



## **Avaya Solution & Interoperability Test Lab**

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# **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.

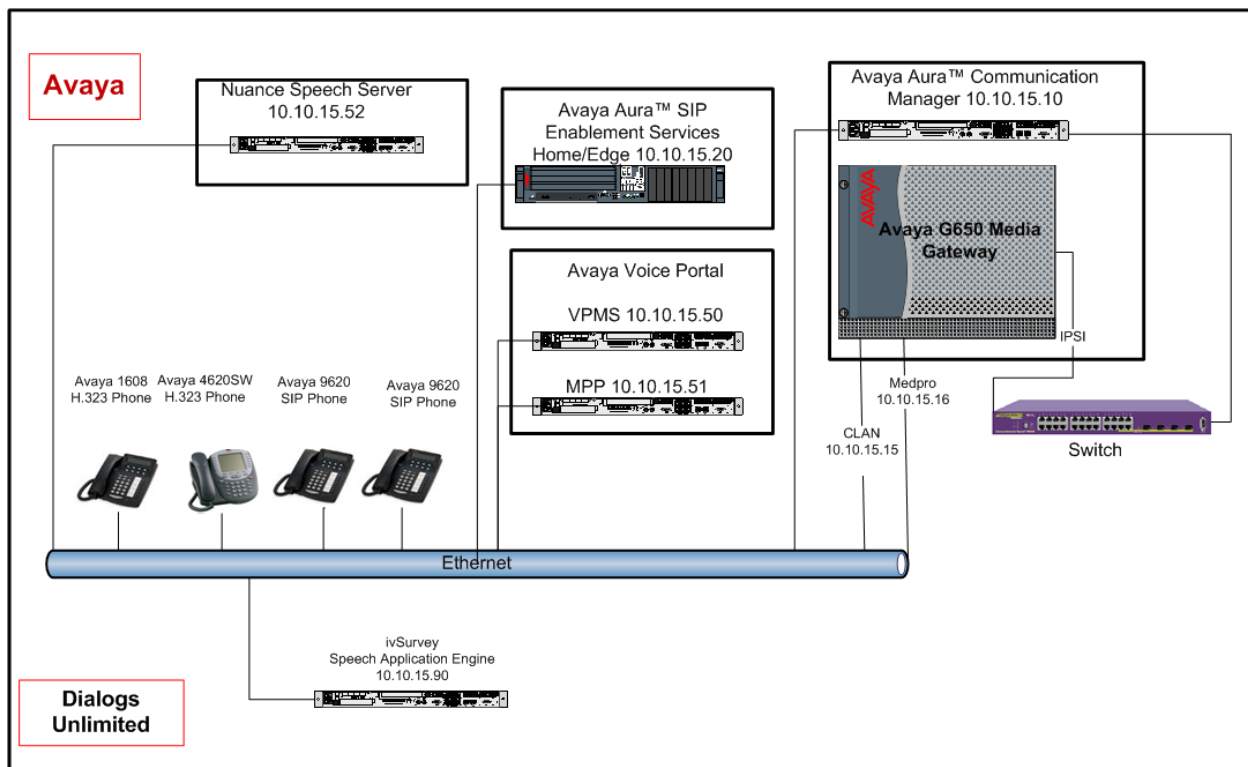
## 1.2. Support

Technical support for the Avaya products can be obtained from Avaya. See the support link at [support.avaya.com](http://support.avaya.com) for contact information. Technical support for the ivSurvey products can be obtained from Dialogs Unlimited. See [www.dialogsunlimited.com](http://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	
<b>Maximum Off-PBX Telephones - OPS: 100</b>	<b>0</b>	
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	
<b>Maximum Administered SIP Trunks: 100</b>	<b>48</b>	
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	
<b>ARS? y</b>	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	<b>Media Encryption Over IP? n</b>	
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	
<b>IP Trunks? y</b>		
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		
Personal Station Access (PSA)? n	System Management Data Transfer? n	
PNC Duplication? n	Tenant Partitioning? n	
Port Network Support? y	Terminal Trans. Init. (TTI)? n	
Posted Messages? n	Time of Day Routing? n	
	TN2501 VAL Maximum Capacity? y	
	<b>Uniform Dialing Plan? y</b>	
<b>Private Networking? y</b>	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:		
<b>Auto Alternate Routing (AAR) Access Code: *8</b>		
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			
<b>clan</b>	<b>10.10.15.15</b>			
default	0.0.0.0			
gateway	10.10.15.1			
<b>medpro</b>	<b>10.10.15.16</b>			
procr	10.10.15.10			
<b>ses</b>	<b>10.10.15.20</b>			

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)	
<b>1: G.711MU</b>	<b>n</b>	<b>2</b>	<b>20</b>	



- 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

```

- 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			
UI 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	
													Intw	
</														

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 String	Total Min Max		Route Pattern	Call Type	Node Num	ANI Reqdn
222		4	4	1	aar		

## 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.

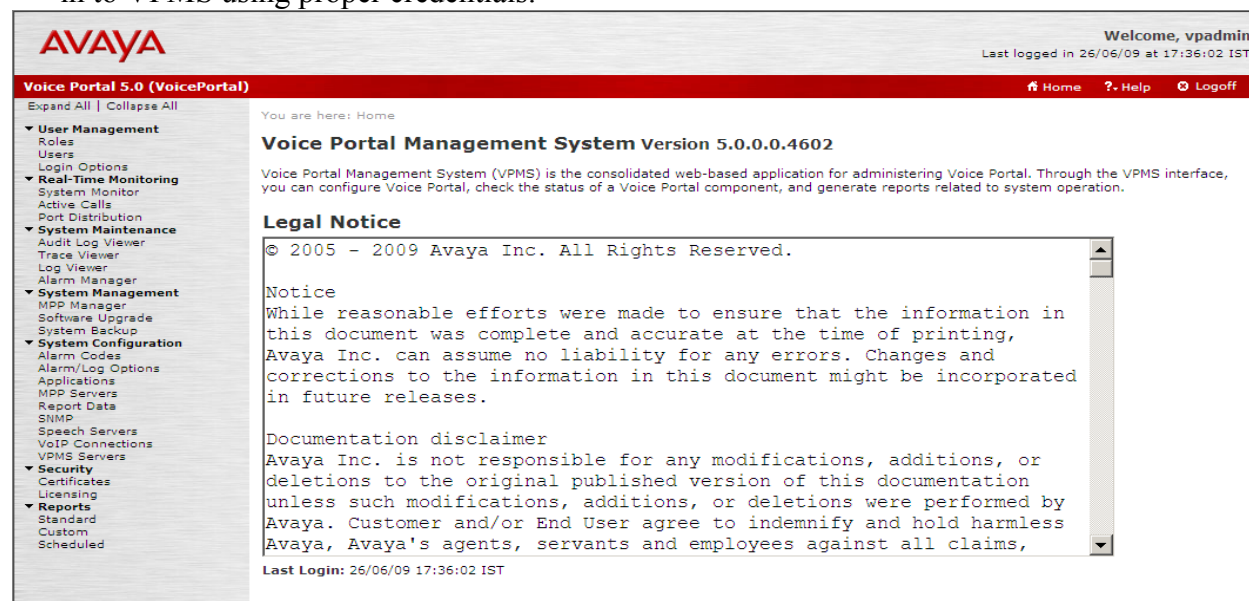
### 5.1. Verify Avaya Voice Portal Licenses

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**.

**AVAYA** Welcome, vpadmin  
Last logged in 26/06/09 at 17:36:02 IST

**Voice Portal 5.0 (VoicePortal)** Home Help Logoff

Expand All | Collapse All

You are here: [Home](#) > Security > Licensing

### Licensing

This page displays the Voice Portal license information that is currently in effect. Voice Portal uses Avaya License Manager (WebLM) to control the number of telephony ports that are used.

#### License Information

License Server URL:	https://10.10.15.50:8443/WebLM/LicenseServer
Telephony Ports:	100
Non Media Ports:	0
Announcement Ports:	0
ASR Connections:	100
TTS Connections:	100
Video Server Connections:	100
Version:	5
Last Changed:	27/04/09 10:47:40 IST
Last Successful Poll:	29/06/09 11:14:58 IST

#### License Settings

License Server URL:  **Verify**

	Minimum	Maximum
Telephony Ports:	<input type="text" value="0"/>	<input type="text" value="5,000"/>
Non Media Ports:	<input type="text" value="0"/>	<input type="text" value="0"/>
Announcement Ports:	<input type="text" value="0"/>	<input type="text" value="0"/>

**Apply** **Cancel** **Help**

## 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.

**AVAYA** Welcome, vpadmin  
Last logged in 26/06/09 at 17:36:02 IST

**Voice Portal 5.0 (VoicePortal)** Home Help Logoff

Expand All | Collapse All

You are here: [Home](#) > System Configuration > VoIP Connections

### VoIP Connections

This page displays a list of Voice over Internet Protocol (VoIP) servers that Voice Portal communicates with. You can configure multiple SIP connections, but only one SIP connection can be enabled at any one given time.

H.323 SIP

<input type="checkbox"/>	Name	Enable	Proxy Transport	Proxy Server Address	Proxy Server Port	Listener Port	SIP Domain	Maximum Simultaneous Calls	Inbound Calls Allowed	Outbound Calls Allowed
<input type="checkbox"/>	SES-SRV	Yes	TCP	10.10.15.20	5060	5060	du.rnd.avaya.com	10	10	10

**Add** **Delete** **Help**

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.

The screenshot displays the Avaya Voice Portal 5.0 (VoicePortal) web interface. The top header includes the Avaya logo, a welcome message for 'vpadmin', and the login time 'Last logged in 26/06/09 at 17:36:02 IST'. The main navigation bar shows 'Voice Portal 5.0 (VoicePortal)' and links for 'Home', 'Help', and 'Logoff'. A left sidebar contains a tree view of system management options, with 'VoIP Connections' highlighted. The main content area shows the breadcrumb 'You are here: Home > System Configuration > VoIP Connections > Add SIP Connection' and the title 'Add SIP Connection'. Below the title, a message states 'Use this page to add a new SIP connection.' The form contains four fields: 'Name' (text box with 'ses-srv'), 'Proxy Transport' (dropdown menu with 'TCP' selected), 'Proxy Server Address' (text box with '10.10.15.20'), and 'Proxy Server Port' (text box with '5060'). To the right of the 'Proxy Server Address' field is a dark button labeled 'Administration'. At the bottom of the form are three buttons: 'Continue', 'Cancel', and 'Help'.

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.

**AVAYA** Welcome, vpadmin  
Last logged in 26/06/09 at 17:36:02 IST

**Voice Portal 5.0 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

- ▼ **User Management**
  - Roles
  - Users
  - Login Options
- ▼ **Real-Time Monitoring**
  - System Monitor
  - Active Calls
  - Port Distribution
- ▼ **System Maintenance**
  - Audit Log Viewer
  - Trace Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Management**
  - MPP Manager
  - Software Upgrade
  - System Backup
- ▼ **System Configuration**
  - Alarm Codes
  - Alarm/Log Options
  - Applications
  - MPP Servers
  - Report Data
  - SHMP
  - Speech Servers
  - VoIP Connections
  - VPMS Servers
- ▼ **Security**
  - Certificates
  - Licensing
- ▼ **Reports**
  - Standard
  - Custom
  - Scheduled

You are here: [Home](#) > [System Configuration](#) > [VoIP Connections](#) > Add SIP Connection

### Add SIP Connection

Use this page to add a new SIP connection.

Name: ses-srv  
Enable: ☒ Yes ☐ No  
Proxy Transport: TCP

Address	Port	Administration
10.10.15.20	5060	Administration Remove

[Additional Proxy Server](#)

Listener Port: 5060  
SIP Domain: du.rnd.avaya.com  
P-Asserted-Identity:

**Call Capacity**  
Maximum Simultaneous Calls: 10  
☒ All Calls can be either inbound or outbound  
☐ Configure number of inbound and outbound calls allowed

**Save Cancel Help**



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

The screenshot shows the Avaya Voice Portal 5.0 (VoicePortal) interface. The top navigation bar includes the Avaya logo, a welcome message for 'vpadmin', and a timestamp 'Last logged in 26/06/09 at 17:36:02 IST'. The main navigation menu on the left lists various system management and monitoring options. The current page is 'VoIP Connections', which displays a table of configured SIP connections. A message indicates that the entered information has been saved.

**VoIP Connections**

This page displays a list of Voice over Internet Protocol (VoIP) servers that Voice Portal communicates with. You can configure multiple SIP connections, but only one SIP connection can be enabled at any one given time.

• The information that you entered has been saved.

	Name	Enable	Proxy Transport	Proxy Server Address	Proxy Server Port	Listener Port	SIP Domain	Maximum Simultaneous Calls	Inbound Calls Allowed	Outbound Calls Allowed
<input type="checkbox"/>	SES- srv	Yes	TCP	10.10.15.20	5060	5060	du.rnd.avaya.com	10	10	10

**Add** **Delete** **Help**

### 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.

Welcome, vpadmin  
Last logged in today at 11:17:12 IST

**Voice Portal 5.0 (VoicePortal)**
[Home](#)
[Help](#)
[Logoff](#)

Expand All | Collapse All

**User Management**  
Roles  
Users  
Login Options

**Real-Time Monitoring**  
System Monitor  
Active Calls  
Port Distribution

**System Maintenance**  
Audit Log Viewer  
Trace Viewer  
Log Viewer  
Alarm Manager

**System Management**  
MPP Manager  
Software Upgrade  
System Backup

**System Configuration**  
Alarm Codes  
Alarm/Log Options  
**Applications**  
MPP Servers  
Report Data  
SNMP  
Speech Servers  
VoIP Connections  
VOMS Servers

**Security**  
Certificates  
Licensing

**Reports**  
Standard  
Custom  
Scheduled

You are here: [Home](#) > System Configuration > Applications

### Applications

This page displays the VoiceXML and CCXML applications that are currently deployed on the Voice Portal system. When a call comes in, Voice Portal compares the called number or URI with the values in the Launch column, starting with the first application in the list and proceeding down the list in order. As soon as it finds a match, it invokes that application to handle the call. If two or more applications have launch values that overlap or duplicate each other, make sure that the application you want Voice Portal to use appears first in the list. To move an application, click Change Launch Order.

[Change Launch Order](#)

	Name	Enable	MIME Type	URL	Launch	ASR	Languages	TTS	Voices	Configurable Application Variables
<input type="checkbox"/>	<a href="#">Test1</a>	Yes	VoiceXML	http://10.10.15.51/mpp/misc/avptestapp/intro.vxml	5007, 2226	Nuance	English(USA) en-us	Nuance	English(USA) en-US Jennifer F.	
<input type="checkbox"/>	<a href="#">ImmediateVoice</a>	Yes	CCXML	http://10.10.15.90:3188/ImmediateVoice/iv.ccxml.jsp	9995	Nuance	Dutch (Netherlands) nl-nl, English(UK) en-gb	Nuance	Dutch (Netherlands) nl-NL Claire F, English(UK) en-GB Daniel M	
<input type="checkbox"/>	<a href="#">ivSurvey_start</a>	Yes	VoiceXML	http://10.10.15.90:3184/nextidm/vt/100001/ivsury/vrs/nl-NL/vxml/uvwv-start.xml	Outbound	No ASR		No TTS		
<input type="checkbox"/>	<a href="#">ivSurvey</a>	Yes	CCXML	http://10.10.15.90:3184/nextidm/vt/100001/ivsury/vrs/nl-NL/vxml/ccxml-uvwv-2.xml	8887	Nuance	Dutch (Netherlands) nl-nl	Nuance	Dutch (Netherlands) nl-NL Claire F	

Add
Delete
Help

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.

**AVAYA** Welcome, vpadmin  
Last logged in today at 11:17:12 IST

**Voice Portal 5.0 (VoicePortal)** Home ? Help Logoff

Expand All | Collapse All

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > Change Application

### Add Application

Use this page to deploy and configure a new VoiceXML or CCXML application on the Voice Portal system.

Name:

Enable: ☒ Yes ☐ No

MIME Type:

VoiceXML URL:

**Speech Servers**

ASR:  TTS:

**Application Launch**

Type: ☐ Inbound ☐ Inbound Default ☒ Outbound

**Speech Parameters** ▶

**Reporting Parameters** ▶

**Advanced Parameters** ▶

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.

**AVAYA** Welcome, vpadmin  
Last logged in today at 11:17:12 IST

**Voice Portal 5.0 (VoicePortal)**

You are here: [Home](#) > [System Configuration](#) > [Applications](#) > [Change Application](#)

### Add Application

Use this page to deploy and configure a new VoiceXML or CCXML application on the Voice Portal system.

Name:

Enable: ☒ Yes ☐ No

MIME Type:

CCXML URL:

#### Speech Servers

ASR: <input type="text" value="Nuance"/>	TTS: <input type="text" value="Nuance"/>
Languages: <input type="text" value="Dutch(Netherlands) nl-nl"/> <input type="text" value="English(UK) en-gb"/> <input type="text" value="English(USA) en-us"/>	Voices: <input type="text" value="Dutch(Netherlands) nl-NL Claire F"/> <input type="text" value="English(UK) en-GB Daniel M"/> <input type="text" value="English(UK) en-GB Serena F"/>

#### Application Launch

Type: ☒ Inbound ☐ Inbound Default ☐ Outbound

☒ Number ☐ Number Range ☐ URI

Called Number:

**Speech Parameters**

**Reporting Parameters**

**Advanced Parameters**

## 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.

The screenshot shows the Avaya Voice Portal 5.0 (VoicePortal) interface. The top header includes the Avaya logo, a welcome message for 'vpadmin', and the login time 'Last logged in today at 11:17:12 IST'. The left sidebar contains a navigation menu with categories like User Management, Real-Time Monitoring, System Maintenance, System Configuration, Security, and Reports. The main content area shows the breadcrumb 'You are here: Home > System Configuration > VPMS Servers' and the title 'VPMS Servers'. Below the title, a message states: 'This page displays the list of primary and secondary VPMS servers in the Voice Portal system.' A table lists the servers:

Name	Type	Host Address
VPMS	Primary	VPMS

Below the table are 'Add' and 'Delete' buttons. At the bottom of the main content area are four buttons: 'VPMS Settings', 'Email Server Settings', 'Report DB Settings', and 'Help'.

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.

The screenshot shows the Avaya Voice Portal 5.0 (VoicePortal) interface, specifically the 'VPMS Settings' page. The top header is the same as the previous screenshot. The left sidebar is also the same. The main content area shows the breadcrumb 'You are here: Home > System Configuration > VPMS Servers > VPMS Settings' and the title 'VPMS Settings'. Below the title, a message states: 'Use this page to configure system parameters that affect the Voice Portal system.' The page contains several configuration sections:

- Voice Portal Name:** A text field with the value 'VoicePortal'.
- Number of Application Server Failover Logs:** A text field with the value '10'.
- Commands to Retain in MPP Configuration History:** A text field with the value '50'.
- Voice Portal Release 3 Application Reporting Enabled:** Radio buttons for 'Yes' (selected) and 'No'.
- Resource Alerting Thresholds (%):** Two tabs, 'High Water' and 'Low Water'. Below them are two text fields for 'Disk' with values '80' and '60'.
- Web Service Authentication:** A section with a dropdown arrow. It contains:
  - Application Reporting:** Three text fields for 'User Name' (value '<Default>'), 'Password' (masked with dots), and 'Verify Password'.
  - Outcall:** Three text fields for 'User Name' (value 'admin'), 'Password' (masked with dots), and 'Verify Password'.

At the bottom of the main content area are four buttons: 'Save', 'Apply', 'Cancel', and 'Help'.

## 6. Configure SIP Enablement Services

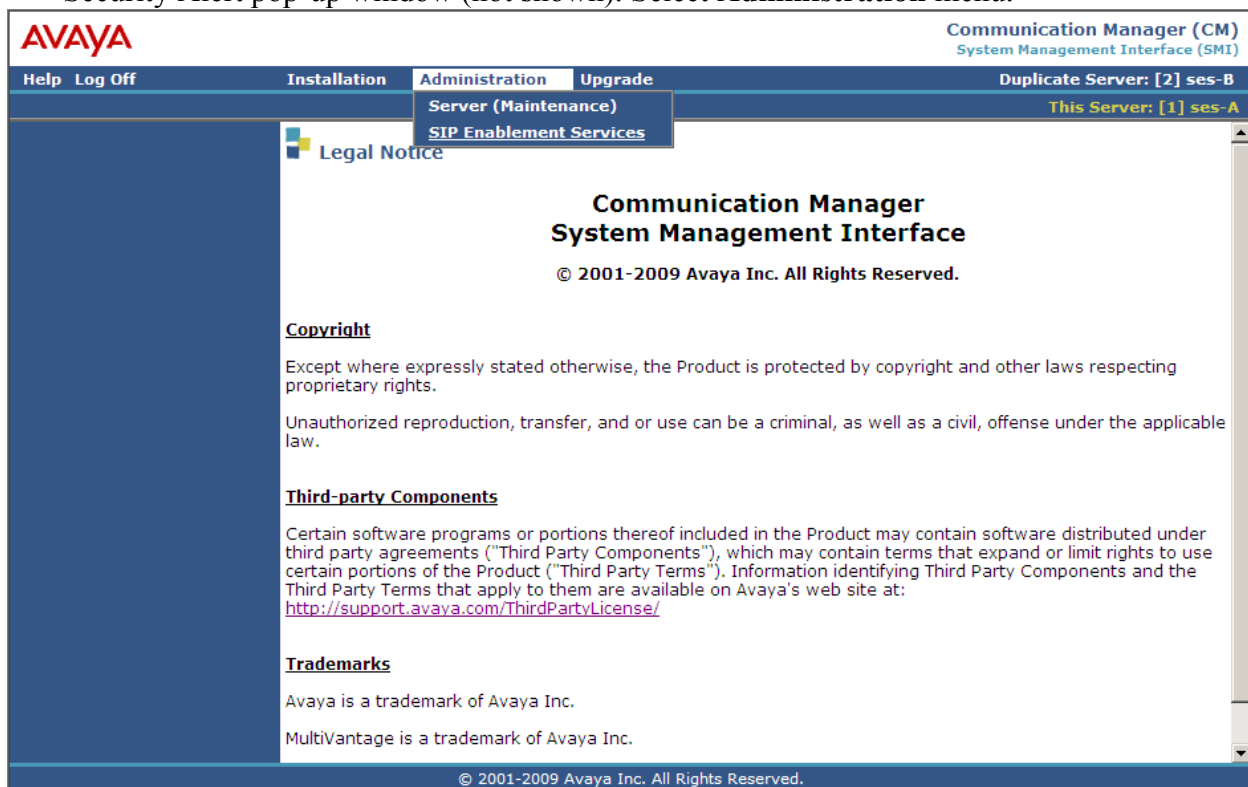
This section only details the configuration required on SIP Enablement Services to set up SIP connectivity to Communication Manager and Avaya Voice Portal and assumes the Avaya SES service is already in place. For additional administration information on Avaya SES, please consult reference [7].

### 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:



**AVAYA** Integrated Management SIP Server Management

Help Exit Primary Server: [1] ses-A Duplicate Server: [2] ses-B

**Top**

- Setup
- Users
  - Address Map Priorities
- Adjunct Systems
- Aggregator
- Certificate Management
- Conferences
- Emergency Contacts
- Export/Import to ProVision
- Hosts
  - IM logs
- Communication Manager Servers
- Communication Manager Extensions
- Server Configuration
- SIP Phone Settings
- Survivable Call Processors
- System Status
- Trace Logger
- Trusted Hosts

Top	
Manage Users	Add and delete Users.
Manage Address Map Priorities	Adjust Address Map Priorities.
Manage Adjunct Systems	Add and delete Adjunct Systems.
Manage Event Aggregators	Add/Delete Event Aggregators.
Certificate Management	Manage Certificates.
Manage Conferencing	Add and delete Conference Extensions.
Manage Emergency Contacts	Add and delete Emergency Contacts.
Export Import to ProVision	Export and import data using ProVision on this host.
Manage Hosts	Add and delete Hosts.
IM logs	Download IM Logs.
Manage Communication Manager Servers	Add and delete Communication Manager Servers.
Manage Communication Manager Extensions	Add and delete Communication Manager Extensions.
Server Configuration	View Properties of the system.
Manage SIP Phone Settings	Add/Delete Phone Settings
Manage Survivable Call Processors	Add and delete Survivable Call Processors.
System Status	View System Status.
Trace Logger	Manage SIP Trace Logs.
Manage Trusted Hosts	Add and delete Trusted Hosts.

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.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The left sidebar contains a navigation menu with options like Top, Setup, Users, Address Map Priorities, Adjunct Systems, Aggregator, Certificate Management, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts, IM logs, Communication Manager Servers (Add, List), Communication Manager Extensions, Server Configuration, SIP Phone Settings, Survivable Call Processors, System Status, Trace Logger, and Trusted Hosts. The main content area is titled 'Add Communication Manager Server Interface' and contains the following fields and options:

- Communication Manager Server Interface Name\***: CMIV
- Host**: 10.10.15.20
- SIP Trunk**
  - SIP Trunk Link Type**: ☐ TCP ☒ TLS
  - SIP Trunk IP Address\***: 10.10.15.15
- Communication Manager Server**
  - Communication Manager Server Admin Address\*** (see Help): 10.10.15.10
  - Communication Manager Server Admin Port\***: 5022
  - Communication Manager Server Admin Login\***: init
  - Communication Manager Server Admin Password\***: [masked]
  - Communication Manager Server Admin Password Confirm\***: [masked]
- SMS Connection Type**: ☒ SSH ☐ Telnet ☐ Not Available

A note at the bottom states: 'Note: If the Communication Manager Server connection type is changed and the admin port value is not also changed, changing connection type to SSH will change the admin port to 5022 when Add or Update is clicked and changing connection type to Telnet will change admin port to 5023 when Add or Update is clicked.'

Fields marked \* are required.

An **Add** button is located at the bottom left of the form.



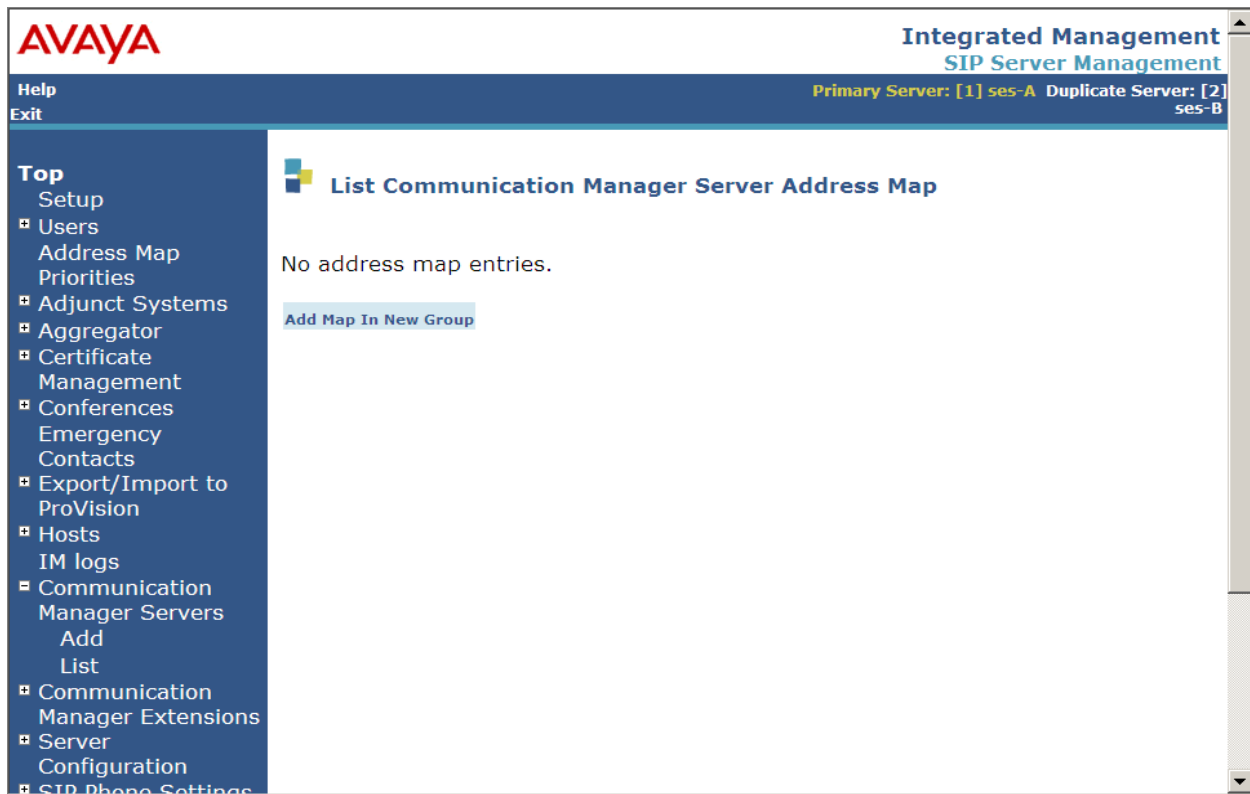
4. Click on **Communication Manager Servers** → **List** to list the configured Communication Manager Server Interface.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo and the title 'Integrated Management SIP Server Management'. Below the header, a status bar indicates 'Primary Server: [1] ses-A Duplicate Server: [2] ses-B'. The left sidebar contains a navigation menu with options like 'Help', 'Exit', 'Top', 'Setup', 'Users', 'Address Map', 'Priorities', 'Adjunct Systems', 'Add', 'List', 'Aggregator', 'Certificate Management', 'Conferences', 'Emergency', 'Contacts', 'Export/Import to ProVision', 'Hosts', 'IM logs', 'Communication Manager Servers', 'Add', 'List', and 'Communication'. The main content area is titled 'List Communication Manager Servers' and features a table with two rows of data. The table has three columns: 'Commands', 'Interface', and 'Host'. The first row shows 'Edit', 'Extensions', 'Map', 'Test-Link', 'Delete' for 'CMIV' on host '10.10.15.20'. The second row shows 'Edit', 'Extensions', 'Map', 'Test-Link', 'Delete' for 'CMipc' on host '10.10.15.20'. Below the table, there is a link to 'Add Another Communication Manager Server Interface'.

Commands					Interface	Host
Edit	Extensions	Map	Test-Link	Delete	CMIV	10.10.15.20
Edit	Extensions	Map	Test-Link	Delete	CMipc	10.10.15.20

[Add Another 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).

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The left sidebar contains a navigation menu with options like Top, Setup, Users, Address Map, Priorities, Adjunct Systems, Aggregator, Certificate Management, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts, IM logs, Communication Manager Servers (Add, List), Communication Manager Extensions, Server Configuration, and SIP Phone Settings. The main content area is titled 'Add Communication Manager Server Address Map'. It contains two input fields: 'Name\*' with the value '71xx' and 'Pattern\*' with the value '^sip:71[0-9]{2}'. Below these fields is a note: 'Fields marked \* are required.' and an 'Add' button.

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

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The left sidebar is the same as in the previous screenshot. The main content area is titled 'List Communication Manager Server Address Map'. It displays a table with the following data:

Commands	Name	Commands	Contact
Edit Delete	71xx	Edit Delete	sip:\$(user) @10.10.15.15:5061;transport=tls

Below the table, there are two buttons: 'Add Another Map' and 'Add Another Contact'. To the right of the table, there is a 'Delete Group' button. At the bottom of the main content area, there is a button labeled 'Add Map In New Group'.

## 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.

The screenshot shows the Avaya SIP Server Management interface. The left sidebar contains a navigation menu with options like Setup, Users, Address Map, Priorities, Adjunct Systems, Add, List, Aggregator, Certificate Management, Conferences, Emergency Contacts, Export/Import to ProVision, Hosts, IM logs, Communication Manager Servers, Communication Manager Extensions, and Server. The main content area is titled 'Add Adjunct System'. It contains a form with the following fields: 'System Name\*' with the value 'avayavoiceportal50', 'Host' with a dropdown menu showing '10.10.15.20', and a 'Replace URI' checkbox which is unchecked. Below the form is an 'Add' button. A note at the bottom of the form states 'Fields marked \* are required.'

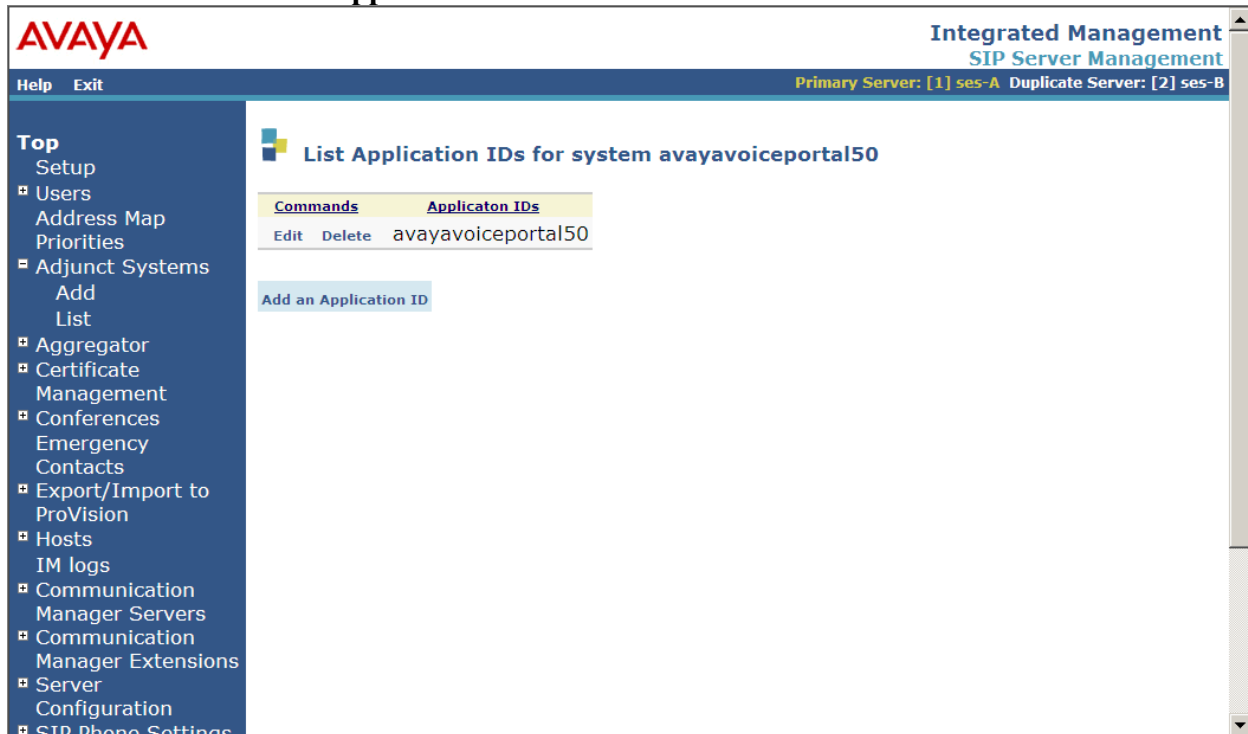
2. Click on the **Adjunct Systems**→**List**→**List Application IDs (1)** link for the adjunct system added (**avayavoiceportal50**) in Step 1 above.

The screenshot shows the Avaya SIP Server Management interface. The left sidebar is the same as in the previous screenshot. The main content area is titled 'List Adjunct Systems'. It contains a table with the following data:

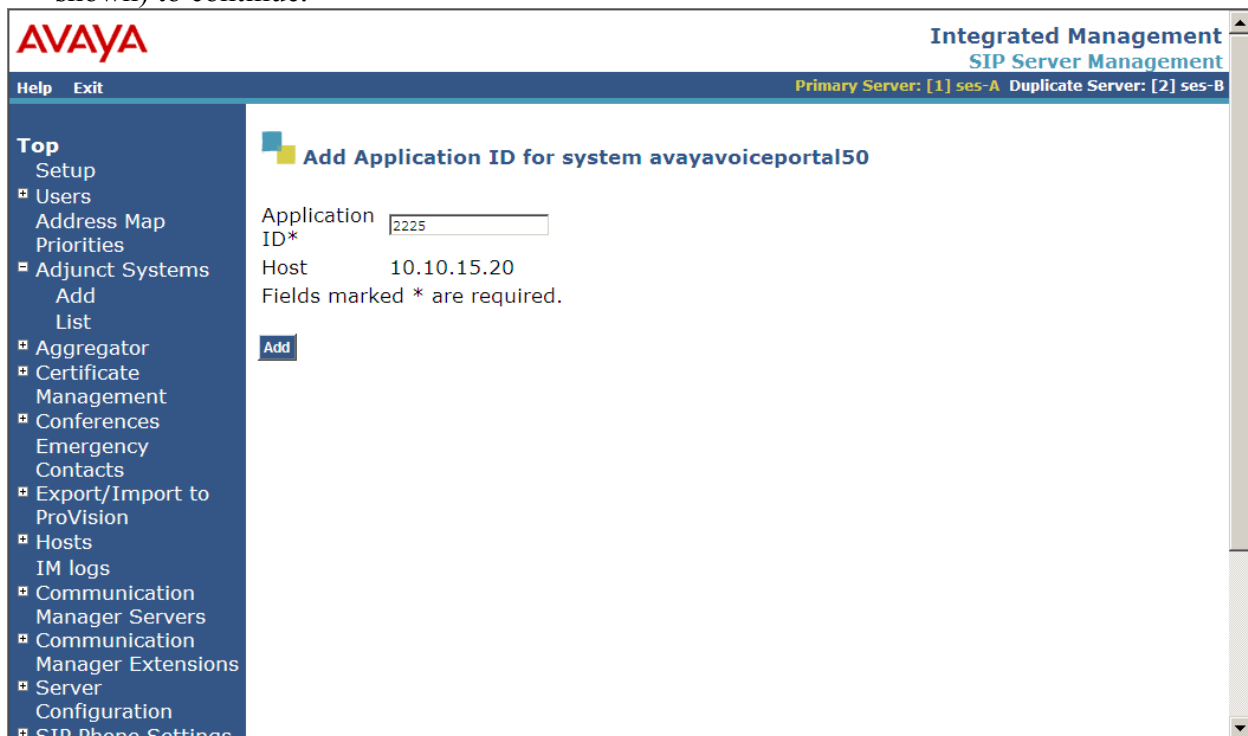
Commands				System	Host
Edit	Delete	List Adjunct Servers(0)	List Application IDs(1)	avayavoiceportal50	10.10.15.20
Edit	Delete	List Adjunct Servers(1)	List Application IDs(2)	silmm	10.10.15.20
Edit	Delete	List Adjunct Servers(0)	List Application IDs(1)	test	10.10.15.20
Edit	Delete	List Adjunct Servers(1)	List Application IDs(1)	voiceportal50	10.10.15.20
Edit	Delete	List Adjunct Servers(1)	List Application IDs(2)	voiceportalsystem	10.10.15.20

Below the table is a button labeled 'Add Another Adjunct System'.

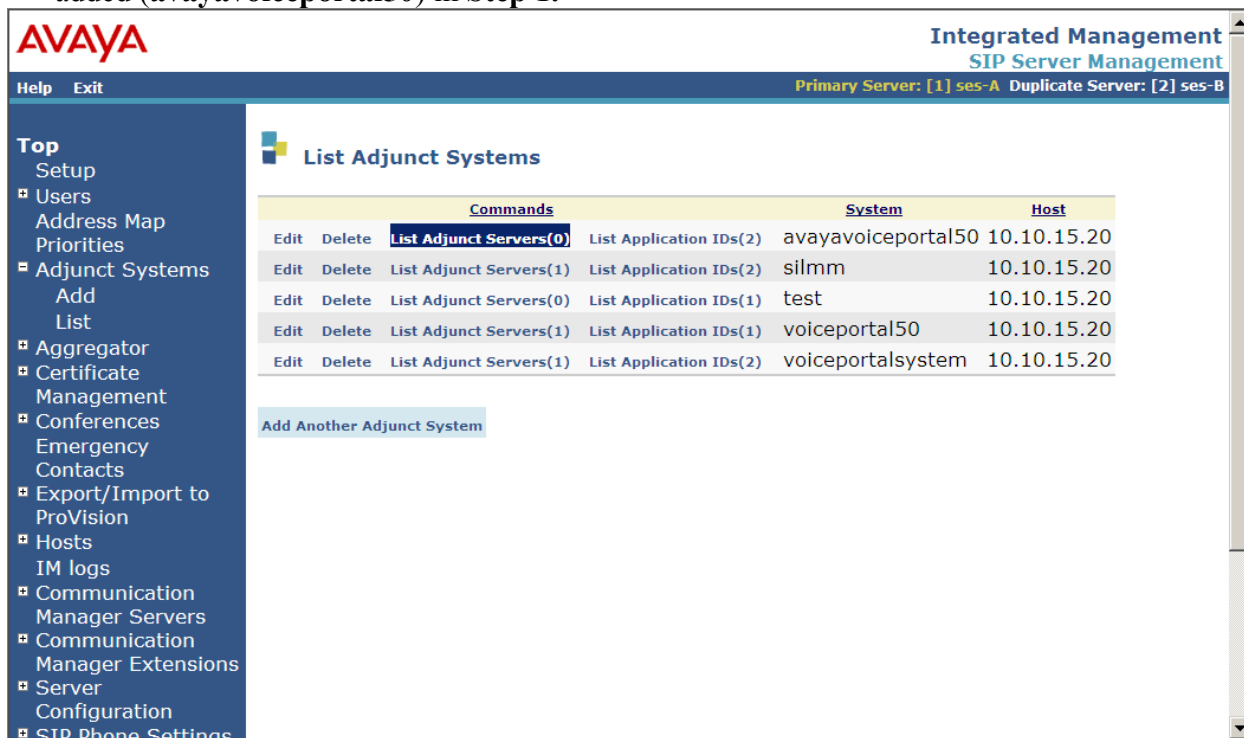
3. Click on the **Add an Application ID** link.



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.



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



**AVAYA** Integrated Management SIP Server Management  
Primary Server: [1] ses-A Duplicate Server: [2] ses-B

Help Exit

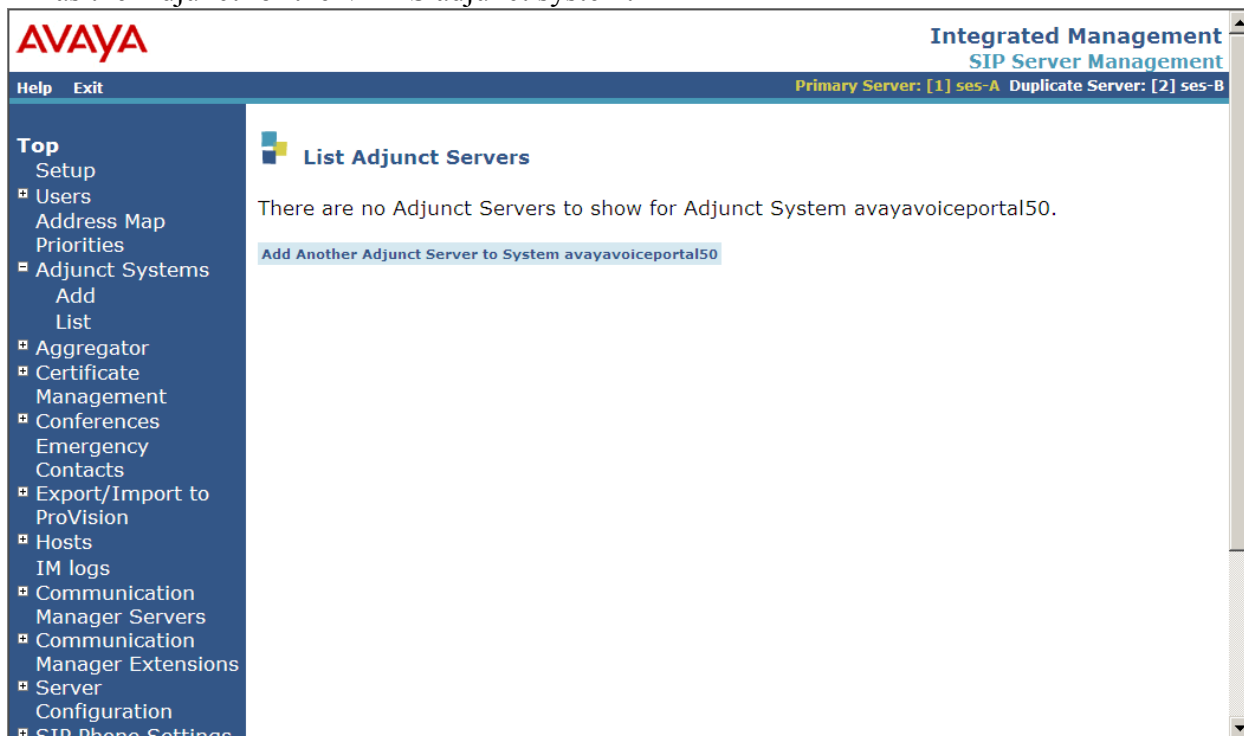
**Top**  
 Setup  
 Users  
 Address Map  
 Priorities  
 Adjunct Systems  
 Add  
 List  
 Aggregator  
 Certificate  
 Management  
 Conferences  
 Emergency  
 Contacts  
 Export/Import to  
 ProVision  
 Hosts  
 IM logs  
 Communication  
 Manager Servers  
 Communication  
 Manager Extensions  
 Server  
 Configuration  
 SIP Phone Settings

**List Adjunct Systems**

Commands		System	Host
Edit	Delete	<b>List Adjunct Servers(0)</b>	List Application IDs(2)
		avayavoiceportal50	10.10.15.20
Edit	Delete	List Adjunct Servers(1)	List Application IDs(2)
		silmm	10.10.15.20
Edit	Delete	List Adjunct Servers(0)	List Application IDs(1)
		test	10.10.15.20
Edit	Delete	List Adjunct Servers(1)	List Application IDs(1)
		voiceportal50	10.10.15.20
Edit	Delete	List Adjunct Servers(1)	List Application IDs(2)
		voiceportalsystem	10.10.15.20

Add Another Adjunct System

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



**AVAYA** Integrated Management SIP Server Management  
Primary Server: [1] ses-A Duplicate Server: [2] ses-B

Help Exit

**Top**  
 Setup  
 Users  
 Address Map  
 Priorities  
 Adjunct Systems  
 Add  
 List  
 Aggregator  
 Certificate  
 Management  
 Conferences  
 Emergency  
 Contacts  
 Export/Import to  
 ProVision  
 Hosts  
 IM logs  
 Communication  
 Manager Servers  
 Communication  
 Manager Extensions  
 Server  
 Configuration  
 SIP Phone Settings

**List Adjunct Servers**

There are no Adjunct Servers to show for Adjunct System avayavoiceportal50.

Add Another Adjunct Server to System avayavoiceportal50

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.

The screenshot displays the Avaya Integrated Management SIP Server Management interface. The top header includes the Avaya logo, the title 'Integrated Management SIP Server Management', and status information: 'Primary Server: [1] ses-A Duplicate Server: [2] ses-B'. A navigation menu on the left lists various configuration options, with 'Add Adjunct Server' selected. The main content area shows the 'Add Adjunct Server' form with the following fields: Host (10.10.15.20), System (avayavoiceportal50), Server Name\* (VP50), Server ID (2228), Link Type (radio buttons for TCP and TLS, with TCP selected), and Server IP Address\* (10.10.15.51). A note states 'Fields marked \* are required.' and an 'Add' button is at the bottom left of the form area.

Add Adjunct Server	
Host	10.10.15.20
System	avayavoiceportal50
Server Name*	VP50
Server ID	2228
Link Type	<input checked="" type="radio"/> TCP <input type="radio"/> TLS
Server IP Address*	10.10.15.51

Fields marked \* are required.

Add

## 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>/ivsSurvey”. 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.

U ser

telefoneren mijn instellingen resultaten ontwikkeling overzicht uitloggen

**PROFIEL AANPASSEN**

Voorletter(s): U \*

Achternaam: ser \*

Telefoonnummer: 7110 \*

Telefoontype: mobiel v

Email: devcon \*

Wachtwoord: ••••• \*

Nieuw Wachtwoord:

Bevestigen Wachtwoord:

opslaan

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

U ser

telefoneren mijn instellingen resultaten ontwikkeling overzicht uitloggen

**PROFIEL AANPASSEN**

Uw profiel is aangepast.

verder

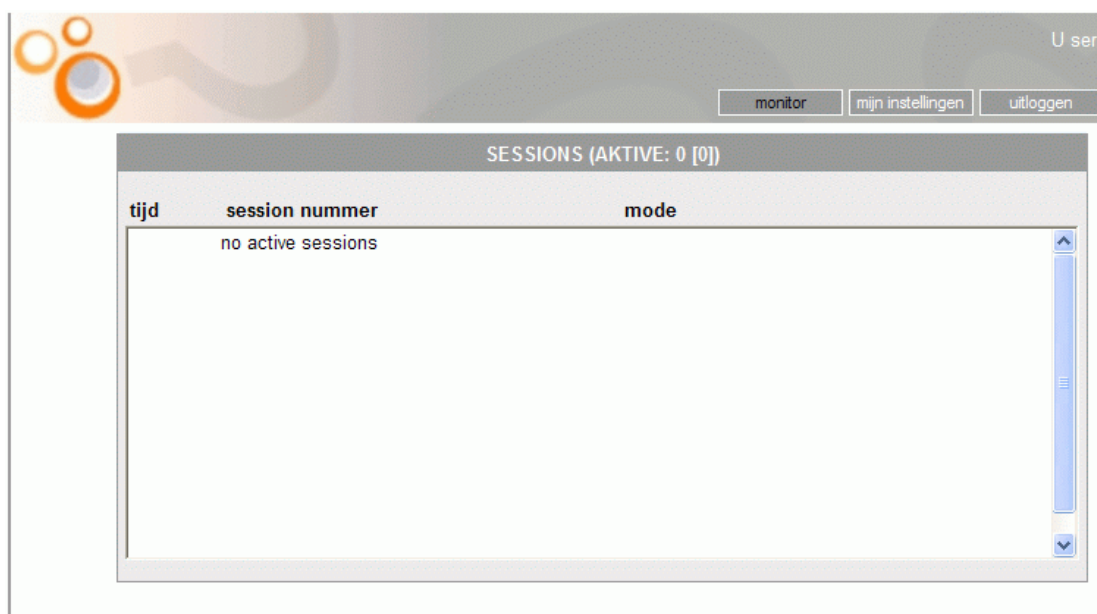
5. Fill in a test phone number (in this sample application, 7112 is used) in **Telefoonnummer** field and click **verbinden** to proceed with the call.

The screenshot shows the ivSurvey application interface. At the top, there is a navigation bar with buttons: 'telefoneren', 'mijn instellingen', 'resultaten', 'ontwikkeling', 'overzicht', and 'uitloggen'. The main content area is divided into two sections. On the left, under 'GEBELDE NUMMERS', there is a table with columns 'tijd', 'nummer', and 'status'. The table is currently empty, showing 'geen nummers'. On the right, under 'NUMMER BELLEN', there is a dropdown menu set to 'telefonie - test'. Below this, there is a text input field labeled 'Telefoonnummer:' containing the value '7112', and a 'verbinden' button. Below the input field, there is a 'STATUS' section with the text 'Voer een telefoonnummer in.' and a telephone icon.

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>/ivsSurvey”. Enter Agent’s **Login** and **Wachtwoord** (Password) and press **inloggen** button.

The screenshot shows the ivSurvey application interface. At the top, there is a navigation bar with buttons: 'telefoneren', 'mijn instellingen', 'resultaten', 'ontwikkeling', 'overzicht', and 'uitloggen'. The main content area is divided into two sections. On the left, under 'GEBELDE NUMMERS', there is a table with columns 'tijd', 'nummer', and 'status'. The table is currently empty, showing 'geen nummers'. On the right, under 'NUMMER BELLEN', there is a dropdown menu set to 'telefonie - test'. Below this, there is a text input field labeled 'Telefoonnummer:' containing the value '7112', and a 'verbinden' button. Below the input field, there is a 'STATUS' section with the text 'Voer een telefoonnummer in.' and a telephone icon.

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



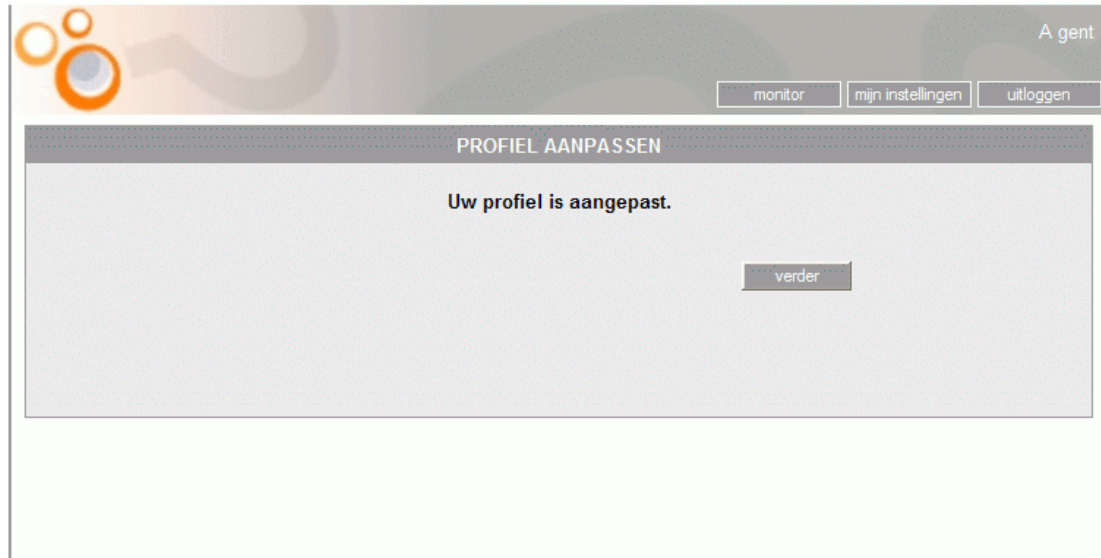
8. Edit **Telefoonnummer** field to a test number (in this sample configuration, 7111 is used), fill in **Wachtwoord** with password and click **opslaan** button to save changes.

The screenshot shows a web interface for editing a profile. At the top right, it says 'A gent'. Below this are three buttons: 'monitor', 'mijn instellingen', and 'uitloggen'. The main content area is titled 'PROFIEL AANPASSEN'. It contains the following form fields:

- Voorletter(s): A \*
- Achternaam: gent \*
- Telefoonnummer: 7111 \*
- Telefoontype: mobiel (dropdown menu)
- Email: agent \*
- Wachtwoord: (masked with dots) \*
- Nieuw Wachtwoord: (empty)
- Bevestigen Wachtwoord: (empty)

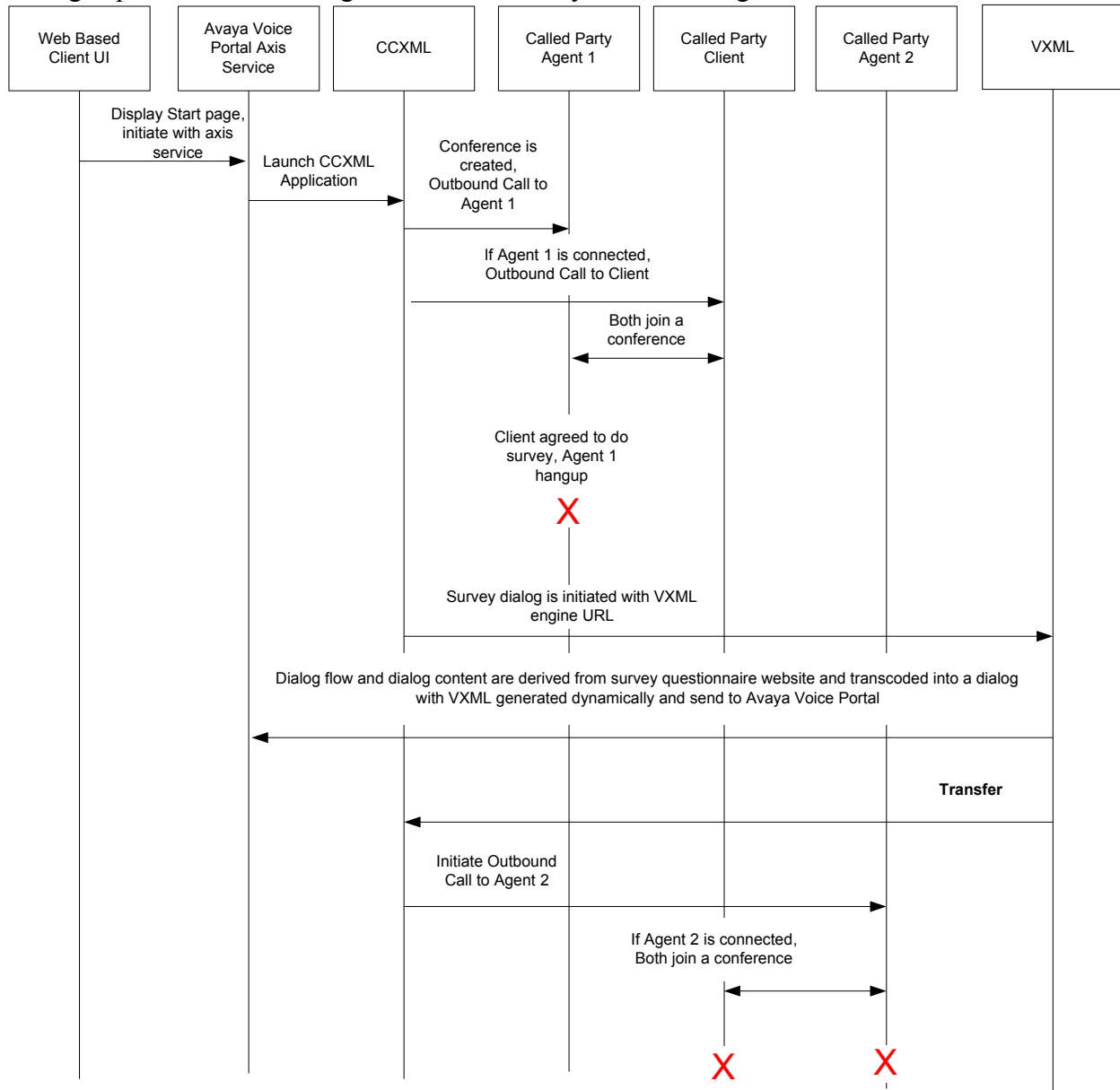
An 'opslaan' button is located at the bottom right of the form.

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

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