Avaya Solution & Interoperability Test Lab

Application Notes for the Nuance Communications OnDemand Hosted IVR MediaServer/SIP Proxy Configuration with Avaya Communication Manager and Avaya SIP Enablement Services – Issue 1.0

Abstract

These Application Notes describe a solution comprised of Avaya Communication Manager, Avaya SIP Enablement Services, and Nuance Communications MediaServer and SIP Proxy. The Nuance Communications SIP Proxy handles all the SIP-related signaling and the Nuance Communications MediaServer is VoIP software for executing Voice Extensible Markup Language (VXML) applications in a hosted environment. The function of VXML is application-specific and provides an IVR-driven menu for access to these applications. For the purpose of compliance testing, several VXML IVR applications provided by Nuance Communications were used to exercise SIP call flows with SIP and non-SIP telephones. Information in these Application Notes has been obtained through 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 a solution comprised of Avaya Communication Manager 4.0.1, Avaya SIP Enablement Services 4.0, and Nuance Communications MediaServer 9.1.1 and SIP Proxy 0.8.14. The Nuance Communications SIP Proxy handles all the SIP-related signaling and the Nuance Communications MediaServer is VoIP software for executing Voice Extensible Markup Language (VXML) applications in a hosted environment. The function of VXML is application-specific and provides an IVR-driven menu for access to these applications. For the purpose of compliance testing, several VXML IVR applications provided by Nuance Communications were used to exercise SIP call flows with SIP and non-SIP telephones.

Figure 1 illustrates a sample configuration consisting of an Avaya S8300 Server, an Avaya G700 Media Gateway, an Avaya SIP Enablement Services (SES) server, and two Nuance Hosted Sites, Primary and Backup, each with a Nuance MediaServer and Nuance SIP Proxy (Nuance Hosted IVR). Avaya Communication Manager was installed on the Avaya S8300 Server. The solution described herein is also extensible to other Avaya Servers and Gateways. For completeness, Avaya 4600-Series SIP IP Telephones, Avaya 4600-Series H.323 IP Telephones, and Avaya 6400 and 8400-Series Digital Telephones are included in **Figure 1** to demonstrate calls from Avaya SIP, H.323, and digital telephones into the Nuance Hosted IVR. The analog PSTN telephone is also included to demonstrate calls routed by Avaya Communication Manager to the Nuance Hosted IVR.

Typical SIP call flows for Avaya Communication Manager and Avaya SES are as follows:

- Calls originating from SIP endpoints are routed to Avaya Communication Manager via Avaya SES over a SIP trunk and are routed to SIP endpoints from Avaya Communication Manager via Avaya SES over a SIP trunk.
- Calls originating from non-SIP endpoints come directly to Avaya Communication Manager and are routed to SIP endpoints from Avaya Communication Manager over SIP trunk via Avaya SES.

The Nuance Hosted IVR at each site is configured as a trusted host with Avaya SES and supports the G711MU codec using RFC2823 for DTMF.

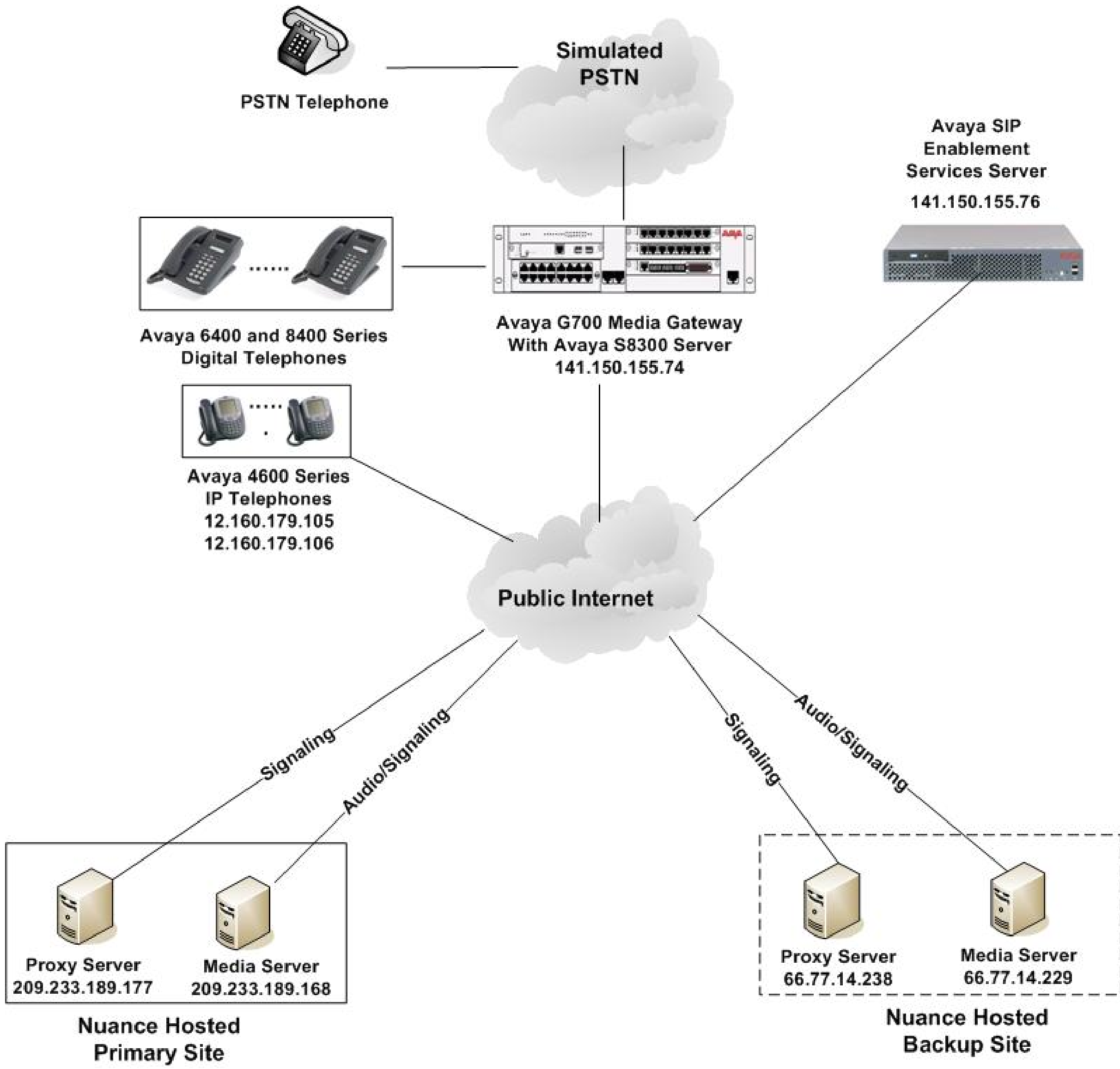


Figure 1: Sample Configuration

2. Equipment and Software Validated

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

Equipment	Software/Firmware
Avaya S8300 Server	Avaya Communication Manager 4.0.1 (R014x.00.1.731.2)
Avaya G700 Media Gateway	26.33.0
MM710P DS1	HW05, FW013
MM717AP DCP	HW03, FW04
MM711AP ANA	HW03, FW17
Avaya SIP Enablement Services Server S8500B	Avaya SIP Enablement Services 4.0 (SES-4.0.0.0-033.6)
Avaya 4600 Series IP Telephones	2.2.3 (4610SW SIP) 2.3 (4602SW H.323) 2.6 (4610SW H.323) 2.5 (4625SW H.323)
Avaya 6400 and 8400 Series Digital Telephones	-
Analog Telephone	-
Nuance Communications MediaServer	9.1.1
Nuance Communications SIP Proxy	0.8.14

3. Configure Avaya Communication Manager

This section describes the steps for configuring Avaya Communication Manager to route the calls properly for interaction with the Nuance Hosted IVR via Avaya SES. The System Access Terminal (SAT) interface is used to configure IP codec set, SIP signaling and trunking between Avaya Communication Manager and Avaya SES and setting up the dial plan for routing the calls destined for Avaya SES. Configuration in the following sections is only for the fields where a value needs to be entered or modified. Default values are used for all other fields. Refer to [1] for additional details.

3.1. IP Codec Set

This section describes the steps for administering an IP codec set in Avaya Communication Manager. This IP codec set is used in the IP network region for communications between Avaya Communication Manager and Avaya SES.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="2">1.</td><td>Enter the change ip-codec-set <c> command, where c is a number between 1 and 7, inclusive. This IP codec set is used in Section 3.2 to specify which IP codecs may be used within and between the associated network regions. For the compliance testing, only G.711MU was used and Media Encryption was set to none as encryption is currently not supported for SIP telephony.</td></tr>
<tr><td><pre>
change ip-codec-set 1   Page   1 of 2

                          IP Codec Set

    Codec Set: 1

    Audio   Silence   Frames   Packet
    Codec   Suppression Per Pkt Size(ms)
 1: G.711MU   n   2   20
 2:
 3:
 4:
 5:
 6:
 7:

     Media Encryption
 1: none
 2:
 3:
</pre></td></tr>
</table>

3.2. IP Network Region

This section describes the steps for administering an IP network region in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description
1.	Enter the **change ip-network-region <n>** command, where **n** is a number between **1** and **250** inclusive and configure the following: • **Authoritative Domain** –This should match the **SIP Domain** value in **Section 4**, **Step 2**. In this example, **devcon.com** is used. • **Intra-region IP-IP Direct Audio** – Set to **no**. Intra-region IP-to-IP connectivity is not supported in this solution. • **Codec Set** – Set the codec set number as provisioned in **Section 3.1**. • **Inter-region IP-IP Direct Audio** – Set to **no**. Inter-region IP-to-IP connectivity is not supported in this solution.

```
change ip-network-region 1                                       Page 1 of 19
                               IP NETWORK REGION
  Region: 1
Location:         Authoritative Domain: devcon.com
    Name:
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: no
      Codec Set: 1              Inter-region IP-IP Direct Audio: no
   UDP Port Min: 2048                      IP Audio Hairpinning? y
   UDP Port Max: 65535
DIFFSERV/TOS PARAMETERS                  RTCP Reporting Enabled? y
 Call Control PHB Value: 46      RTCP MONITOR SERVER PARAMETERS
        Audio PHB Value: 46       Use Default Server Parameters? y
        Video PHB Value: 26
802.1P/Q PARAMETERS
 Call Control 802.1p Priority: 6
        Audio 802.1p Priority: 6
        Video 802.1p Priority: 5      AUDIO RESOURCE RESERVATION PARAMETERS
H.323 IP ENDPOINTS                                        RSVP Enabled? n
  H.323 Link Bounce Recovery? y               RSVP Refresh Rate(secs): 15
 Idle Traffic Interval (sec): 20      Retry upon RSVP Failure Enabled: y
   Keep-Alive Interval (sec): 5                 RSVP Profile: guaranteed-service
            Keep-Alive Count: 5       RSVP unreserved (BBE) PHB Value: 46
```

3.3. IP Node Names

This section describes the steps for defining the IP node name for Avaya SES in Avaya Communication Manager.

Step	Description
1.	Enter the **change node-names ip** command and add a node name for Avaya SES along with its IP address. Take note of the IP address of **procr** as this will be used in **Section 4**.

```
change node-names ip                                               Page 1 of 1
                                  IP NODE NAMES
    Name              IP Address
procr               141.150.155.74
default             0.0.0.0
SES                 141.150.155.76
```

3.4. SIP Signaling

This section describes the steps for administering a signaling group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td rowspan="3">1.</td><td>Issue the command add signaling-group <s>, where s is an available signaling group, and configure the following:

Group Type – Set to sip.
Transport Method – Set to tls.
Near-end Node Name - Set to procr as displayed in Section 3.3.
Far-end Node Name - Set to Avaya SES name configured in Section 3.3.
Far-end Network Region - Set to the region configured in Section 3.2.
Far-end Domain - Set to the devcon.com. This should match the SIP Domain value in Section 4, Step 2.
DTMF over IP - Set to rtp-payload.
Direct IP-IP Audio Connections – Set to n.
</td></tr>
<tr><td><pre>
add signaling-group 1   Page 1 of 1
                               SIGNALING GROUP

Group Number: 1   Group Type: sip
                      Transport Method: tls

  Near-end Node Name: procr   Far-end Node Name: SES
Near-end Listen Port: 5061   Far-end Listen Port: 5061
                                       Far-end Network Region: 1
      Far-end Domain:devcon.com

                                            Bypass If IP Threshold Exceeded? n

        DTMF over IP: rtp-payload   Direct IP-IP Audio Connections? n
                                                       IP Audio Hairpinning? y
Enable Layer 3 Test? n
Session Establishment Timer(min): 120
</pre></td></tr>
<tr><td></td></tr>
</table>

3.5. SIP Trunking

This section describes the steps for administering a trunk group in Avaya Communication Manager for communication between Avaya Communication Manager and Avaya SES.

Step	Description
1.	Issue the command **add trunk-group <t>**, where **t** is an available trunk group number, and configure the following: • **Group Type** – Set to **sip**. • **Group Name –** Enter any descriptive name. • **TAC** (Trunk Access Code) – Set to any available trunk access code. • **Signaling Group** – Set to the signaling group configured in **Section 3.4**. • **Number of Members –** Allowed values are between **0** and **255**. Set to a value large enough to accommodate the number of SIP telephone extensions being used. **Note:** *Each SIP call between two SIP endpoints (whether internal or external) requires two SIP trunks for the duration of the call. The license file installed on the system controls the maximum permitted.*

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

Group Number: 1                    Group Type: sip           CDR Reports: y
  Group Name: To SES                      COR: 1        TN: 1        TAC: 101
   Direction: two-way       Outgoing Display? n
 Dial Access? n                                          Night Service:
Queue Length: 0
Service Type: tie                   Auth Code? n

                                                       Signaling Group: 1
                                                     Number of Members: 90
```

3.6. Dial Plan/AAR/Route Pattern

This section describes the steps for setting the dial plan, AAR digit analysis and Route Pattern in Avaya Communication Manager for proper routing of calls from Avaya Communication Manager to Avaya SES. These calls are ultimately destined for the Nuance Hosted IVR.

Step	Description
1.	Issue the command **change dialplan analysis** and add the following entries: • **Dialed String** – Set it to a value for routing calls to Avaya SES for proper AAR digit analysis. Set to **41212** in this example. • **Total Length** – Set to a value equal to the length of the **Dialed String**. • **Call Type** – Set to **aar**.

```
change dialplan analysis                                    Page 1 of 12
                             DIAL PLAN ANALYSIS TABLE
                                                       Percent Full:    2
      Dialed  Total  Call      Dialed  Total  Call      Dialed  Total  Call
      String  Length Type      String  Length Type      String  Length Type
        0       1    fac
        1       3    dac
        41212   5    aar
        2       4    ext
        3       5    ext
        4       5    ars
        5       5    ext
        8       1    fac
        9       1    fac
        *       3    fac
        #       3    fac
```

Step	Description
2.	Issue the command **change public-unknown-numbering <e>**, where **e** is extension code to be administered, and add an entry with the following values: • **Ext Len** – Set to the length of the calling party number for extensions that will be calling the Nuance Hosted IVR. • **Ext Code** – Prefix of the calling party. Set to **5** in this example. • **Trk Grp<s>** - Trunk Group(s) where calls from these extensions would be routed. • **Total CPN Len** – Length of the outbound calling party number.

```
change public-unknown-numbering 5                               Page   1  of  2
                    NUMBERING – PUBLIC/UNKNOWN FORMAT
                                                  Total
Ext Ext            Trk       CPN                   CPN
Len Code           Grp<s>    Prefix                Len

5   5              1                               5
```

Step	Description
3.	Issue the command **change route-pattern <r>**, where <**r**> is the number of an available route pattern to be administered. • **Grp No –** Set to the Trunk Group provisioned in **Section 3.5**. • **FRL –** Set to **0**. **Note**: In the example below, **Inserted Digits** will be prepended to the called number.

```
change route-pattern 10                                          Page   1  of  3
                    Pattern Number: 1   Pattern Name: SES SIP
                             SCCAN? n     Secure SIP? n
    Grp FRL NPA Pfx Hop Toll No.  Inserted                              DCS/ IXC
    No          Mrk Lmt List Del  Digits                                QSIG
                             Dgts                                       Intw
 1: 1   0                      0     33344                               n   user
 2:                                                                      n   user
 3:                                                                      n   user
 4:                                                                      n   user
 5:                                                                      n   user
 6:                                                                      n   user

     BCC VALUE  TSC CA-TSC    ITC BCIE Service/Feature PARM  No. Numbering LAR
    0 1 2 3 4 W     Request                                Dgts Format
                                                         Subaddress
 1: y y y y y n  n            rest                                         none
 2: y y y y y n  n            rest                                         none
 3: y y y y y n  n            rest                                         none
 4: y y y y y n  n            rest                                         none
 5: y y y y y n  n            rest                                         none
 6: y y y y y n  n            rest                                         none
```

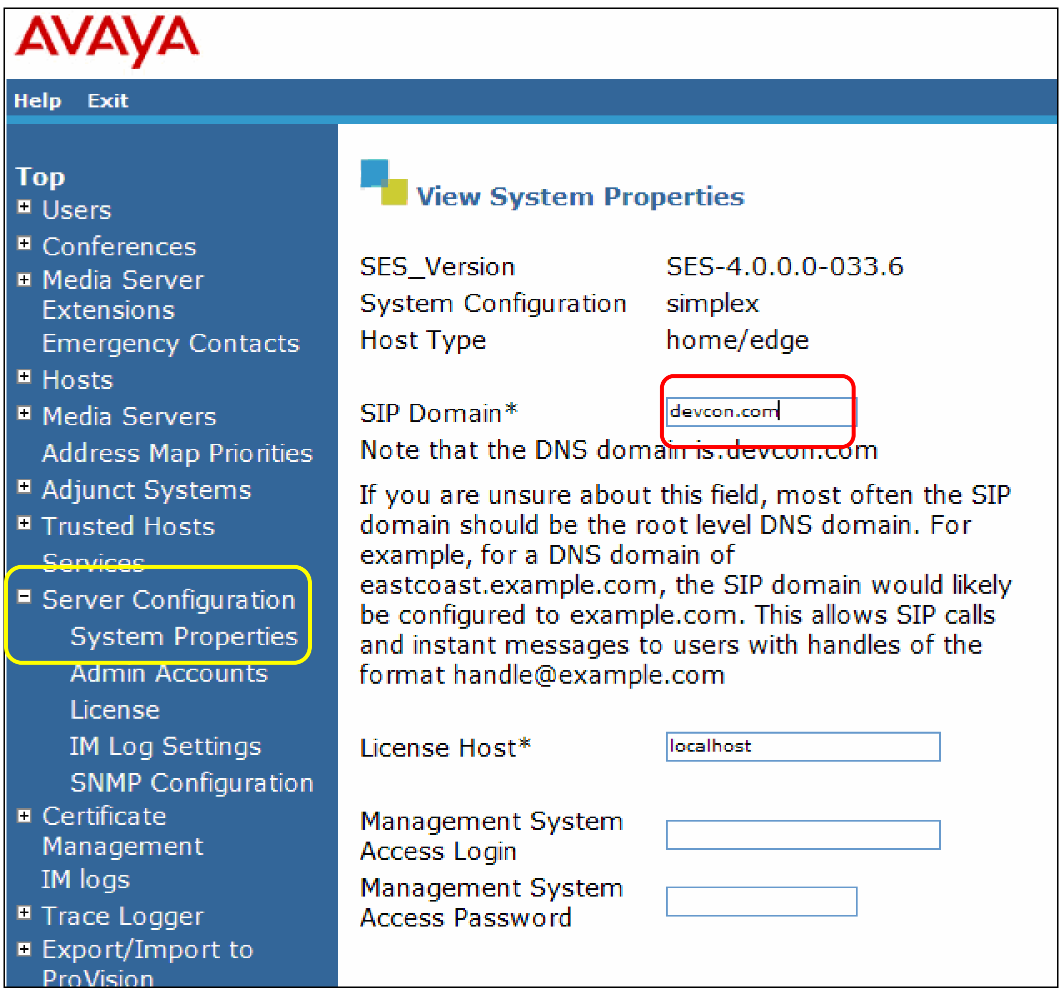
Step	Description
4.	Issue the command **change aar analysis 4** and add the following entries: • **Dialed String –** Set to match the digits dialed. Set to **41212** in this example. • **Total Min and Max –** Set to the value of the dialed digits**.** • **Route Pattern –** Set the value to the route pattern defined in **Step 2**. • **Call Type –** Set to **aar.**

```
change aar analysis 4                                             Page  1  of  2
                             AAR DIGIT ANALYSIS TABLE
                                                            Percent Full:   2

          Dialed           Total     Route    Call   Node   ANI
          String          Min  Max  Pattern   Type   Num    Reqd
    41212                 5    5     10       aar           n
    2                     5    5     15       aar           n
    2                     7    7     999      aar           n
    245                   5    5     33       aar           n
```

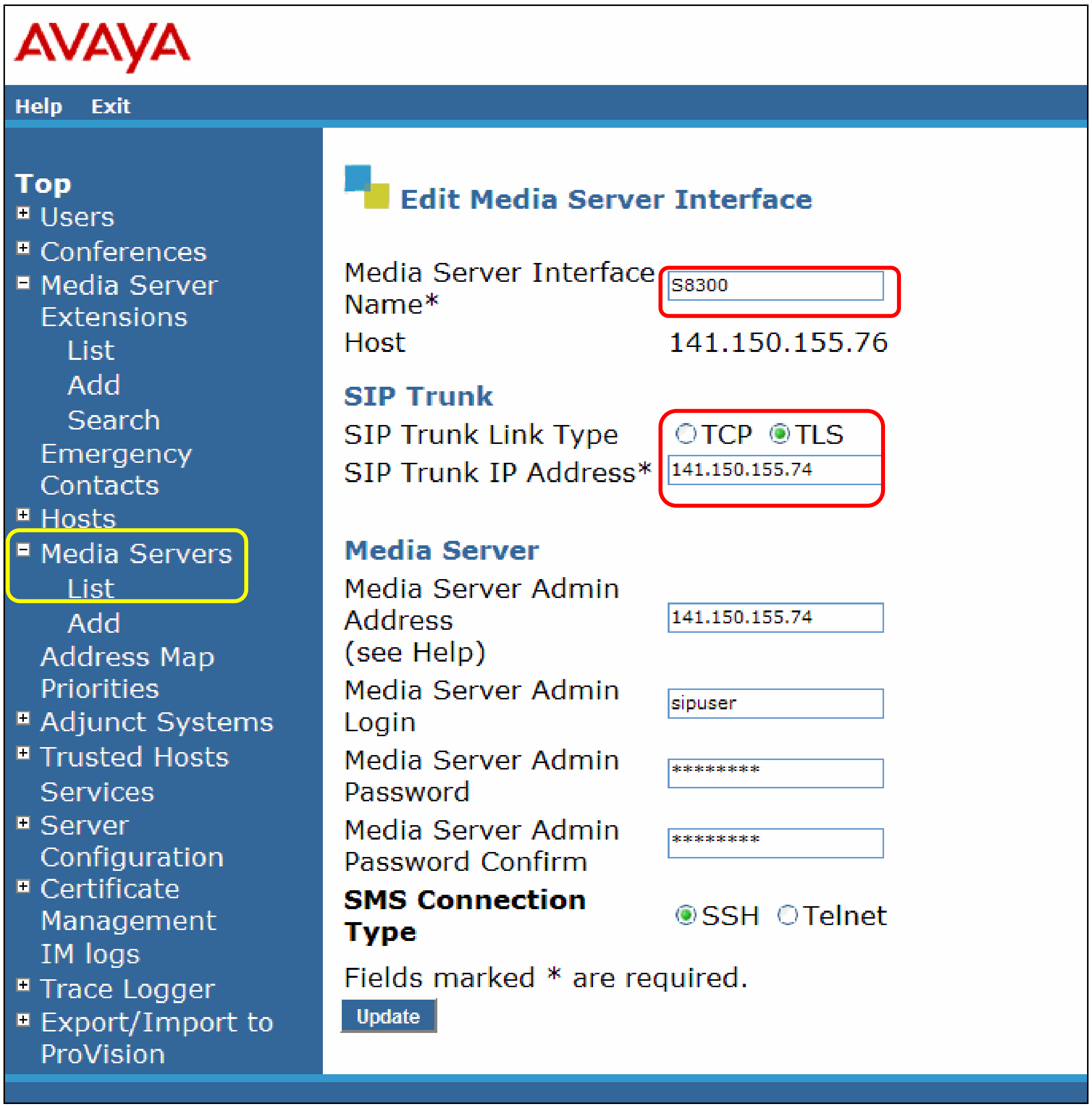
4. Configure Avaya SIP Enablement Services

This section describes steps to verify that SIP trunking has been properly configured between Avaya Communication Manager and Avaya SES. Also, additional configuration is done to add the Nuance Hosted IVR as a trusted host in Avaya SES. Configuration in the following steps is only for the fields where a value needs to be entered or modified. Default values are used for all other fields. Refer to [3, 4] for additional details.

Step	Description
1.	Open a web browser, enter http://<IP address of Avaya SES server>/admin for the URL, and log in with the appropriate credentials. Click on the **Launch Administration Web Interface** link upon successful login.
2.	On the **SIP Server Management** page: • Click the + sign to expand the options under **Server Configuration**. • Click **System Properties**. • Verify the **SIP Domain** matches the **Far-end Domain** field value configured for the signaling group on Avaya Communication Manager in **Section 3.4**. 

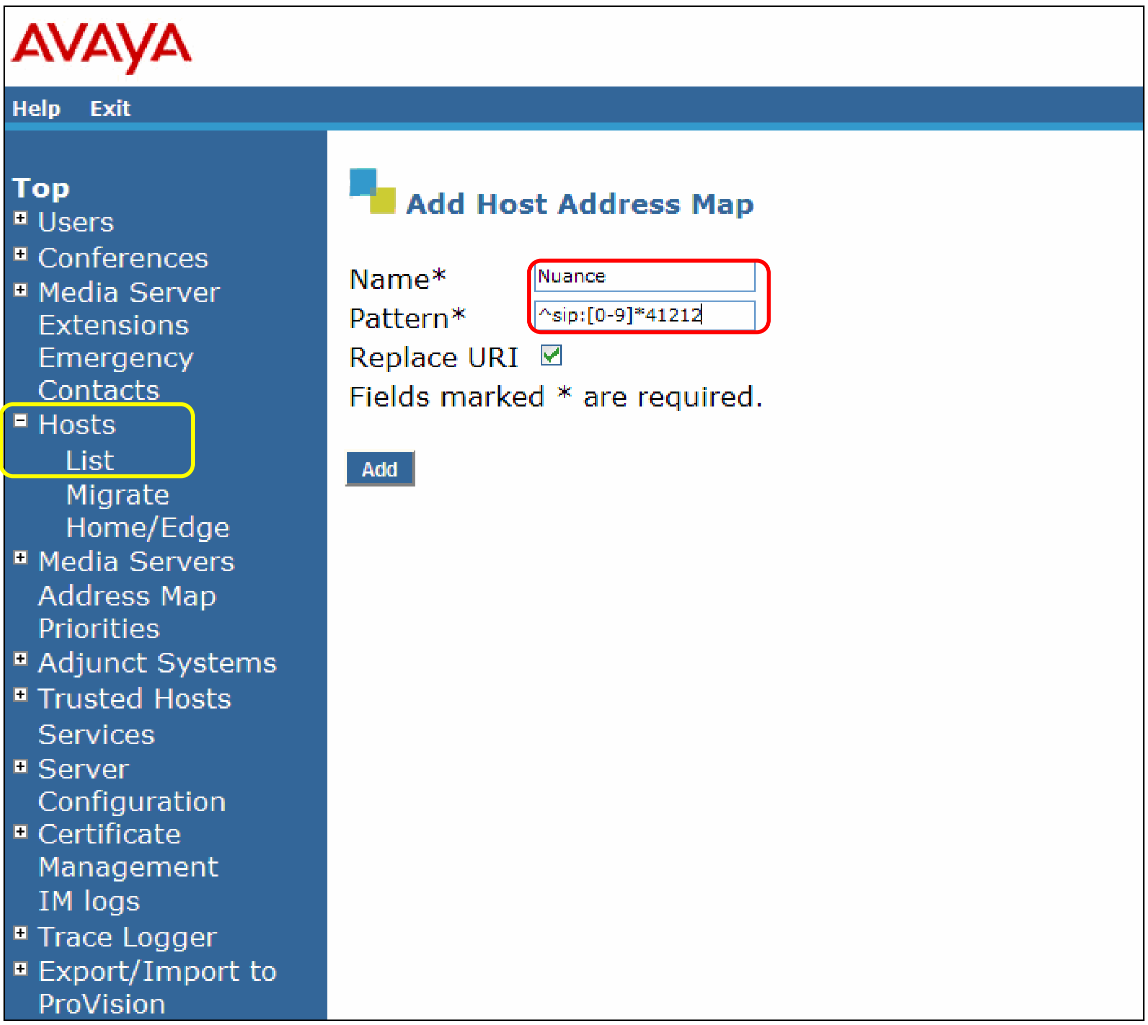
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>
To enable secure SIP trunking between Avaya SES and Avaya Communication Manager, verify Media Server Interface is properly configured:

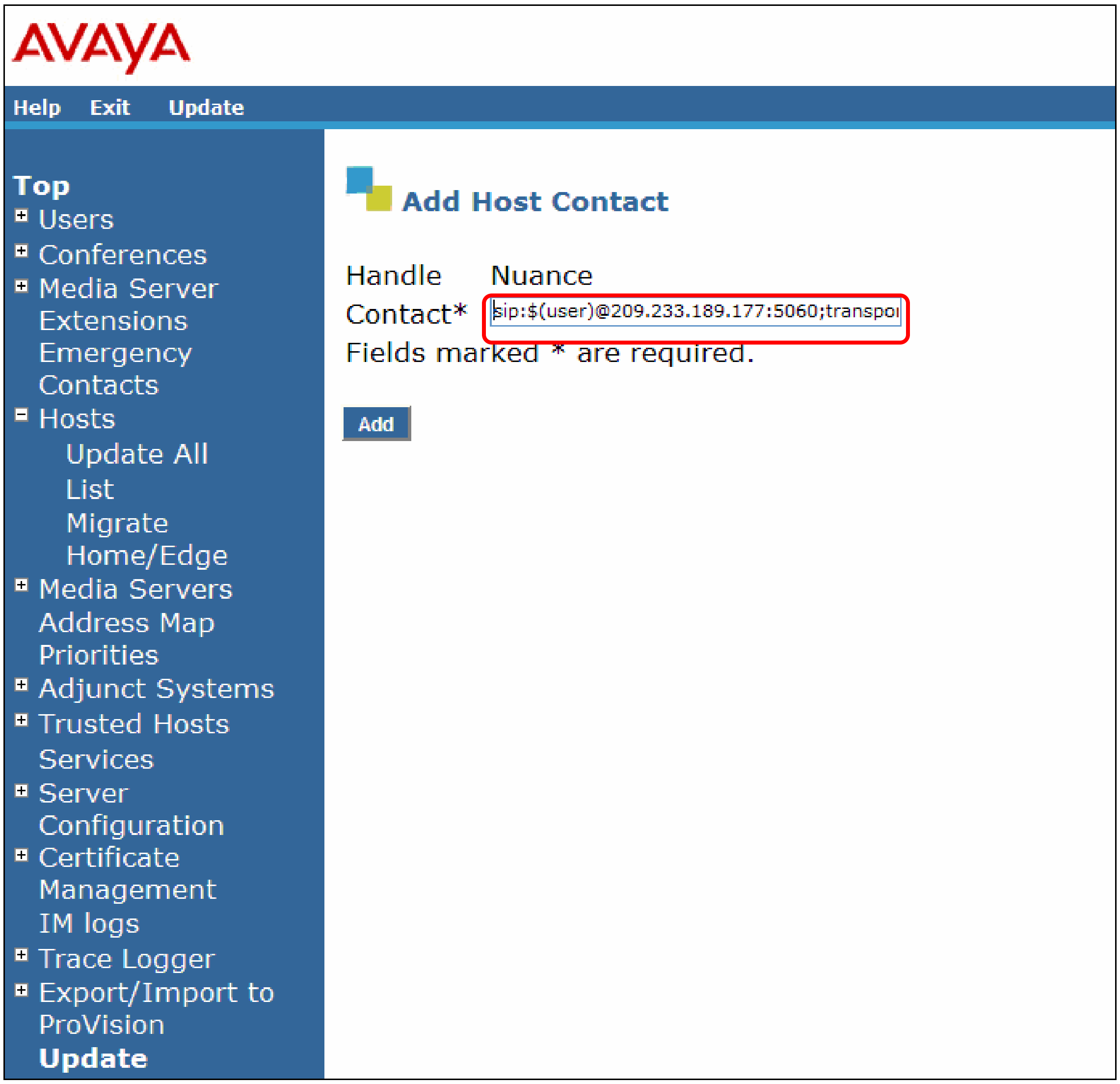
Click the + sign to expand the options under Media Servers
Click List and the following screen appears.
Verify that the Host matches the Avaya SES IP Address configured in Section 3.3.
Verify that the SIP Trunk Link Type matches the Transport Method configured in Section 3.4.
Verify that SIP Trunk IP Address matches the IP Address of procr shown in Section 3.3, Step 1.

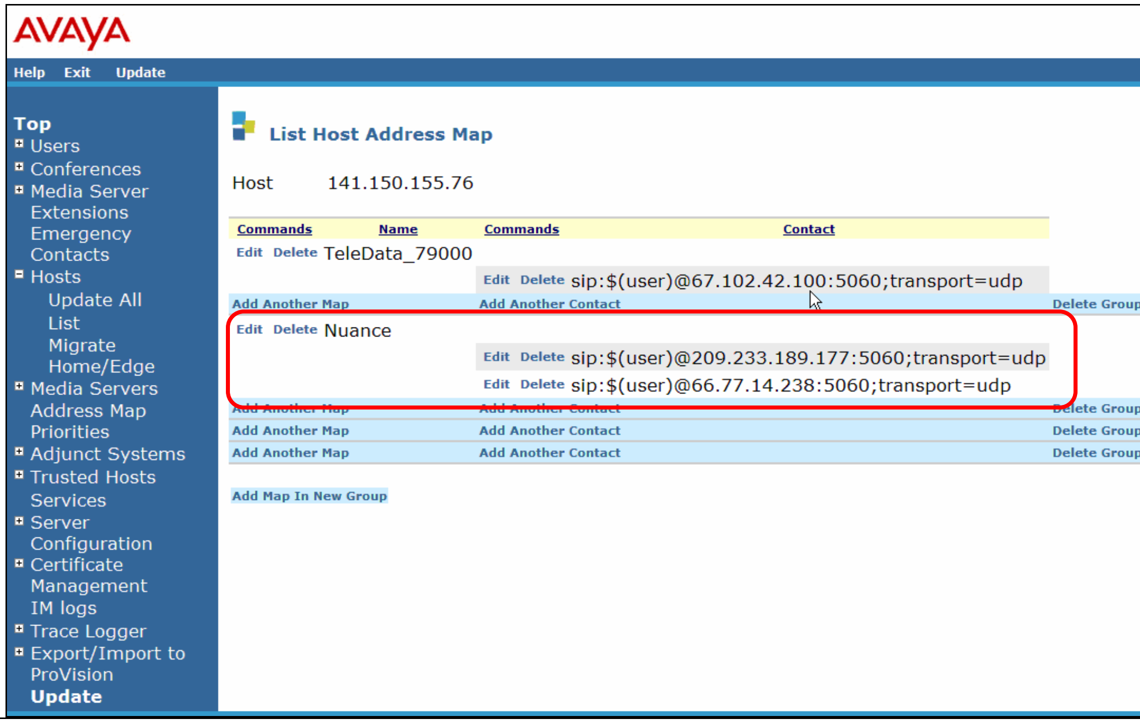


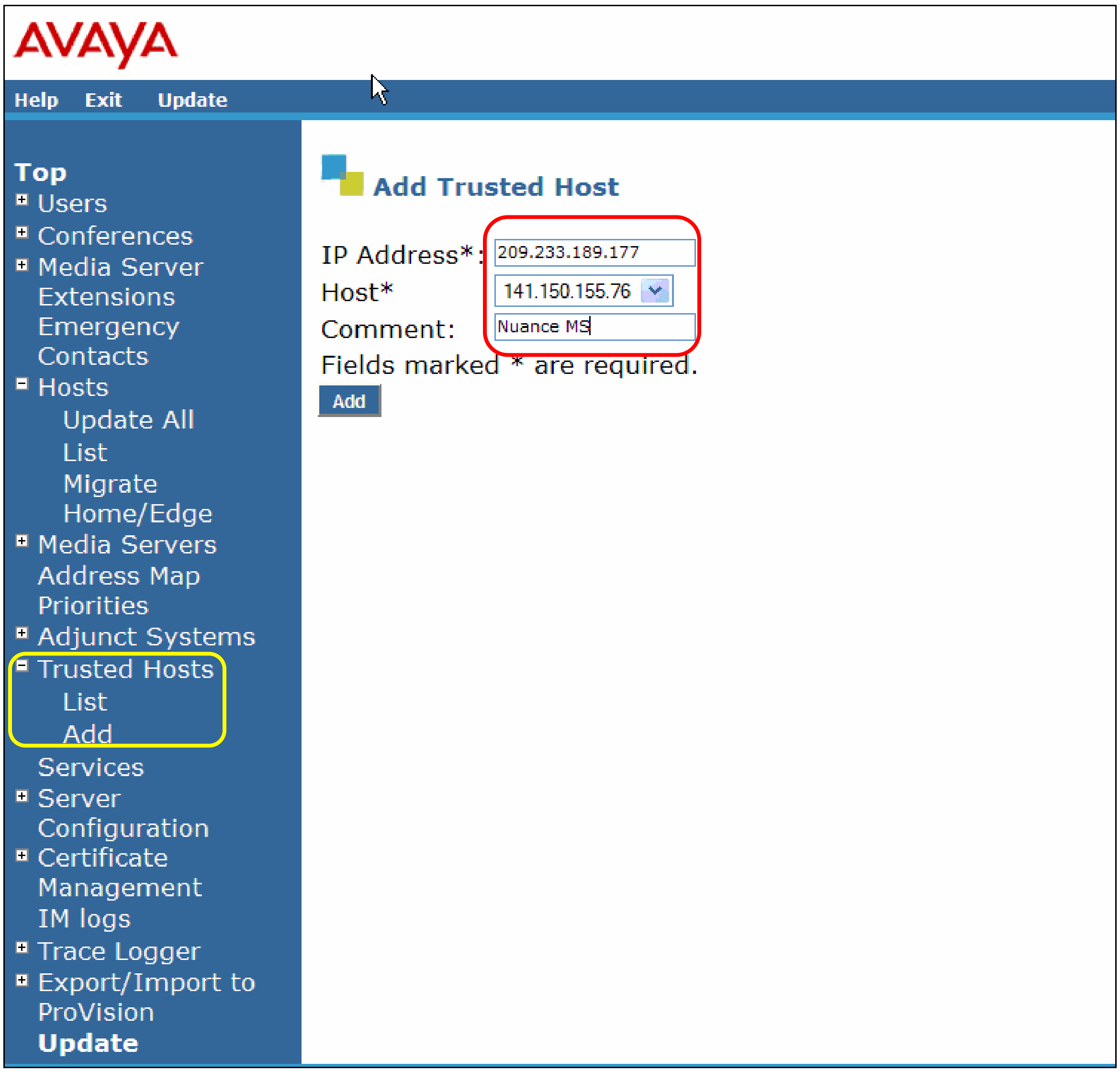
</td></tr>
</table>

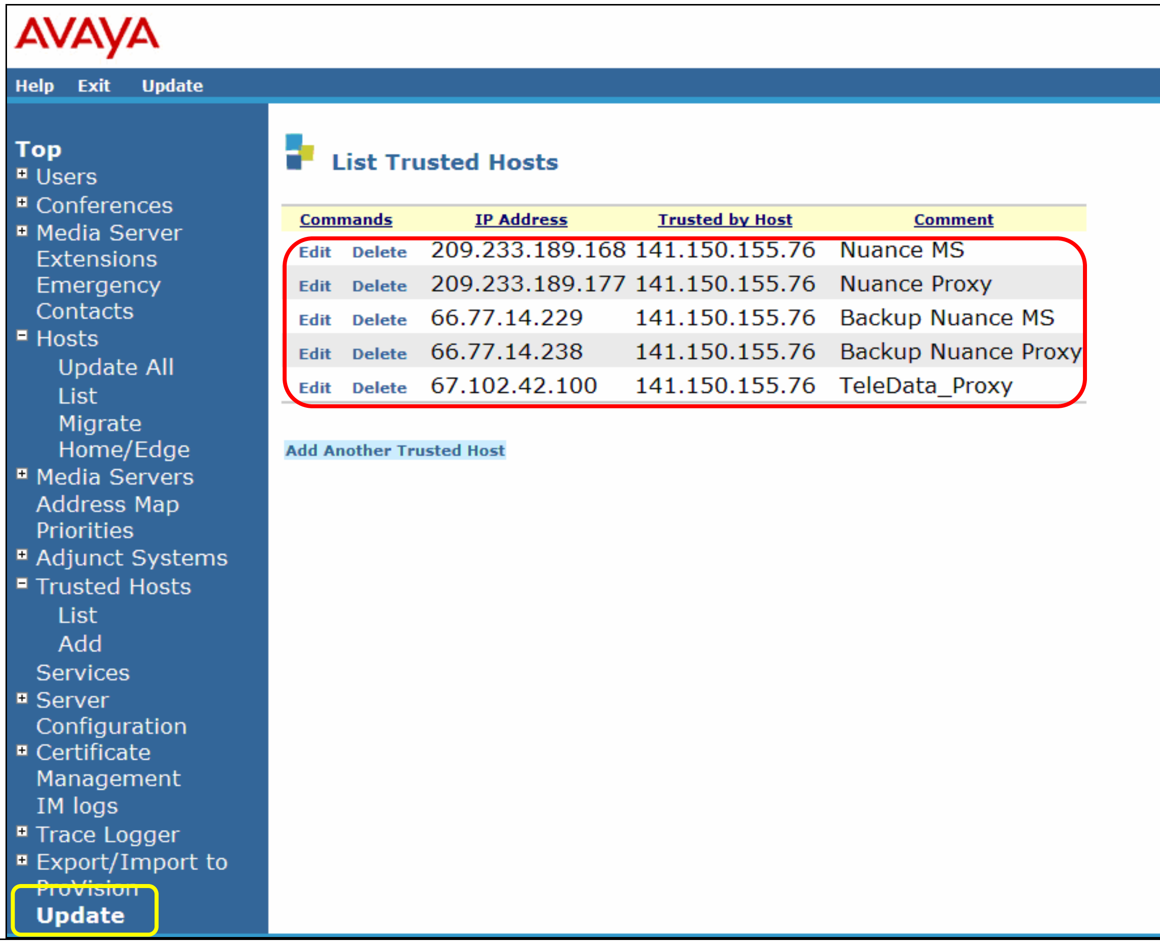
Step	Description

4.	A Host Address Map is required on Avaya SES to direct outbound calls from Avaya Communication Manager to the Nuance Hosted IVR. An Address Map is used to route the calls based on the contents of SIP INVITE URI. To configure a Host Address Map, do the following:
	• Click the + sign to expand the options under **Hosts** and then select **List**.
	• At the next screen [not shown], click **Add Map In New Group** and the screen shown here is displayed. Enter the following fields:
	    o **Name** – Any Descriptive Name
	    o **Pattern** – Expression to match the beginning of URI. In this example, the value is set to **^sip:[0-9]*41212**.
	• Click **Add**.
	
	Note: Additional Host Address Maps can be added by repeating this step.
	
	

Step	Description
5.	At the next screen (not shown), click **Add Host Contact**. On the screen that follows (see below), make the following entry: • **Contact** – Enter the destination IP address (*ip_addr*), port number (*port*) and transport protocol (*protocol*) as follows: o **sip:$(user)@***ip_addr*:*port*;**transport=***protocol* In this example, **sip:$(user)209.233.189.177;transport=udp** is entered. The IP address corresponds to the Nuance SIP Proxy IP address shown in **Figure 1**. Click on **Add**. **Note**: Additional host contacts can be added by repeating this step. 

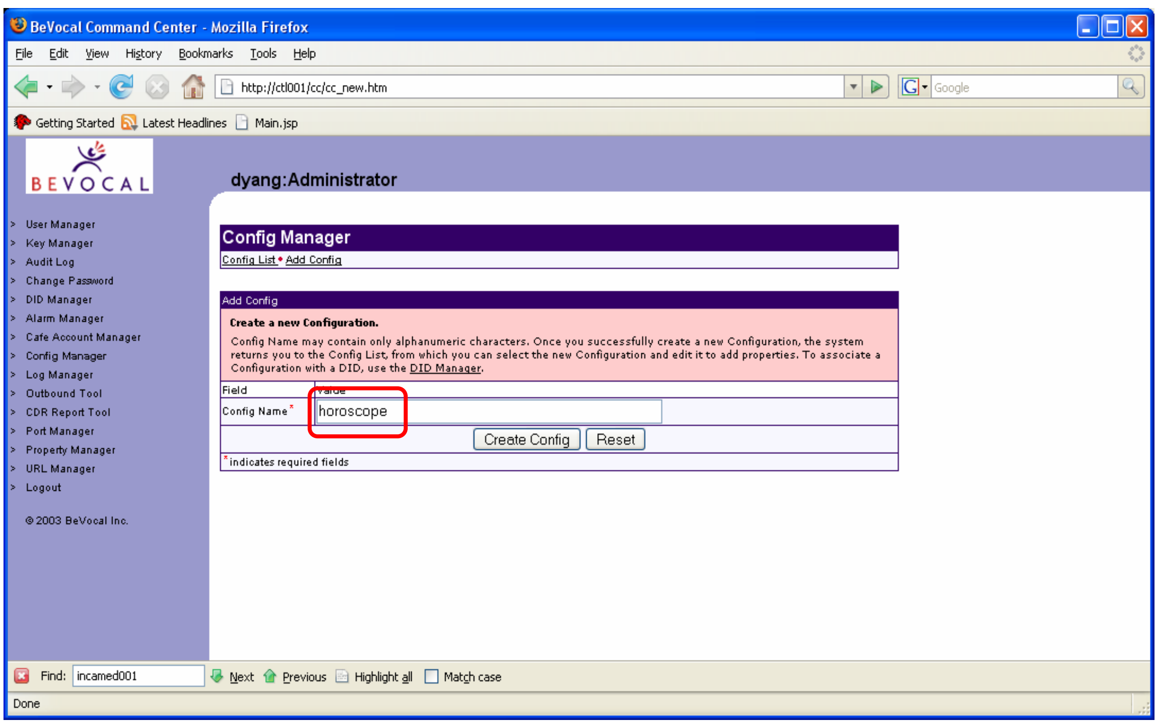
<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>6.</td><td>This is the confirmation screen after adding host contacts for both the primary and backup Nuance SIP Proxies.

</td></tr>
</table>

Step	Description
7.	The Nuance SIP Proxy and the Nuance MediaServer do not register with Avaya SES and are configured as trusted hosts as follows: • Click the + sign to expand options under **Trusted Hosts** and select **Add** to display the following screen. Enter the following fields: o **IP Address** – IP Address of the Nuance SIP Proxy or the Nuance MediaServer. o **Host** – Select the IP Address of the Avaya SES server where the trusted host is to be configured. In this configuration, it should match the value set in **Section 3.3**. o **Comment** – Any Descriptive Information. • Click **Add**. **Note**: Repeat this step to add additional trusted hosts. 

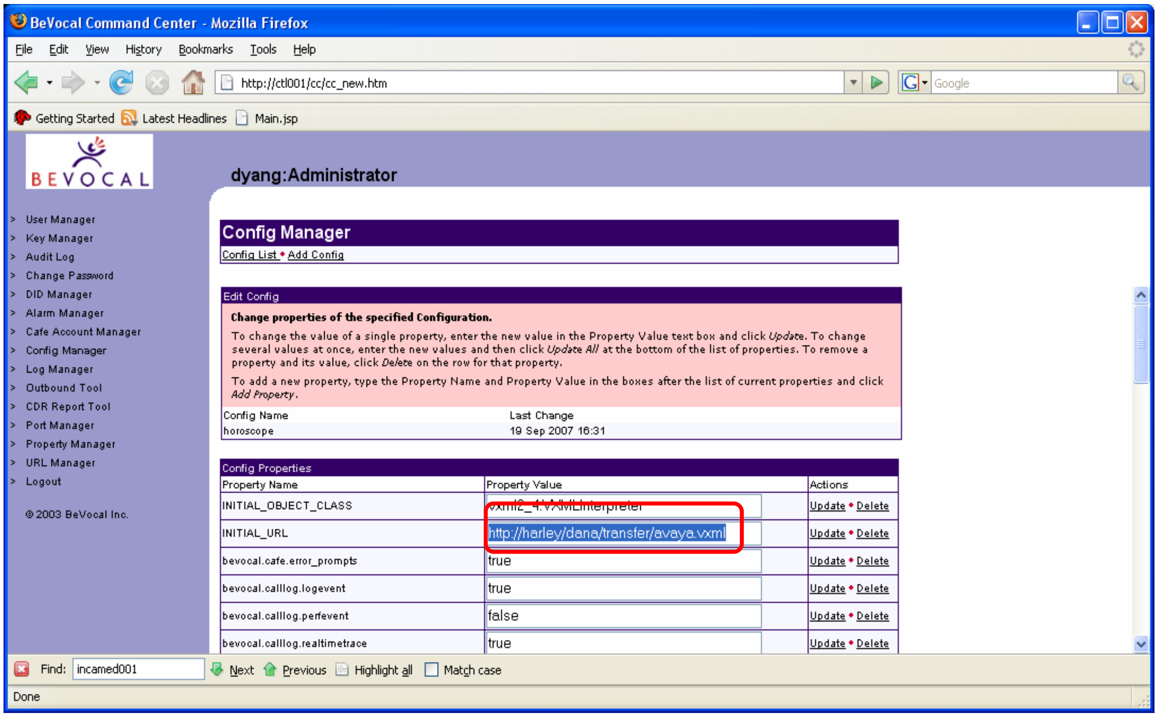
Step	Description
8.	This is the confirmation screen presented after adding trusted hosts associated with both primary and backup sites. Click **Update** in the left pane to effect configuration in Avaya SES. 

5. Configure the Nuance Hosted IVR

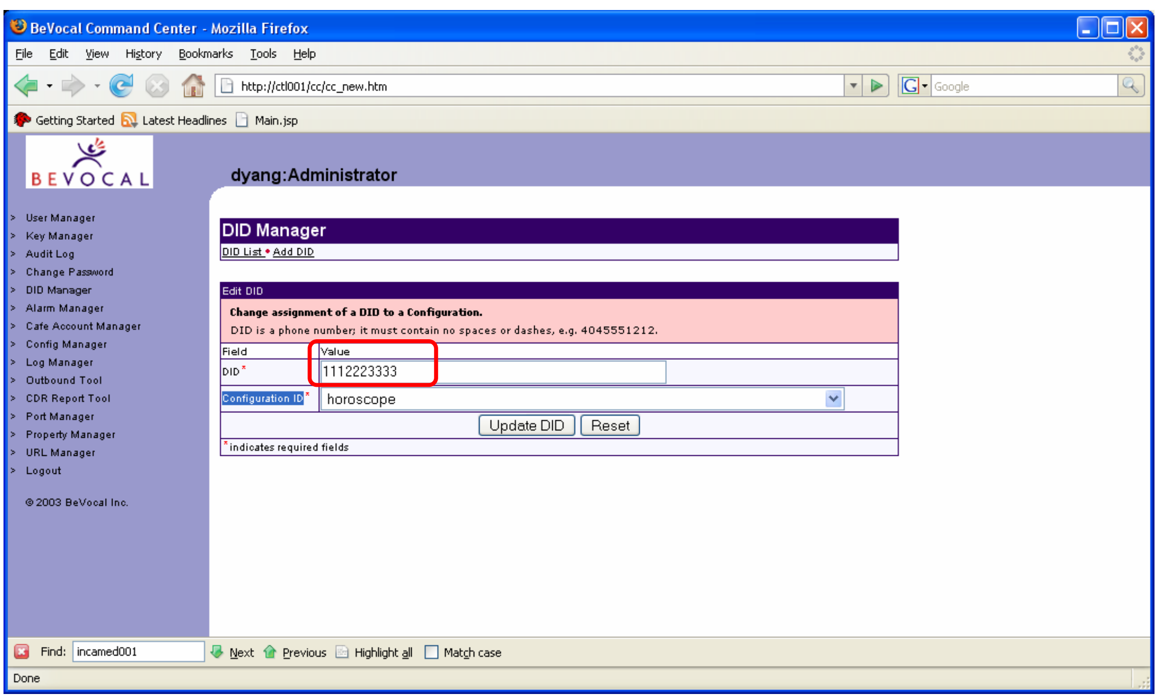
This section describes the steps for configuring the Nuance Hosted IVR. This section assumes that the Nuance SIP Proxy and the Nuance MediaServer software at each site (Primary and Backup) are already installed and their IP addresses are set. Additionally, the Nuance solution is a hosted solution and it is assumed is that the Nuance SIP Proxy and the Nuance MediaServer are properly configured. Voice applications hosted by the Nuance Hosted IVR can be accessed by valid DID numbers. This document provides configuration steps for associating a DID with a voice application. Configuration steps described in this section apply only to the fields where a value needs to be modified or entered. Default values are used for all other fields. Refer to [5] for additional details.

Step	Description
1.	Using a Web browser, navigate to http://www.bevocal.com and login with proper credentials to add a new VXML application.
2.	Select **Config Manager** in the left pane. On the **Config Manager** screen, enter a valid name in the **Config Name** field and click **Create Config**. 

<table>
<tr><th>Step</th><th>Description</th></tr>
<tr><td>3.</td><td>At the next screen, set the INITIAL_URL field to the voice application URL, then scroll down and click Add (not shown).



</td></tr>
<tr><td>4.</td><td>Select DID Manager in the left pane. Enter the number to reach the application configured in Step 2 and click Update DID.



</td></tr>
</table>

6. Interoperability Compliance Testing

The focus of the interoperability compliance testing was primarily to exercise the Nuance Hosted IVR solution using speech and DTMF. Tests were done to verify that the Nuance Hosted IVR is able to recognize speech and DTMF commands and take appropriate action for calls originating from Avaya SIP, H.323, digital, analog and PSTN telephones using Avaya SES and Avaya Communication Manager.

6.1. General Test Approach

The general test approach was to place calls from any phone to establish a call into the Nuance Hosted IVR and exercise the supported features. The main objectives were to verify that:

- The Nuance Hosted IVR recognizes DTMF tones and takes appropriate action based upon the option entered.
- The Nuance Hosted IVR recognizes speech and takes appropriate action based upon the option entered.
- The Nuance Hosted IVR successfully executes a blind transfer.
- The Nuance Hosted IVR successfully terminates the call.

For serviceability testing, failures such as cable disconnections and hardware resets were applied. For redundancy testing, failures of the Primary Nuance SIP Proxy and Primary Nuance MediaServer were created to verify that calls failed over to the Backup Nuance SIP Proxy/MediaServer pair.

6.2. Test Results

The test objectives of **Section 6.1** were verified. For serviceability testing, the Nuance Hosted IVR operated properly after recovering from failures such as cable disconnects, and resets of the Nuance Hosted IVR, Avaya SES server, and Avaya Communication Manager. For redundancy testing, the calls were successfully handled by the redundant Nuance SIP Proxy/MediaServer pair. Calls placed into the Nuance Hosted IVR completed successfully.

The following observation was made during testing:

- The Nuance MediaServer needs to be rebooted after its Ethernet cable is disconnected and re-connected.

Nuance Communications expects to resolve the above issue in a future release. Contact Nuance Communications (www.bevocal.com) for further updates.

7. Support

For technical support on the Nuance Hosted IVR solution, consult the support pages at http://www.bevocal.com or contact Nuance Communications technical support at:

- Phone: (650) 210-8600
- E-mail: support@bevocal.com

8. Conclusion

These Application Notes describe a solution comprised of Avaya Communication Manager 4.0.1, Avaya SIP Enablement Services 4.0, and Nuance Communications MediaServer 9.1.1 and SIP Proxy 0.8.14. The Nuance Communications SIP Proxy handles all the SIP-related signaling and the Nuance Communications MediaServer is VoIP software for executing Voice Extensible Markup Language (VXML) applications in a hosted environment. The function of VXML is application-specific and provides an IVR-driven menu for access to these applications. For the purpose of compliance testing, several VXML IVR applications provided by Nuance Communications were used to exercise SIP call flows with SIP and non-SIP telephones. The compliance testing was successful with the exception of one minor issue noted in **Section 6.2**.

9. Additional References

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

[1] *Administrator Guide for Avaya Communication Manager*, Issue 3.1, February 2007, Document Number 03-300509.
[2] *Administration for Network Connectivity for Avaya Communication Manager*, Issue 12, February 2007, Document Number 555-233-504.
[3] *SIP Support in Avaya Communication Manager Running on the Avaya S8300, S8400, S8500 series, and S8700 Series Media Server*, Issue 7, May 2007, Document Number 555-245-206.
[4] *Installing and Administering SIP Enablement Services*, Issue 4, May 2007, Document Number 03-600768.
[5] *Nuance On-Demand Hosting Service Overview*, Version 9.1.1.

Additional product documentation for Nuance Communications products may be found at http://www.bevocal.com/.

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.