Avaya Solution & Interoperability Test Lab

Application Notes for Avaya Voice Portal 5.0 with Dialogs Unlimited ivSurvey v2.0 Speech Application Engine - Issue 1.0

Abstract

These Application Notes describe the steps for configuring Avaya Voice Portal 5.0 with Dialogs Unlimited ivSurvey v2.0. Dialogs Unlimited has developed the ImmediateVoice platform and technology which enables the development and operation of Speech Applications based on existing or new e-commerce web sites. ivSurvey utilizes the power of ImmediateVoice with regard to extracting interactions from the web pages, converting them into customer speech dialog and converting the speech back to the required data field inputs, which in combination with Avaya Voice Portal advanced CCXML based call control features delivers a state-of-the-art solution for Customer or Employee Satisfaction Surveys, Performance Management, Opinion Polls and Market Research. ivSurvey presents questionnaires to customers/clients/employees via web and speech. Avaya Voice Portal is a speech-enabled interactive voice response system that allows enterprises to provide multiple self and assisted service resources to their customers in a flexible and customizable manner.

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

1. Introduction

These Application Notes describe the steps for configuring Avaya Voice Portal 5.0 with Dialogs Unlimited ivSurvey v2.0 and Avaya Aura™ SIP Enablement Services based on SIP connectivity. Dialogs Unlimited has developed the ImmediateVoice platform and technology which enables the development and operation of Speech Applications based on existing or new e-commerce web sites. ImmediateVoice is a web-based speech engine including web-based speech application design tools making it possible to design, develop and run speech applications based on existing self-service websites. ImmediateVoice has 2 basic components: Voice Transcoder (VT) that interprets webpages and converts to XML, and Dialogs Manager (NextIDM) that generates VXML 2.1 based on inputs from VT. The NextIDM component of ImmediateVoice interfaces with the Avaya Voice Portal via http and by submitting VXML2.1 scripts.

ivSurvey utilizes the power of ImmediateVoice with regard to extracting interactions from the web pages, converting them into customer speech dialog and converting the speech back to the required data field inputs, which in combination with Avaya Voice Portal advanced CCXML based call control features, delivers a state-of-the-art solution for Customer or Employee Satisfaction Surveys, Performance Management, Opinion Polls and Market Research.
Avaya Voice Portal is a web-based and speech enabled interactive voice response system that is configured as an adjunct system on the SIP Enablement Services Home Server. A SIP Enablement Services adjunct is an entity that provides services to the SIP Enablement Services Home Server via a SIP interface. The Avaya Voice Portal system is composed of a Voice Portal Management System (VPMS) server, one or more Media Processing Platform (MPP) servers, and typically includes web-based application servers that provide automated speech recognition and text-to-speech services. The MPP provides media processor resources and is the proxy interface to the web-based application servers. In these Application Notes, the VPMS with MPP uses the SIP protocol to communicate with Communication Manager via the SIP Enablement Services Home Server. The VPMS provides centralized management for the MPP(s) and provides a web interface for administering the Avaya Voice Portal system. For Avaya Voice Portal, SIP Enablement Services delivers invites to the VPMS. ivSurvey utilises Avaya Voice Portal voice axis capability.

1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying interoperability of the Dialogs Unlimited ivSurvey 2.0 with Avaya Voice Portal 5.0 based on SIP connectivity. The focus of testing is VXML (CCXML) interoperability and support of major call functions like making outbound call, extracting interactions from web pages, converting them into customer speech dialog and converting the speech back to the required data field inputs, presenting questionnaires to customers and call transfer.

1.2. Support

Technical support for the Avaya products can be obtained from Avaya. See the support link at support.avaya.com for contact information. Technical support for the ivSurvey products can be obtained from Dialogs Unlimited. See www.dialogsunlimited.com for contact information.

2. Reference Configuration

The sample configuration used in these Application Notes is shown in **Figure 1** and consists of several components:

- Avaya Voice Portal provides interactive voice response services to inbound callers. Avaya Voice Portal consists of one or more Media Processing Platform (MPP) servers and a Voice Portal Management System (VPMS) server.
- Avaya Aura™ Communication Manager provides the enterprise voice communication services. In this sample configuration, Communication Manager runs on an Avaya S8500C Server. This solution is extensible to other Avaya S8xxx Servers.
- Avaya Media Gateway provides the physical interfaces and resources for enterprise voice communications. In this sample configuration, an Avaya G650 Media Gateway is used. This solution is extensible to other Avaya Media Gateways.
- SIP Enablement Services creates a SIP communication network supporting telephony. In this sample configuration, SIP Enablement Services runs on an Avaya S8510 Server.
- Avaya phones are represented with Avaya 1600-Series, 4600-Series and 9600-Series IP Telephones running H.323 software or SIP software.
- The Nuance Speech Server consists of Nuance OpenSpeech Recognizer and Nuance RealSpeak. Avaya Voice Portal uses the Speech Server for Text-To-Speech (TTS) and Automatic Speech Recognition (ASR) capabilities.
- Dialogs Unlimited ivSurvey Speech Application Engine comes as a complete platform supporting Speech Application Creation, Service Monitoring, Content Management and Speech Engine.

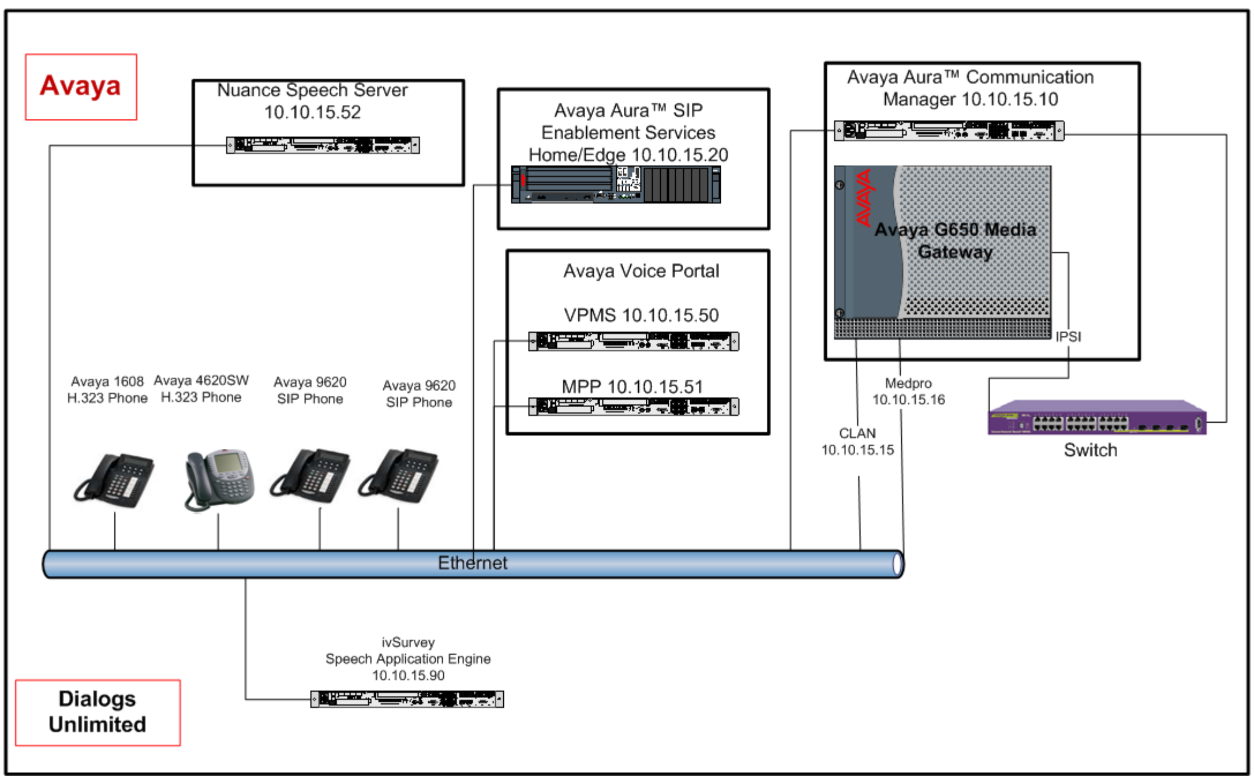


Figure 1: Sample Configuration

3. Equipment and Software Validated

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

Equipment	Software
Avaya S8500C Media Server with an Avaya G650 Media Gateway	Avaya Aura™ Communication Manager 5.2 (R015x.02.0.947.3-17250)
Avaya Voice Portal System VPMS Server MPP Server	Avaya Voice Portal 5.0 VPMS 5.0.0.0.4602 MPP 5.0.0.0.4603
Avaya S8510 SIP Enablement Service (SES) Home/Edge combined Server	Avaya Aura™ SIP Enablement Services 5.2 SES-5.2.0.0-947.3b
Avaya 4620SW H.323 Phone	H.323 Release 2.9
Avaya 1608 H.323 Phone	H.323 Release 1.2
Avaya 9620 SIP Phone	Avaya one-X Deskphone Edition SIP Release 2.4.1
Dialogs Unlimited ivSurvey Speech Application Engine	ivSurvey Speech Application Engine 2.0
Nuance Speech Server (co-resident) Nuance RealSpeak Nuance OpenSpeech Recognizer	NSS 5.0.2 RealSpeak4.5 sp2 NRec9.0.3

4. Configure Avaya Aura™ Communication Manager

This section describes the administration steps for Communication Manager in support of the sample configuration described in these Application Notes. The steps are performed from the Communication Manager System Access Terminal (SAT) interface. These Application Notes assume that basic Communication Manager administration, including C-LAN, Media Processor, etc., has already been performed.

4.1. Communication Manager System Parameters

This section reviews the Communication Manager system parameters and features that are required for the sample configuration described in these Application Notes.

1. Enter the **display system-parameters customer-options** command.
   On **Page 1** of the form, verify that **Maximum Off-PBX Telephones - OPS:** is sufficient for the number of expected off-pbx stations.

```
display system-parameters customer-options                      Page   1 of  10
                                OPTIONAL FEATURES

     G3 Version: V15                              Software Package: Standard
       Location: 1                             RFA System ID (SID): 1
       Platform: 12                            RFA Module ID (MID): 1

                                                              USED
                                 Platform Maximum Ports: 44000 92
                                       Maximum Stations: 36000 15
                               Maximum XMOBILE Stations: 0     0
                     Maximum Off-PBX Telephones - EC500: 0     0
                     Maximum Off-PBX Telephones -   OPS: 100   0
                     Maximum Off-PBX Telephones - PBFMC: 0     0
                     Maximum Off-PBX Telephones - PVFMC: 0     0
                     Maximum Off-PBX Telephones - SCCAN: 0     0
```

   On **Page 2** of the form, verify that **Maximum Administered SIP Trunks** number is sufficient for the number of expected SIP trunks.

```
display system-parameters customer-options                      Page   2 of  10
                                OPTIONAL FEATURES

IP PORT CAPACITIES                                            USED
                     Maximum Administered H.323 Trunks: 100   0
           Maximum Concurrently Registered IP Stations: 18000 8
             Maximum Administered Remote Office Trunks: 0     0
Maximum Concurrently Registered Remote Office Stations: 0     0
              Maximum Concurrently Registered IP eCons: 0     0
  Max Concur Registered Unauthenticated H.323 Stations: 0     0
                  Maximum Video Capable H.323 Stations: 0     0
                   Maximum Video Capable IP Softphones: 0     0
                       Maximum Administered SIP Trunks: 100   48
  Maximum Administered Ad-hoc Video Conferencing Ports: 0     0
   Maximum Number of DS1 Boards with Echo Cancellation: 0     0
                             Maximum TN2501 VAL Boards: 10    0
                     Maximum Media Gateway VAL Sources: 0     0
           Maximum TN2602 Boards with 80 VoIP Channels: 128   0
          Maximum TN2602 Boards with 320 VoIP Channels: 128   1
   Maximum Number of Expanded Meet-me Conference Ports: 0     0
```

On **Page 3** of the **system-parameters customer-options** form, verify that the bolded field in the following screenshot is set to **y**.

```
display system-parameters customer-options                      Page   3 of  10
                                OPTIONAL FEATURES

    Abbreviated Dialing Enhanced List? n          Audible Message Waiting? n
        Access Security Gateway (ASG)? n              Authorization Codes? n
        Analog Trunk Incoming Call ID? n                       CAS Branch? n
  A/D Grp/Sys List Dialing Start at 01? n                         CAS Main? n
Answer Supervision by Call Classifier? n                Change COR by FAC? n
                                  ARS? y  Computer Telephony Adjunct Links? n
                 ARS/AAR Partitioning? y   Cvg Of Calls Redirected Off-net? n
          ARS/AAR Dialing without FAC? y                       DCS (Basic)? y
          ASAI Link Core Capabilities? n                 DCS Call Coverage? n
          ASAI Link Plus Capabilities? n                DCS with Rerouting? n
       Async. Transfer Mode (ATM) PNC? n
  Async. Transfer Mode (ATM) Trunking? n    Digital Loss Plan Modification? n
              ATM WAN Spare Processor? n                          DS1 MSP? n
                                 ATMS? n             DS1 Echo Cancellation? n
                  Attendant Vectoring? n
```

On **Page 4** of the **system-parameters customer-options** form, verify that **IP Trunks** is set to **y**. If **Media Encryption Over IP** is set to **y**, it will allow for SRTP media encryption on calls routed to Avaya Voice Portal system, in this case the respective security certificates need to be installed. In this sample configuration, it is set to **n**.

```
display system-parameters customer-options                      Page   4 of  10
                                OPTIONAL FEATURES

   Emergency Access to Attendant? y                            IP Stations? y
           Enable 'dadmin' Login? y
           Enhanced Conferencing? y                      ISDN Feature Plus? n
                  Enhanced EC500? y      ISDN/SIP Network Call Redirection? n
    Enterprise Survivable Server? n                        ISDN-BRI Trunks? n
       Enterprise Wide Licensing? n                               ISDN-PRI? y
              ESS Administration? n             Local Survivable Processor? n
          Extended Cvg/Fwd Admin? n                   Malicious Call Trace? n
     External Device Alarm Admin? n               Media Encryption Over IP? n
  Five Port Networks Max Per MCC? n    Mode Code for Centralized Voice Mail? n
                Flexible Billing? n
   Forced Entry of Account Codes? n                Multifrequency Signaling? y
      Global Call Classification? y        Multimedia Call Handling (Basic)? n
             Hospitality (Basic)? y     Multimedia Call Handling (Enhanced)? n
 Hospitality (G3V3 Enhancements)? n              Multimedia IP SIP Trunking? n
                       IP Trunks? y

           IP Attendant Consoles? n
```

On **Page 5** of the **system-parameters customer-options** form, verify that **Private Networking** and **Uniform Dialing Plan** are set to **y**.

```
display system-parameters customer-options                      Page   5 of  10

                                OPTIONAL FEATURES

                Multinational Locations? n             Station and Trunk MSP? n
 Multiple Level Precedence & Preemption? n      Station as Virtual Extension? n
                     Multiple Locations? n
                                             System Management Data Transfer? n
          Personal Station Access (PSA)? n               Tenant Partitioning? n
                        PNC Duplication? n       Terminal Trans. Init. (TTI)? n
                   Port Network Support? y               Time of Day Routing? n
                        Posted Messages? n       TN2501 VAL Maximum Capacity? y
                                                        Uniform Dialing Plan? y
                     Private Networking? y     Usage Allocation Enhancements? y
               Processor and System MSP? n
                     Processor Ethernet? y                Wideband Switching? n
                                                                    Wireless? n
                          Remote Office? n
          Restrict Call Forward Off Net? y
                  Secondary Data Module? y
```

2. Enter the **display feature-access-codes** command.
   On **Page 1** of the form, verify that **Auto Alternate Routing (AAR) Access Code** is set, in this sample configuration, it is set to ***8**.

```
display feature-access-codes                                    Page   1 of   6
                               FEATURE ACCESS CODE (FAC)
         Abbreviated Dialing List1 Access Code:
         Abbreviated Dialing List2 Access Code:
         Abbreviated Dialing List3 Access Code:
Abbreviated Dial - Prgm Group List Access Code:
                      Announcement Access Code:
                       Answer Back Access Code:
                         Attendant Access Code:
      Auto Alternate Routing (AAR) Access Code: *8
    Auto Route Selection (ARS) - Access Code 1:      Access Code 2:
             Automatic Callback Activation:        Deactivation:
Call Forwarding Activation Busy/DA:       All:        Deactivation:
   Call Forwarding Enhanced Status:       Act:        Deactivation:
                         Call Park Access Code:
                       Call Pickup Access Code:
CAS Remote Hold/Answer Hold-Unhold Access Code:
                  CDR Account Code Access Code:
                        Change COR Access Code:
                   Change Coverage Access Code:
            Conditional Call Extend Activation:        Deactivation:
                   Contact Closure   Open Code:          Close Code:
```

4.2. Configure SIP trunk to SIP Enablement Services

This section describes the administration on Communication Manager that need be performed to set up a SIP trunk to SIP Enablement Services Home Server.

1. Enter command **change node-name ip** to assign the node names for the C-LAN board and SIP Enablement Services Home Server. The following values were entered for the **Name** and **IP Address** fields for this configuration example: **ses** and **10.10.15.20** for the SIP Enablement Services Home Server, **clan** and **10.10.15.15** for the C-LAN board, **medpro** and **10.10.15.16** for the IP Media Resource board.

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

                                 IP NODE NAMES

   Name              IP Address

clan                10.10.15.15
default             0.0.0.0
gateway             10.10.15.1
medpro              10.10.15.16
procr               10.10.15.10
ses                 10.10.15.20
```

2. Enter command **change ip-codec-set n**, where **n** is the IP codec set number used for the SIP connectivity to Avaya Voice Portal. For this sample configuration, IP codec set **1** is used. Enter **G.711MU** as the Audio Codec to be supported.

```
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
```

3. Enter command **change ip-network-region n**, where **n** is the IP network region. **1** is used for this sample configuration (assume all servers residing on the same network region). Enter **du.rnd.avaya.com** for the **Authoritative Domain** field, **1** for the **Codec Set** field (refer to **Step 2**), and **yes** for the **Intra-region IP-IP Direct Audio** and **Inter-region IP-IP Direct Audio** fields. The IP-IP Direct Audio setting ensures the most efficient use of the media processor resources.

```
change ip-network-region 1                                     Page   1 of  19
                               IP NETWORK REGION
  Region: 1
Location: 1       Authoritative Domain: du.rnd.avaya.com
    Name: ImmediateVoice
MEDIA PARAMETERS                Intra-region IP-IP Direct Audio: yes
      Codec Set: 1              Inter-region IP-IP Direct Audio: yes
   UDP Port Min: 2048                      IP Audio Hairpinning? n
   UDP Port Max: 3329
DIFFSERV/TOS PARAMETERS                  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
 Idle Traffic Interval (sec): 20
   Keep-Alive Interval (sec): 5
            Keep-Alive Count: 5
```

4. Enter command **add signaling-group n**, where **n** is an available signaling group number. Signaling group **1** is used for the sample configuration. Enter the following fields:
   - **Group Type:** set to **sip**
   - **Transport Method:** set to **tls**
   - **Near-end Node Name:** set to **clan**
   - **Far-end Node Name:** set to **ses**
   - Set **Near-end Listen Port** and **Far-end Listen Port** to be **5061**
   - **Far-end Network Region:** set to **1**
   - **Far-end Domain:** set to **du.rnd.avaya.com**, match the **Authoritative Domain** entered in **Step 3**.
   - To support shuffling, set **Direct IP-IP Audio Connections** to be **y**.

```
add signaling-group 1                                       Page   1 of   1
                                SIGNALING GROUP
 Group Number: 1              Group Type: sip
                       Transport Method: tls
  IMS Enabled? n
   Near-end Node Name: clan                   Far-end Node Name: ses
 Near-end Listen Port: 5061                 Far-end Listen Port: 5061
                                         Far-end Network Region: 1
Far-end Domain: du.rnd.avaya.com
                                            Bypass If IP Threshold Exceeded? n
         DTMF over IP: rtp-payload           Direct IP-IP Audio Connections? y
Session Establishment Timer(min): 3                    IP Audio Hairpinning? y
         Enable Layer 3 Test? n                    Direct IP-IP Early Media? n
H.323 Station Outgoing Direct Media? n           Alternate Route Timer(sec): 6
```

5. Enter command **add trunk-group n**, where **n** is an available trunk group number. Trunk group **1** is used for the sample configuration. Enter the following fields:
   - **Group Type:** set to **sip**
   - **Group Name:** set to a descriptive name
   - **TAC:** set to an available trunk access code
   - **Service Type:** set to **tie**
   - **Signaling Group:** set to **1**
   - **Number of Members:** set to **48** for this sample configuration.

```
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: 100
   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: 48
```

On **Page 3,** verify **Numbering Format** is set to **public**.

```
add trunk-group 1                                           Page   3 of  21
TRUNK FEATURES
          ACA Assignment? n            Measured: none
                                                         Maintenance Tests? y



                     Numbering Format: public
                                               UUI Treatment: service-provider

                                                Replace Restricted Numbers? n
                                               Replace Unavailable Numbers? n


 Show ANSWERED BY on Display? y
```

4.3. Configure Call Routing to the Avaya Voice Portal System

This section describes the administration on Communication Manager that need be performed to set up call routing to the Avaya Voice Portal system.

1. Enter command **change route-pattern n**, where **n** is an available route pattern number. Route pattern **1** is used for the sample configuration. For the **Pattern Name** field, enter a descriptive name for the route pattern. Enter the trunk number assigned in **Section 4.2 Step 5** for the **Grp No** field, and assign an appropriate Facility Restriction Level (**0** is the least restrictive) for the **FRL** field.

```
change route-pattern 1                                          Page   1 of   3
                    Pattern Number: 1   Pattern Name: TO VP-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                                                                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 M 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
```

2. Enter command **change uniform-dialplan n**, where **n** is the leading digits to be assigned for routing calls to Avaya Voice Portal. In this sample configuration, the extension range **222x** is set aside for assigning extensions to various self-services applications served by Avaya Voice Portal. To add a new entry to the table, enter the following fields:
   - **Matching Pattern:** is set to **222** (Leading digits dialed)
   - **Len:** is set to **4** (The length of the extension)
   - **Del:** is set to **0** (No digit deletion)
   - **Net:** is set to **aar** (Sent to the Automatic Alternate Routing table)
   - **Conv:** is set to **n** (No conversion)

```
 change uniform-dialplan 2                                        Page   1 of   2
                         UNIFORM DIAL PLAN TABLE
                                                              Percent Full: 0

   Matching                      Insert             Node
   Pattern       Len Del         Digits      Net Conv Num
  222              4   0                     aar  n
```

3. Enter command **change aar analysis *n***, where ***n*** matches the same leading digits used in **Step 2** above. The aar digit analysis table matches the leading digits dialed (as per the Uniform Dial Plan) to the desired route pattern. To add a new entry to the table, enter the following fields:
   - **Dialed String:** is set to **222** (Leading digits dialed)
   - **Total Min:** is set to **4** (Minimum number of digits expected)
   - **Total Max:** is set to **4** (Maximum number of digits expected)
   - **Route Pattern:** is set to **1** (Refer to **Step 1**)
   - **Call Type:** is set to **aar** (Automatic Alternate Routing)

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

          Dialed           Total     Route     Call   Node   ANI
          String         Min  Max   Pattern    Type   Num    Reqd
    222                  4    4      1         aar           n
```

5. Configure Avaya Voice Portal

This section details the administration on Avaya Voice Portal that must be performed to setup SIP connectivity to Communication Manager using SIP Enablement Services. For additional information on how to configure Avaya Voice Portal with web-based application services to provide voice and speech response self-service applications, please consult references **[2]** and **[3]** of these Application Notes.

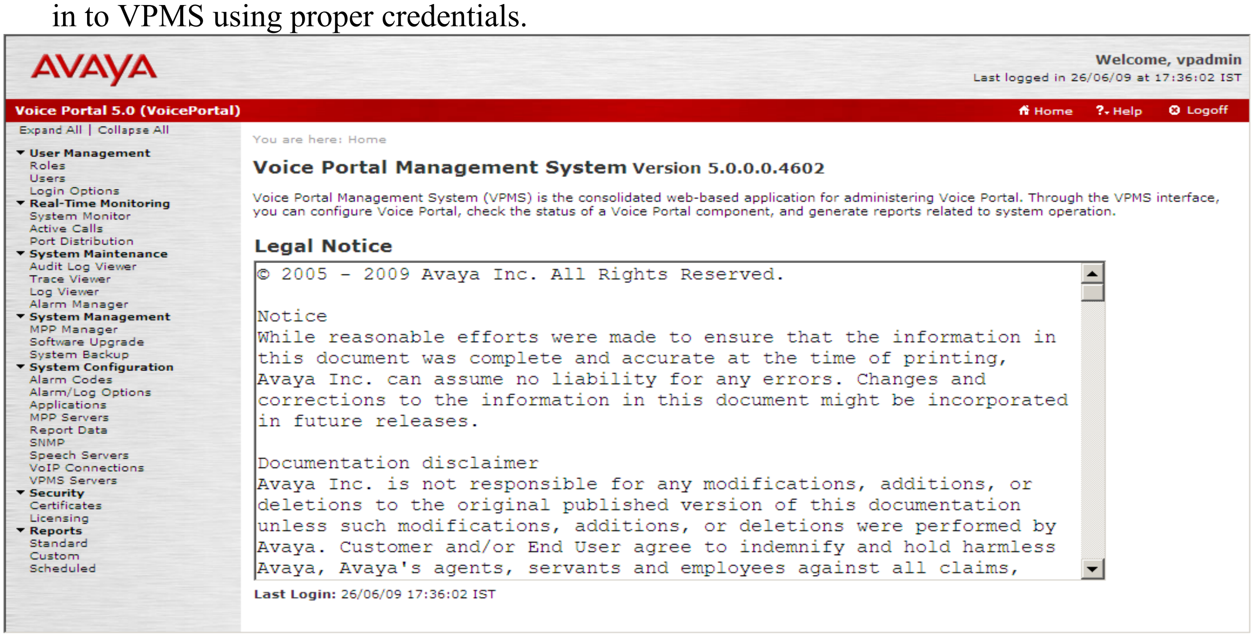
5.1. Verify Avaya Voice Portal Licenses

Avaya Voice Portal utilizes primary and secondary WebLM servers for implementation of feature licenses. If activation of additional features or ports is required, an updated license file must be obtained and installed on the WebLM server. The license file contains details about the features and number of ports purchased. To obtain an updated license file, please contact the Avaya Authorized Sales representative. To configure TLS as the proxy transport for SIP signaling between Avaya Voice Portal, SIP Enablement Services and Communication Manager, a security certificate must be installed on Avaya Voice Portal and a complementary trusted certificate on the SIP Enablement Services Home server.

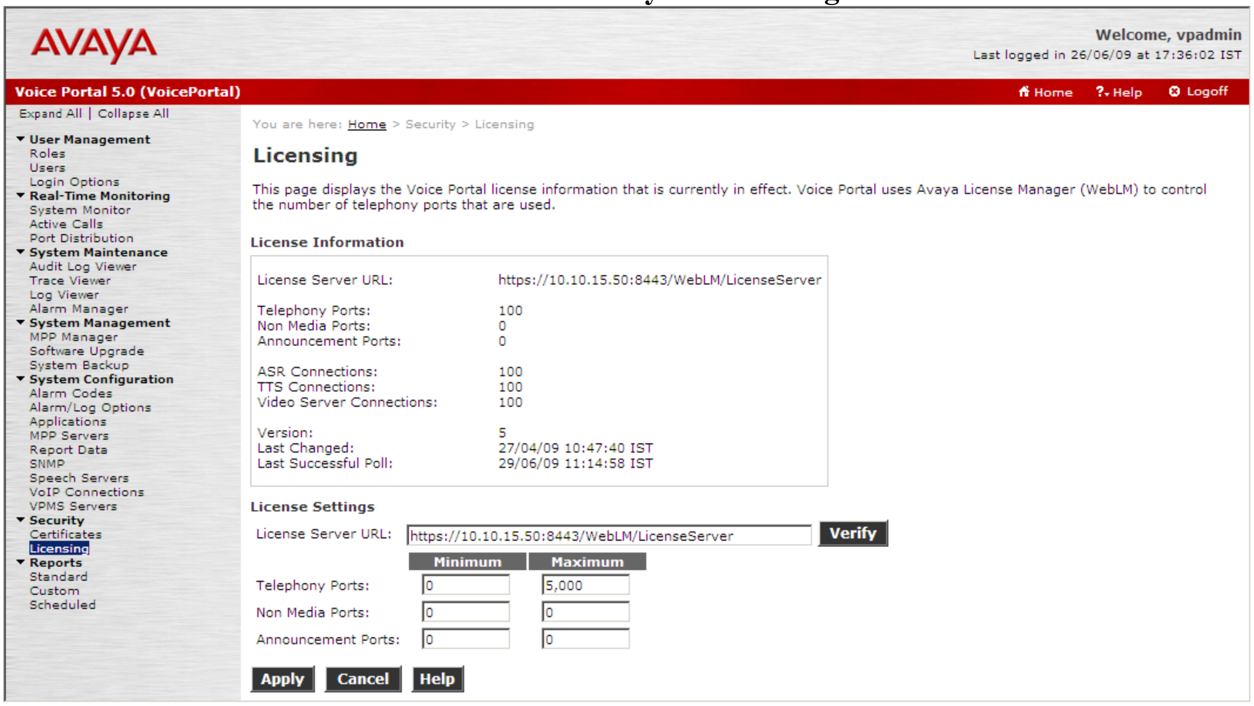
Note: For simplicity reasons, in this sample application, TCP is used as the proxy transport for SIP signaling between Avaya Voice Portal and SIP Enablement Services. Thus the security certificates do not need to be installed (on Avaya Voice Portal and a complementary trusted certificate on the Avaya SES Home/Edge server). For more information about security on Avaya Voice Portal, please consult reference **[6]** in **Section 11** of these Application Notes.

The following section illustrates how to verify the licenses.

1. Access the VPMS web interface by typing the following URL on a web browser and then pressing <**enter**>: "**http://<hostname or IP address of VPMS server>/VoicePortal**" Log in to VPMS using proper credentials.



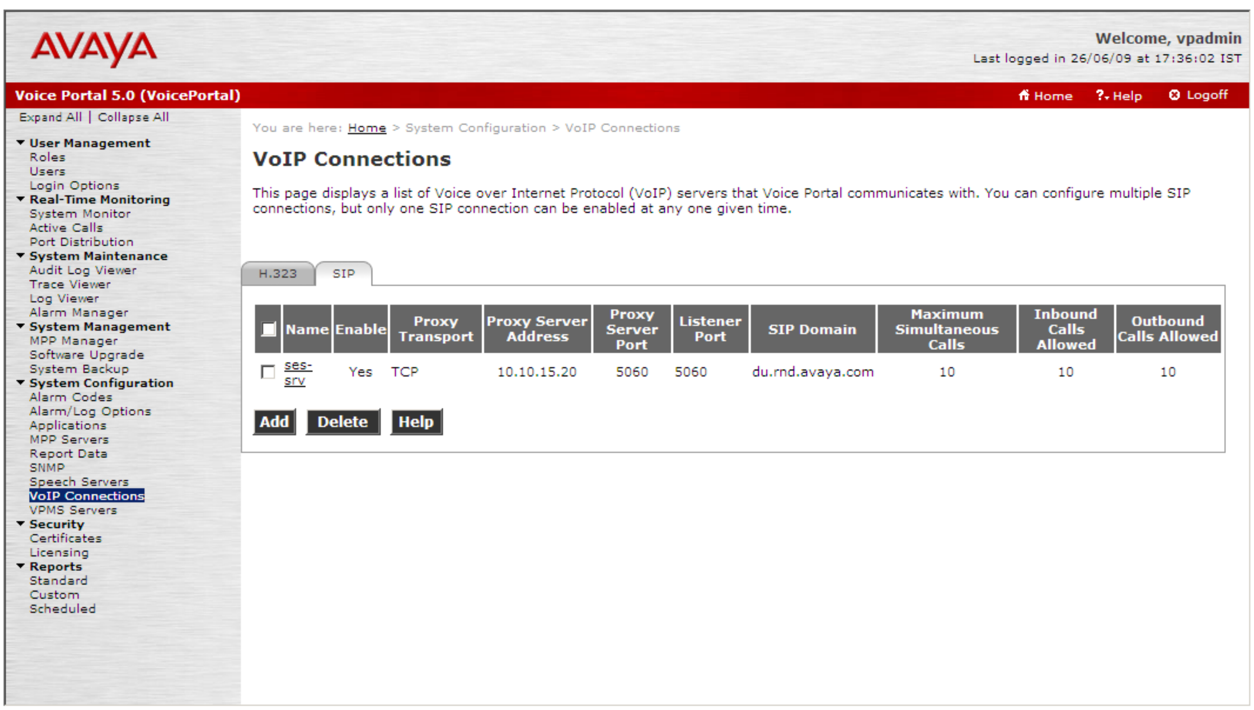
2. In the left pane of the VPMS window that appears, click on **Expand All** to show all options available under each section. Click on **Security → Licensing**.



5.2. Configure a SIP Connection

The following section illustrates how to configure a SIP connection on the Avaya Voice Portal.

1. Click on **System Configuration → VoIP Connections**. Click on the **SIP** tab to continue.



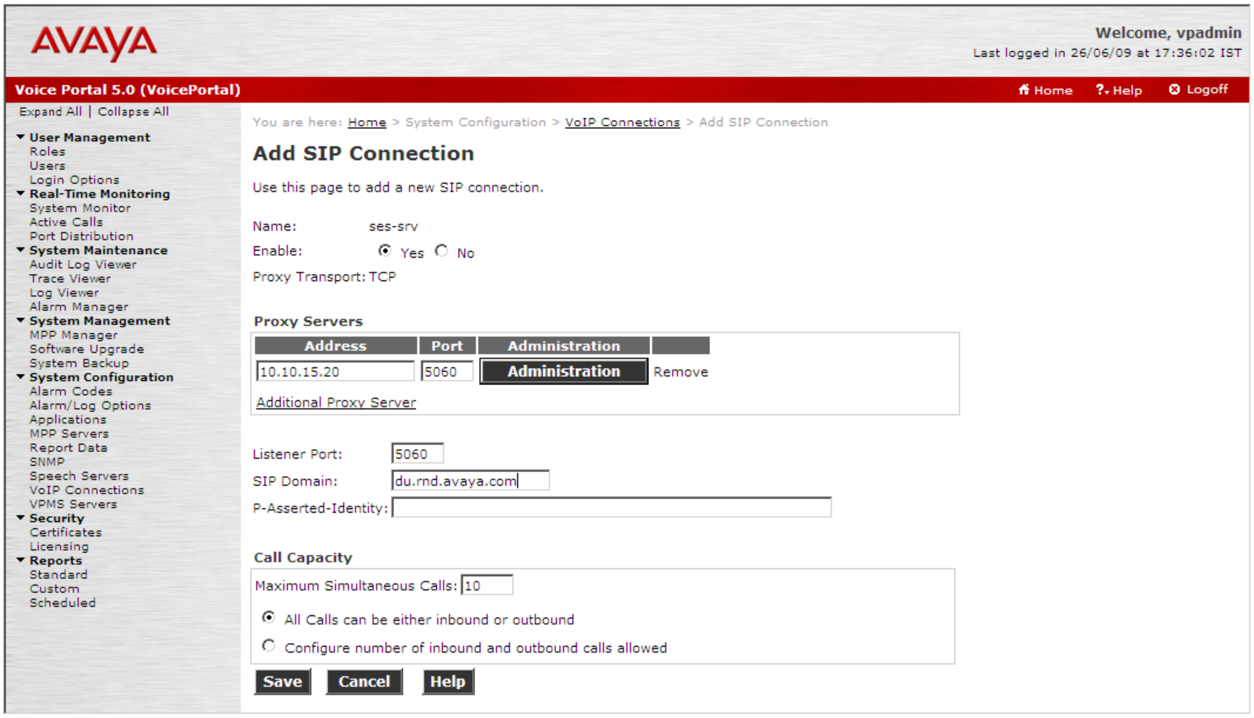
2. Click on the **Add** button to add a SIP connection. Enter the following fields to add a new SIP connection:

- **Name :** set to **ses-srv** (A unique identifier for the SIP connection)
- **Proxy Transport :** set to **TCP** (use unencrypted connections for simplicity)
- **Proxy Server Address :** set to **10.10.15.20** (Full qualified domain name or an IP address of SIP Enablement Services)

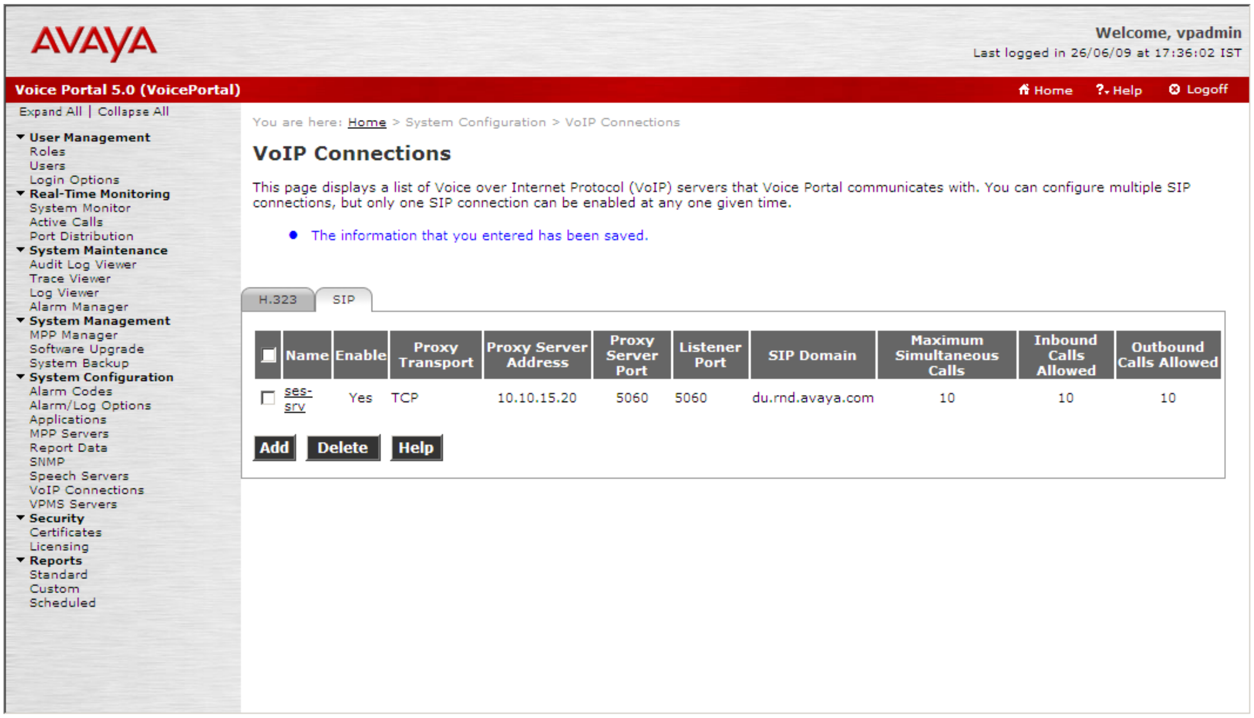
Note: The **Proxy Server Port** default for TCP is 5060, and the field changes accordingly when the proxy transport is selected.

3. Click on the **Continue** button to continue the SIP Connection configuration. Enter the following fields :
   - **SIP Domain :** set to **du.rnd.avaya.com** (must match the domain configured on **Section 4.2 Step 3** and **Step 4**)
   - **Maximum Simultaneous Calls :** set to **10** (The maximum number of calls that this SIP trunk can handle at one time)

   Click on the **Save** button.



4. The screen below illustrates that the SIP connection has been successfully configured and saved.

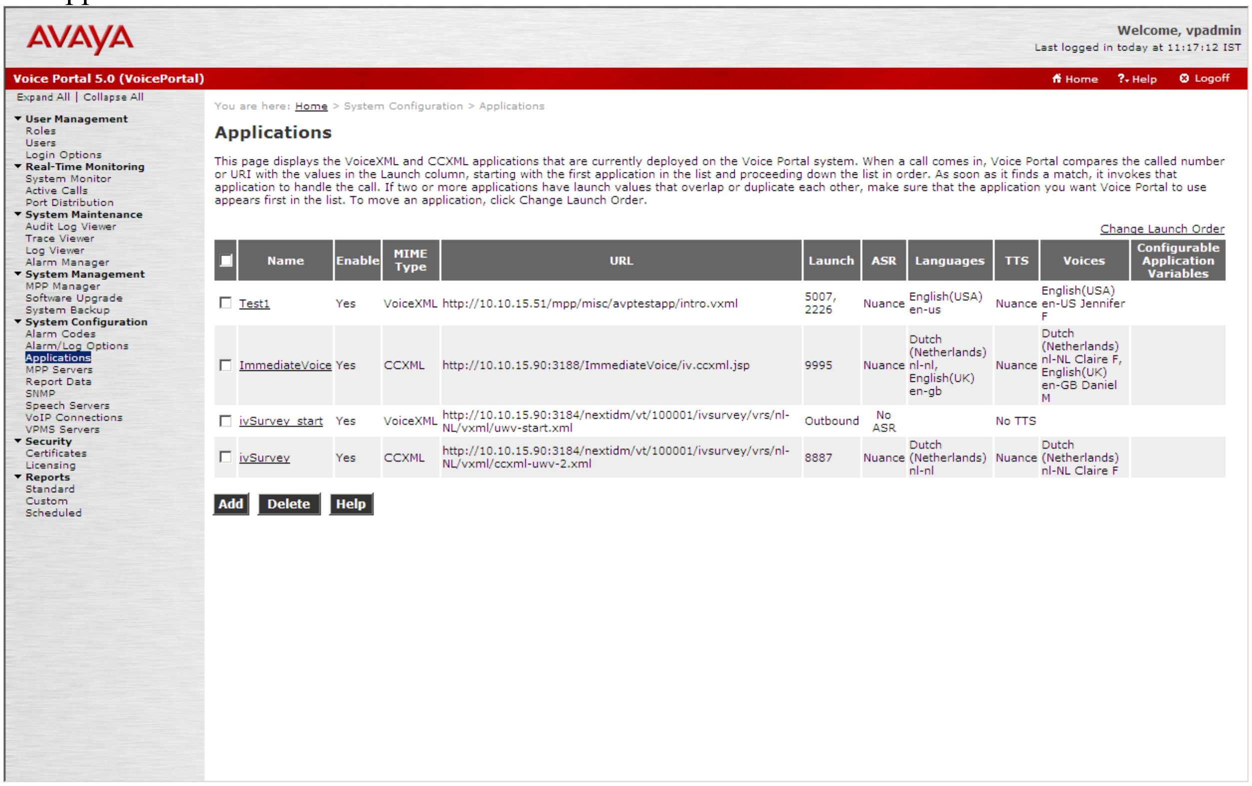


5.3. Configure Applications for ivSurvey

The following section illustrates how to configure a test VXML application to verify the SIP connection on Avaya VPMS. Please consult reference **[2]** in **Section 11** to perform the following configuration tasks prior to configuring the test application:

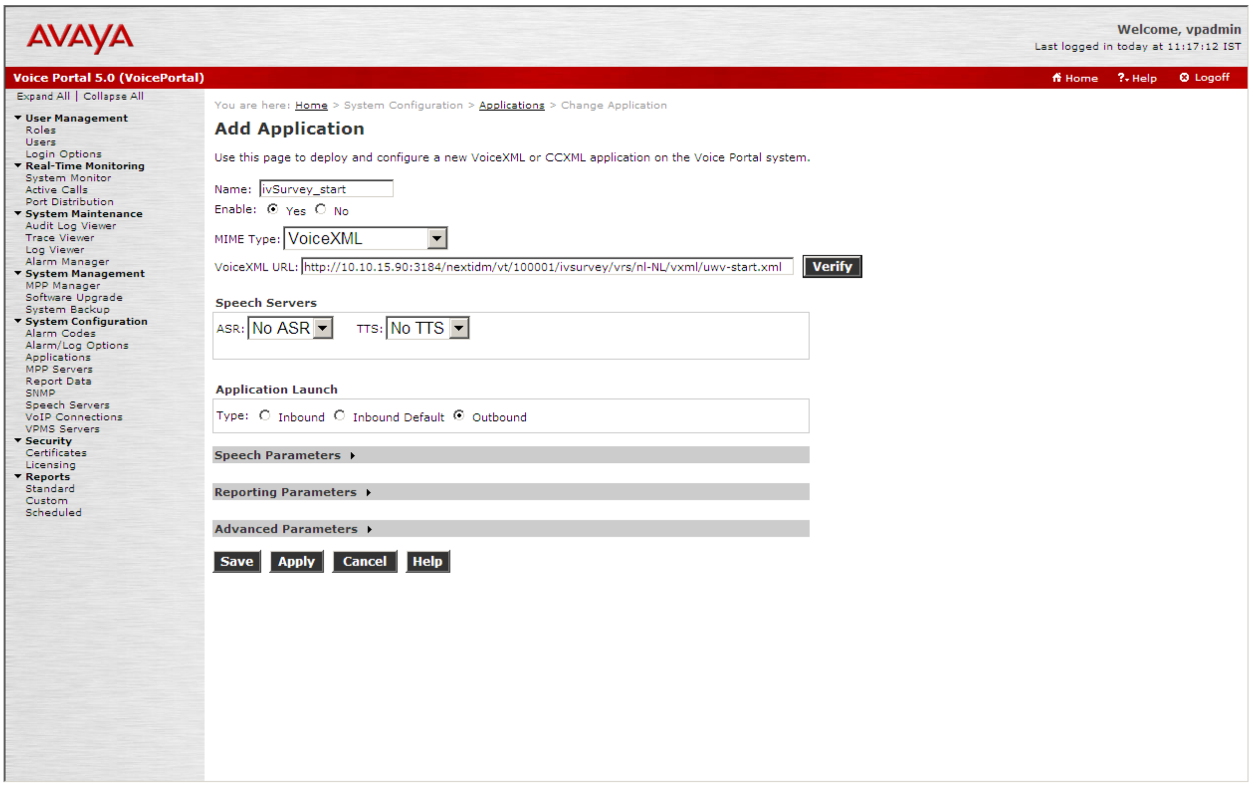
- Adding the MPP server(s)
- Adding an ASR server
- Adding a TTS server

1. Click on **System Configuration → Applications**. Click the **Add** button to configure a new application.



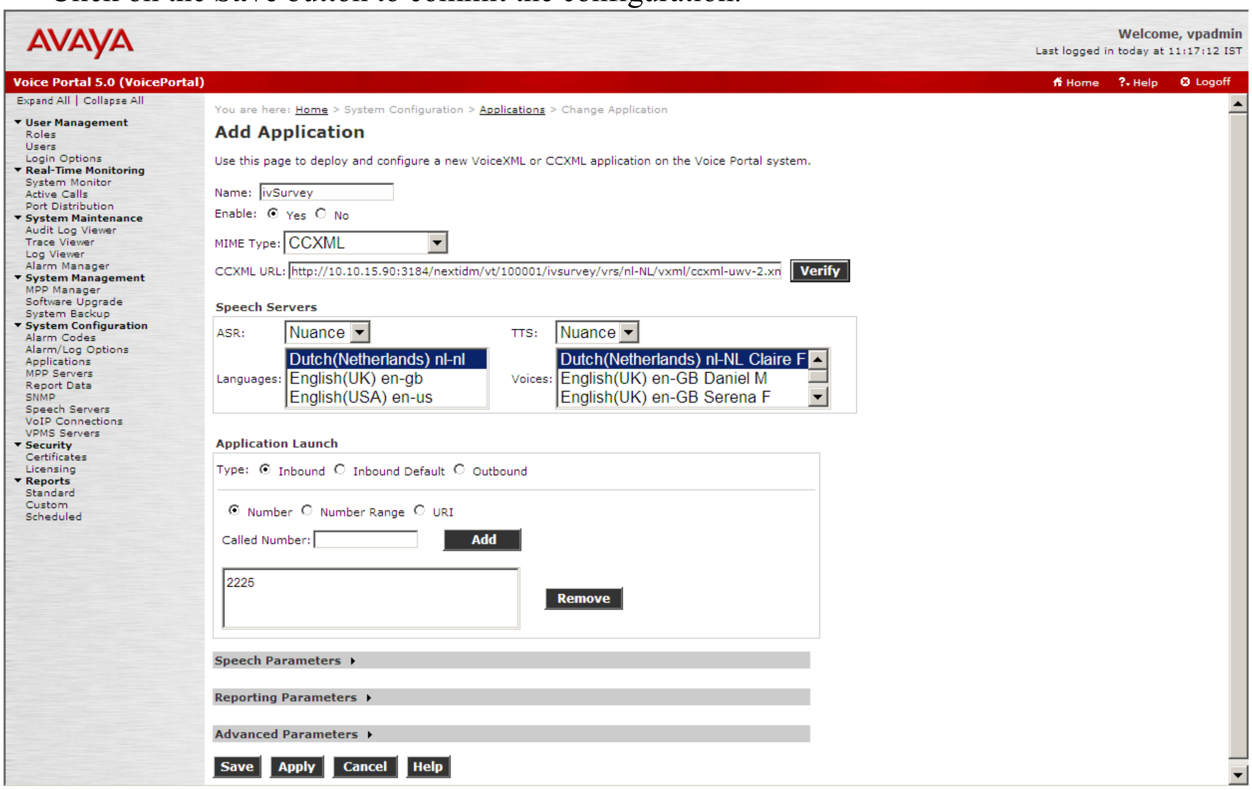
2. Enter the following fields to add a new application:
   - **Name:** set to **ivSurvey_start** (A descriptive name for the test VXML application)
   - **Enable:** select the **Yes**
   - **MIME Type:** select **VoiceXML** according to the application type
   - **VoiceXML URL:** enter the necessary URL(s) to access the VXML application on the application server.
   - **Speech Servers ASR** and **TTS: No ASR** and **No TTS**
   - **Application Launch Type:** select **Outbound**

   Click on the **Save** button to commit the configuration.



3. Enter the following fields to add another new application:
   - **Name:** set to **ivSurvey** (A descriptive name for the test CCXML application)
   - **Enable:** select the **Yes**
   - **MIME Type:** select **CCXML** according to the application type
   - **CCXML URL:** enter the necessary URL(s) to access the CCXML application on the application server.
   - **Speech Servers ASR** and **TTS:** add the ASR and TTS servers as shown below (if available)
   - **Application Launch Type:** select **Inbound**
   - Select the **Number** radio button
   - **Called Number:** enter an extension **2225** for the test application, click on the **Add** button. SIP Enablement Services will have a corresponding Application ID to route this extension (refer to **Step 4** in **Section 6.2**) to Avaya Voice Portal.

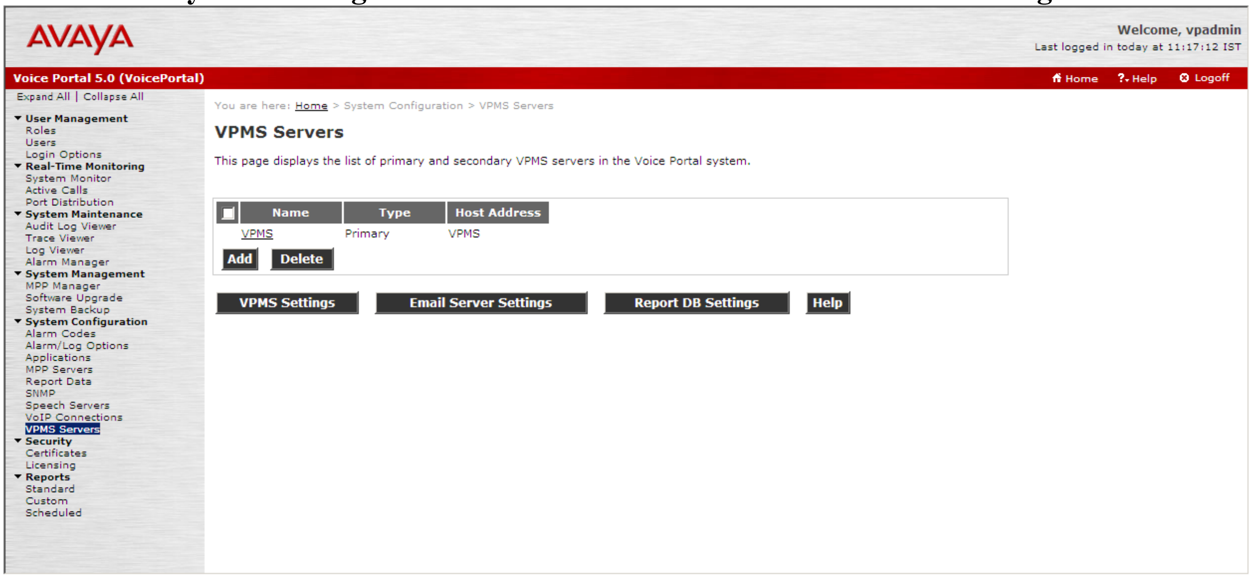
   Click on the **Save** button to commit the configuration.



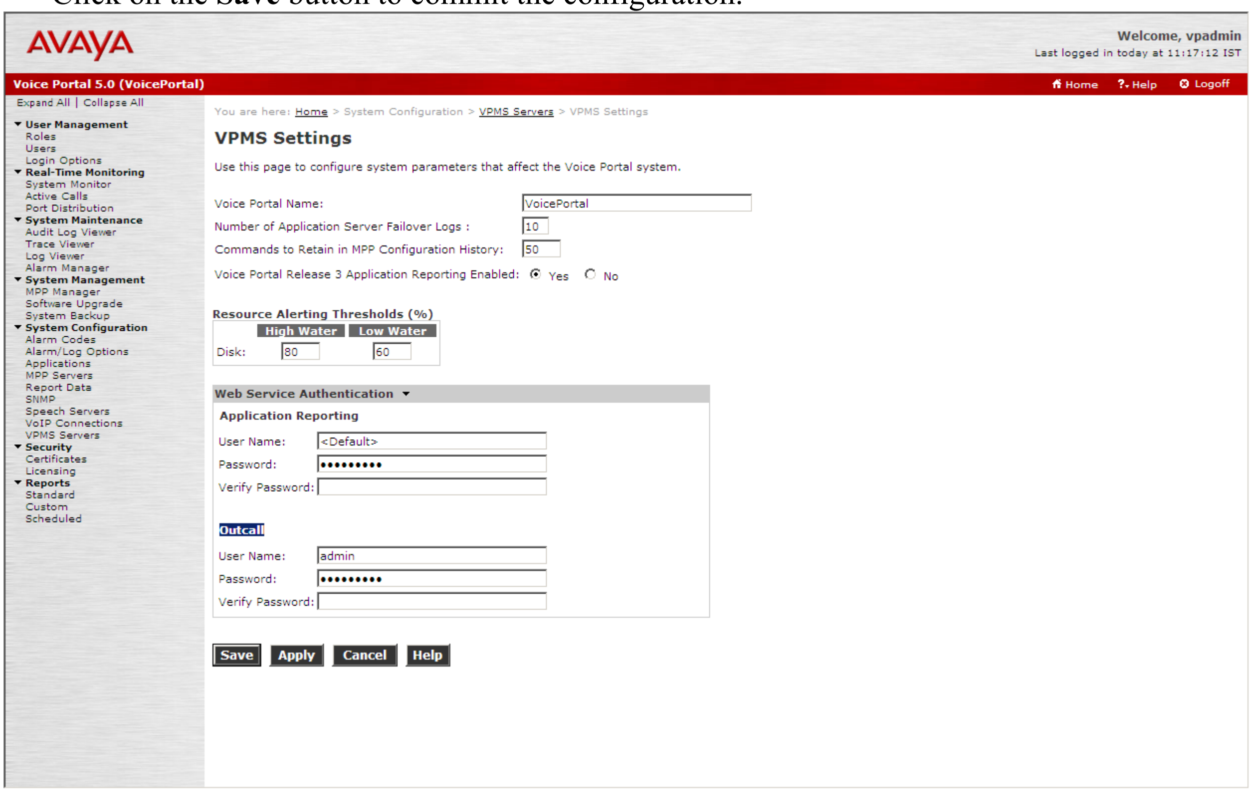
5.4. Configure the Outcall Username and Password

The following section illustrates how to configure the user name and password for Web Service Authentication.

1. Click on **System Configuration → VPMS Servers**. Click the **VPMS Settings** button.



2. Scroll to the **Web Service Authentication** section to view the **Outcall User Name and Password** fields. The User name and Password must match configuration in ivSurvey speech application engine.
   Click on the **Save** button to commit the configuration.



6. Configure SIP Enablement Services

This section only details the configuration required on SIP Enablement Services to set up SIP connectivity to Communication Manager and Avaya Voice Portal and assumes the Avaya SES service is already in place. For additional administration information on Avaya SES, please consult reference **[7]**.

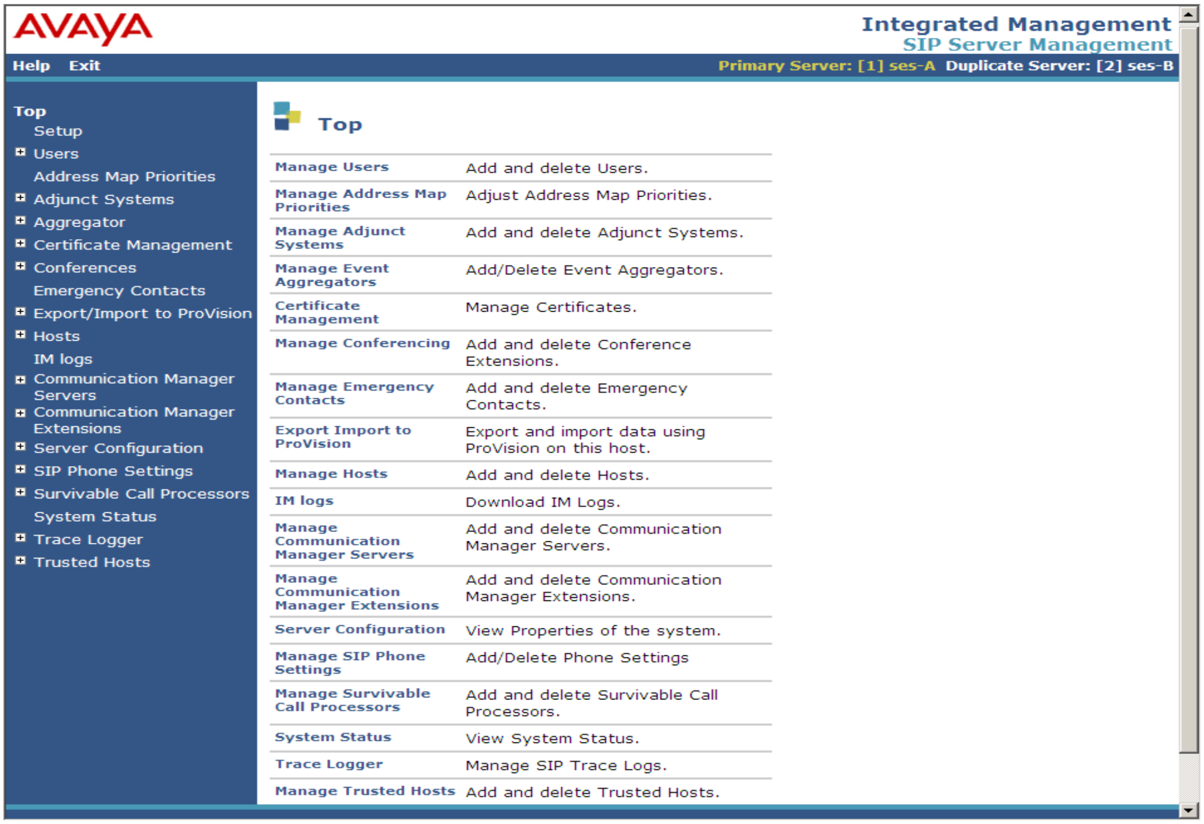
6.1. Configuring Media Server Interface

1. Access SIP Enablement Services web interface by typing the following URL on a web browser and then pressing enter:

   http://<hostname or IP address of SIP Enablement Services>/admin

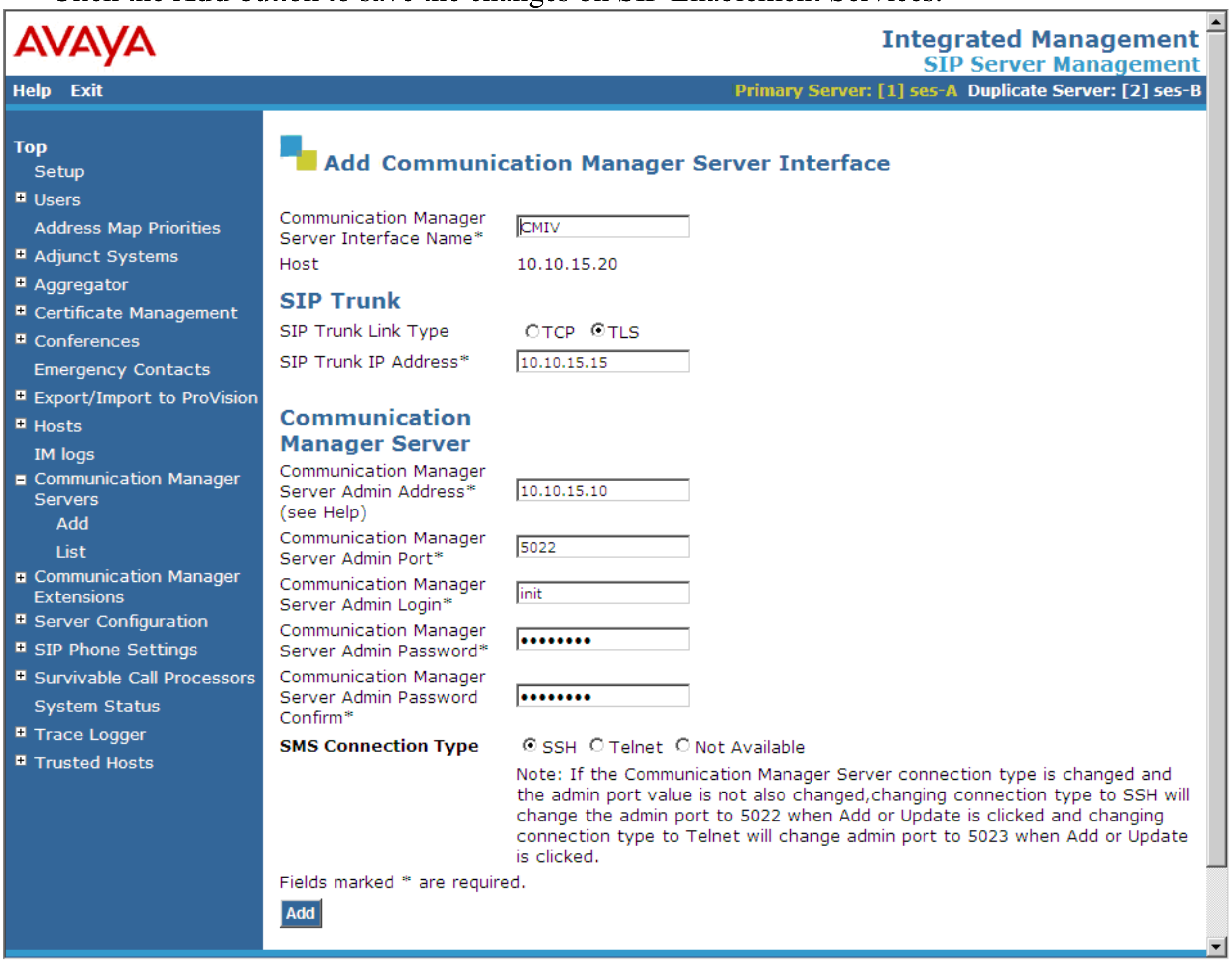
   Press the Continue button on the Welcome web page and then press the **Yes** button in the Security Alert pop-up window (not shown). Select **Administration** menu.

2. Click the **SIP Enablement Services** from Administration Menu. The screen will be shown as below:

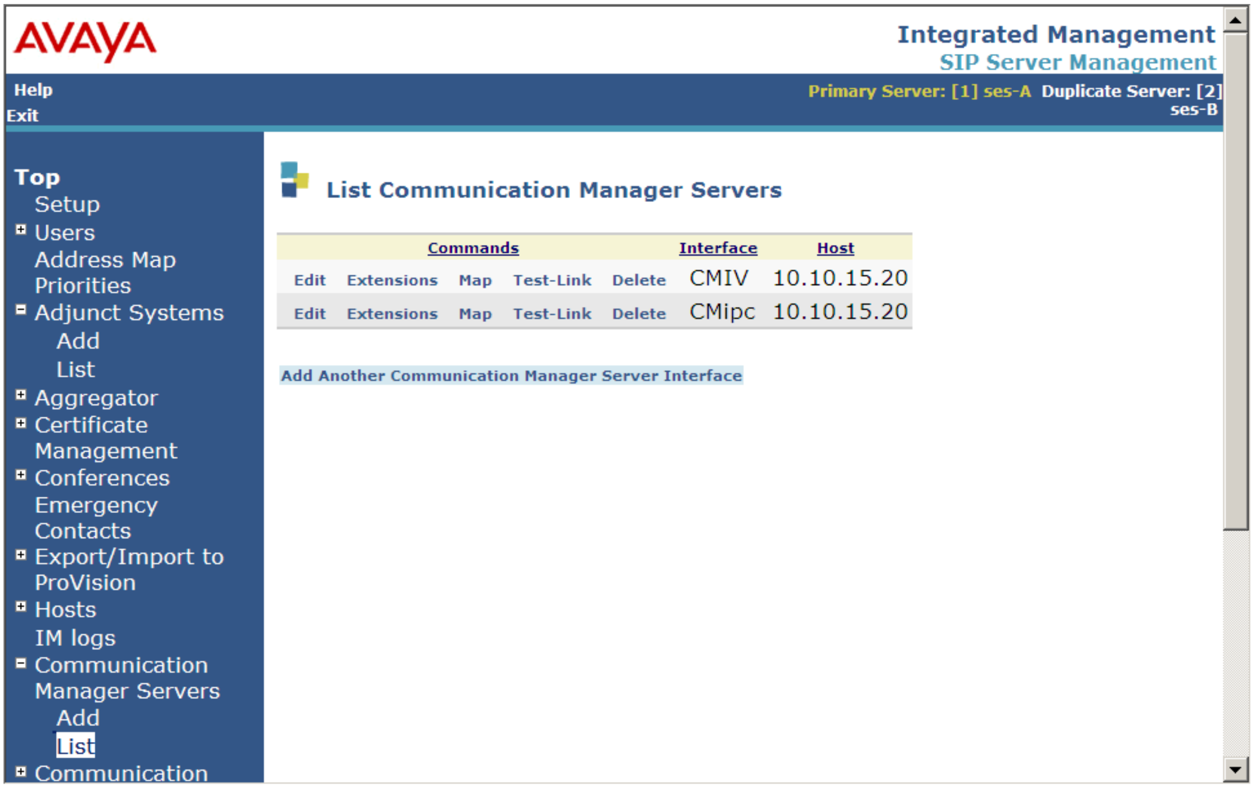


3. Click on **Communication Manager Servers → Add** to add the Communication Manager Server Interface. Enter the following:
   - **Communication Manager Server Interface Name:** set to **CMIV**
   - **Host:** set to **10.10.15.20**
   - **SIP Trunk Link Type:** set to **TLS** (refer to **Transport Method** in **Step 4** in **Section 4.2**)
   - **SIP Trunk IP Address:** set to **10.10.15.15** (refer to **clan IP** in **Step 1** in **Section 4.2**)
   - **Communication Manager Server Admin Address:** set to **10.10.15.10** (IP address of Communication Manager)
   - **Communication Manager Server Admin Login:** set to be Communication Manager administration account with SAT access)
   - **Communication Manager Server Admin Password:** *<password>*
   - **Communication Manager Server Admin Password Confirm:** *<password>*

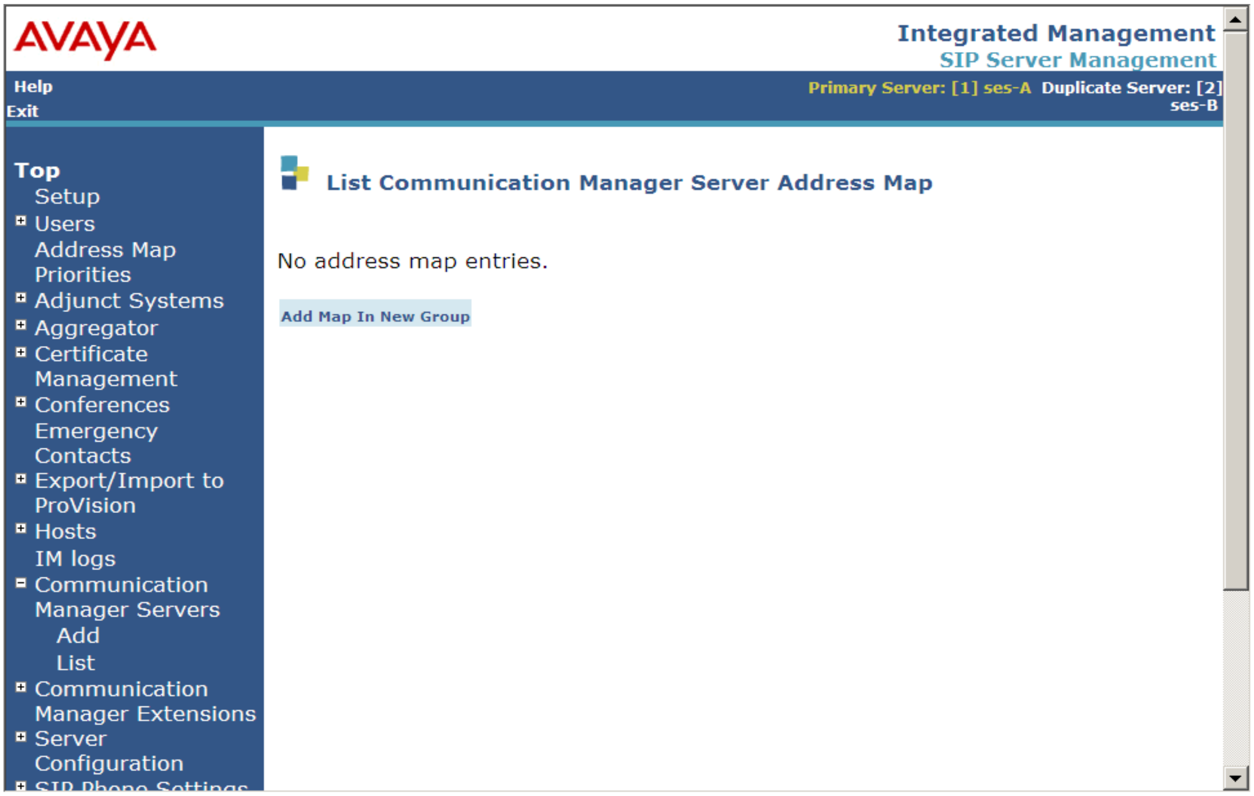
   Click the **Add** button to save the changes on SIP Enablement Services.



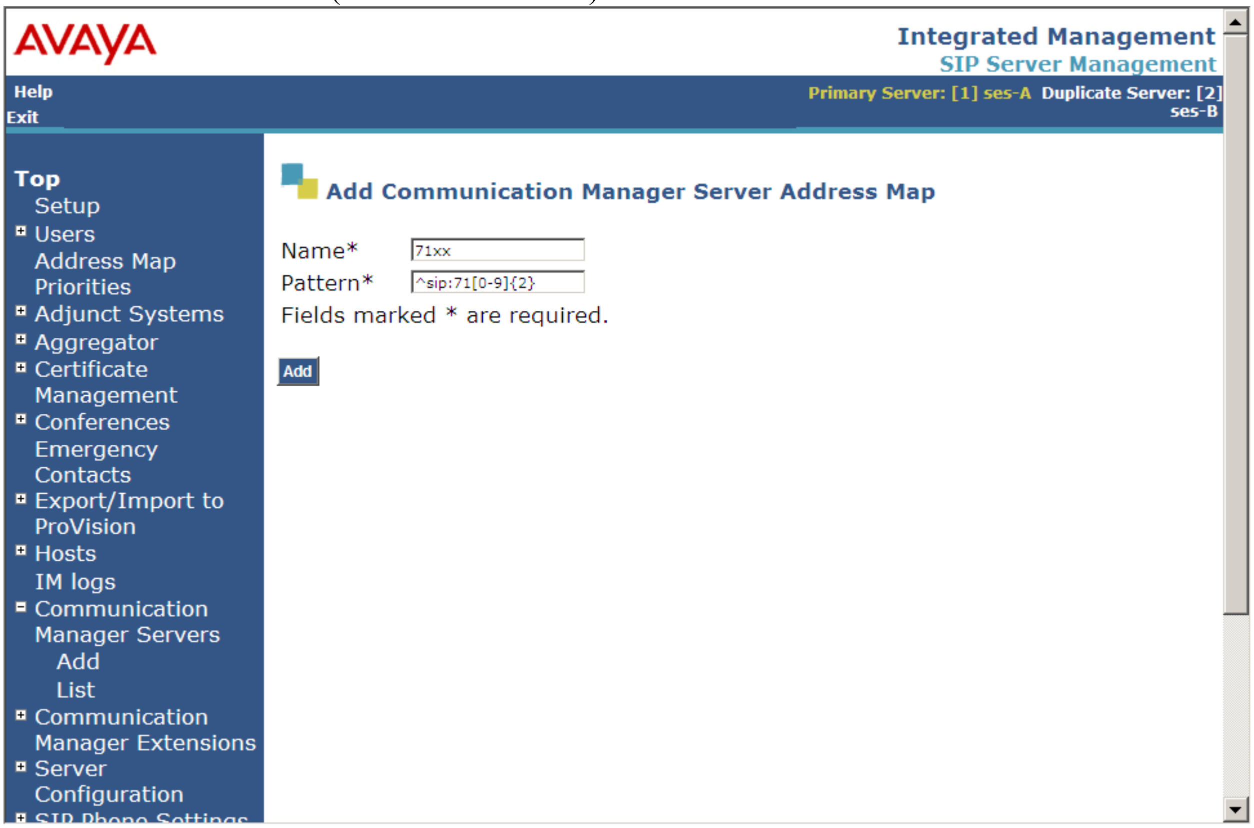
4. Click on **Communication Manager Servers → List** to list the configured Communication Manager Server Interface.



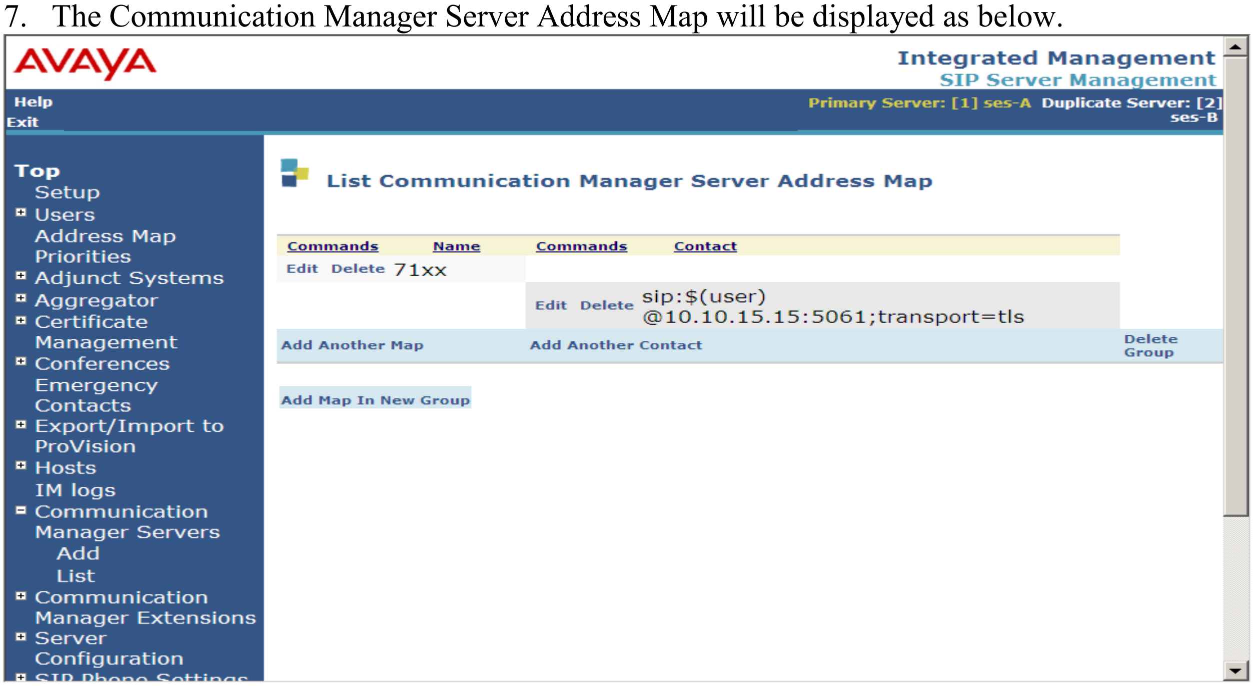
5. Click on **Map** button of the Communication Manager Server configured. The **List Communication Manager Server Address Map** window is displayed. Click on **Add Map in New Group**.



6. Add **Name** field and **Pattern** filed in the **Add Communication Manager Server Address Map**. **Pattern** must match Avaya extensions (in this sample configuration, 71xx are used as Avaya extensions), otherwise outbound call will not be routed through SIP. Click **Add** button and **Continue** button (which is not shown).

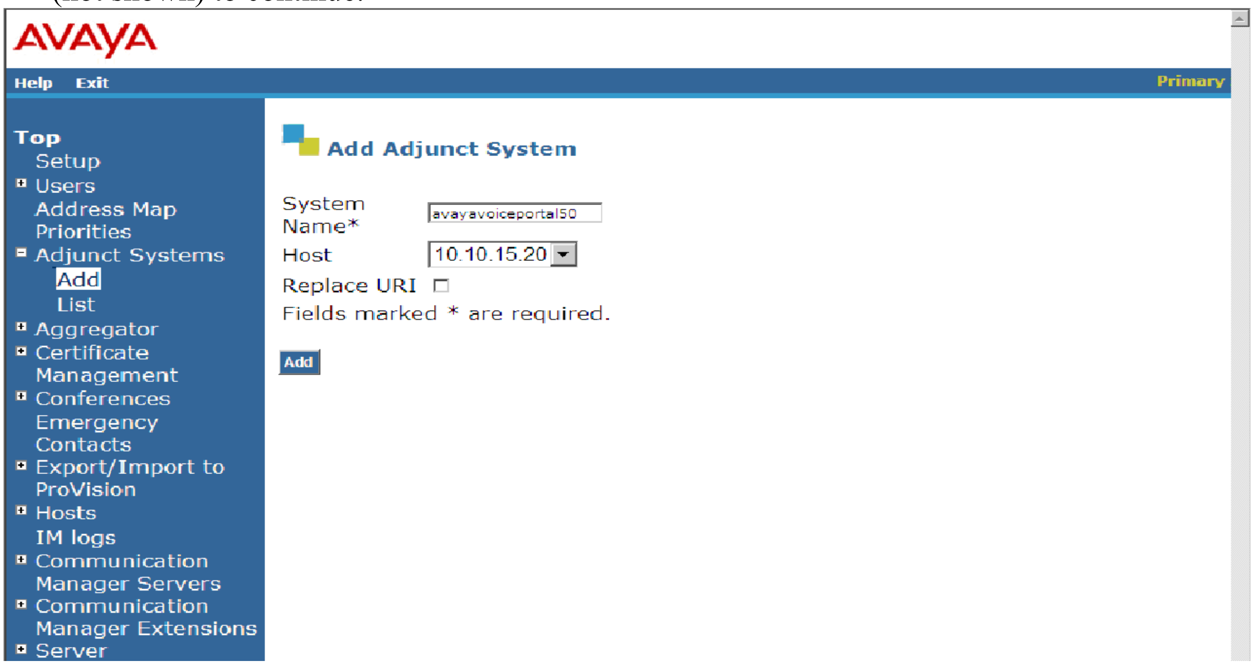


7. The Communication Manager Server Address Map will be displayed as below.

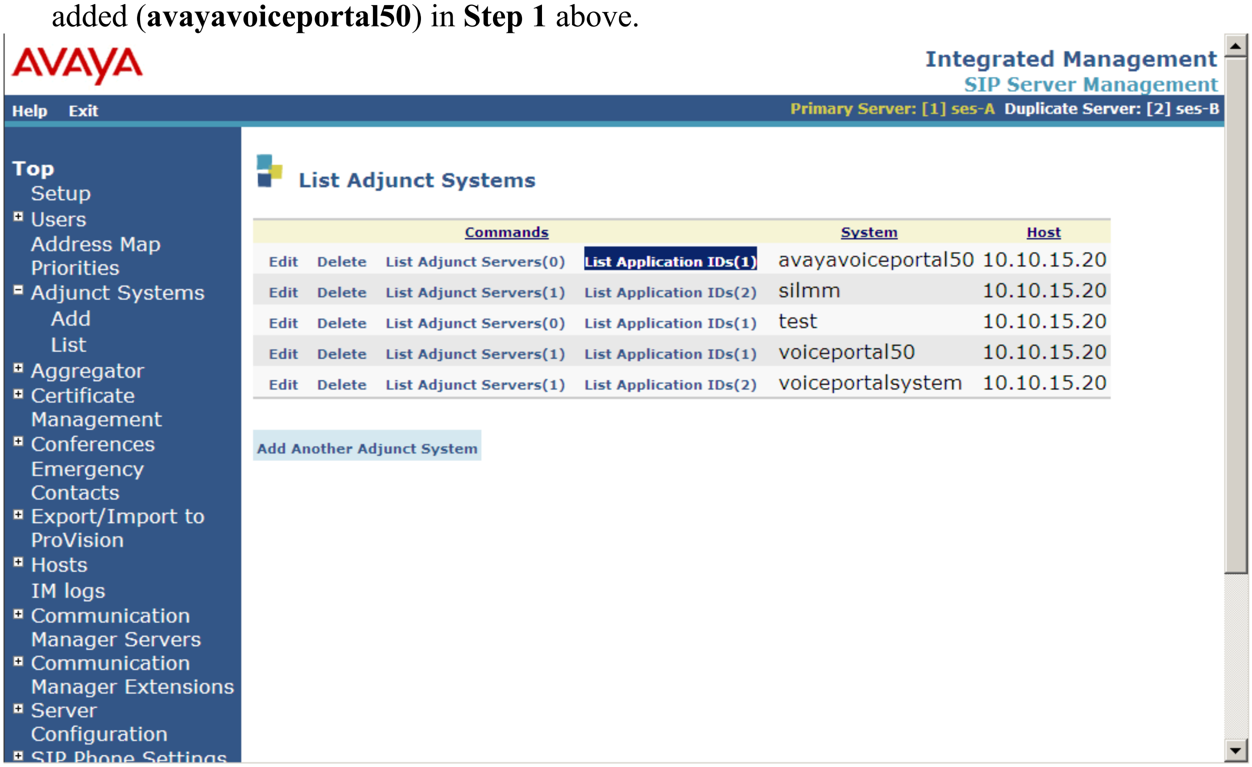


6.2. Configuring Avaya Voice Portal as an Adjunct System

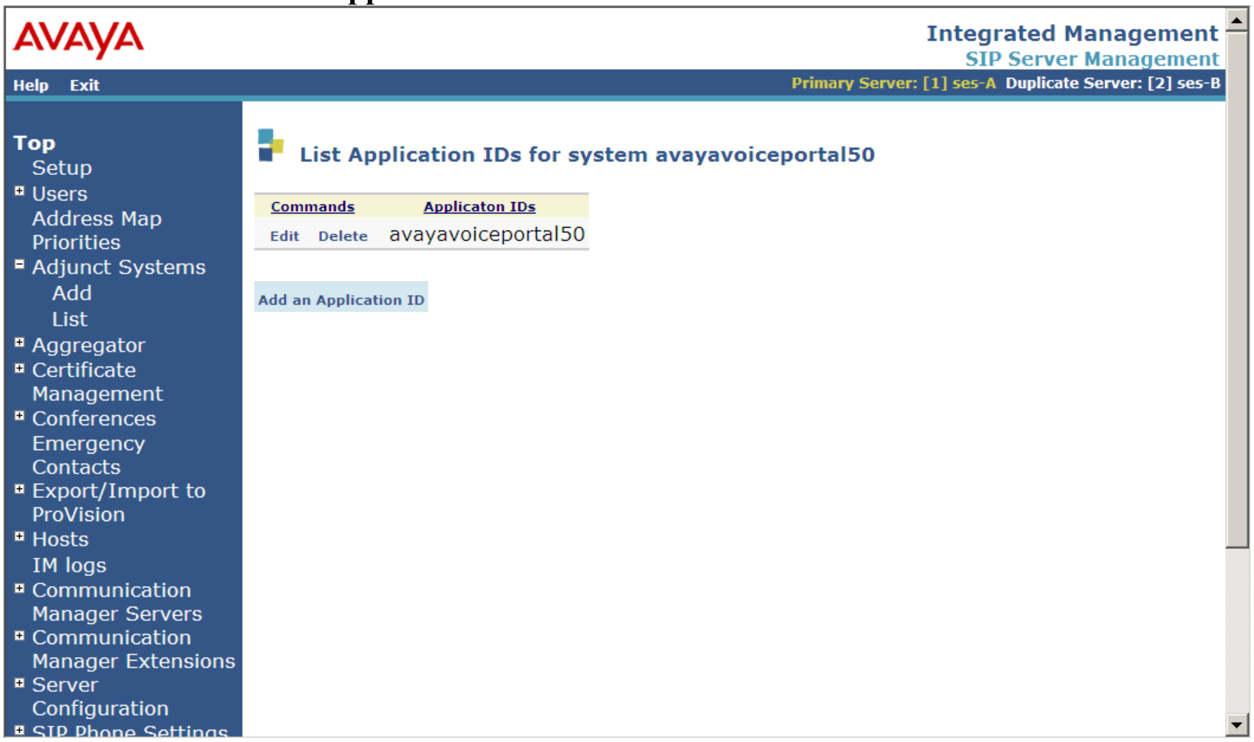
1. Click on **Adjunct Systems→Add** to add the Avaya VPMS as an adjunct system. Enter **avayavoiceportal50** for the VPMS name in the **System Name** field. For the **Host** field, select **10.10.15.20** which is the Avaya SES Home Server IP address with which the VPMS will be integrated. Click the **Add** button to submit the change. Click on the **Continue** button (not shown) to continue.



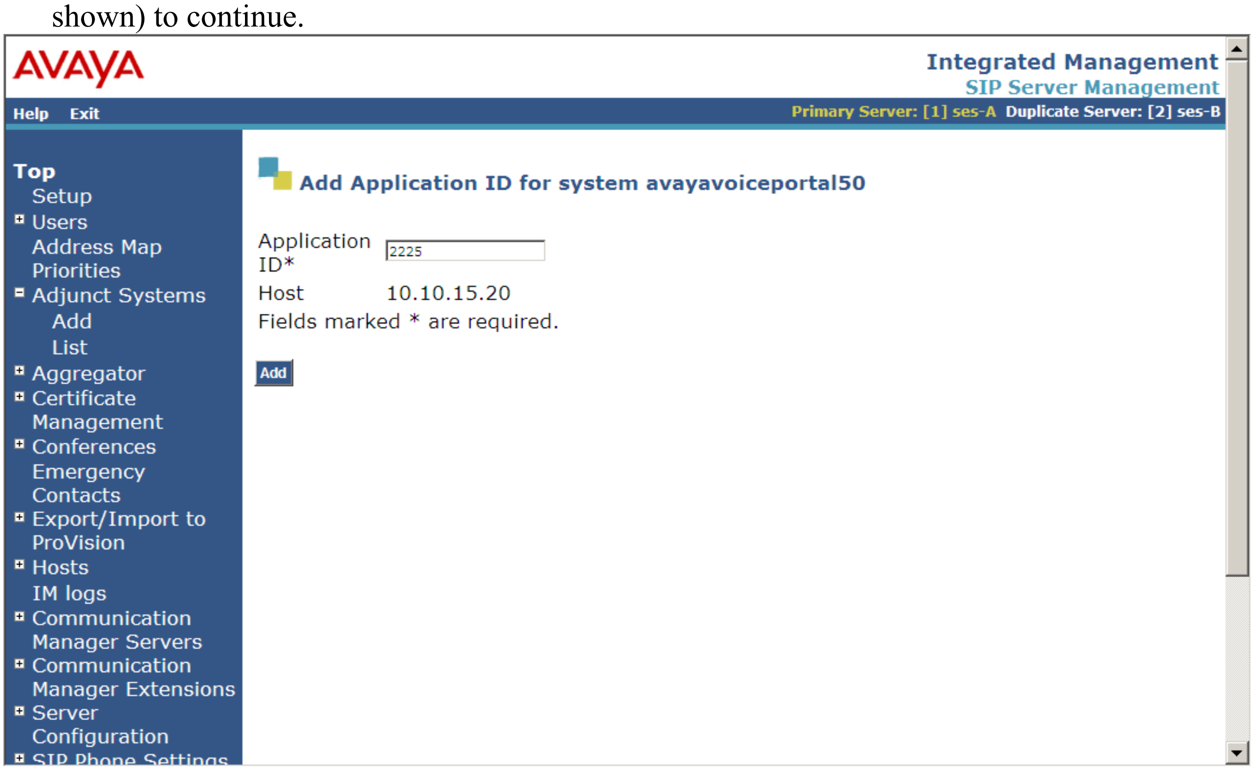
2. Click on the **Adjunct Systems→List→List Application IDs (1)** link for the adjunct system added (**avayavoiceportal50**) in **Step 1** above.



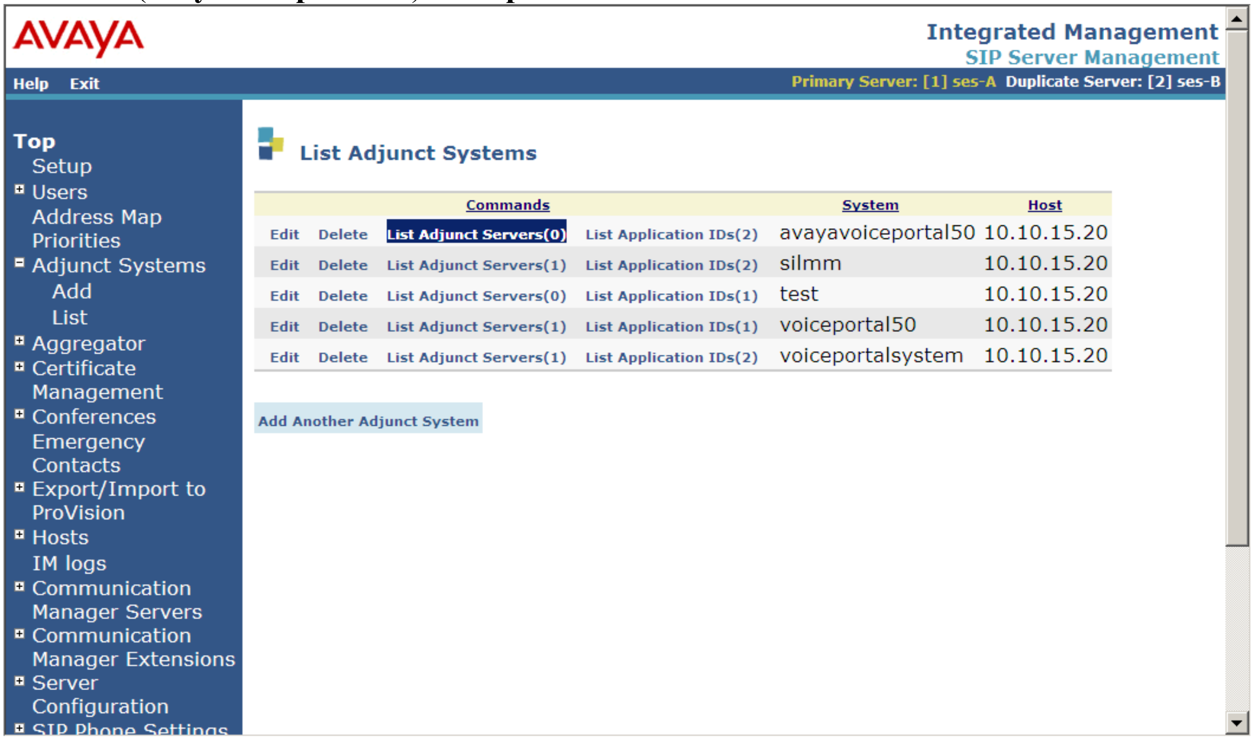
3. Click on the **Add an Application ID** link.



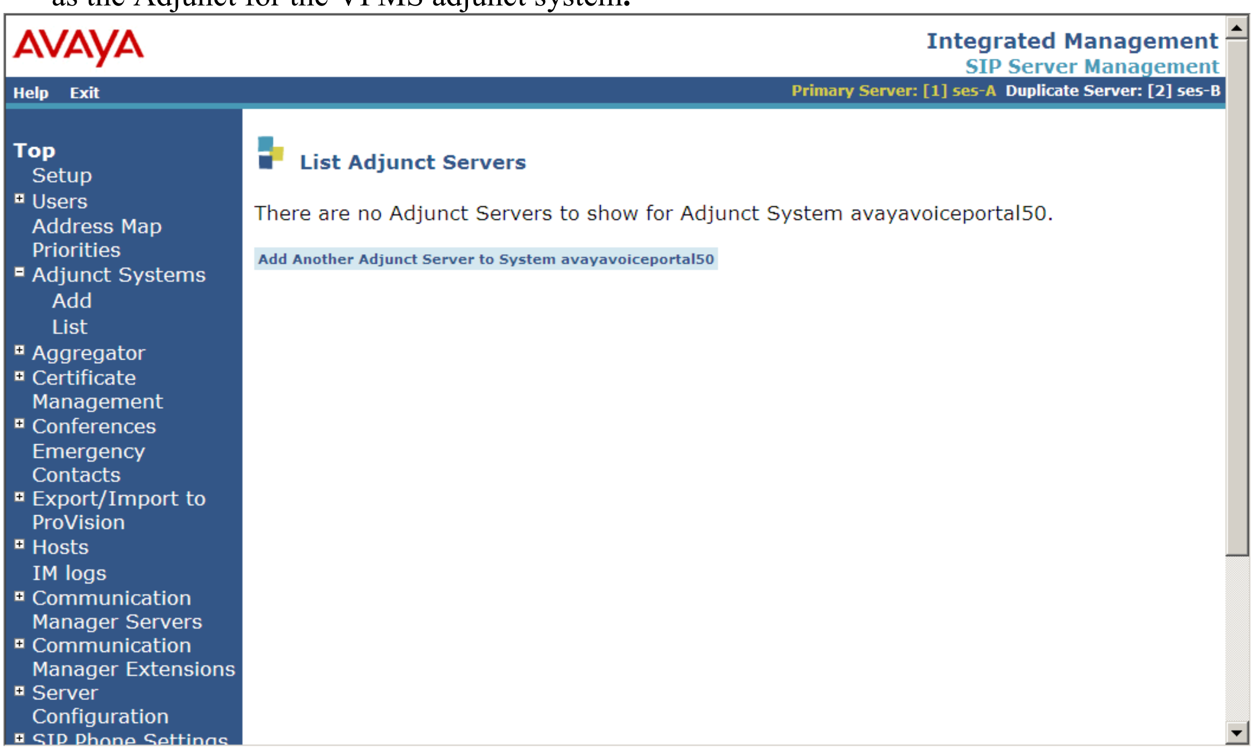
4. Enter the extension **2225** for the **Application ID** field (refer to **Step 3** in **Section 5.3**). Repeat **Step 3** and **Step 4** to add additional application IDs for each application extension on Avaya Voice Portal. Click the **Add** button to submit the change. Click on the **Continue** button (not shown) to continue.



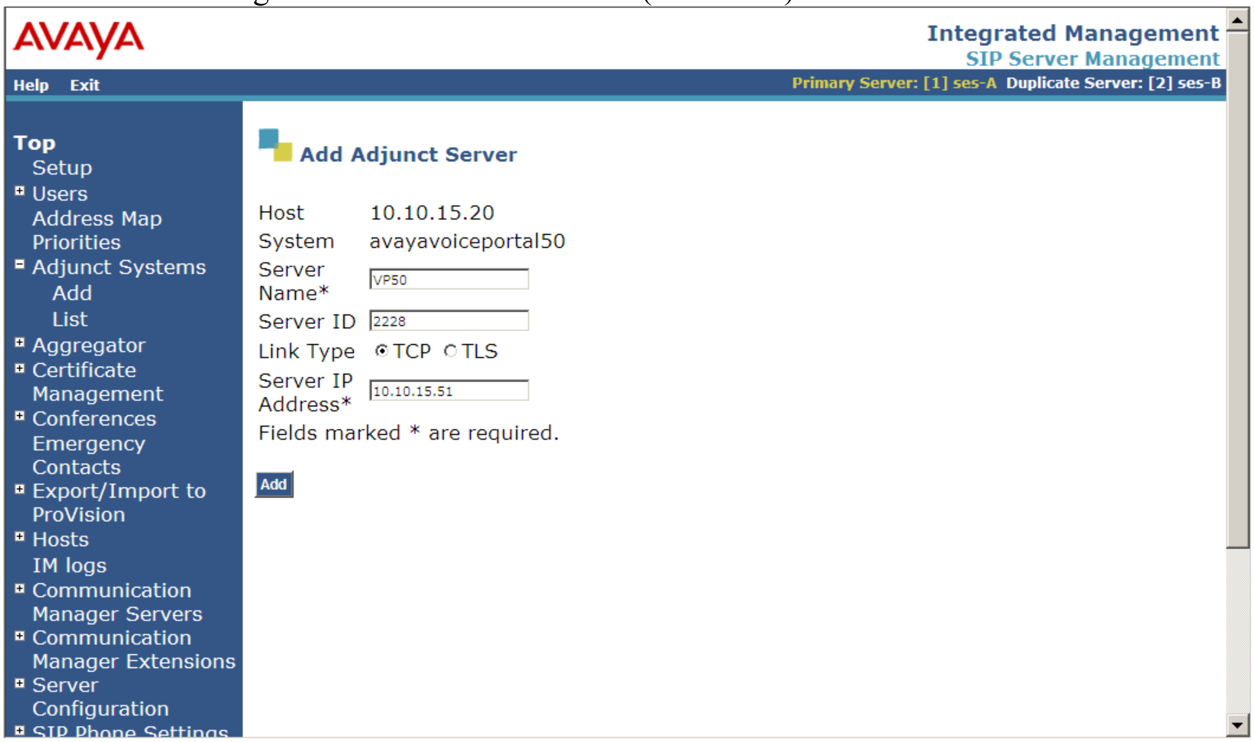
5. Click on **Adjunct Systems→List→List Adjunct Servers (0)** link for the adjunct system added (**avayavoiceportal50**) in **Step 1.**



6. Click on **Add Another Adjunct Server to System avayavoiceportal50** link to add the MPP as the Adjunct for the VPMS adjunct system**.**

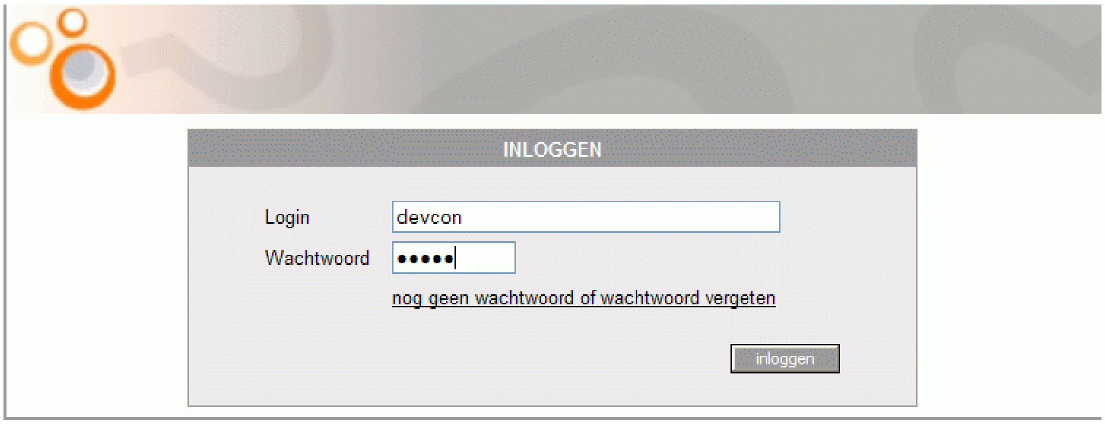


7. Enter **VP50** (a unique name for the MPP adjunct server) in the **Server Name** field**.** Enter the extension **2228** that Communication Manager will use to reach the MPP adjunct server for the **Server ID** field (SIP Enablement Services uses this extension to form the SIP URI for the adjunct system). The **Link Type** field is set to **TCP** in this sample configuration**.** Enter the MPP server IP address **10.10.15.51** for the **Server IP Address** field. Click the **Add** button to submit the change. Click on **Continue** button (not shown) to continue.



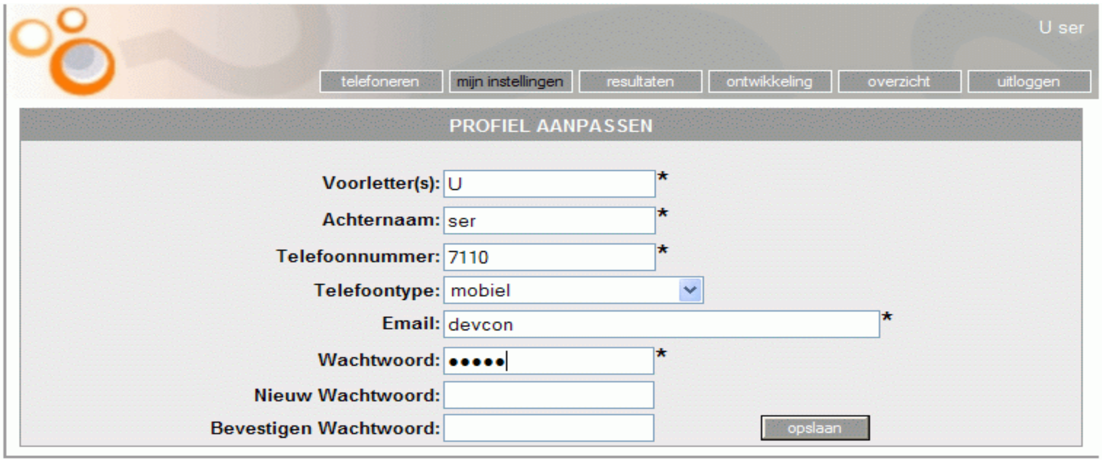
7. Configuring ImmediateVoice Application Server

1. Access the ivSurvey Speech Application Engine by typing the following URL on a web browser and then pressing enter: “http://<hostname or IP address of ivSurvey server>/ivsurvey”. Enter User’s **Login** and **Wachtwoord** (Password) and press **inloggen** button.

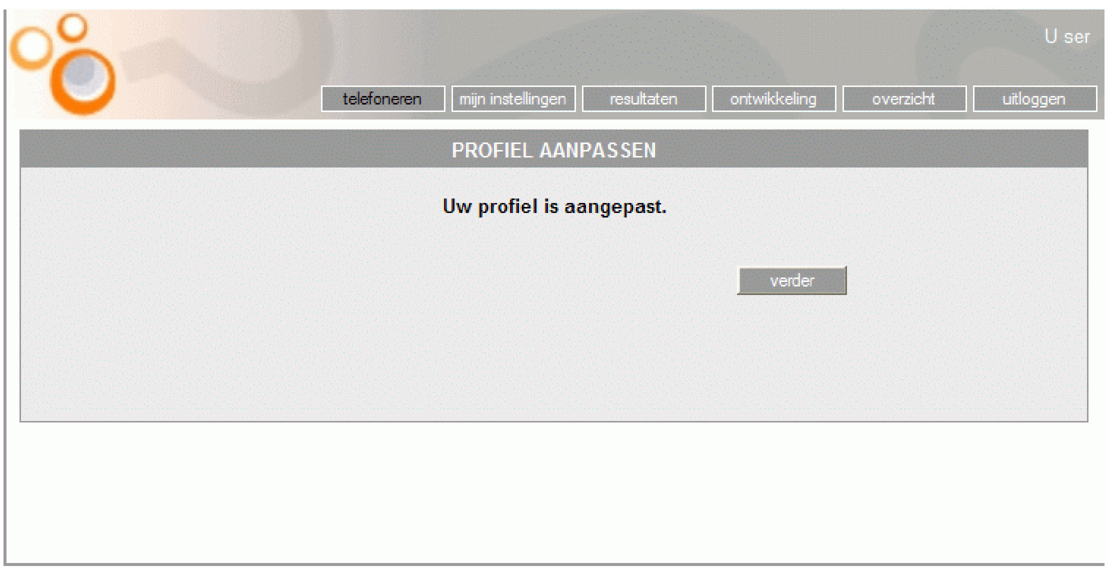


2. The telephony board page will be displayed as shown blow. Click on **mijn instellingen** button to edit user profile.

3. Change **Telefoonnummer** field to a test phone number (in this sample application, **7110** is used), fill in **Wachtwoord** with password and click **opslaan** to save the changes.

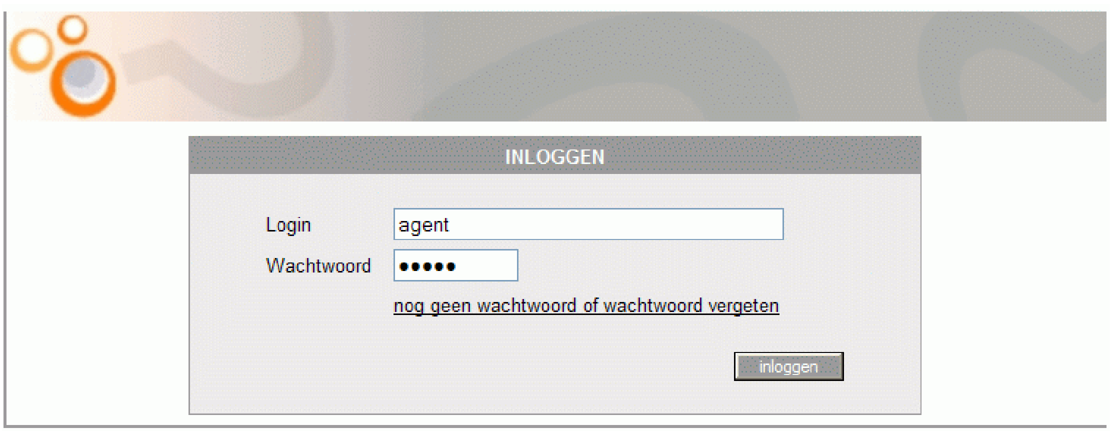


4. Once confirmation page appears, click **verder** to return to the main telephony page.

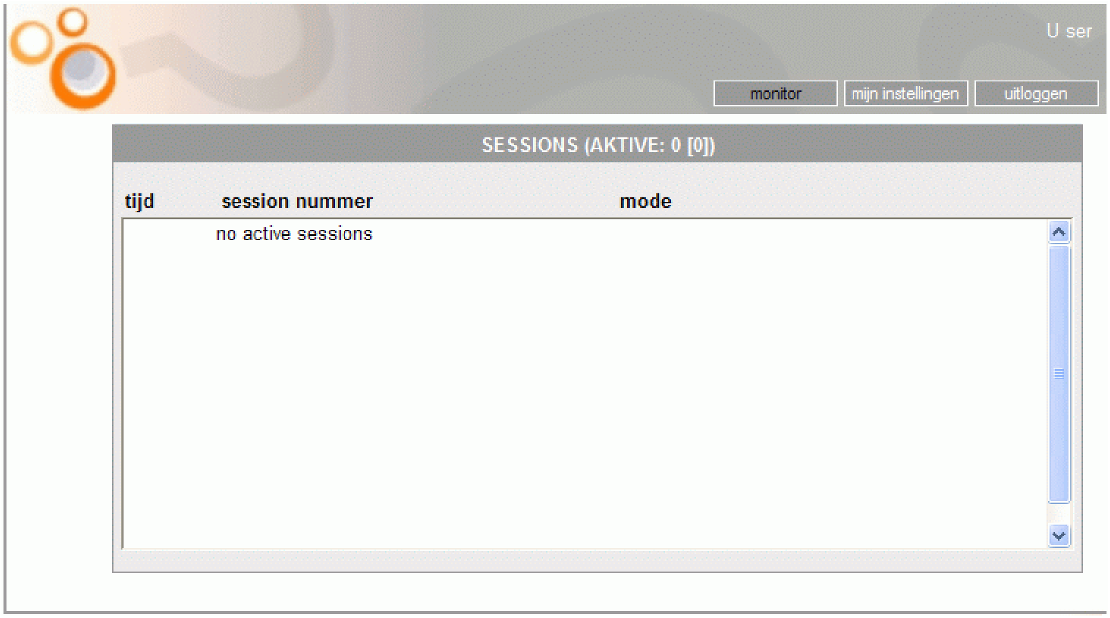


5. Fill in a test phone number (in this sample application, 7112 is used) in **Telefoonnummer** field and click **verbinden** to proceed with the call.

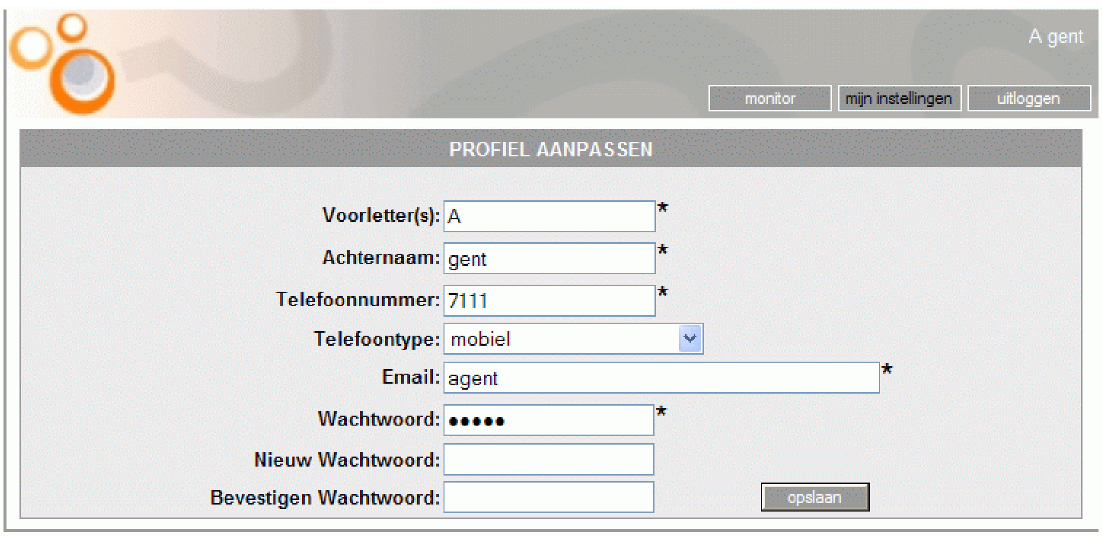
6. Access the ivSurvey Speech Application Engine by typing the following URL on another web browser and then pressing enter: “http://<hostname or IP address of ivSurvey server>/ivsurvey”. Enter Agent’s **Login** and **Wachtwoord** (Password) and press **inloggen** button.



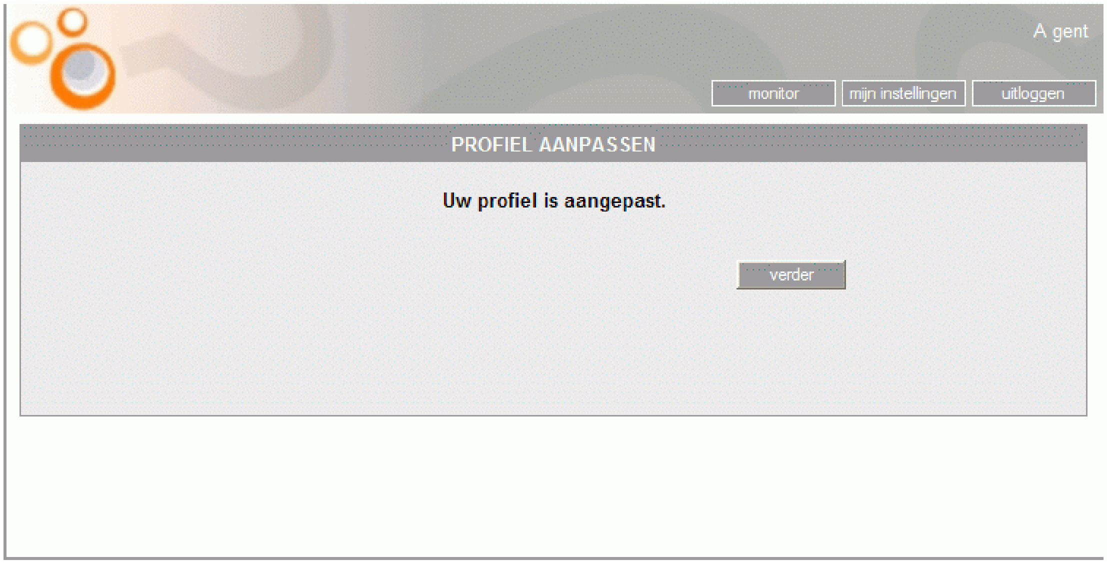
7. Session Monitoring Window will be displayed after login as an agent. Click on **mijn instellingen** button on the Session Monitoring Window.



8. Edit **Telefoonnummer** field to a test number (in this sample configuration, 7111 is used), fill in **Wachtwoord** with password and click **opslaan** button to save changes.



9. Once confirmation page appears, click **verder** to return to the monitor page.



8. General Test Approach and Test Results

The interoperability compliance test included feature and serviceability testing. The feature testing is performed following the below ivSurvey call flow diagram.

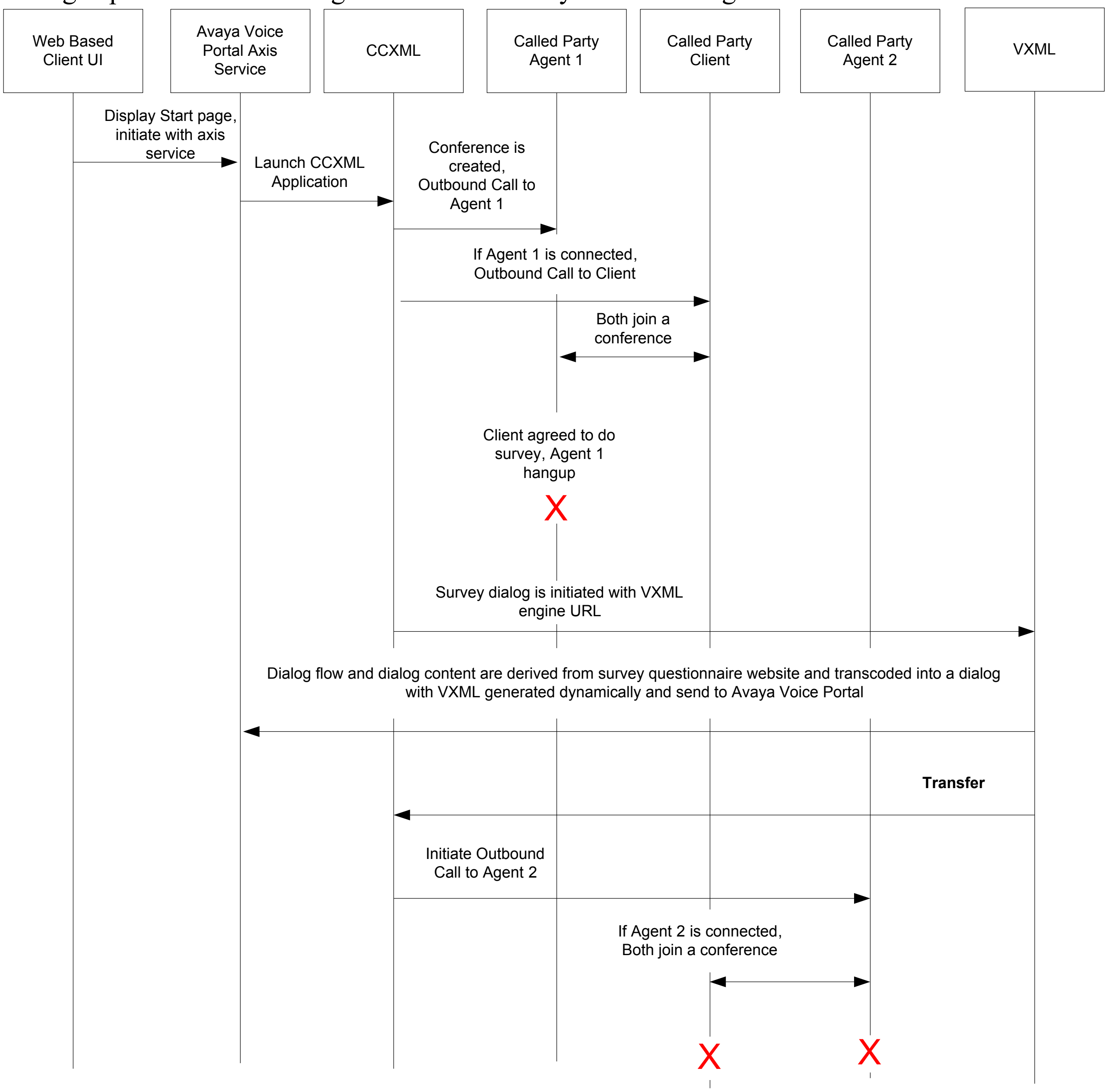


Figure 2: Call Flow

The focus of testing is VXML (CCXML) interoperability and support of major call functions like making outbound call, extracting interactions from web pages, converting them into customer speech dialog and converting the speech back to the required data field inputs, presenting questionnaires to customers and call transfer. The serviceability testing focused on verifying the solution's ability to recover from an outage condition, such as busying out the SIP trunk and disconnecting the cable for the ivSurvey Speech Application Engine or Avaya Voice Portal. All feature and serviceability test cases were performed manually. Tests are performed by using Avaya Voice Portal axis services to make outbound calls and monitoring active sessions in the Agent Session Monitoring window. The monitoring function is supported by ivSurvey speech application engine. Inputs made into website form and browsing which conversations are in progress are also monitored through web browser.

All test cases were executed successfully and no observations were made.

9. Verification Steps

The following steps may be used to verify the configuration:

1. Click **verbinden** in **Section 7 Step 5** to proceed with the call after ivSurvey is fully configured. Verify User number 7110 will ring.
2. Pick up the phone, press 1 which means it's not an auto answer machine to pick up the phone. Verify customer number 7112 will ring.
3. Pick up the phone, and verify there is an audio path between these two phones. Verify on the **Avaya Voice Portal→Real-Time Monitoring→Active Calls** menu, there are two outbound calls, both are through SIP trunk.
4. Talk at least 20 seconds, then hang up phone 7110, and verify if 7112 is asked to do a survey. Verify there is an active session in the Agent Session Monitoring window.

10. Conclusion

These Application Notes describe the steps for configuring Avaya Voice Portal 5.0 with Dialogs Unlimited ivSurvey v2.0 and SIP Enablement Services based on SIP connectivity.

11. Additional References

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

[1] *"Administering Avaya Aura™ Communication Manager"*, Issue 5, May 2009; Doc ID: 03-300509

[2] *"Configuring Avaya Voice Portal with Avaya Communication Manager and Designing a Sample Speech Application using Avaya Dialog Designer"*, Issue 1.0, September 2006

[3] *"Administering Avaya Voice Portal"*, March 2009

[4] *"Configuration Note 3911 Avaya Voice Portal (Software application) – SIP Integration"*, December 2008

[5] *"Application Notes for Configuring SIP Connectivity between Avaya Voice Portal and Avaya Communication Manager using Avaya SIP Enablement Services"*, Issue 1.0, November 2007

[6] *"Avaya Voice Portal 5.0 Security White Paper"*, March 2009

[7] "*Installing, Administering, Maintaining, and Upgrading Avaya Aura™ SIP Enablement Services*", Issue 7, May 2009; Doc ID 03-600768

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.