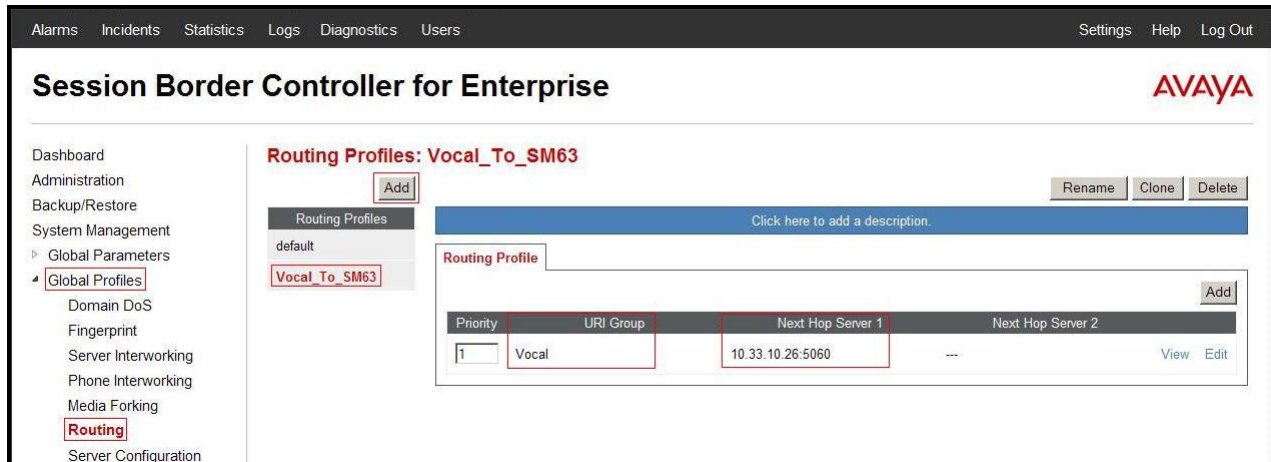
Figure 70 - Routing to Avaya

7.2.5. Configure Routing – Vocal IP Networx site

The Routing Profile allows one to manage parameters related to routing SIP signaling messages.

From the menu on the left-hand side, select **Global Profiles → Routing →Add**
Enter Profile Name: **SM63_To_Vocal**

- **URI Group**: **Vocal**
- **Next Hop Server 1**: **192.168.7.50:5060** (Vocal IP Networx SIP Signaling server IP address)
- Check **Routing Priority based on Next Hop Server** (not shown)
- **Outgoing Transport**: **UDP** (not shown)
- Click **Finish** (not shown).
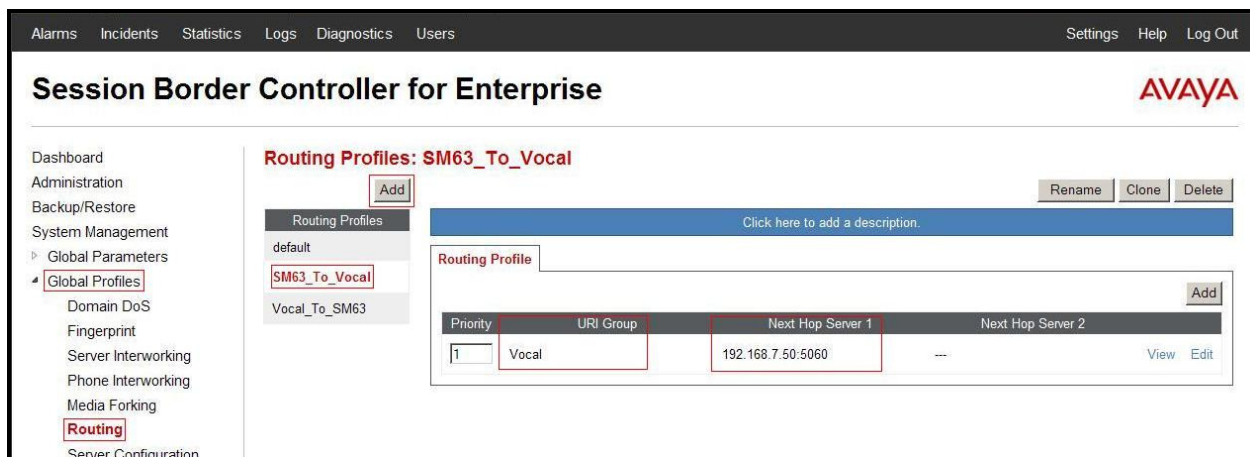
Figure 71 - Routing to Vocal IP Networx

7.2.6. Configure Signaling Manipulation

The Avaya's SIP signaling header manipulation feature is use for the Avaya SBCE product. This feature adds the ability to add, change and delete any of the headers and other information in a SIP message

- Select **Global Profiles** from the menu on the left-hand side
- Select the **Signaling Manipulation**
- Select **Add**. Enter script Title: **Vocal**
  - Edit the script to correct the FROM header of SIP OPTIONS message
  - Edit the script to remove unwanted SIP headers
  - Click **Save** (not shown)
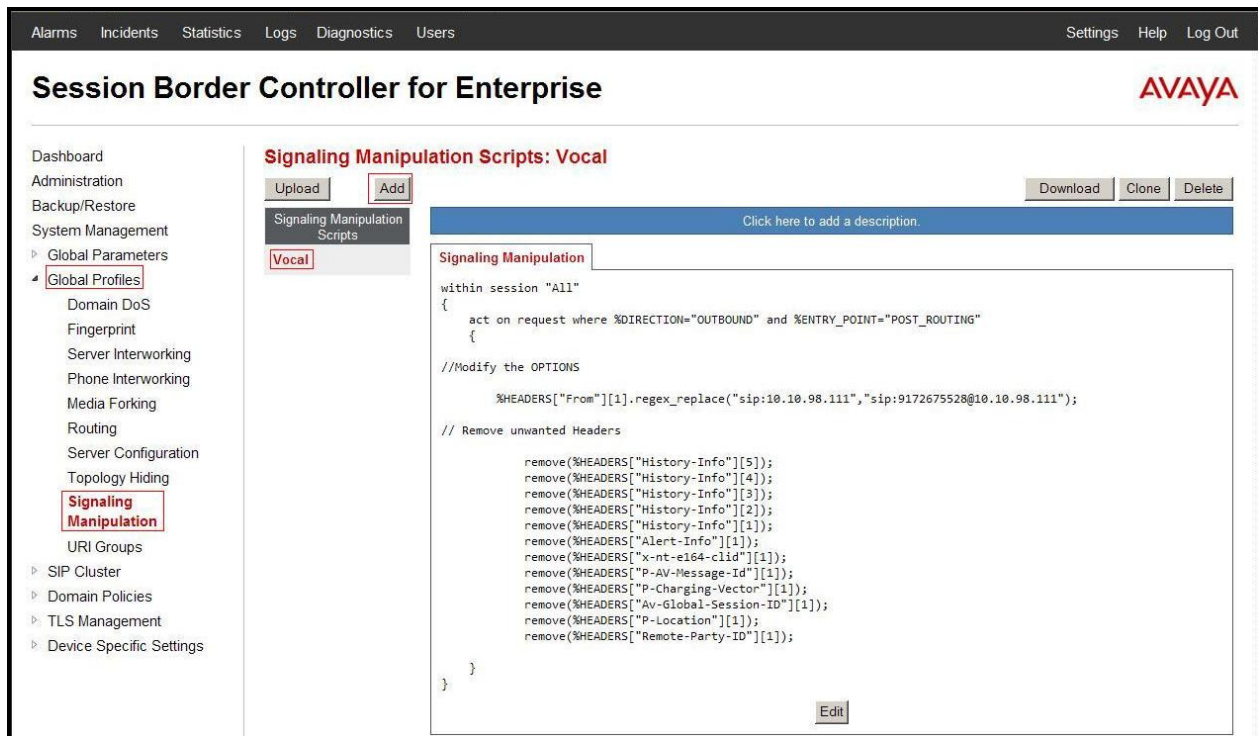
Figure 72 – Signaling Manipulation Vocal IP Networx

7.2.7. Configure Server – Session Manager

The **Server Configuration** screen contains four tabs: **General**, **Authentication**, **Heartbeat**, and **Advanced**. Together, these tabs allow one to configure and manage various SIP call server-specific parameters such as UDP port assignment, IP Server type, heartbeat signaling parameters and some advanced options.

From the menu on the left-hand side, select **Global Profiles → Server Configuration →Add**.

Enter profile name: **SM63**

On **General** tab, enter the following:
- **Server Type**: Select **Call Server**
- **IP Address/FQDNs**: **10.33.10.26** (Session Manager IP Address)
- **Supported Transports**: **UDP**
- **UDP Port**: **5060**
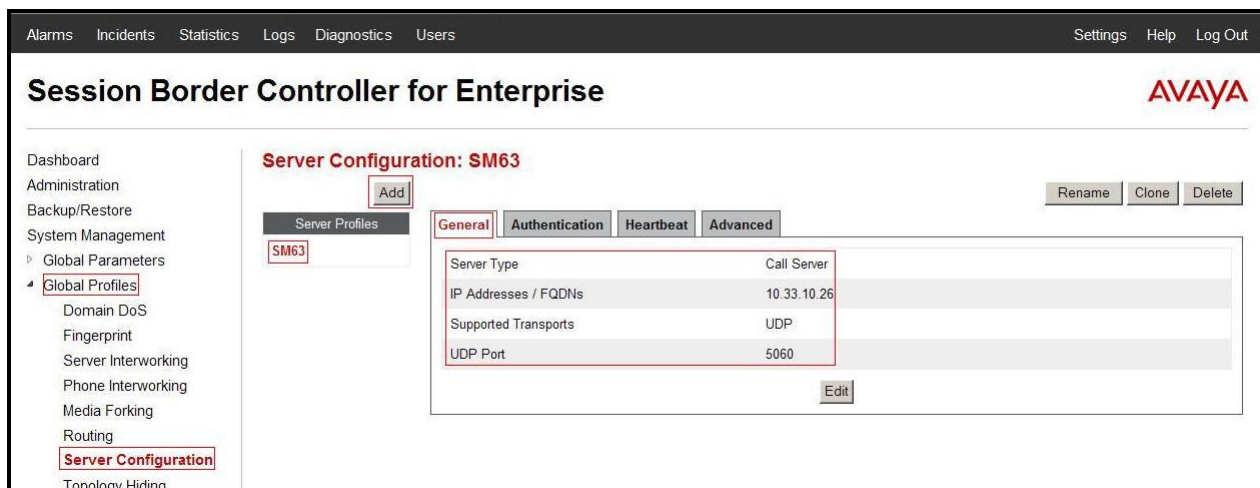
Figure 73 - Session Manager General Server Configuration

On the **Advanced** tab:
- Select **SM63** for **Interworking Profile.**

Click **Finish** (not shown).
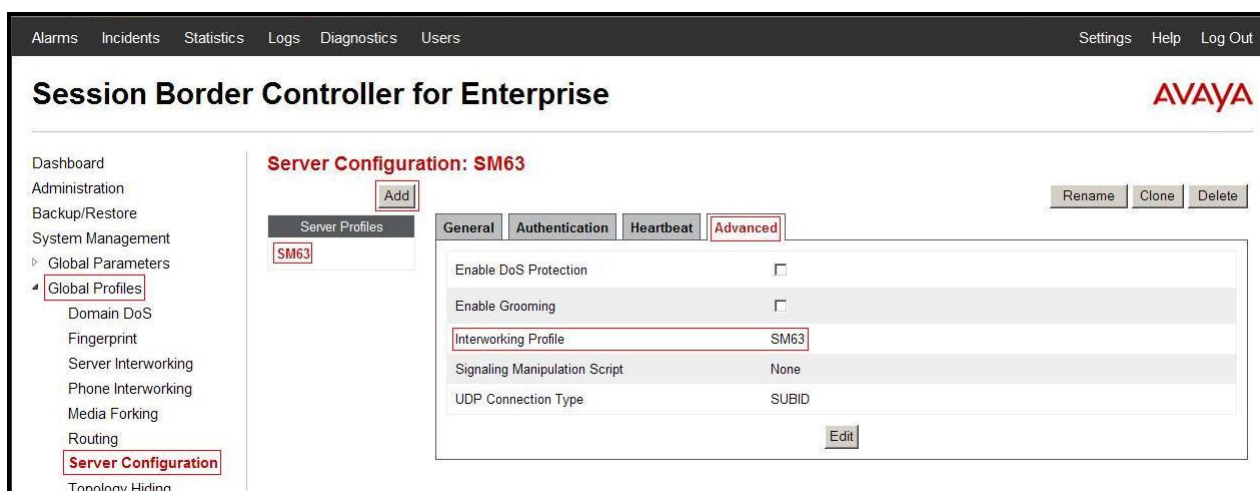
Figure 74 - Session Manager Advanced Server Configuration

7.2.8. Configure Server – Vocal IP Networx

From the menu on the left-hand side, select **Global Profiles** → **Server Configuration** → **Add**.

Enter profile name: **Vocal**

On **General** tab, enter the following:
- **Server Type:** Select **Trunk Server**
- **IP Address: 192.168.7.50** (Vocal IP Networx Signaling server IP Address)

- **Supported Transports**: **UDP**
- **UDP Port: 5060**

Figure 75 - Vocal IP Networx General Server Configuration

On the **Advanced** tab, enter the following:
- **Interworking Profile**: select **Vocal**
- **Signaling Manipulation Script**: select **Vocal**

Click **Finish** (not shown).

Figure 76 - Vocal IP Networx Advanced Server Configuration

7.2.9. Configure Topology Hiding – Avaya site

The Topology Hiding screen allows one to manage how various source, destination and routing information in SIP and SDP message headers are substituted or changed to maintain the integrity of the network. It hides the topology of the enterprise network from external networks

From the menu on the left-hand side, select **Global Profiles → Topology Hiding**.

Select **Add**, enter Profile Name: **Vocal_To_SM63**.
- For the Header **To,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
    In the **Overwrite Value** column**: bvwdev7.com**
- For the Header **Request-Line,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: bvwdev7.com**
- For the Header **From,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
    In the **Overwrite Value** column**: bvwdev7.com**

Click **Finish** (not shown).

Figure 77 - Topology Hiding Session Manager

7.2.10.   Configure Topology Hiding – Vocal IP Networx site

From the menu on the left-hand side, select **Global Profiles → Topology Hiding**.

Select **Add Profile**, enter Profile Name: **SM63_To_Vocal**.
- For the Header **To,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: 192.168.7.50**
- For the Header **Request-Line,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: 192.168.7.50**
- For the Header **From,**
  - In the **Criteria** column select **IP/Domain**
  - In the **Replace Action** column select**: Overwrite**
  - In the **Overwrite Value** column**: 10.10.98.111**

Click **Finish** (not shown).
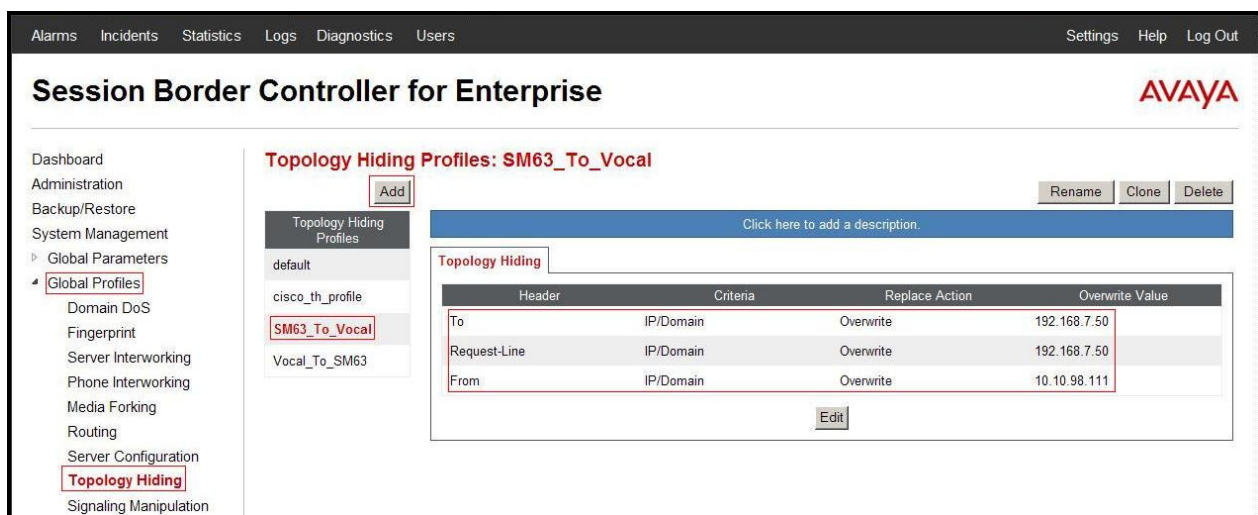
Figure 78 - Topology Hiding Vocal IP Networx

7.3. Domain Policies

The Domain Policies feature allows one to configure, apply, and manage various rule sets (policies) to control unified communications based upon various criteria of communication sessions originating from or terminating in the enterprise. These criteria can be used to trigger different policies which will apply on call flows, change the behavior of the call, and make sure the call does not violate any of the policies. There are default policies available to use, or one can create a custom domain policy.

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7.3.1. Create Application Rules

Application Rules allow one to define which types of SIP-based Unified Communications (UC) applications the Avaya SBCE security device will protect: voice, video, and/or Instant Messaging (IM). In addition, one can determine the maximum number of concurrent voice and video sessions so that the network will process to prevent resource exhaustion.

From the menu on the left-hand side, select **Domain Policies → Application Rules**.
- Select the **default** Rule
- Select **Clone** button
  - Name**: SM63_Vocal_AppR**
  - Click **Finish** (not shown).

Figure 79 - Session Manager Application Rule

From the menu on the left-hand side, select **Domain Policies → Application Rules**.
- Select the **default** Rule
- Select **Clone** button
  - Name**: Vocal_AppR**
  - Click **Finish** (not shown).
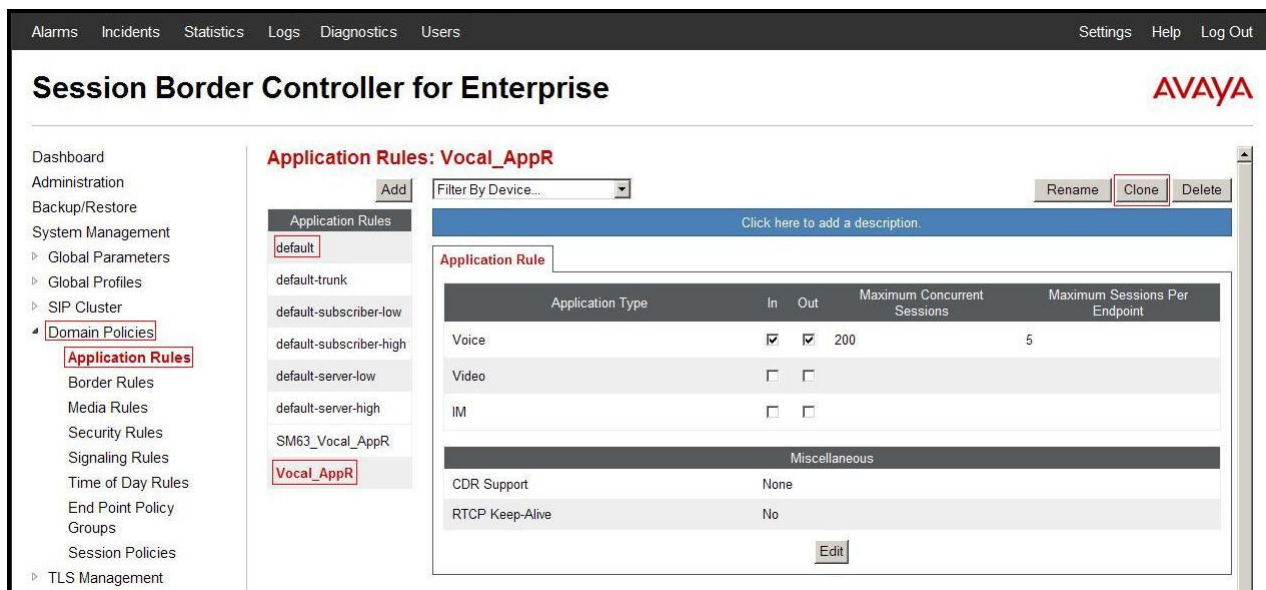
Figure 80 - Vocal IP Networx Application Rule

7.3.2. Create Border Rules

Border Rules allow one to control NAT Traversal. The NAT Traversal feature allows one to determine whether or not call flow through the DMZ needs to traverse a firewall and the manner in which pinholes will be kept open in the firewall to accommodate traffic.

From the menu on the left-hand side, select **Domain Policies** →**Border Rules**.
- Select the **default** Rule
- Select **Clone** button
    - Enter Clone Name: **SM63_Vocal_BorderR**
    - Click **Finish** (not shown).
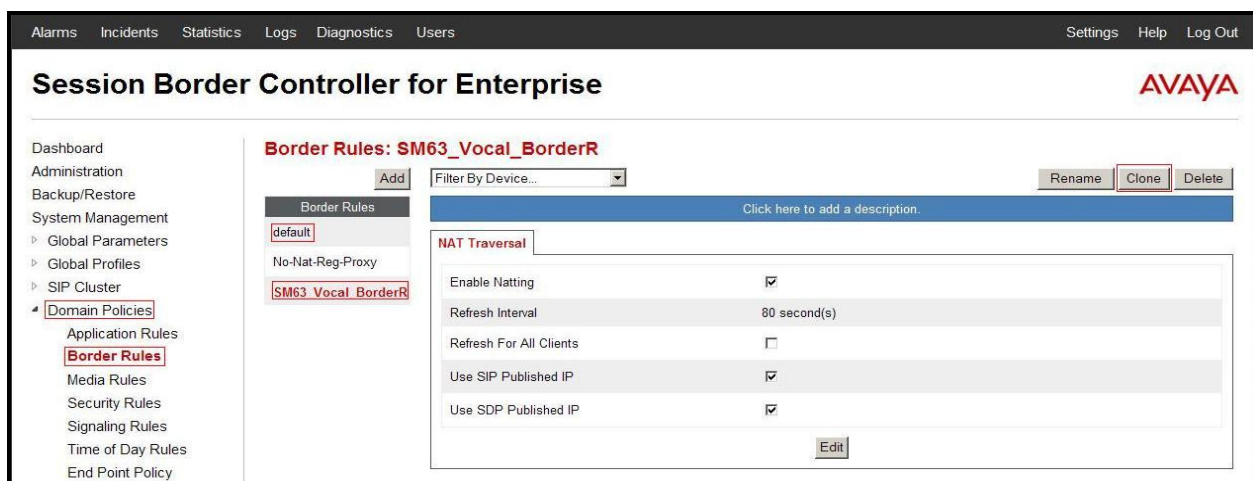
Figure 81 - Session Manager Border Rule

From the menu on the left-hand side, select **Domain Policies → Border Rules**.
- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name: **Vocal_BorderR**
  - Click **Finish** (not shown).
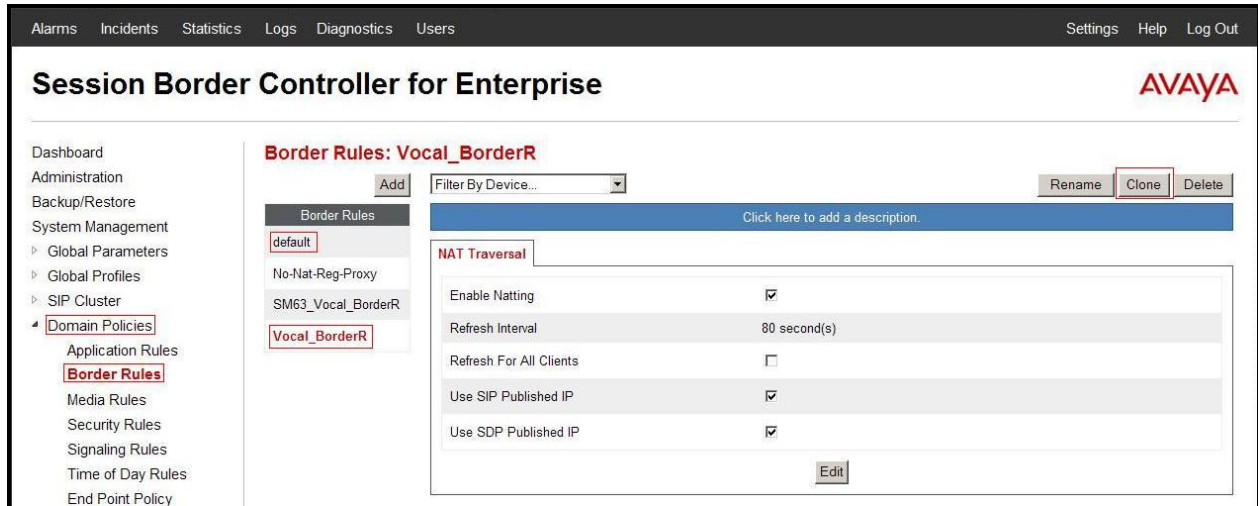
Figure 82 - Vocal IP Networx Border Rule

7.3.3. Create Media Rules

Media Rules allow one to define RTP media packet parameters such as prioritizing encryption techniques and packet encryption techniques. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criteria will be handled by the Avaya SBCE security product.

From the menu on the left-hand side, select **Domain Policies → Media Rules**.
- Select the **default-low-med** Rule
- Select **Clone** button
  - Enter Clone Name: **SM63_Vocal_MediaR**
  - Click **Finish** (not shown).

Figure 83 - Session Manager Media Rule

From the menu on the left-hand side, select **Domain Policies → Media Rules**.
- Select the **default-low-med** Rule
- Select **Clone** button
  - Enter Clone Name**: Vocal_MediaR**
  - Click **Finish** (not shown).
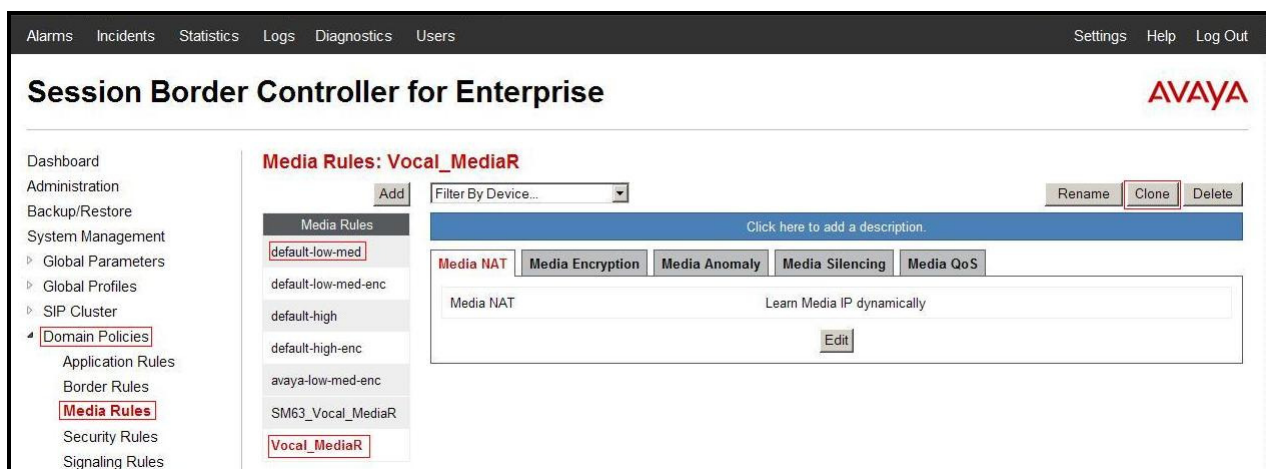
Figure 84 – Vocal IP Networx Media Rule

7.3.4. Create Security Rules

Security Rules allow one to define which enterprise-wide VoIP and Instant Message (IM) security features will be applied to a particular call flow. Security Rules allows one to configure Authentication, Compliance, Fingerprinting, Scrubber, and Domain DoS. In addition to determining which combination of security features are applied, one can also define the security feature profile, so that the feature is applied in a specific manner to a specific situation.
From the menu on the left-hand side, select **Domain Policies → Security Rules**.
- Select the **default-med** Rule
- Select **Clone** button

- Enter Clone Name: **SM63_Vocal_SecR**
- Click **Finish** (not shown).

Figure 85 - Session Manager Security Rule

From the menu on the left-hand side, select **Domain Policies → Security Rules**.
- Select the **default-med** Rule
- Select **Clone** button
    - Enter Clone Name: **Vocal_SecR**
    - Click **Finish** (not shown).

Figure 86 - Vocal IP Networx Security Rule

7.3.5. Create Signaling Rules

Signaling Rules allow one to define the action to be taken (Allow, Block, Block with Response, etc.) for each type of SIP-specific signaling request and response message. When SIP signaling packets are received by the Avaya SBCE, they are parsed and "pattern matched" against the particular signaling criteria defined by these rules. Packets matching the criteria defined by the Signaling Rules are tagged for further policy matching.

From the menu on the left-hand side, select **Domain Policies → Signaling Rules**.
- Select the **default** Rule
- Select **Clone** button
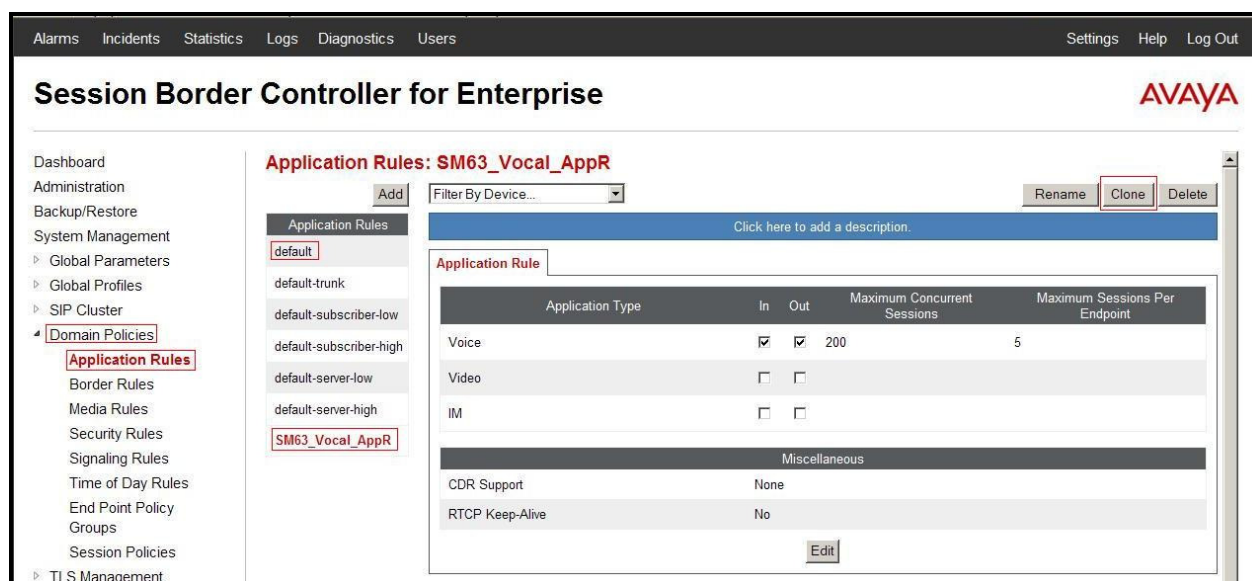 - Enter Clone Name**: SM63_Vocal_SigR**
  - Click **Finish** (not shown).
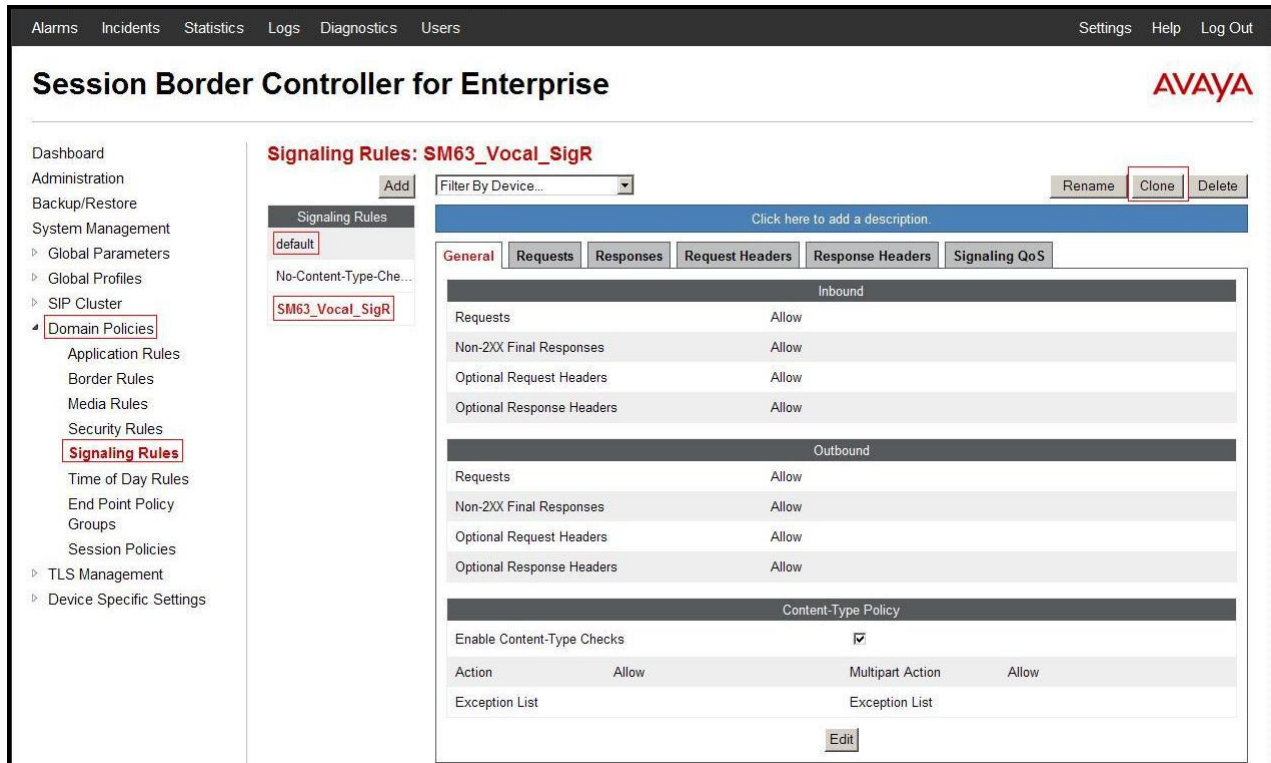
Figure 87 - Session Manager Signaling Rule

From the menu on the left-hand side, select **Domain Policies → Signaling Rules**.
- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name**: Vocal_SigR**
 - Click **Finish** (not shown).
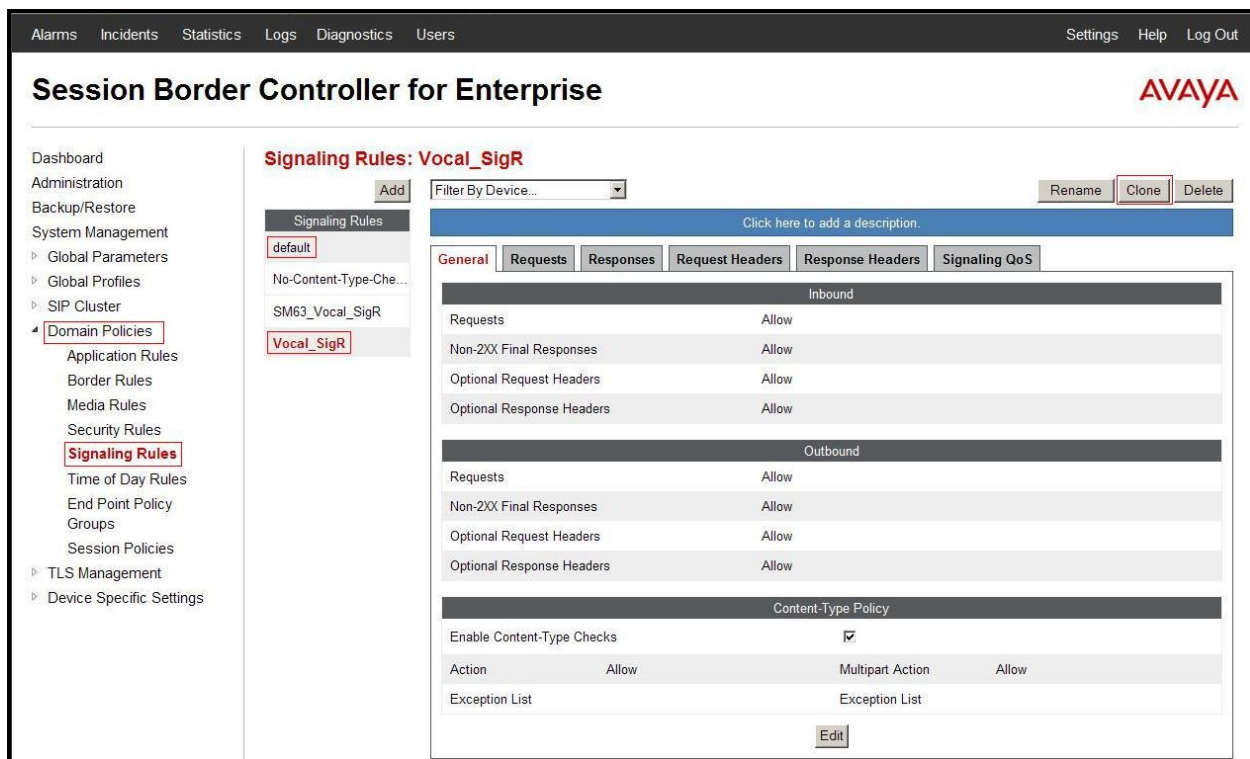
Figure 88 - Vocal IP Networx Signaling Rule

7.3.6. Create Time of Day Rules

A Time-of-day (ToD) Rule allows one to determine when the domain policy which is assigned to will be in effect. ToD Rules provide complete flexibility to fully accommodate the enterprise by, not only determining when a particular domain policy will be in effect, but also to whom it will apply, and for how long it will remain in effect.

From the menu on the left-hand side, select **Domain Policies → Time of Day Rules**.
- Select the **default** Rule
- Select **Clone** button
  - Enter Clone Name**: SM63_Vocal_ToDR**
  - Click **Finish** (not shown).

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Figure 89 - Session Manager Time of Day Rule

From the menu on the left-hand side, select **Domain Policies → Time of Day Rules**.
- Select the **default** Rule
- Select **Clone** button
 - Enter Clone Name**: Vocal_ToDR**
  - Click **Finish** (not shown).
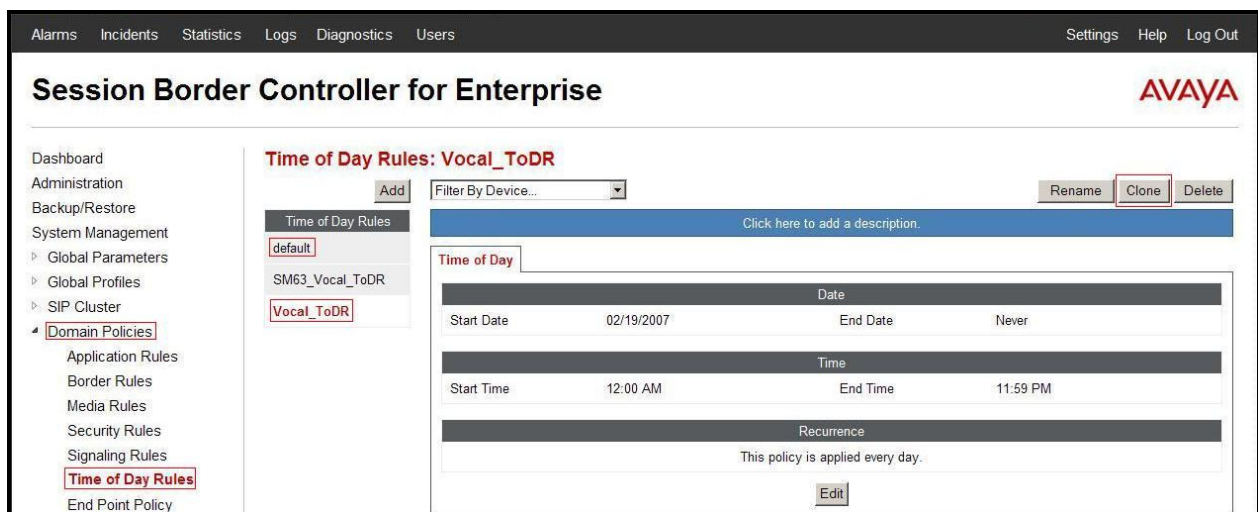
**Figure 90 - Vocal IP Networx Time of Day Rule**

7.3.7. Create Endpoint Policy Groups

The End-Point Policy Group feature allows one to create Policy Sets and Policy Groups. A Policy Set is an association of individual, SIP signaling-specific security policies (rule sets): application, border, media, security, signaling, and ToD, each of which was created using the procedures contained in the previous sections.) A Policy Group is comprised of one or more Policy Sets. The purpose of Policy Sets and Policy Groups is to increasingly aggregate and simplify the application of Avaya SBCE security features to very specific types of SIP signaling messages traversing through the enterprise.

From the menu on the left-hand side, select **Domain Policies → End Point Policy Groups**.
- Select **Add**
- Enter **Group Name: SM63_Vocal_PolicyG**
  - **Application Rule: SM63_Vocal_AppR**
  - **Border Rule: SM63_Vocal_BorderR**
  - **Media Rule: SM63_Vocal_MediaR**
  - **Security Rule: SM63_Vocal_SecR**
  - **Signaling Rule: SM63_Vocal_SigR**
  - **Time of Day: SM63_Vocal_ToDR**
- Select **Finish** (not shown).
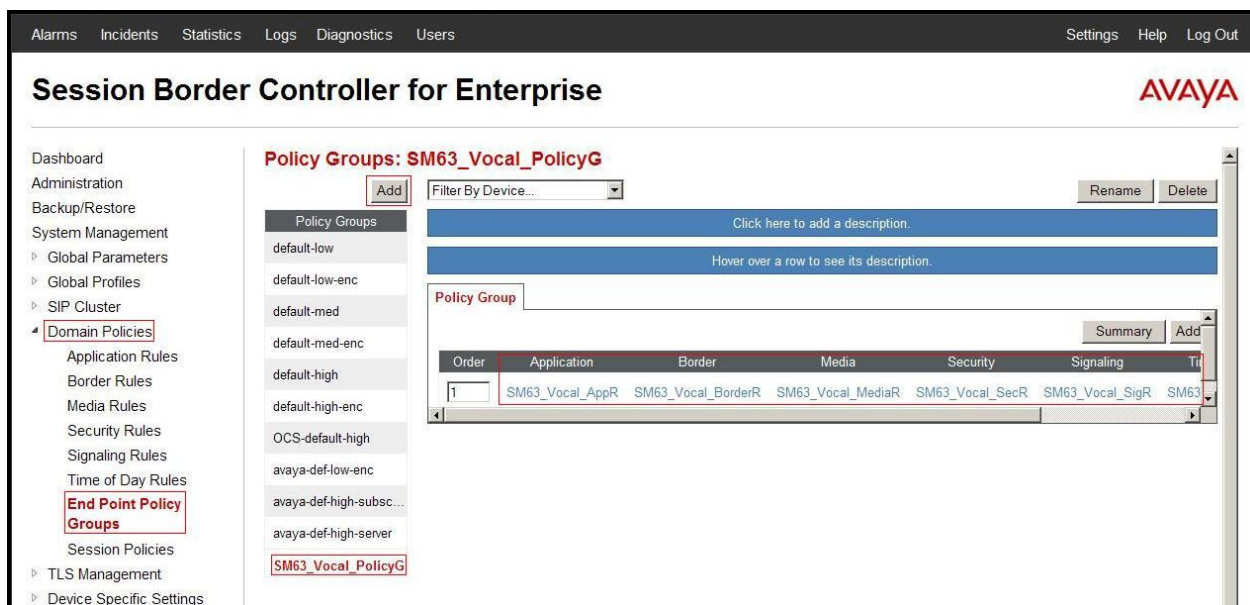
Figure 91 - Session Manager End Point Policy Group

From the menu on the left-hand side, select **Domain Policies → End Point Policy Groups**.
- Select **Add**
- Enter **Group Name: Vocal_PolicyG**
  - **Application Rule: Vocal_AppR**
  - **Border Rule: Vocal_BorderR**

- **Media Rule: Vocal_MediaR**
- **Security Rule: Vocal_SecR**
- **Signaling Rule: Vocal_SigR**
- **Time of Day: Vocal_ToDR**
- Select **Finish** (not shown).

Figure 92 - Vocal IP Networx End Point Policy Group

7.3.8. Create Session Policy

Session Policies allow users to define RTP media packet parameters such as codec types (both audio and video) and codec matching priority. Together these media-related parameters define a strict profile that is associated with other SIP-specific policies to determine how media packets matching these criterion will be handled by the Avaya SBCE security product.

- From the menu on the left-hand side, select **Domain Policies** → **Session Policies**.
- Select the **default** policy
- Select **Clone** button
    - Enter Clone Name**: Vocal**
    - Click **Finish** (not shown).
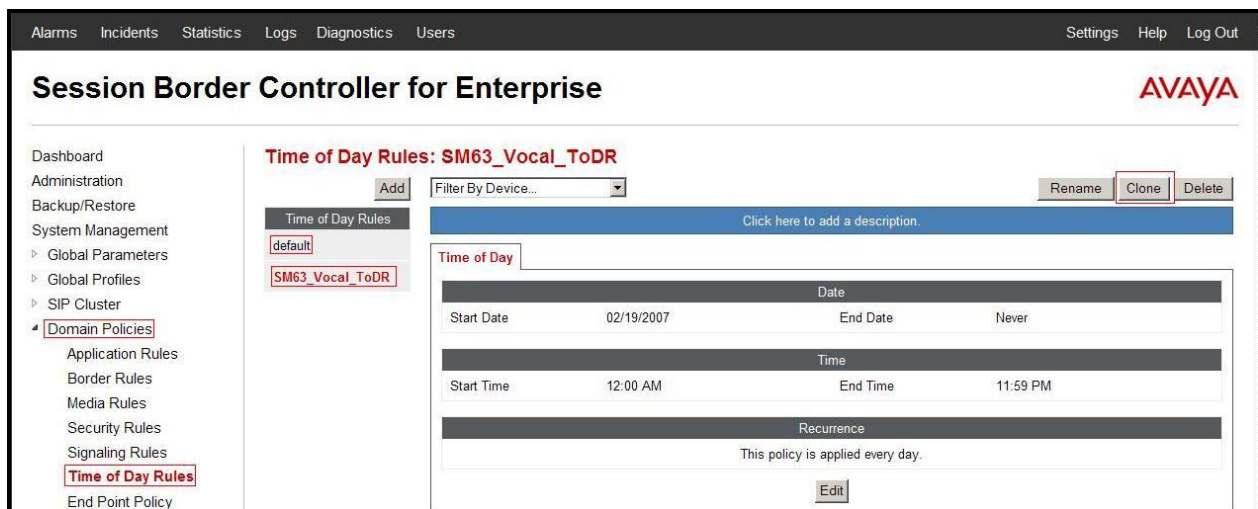- Click **Edit** button on **Media** tab
    - Check **Media Anchoring**
    - Select **Finish** (not shown)
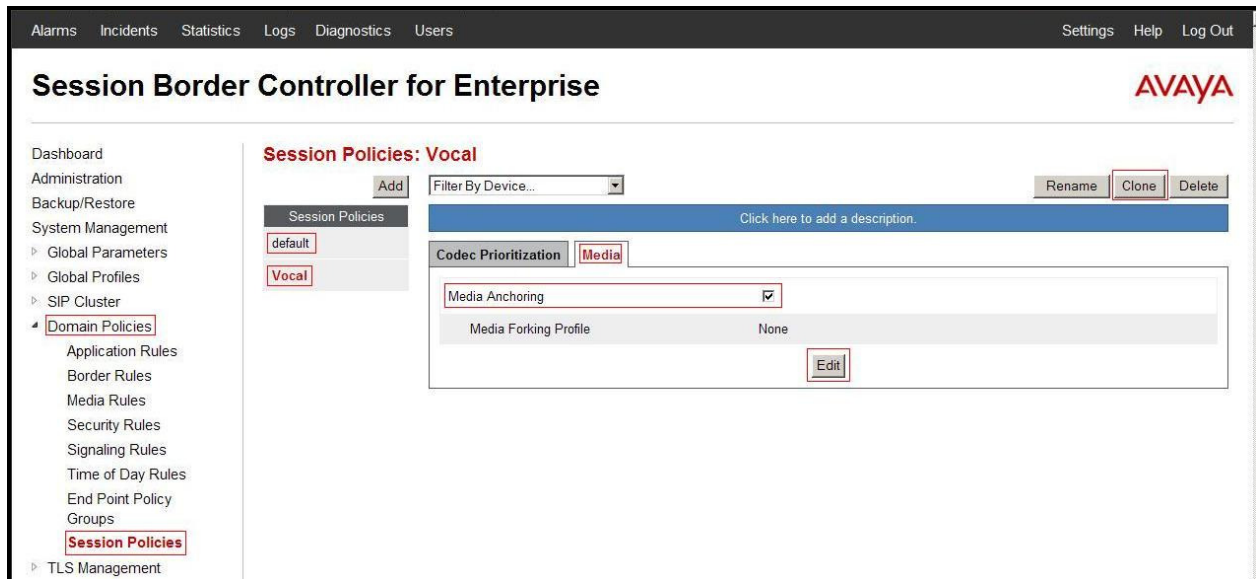
Figure 93 - Vocal IP Networx Session Policy – Anchoring Media

7.4. Device Specific Settings

The Device Specific Settings feature for SIP allows one to view aggregate system information, and manage various device-specific parameters which determine how a particular device will function when deployed in the network. Specifically, one has the ability to define and administer various device-specific protection features such as Message Sequence Analysis (MSA) functionality, end-point and session call flows and Network Management.

7.4.1. Manage Network Settings

From the menu on the left-hand side, select **Device Specific Settings → Network Management**.
- Enter the **IP Address** and **Gateway Address** for both the Inside and the Outside interfaces:
    - **IP Address** for Inside interface: **10.10.98.13**; **Gateway: 10.10.98.1**
    - **IP Address** for Outside interface: **10.10.98.111**; **Gateway: 10.10.98.97**
- Select the physical interface used in the Interface column:
    - **Inside Interface**: **A1**
    - **Outside Interface**: **B1**.

Figure 94 - Network Management

- Select the **Interface Configuration** Tab.
- Toggle the State of the physical interfaces being used to **Enabled**.
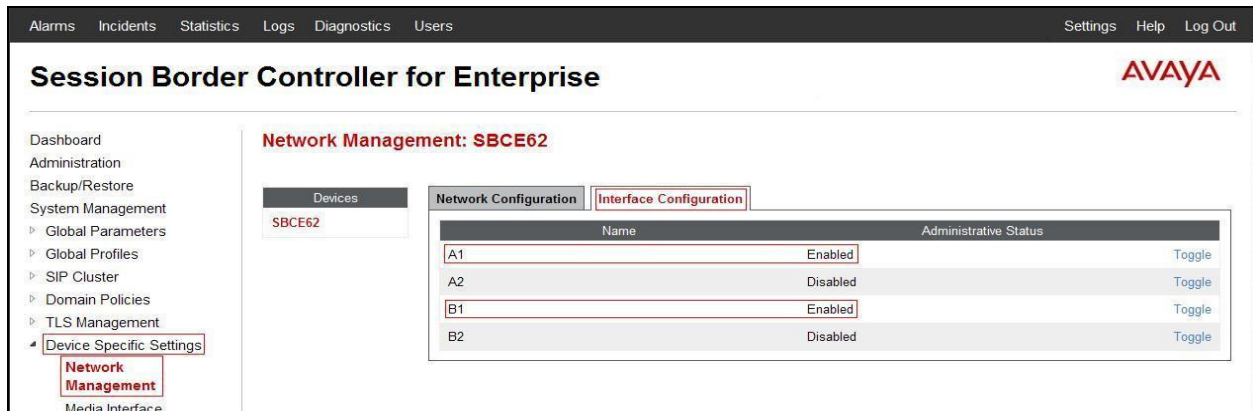
Figure 95 - Network Interface Status

7.4.2. Create Media Interfaces

Media Interfaces define the type of signaling on the ports. The default media port range on the Avaya can be used for both inside and outside ports.

From the menu on the left-hand side, **Device Specific Settings** → **Media Interface**.
- Select **Add**
  - **Name: InsideMedia**
  - **Media IP: 10.10.98.13** (Internal IP Address toward Session Manager)
  - **Port Range: 35000 - 40000**
  - Click **Finish** (not shown)
- Select **Add**
  - **Name: OutsideMedia**
  - **Media IP: 10.10.98.111** (External IP Address toward Vocal IP Networx trunk)
  - **Port Range: 35000 - 40000**
  - Click **Finish** (not shown).

Figure 96 - Media Interface

7.4.3. Create Signaling Interfaces

Signaling Interfaces define the type of signaling on the ports.

From the menu on the left-hand side, select **Device Specific Settings → Signaling Interface**.
- Select **Add**
  - **Name: InsideUDP**
  - **Media IP: 10.10.98.13** (Internal IP Address toward Session Manager)
  - **UDP Port: 5060**
  - Click **Finish** (not shown).

From the menu on the left-hand side, select **Device Specific Settings → Signaling Interface.**
- Select **Add**
  - **Name: OutsideUDP**
  - **Media IP: 10.10.98.111** (External IP Address toward Vocal IP Networx trunk)
  - **UDP Port: 5060**
  - Click **Finish** (not shown).

Figure 97 - Signaling Interface

7.4.4. Configuration Server Flows

Server Flows allow to categorize trunk-side signaling and to apply a policy.

7.4.4.1 Create End Point Flows – From Vocal IP Networx

From the menu on the left-hand side, select **Device Specific Settings → End Point Flows**.
- Select the **Server Flows** Tab
- Select **Add**, enter **Flow Name: From Vocal**
  - **Server Configuration**: **Vocal**
  - **URI Group: Vocal**
  - **Transport: ***
  - **Remote Subnet: ***
  - **Received Interface**: **InsideUDP**
  - **Signaling Interface: OutsideUDP**
  - **Media Interface**: **OutsideMedia**

- **End Point Policy Group: Vocal_PolicyG**
- **Routing Profile: Vocal_To_SM63**
- **Topology Hiding Profile: SM63_To_Vocal**
- Click **Finish**.
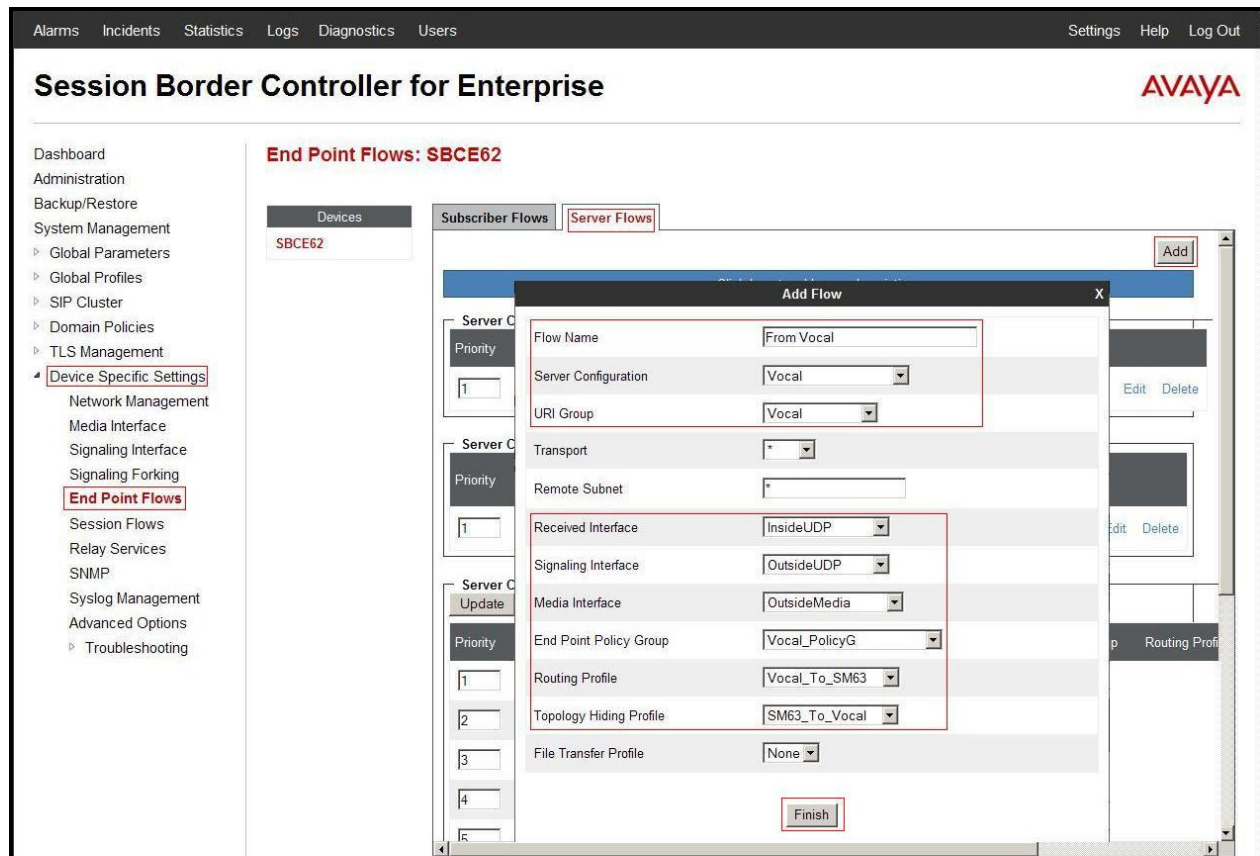
Figure 98 - End Point Flows 1

7.4.4.2 Create End Point Flows – To Vocal IP Networx

From the menu on the left-hand side, select **Device Specific Settings → End Point Flows**.

- Select the **Server Flows** Tab
- Select **Add**, enter **Flow Name: To Vocal**
  - **Server Configuration**: **SM63**
  - **URI Group: Vocal**
  - **Transport: ***
  - **Remote Subnet: ***
  - **Received Interface**: **OutsideUDP**
  - **Signaling Interface: InsideUDP**
  - **Media Interface**: **InsideMedia**
  - **End Point Policy Group: SM63_Vocal_PolicyG**
  - **Routing Profile: SM63_To_Vocal**
  - **Topology Hiding Profile: Vocal_To_SM63**

- Click **Finish**.
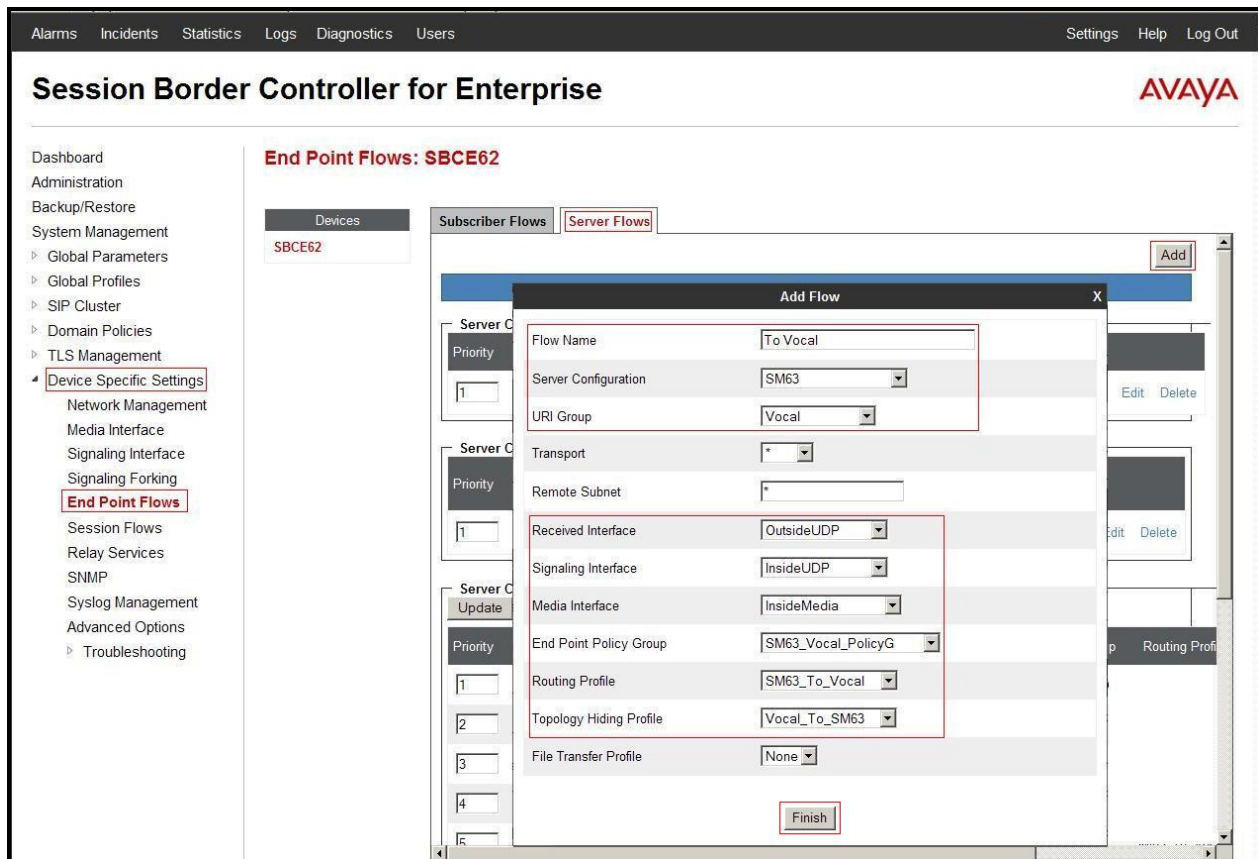
Figure 99 - End Point Flows 2

7.4.5. Create Session Flows

Session Flow determines the media (audio/video) sessions in order to apply the appropriate session policy.

- Select **Device Specific Settings** from the menu on the left-hand side
- Select the **Session Flows**
- Select **Add**
- Enter **Flow Name: Vocal**
  - **URI Group#1**: **Vocal**
  - **URI Group#2**: **Vocal**
  - **Session Policy**: **Vocal**
- Select **Finish** (not shown)
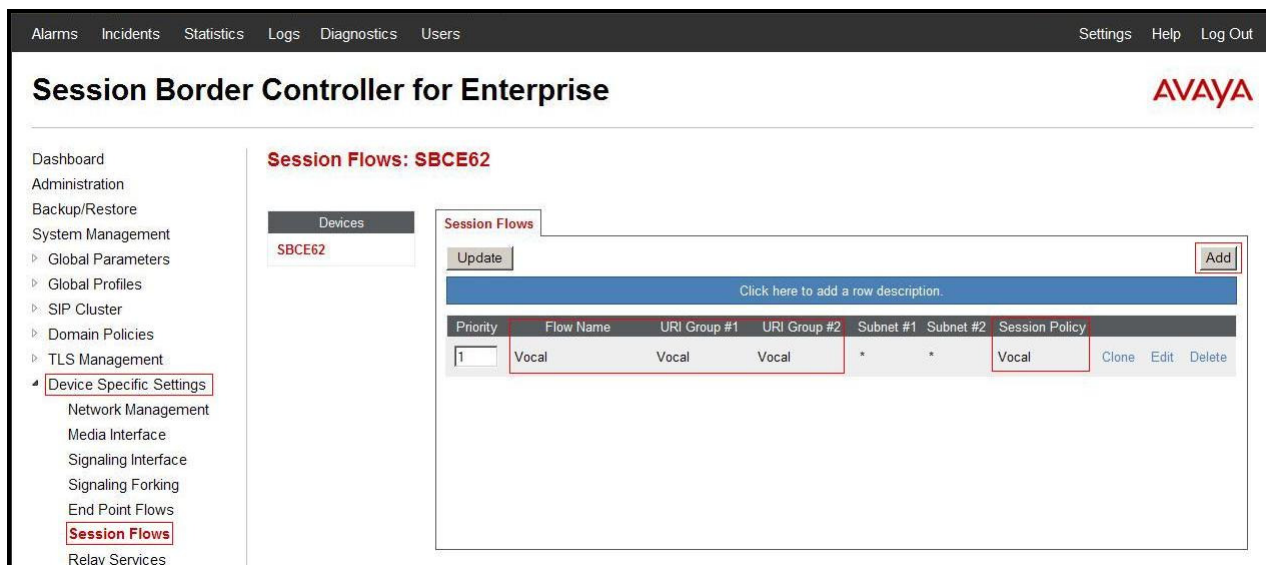
Figure 100 – Session Flows

8. Vocal IP Networx SIP Trunking service Configuration

Vocal IP Networx is responsible for the network configuration of the Vocal IP Networx SIP Trunking service. Vocal IP Networx will require that the customer provide the public IP address used to reach the Avaya SBCE public interface at the edge of the enterprise. Vocal IP Networx will provide the IP address of Vocal IP Networx's SIP proxy/SBC, IP addresses of media sources and Direct Inward Dialed (DID) numbers assigned to the enterprise. This information is used to complete configurations for Communication Server 1000, Session Manager, and the Avaya SBCE discussed in the previous sections.

The configuration between Vocal IP Networx and the enterprise is a static configuration. There is no registration of the SIP trunk or enterprise users to the Vocal IP Networx's network.

9. Verification Steps

The following steps may be used to verify the configuration.

9.1. General

Place an inbound call from a PSTN phone to an internal Avaya phone, answer the call, and verify that two-way speech path exists. Verify that the call remains stable for several minutes and disconnects properly.

9.2. Verification of an Active Call on Communication Server 1000

Active Call Trace (LD 80)
The following is an example of one of the commands available on the Communication Server 1000 to trace the DN for which the call is in progress or idle (5528). The call scenario involved PSTN phone number 6139675206 calling 9172675528 (which is translated to phone 5528).

- Login into Communication Server 1000 Signaling Server 10.10.97.177 with admin account and password.
- Issue a command "cslogin" to login on to the Communication Server 1000 Call Server.
- Log in to the Overlay command prompt, issue the command **LD 80** and then **trace 0 5528**.
- After the call is released, issue command **trac 0 5528** again to see if the DN is released back to idle state.

Below is the actual output of the Communication Server 1000 Call Server Command Line mode when the **5528** is in call state:

```
>ld 80
TRA000
.trac 0 5528

ACTIVE  VTN 096 0 00 02

ORIG   VTN 100 0 00 00   VTRK IPTI  RMBR  100 1 INCOMING VOIP GW CALL
  FAR-END SIP SIGNALLING IP: 10.10.98.13
  FAR-END MEDIA ENDPOINT IP: 10.10.98.13  PORT: 37000
  FAR-END VendorID: AVAYA-SM-6.3.4.0.634014
TERM   VTN 096 0 00 02  KEY 0  SCR MARP  CUST 0  DN 5528  TYPE 2002P2
  SIGNALLING ENCRYPTION: INSEC
  MEDIA ENDPOINT IP: 10.33.5.16  PORT: 5200
MEDIA PROFILE: CODEC G.711 MU-LAW  PAYLOAD 20 ms  VAD OFF
RFC2833: RXPT  101  TXPT  101   DIAL DN 5528
MAIN_PM  ESTD
TALKSLOT  ORIG  6   TERM  11
EES_DATA:
NONE
QUEU  NONE
CALL ID 501 40

---- ISDN ISL CALL (ORIG) ----
CALL REF # =  484
BEARER CAP =  VOICE
HLC =
CALL STATE = 10    ACTIVE
CALLING NO = 16139675206  NUM_PLAN:E164   TON:INTERNATIONAL  ESN:UNKNOWN
CALLED NO  = 9172675528 NUM_PLAN:UNKNOWN    TON:UNKNOWN  ESN:UNKNOWN
```

And this is the example after the call to 5528 is finished.

```
>ld 80
TRA000
.trac 0 5528
IDLE VTN 96 0 00 02   MARP
```

SIP Trunk monitoring (LD 32)
Place a call inbound from PSTN (6139675206) to an internal device (9172675528). Then check the SIP trunk status by using LD 32, one trunk is BUSY.
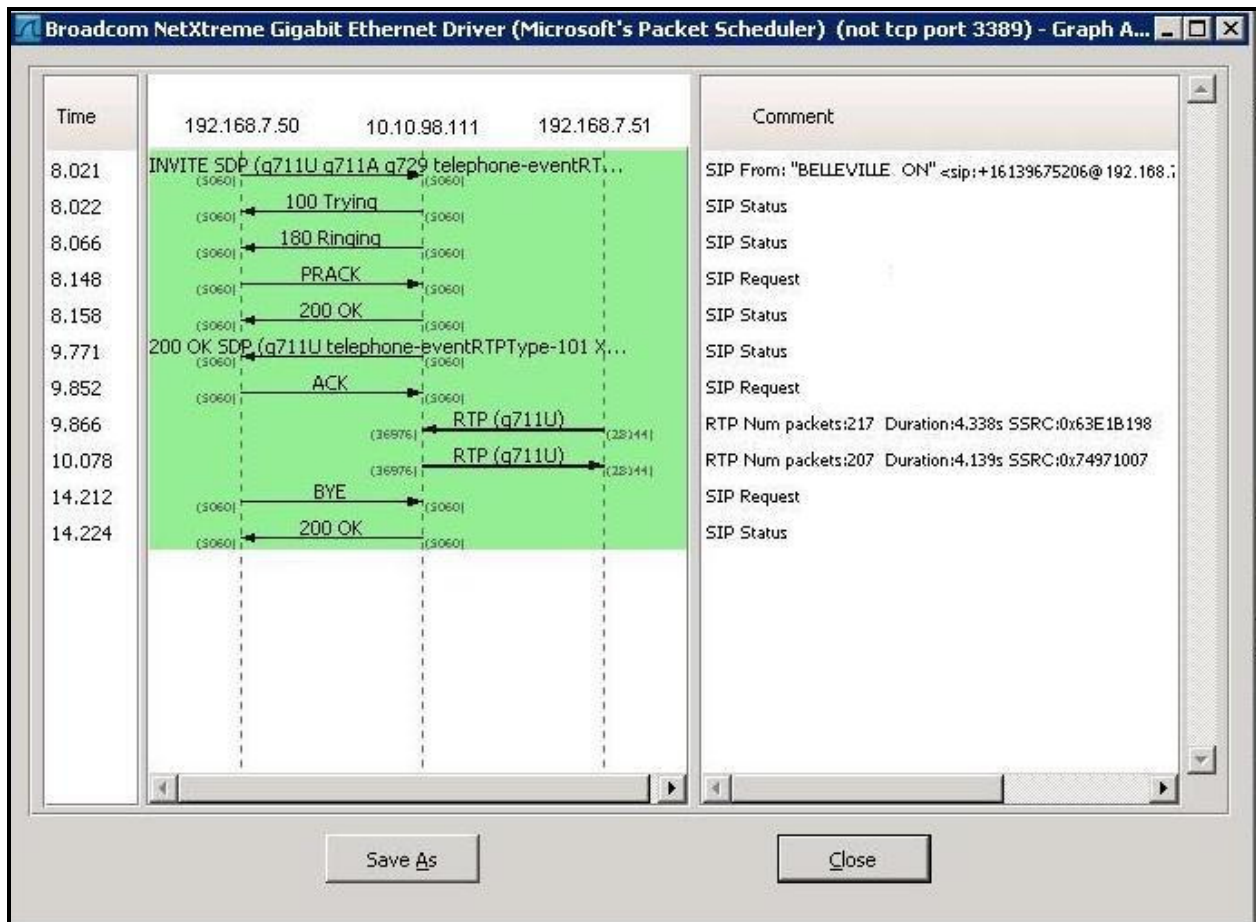
```
>ld 32
NPR000
.stat 100 0
091 UNIT(S) IDLE
001 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
```

After the call is released, check that SIP trunk status changed to the IDLE state.

```
>ld 32
NPR000
.stat 100 0
092 UNIT(S) IDLE
000 UNIT(S) BUSY
000 UNIT(S) DSBL
000 UNIT(S) MBSY
```

9.3. Protocol Trace

Below is a wireshark trace of the same call scenario described in **Section 9.2**.

10. Conclusion

All of the test cases have been executed. Despite observations seen during the testing, as noted in **Section 2.2**, the test met the objectives outlined in **Section 2.1**. The Vocal IP Networx SIP Trunk service is considered **compliant** with Avaya Communication Server 1000 Release 7.6, Avaya Aura® Session Manager Release 6.3 FP3 and Avaya Session Border Controller for Enterprise Release 6.2.0 Q48

11. References

This section references the documentation relevant to these Application Notes.

Product documentation for Avaya products, including the following, is available at:
http://support.avaya.com/

[1] Network Routing Service Fundamentals, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-130, Issue 04.01, March 2013.

[2] IP Peer Networking Installation and Commissioning, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-313, Issue 06.01, March 2013.

[3] Communication Server 1000E Overview, Avaya Communication Server 1000, Release 7.6, Document Number NN43041-110, Issue 06.01, March 2013.

[4] Unified Communications Management Common Services Fundamentals, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-116, Issue 06.01, March 2013.

[5] Dialing Plans Reference, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-283, Issue 06.01, March 2013.

 [6] Product Compatibility Reference, Avaya Communication Server 1000, Release 7.6, Document Number NN43001-256, Issue 06.01 Standard, March 2013.

[7] Avaya Aura® System Manager Overview and Specification, Release 6.3, Issue 2, May 2013.

[8] Administering Avaya Aura® Session Manager, Release 6.3, Issue 2, June 2013.

[9] Maintaining and Troubleshooting Avaya Aura® Session Manager, Release 6.3, Issue 2, May 2013.

[10] Administering Avaya Session Border Controller for Enterprise, Release 6.2, Issue 2, May 2013.

[11] Installing Avaya Session Border Controller for Enterprise, Release 6.2, Issue 3, June 20 2013.

[12] Upgrading Avaya Session Border Controller for Enterprise, Release 6.2, Issue 3, July 2013.

Other resources:

[13] RFC 3261 SIP: Session Initiation Protocol, http://www.ietf.org/

[14] RFC 2833 RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals, http://www.ietf.org/

Appendix A: SigMa Script

The following is the Signaling Manipulation script used in the configuration of the Avaya SBCE, **Section 7.2.6**:

```
within session "All"
{
   act on request where %DIRECTION="OUTBOUND" and %ENTRY_POINT="POST_ROUTING"
   {

// Modify the SIP OPTIONS

   %HEADERS["From"][1].regex_replace("sip:10.10.98.111"," sip:9172675528@10.10.98.111");

// Remove unwanted SIP Headers

      remove(%HEADERS["History-Info"][5]);
      remove(%HEADERS["History-Info"][4]);
      remove(%HEADERS["History-Info"][3]);
      remove(%HEADERS["History-Info"][2]);
      remove(%HEADERS["History-Info"][1]);
      remove(%HEADERS["Alert-Info"][1]);
      remove(%HEADERS["x-nt-e164-clid"][1]);
      remove(%HEADERS["P-AV-Message-Id"][1]);
      remove(%HEADERS["P-Charging-Vector"][1]);
      remove(%HEADERS["Av-Global-Session-ID"][1]);
      remove(%HEADERS["P-Location"][1]);
      remove(%HEADERS["Remote-Party-ID"][1]);
   }
}
```