



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

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# **Application Notes for Avaya Voice Portal 4.1 with Dialogs Unlimited ImmediateVoice v2.0 Speech Application Engine - Issue 1.0**

### **Abstract**

These Application Notes describe the steps for configuring Avaya Voice Portal 4.1 with Dialogs Unlimited ImmediateVoice 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. ImmediateVoice extracts the e-commerce interactions from the web pages, converts them into customer speech dialog and converts the speech back to the required data field inputs. This allows the original web e-commerce transaction to be completed by voice. In addition, ImmediateVoice has powerful online Service Creation tools that discover the customer interaction types automatically and allow for a Rapid Application Development with iterative quality improvements of the Speech Application. 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 4.1 with Dialogs Unlimited ImmediateVoice 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.

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 co-resident with MPP uses the SIP protocol to communicate with Avaya Aura™ 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. ImmediateVoice utilises Avaya Voice Portal voice browser capability.

## 1.1. Interoperability Compliance Testing

The interoperability compliance testing focused on verifying interoperability of the Dialogs Unlimited ImmediateVoice with Avaya Voice Portal based on SIP connectivity. The focus of testing is VXML (CCXML) interoperability and support of major call functions like call pickup, call transfer, making inputs in the web forms by voice and web browsing by voice as per selected ImmediateVoice Speech Application scope.

## 1.2. Support

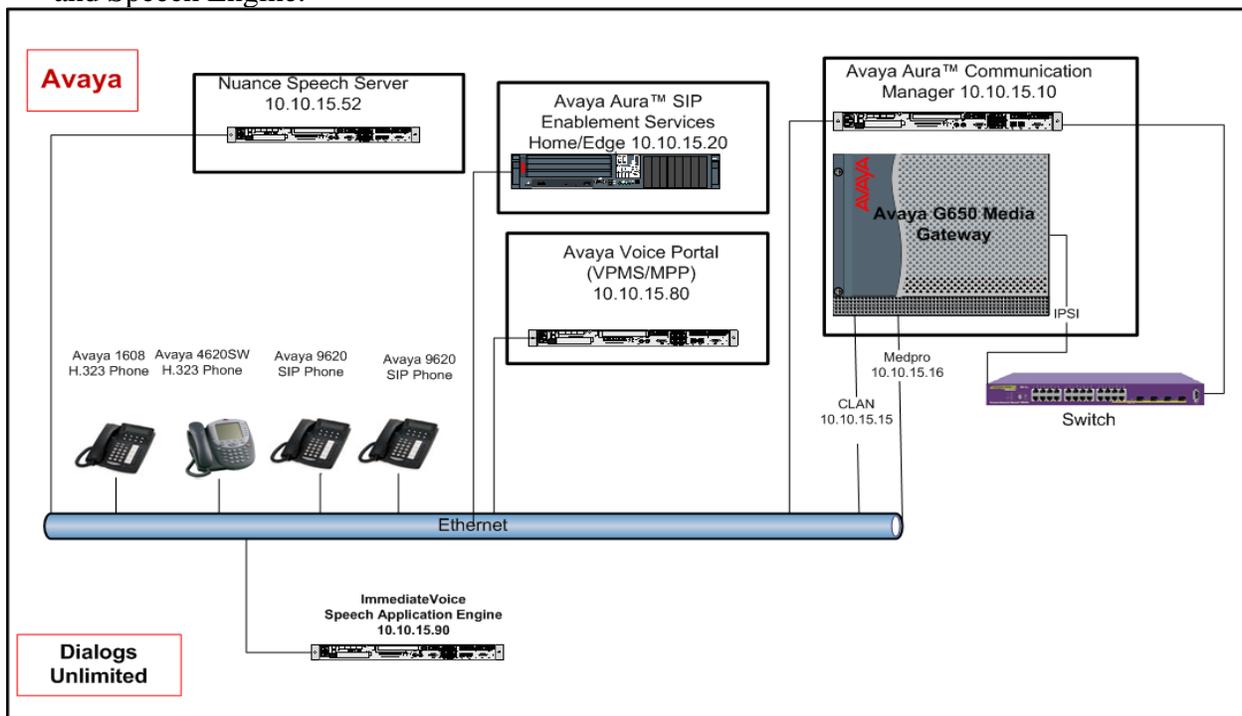
Technical support for the Avaya products can be obtained from Avaya. See the support link at [support.avaya.com](http://support.avaya.com) for contact information.

Technical support for the ImmediateVoice products can be obtained from Dialogs Unlimited. See the support link at [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 16xx, 46xx and 96xx 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 ImmediateVoice 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 and MPP co-resident on a single server)	Avaya Voice Portal 4.1 sp3 VPMS 4.1.0.3.0111 MPP 4.1.0.3-0002
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 ImmediateVoice Speech Application Engine	ImmediateVoice 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:</b>	<b>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:</b>	<b>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
                                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
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
                                                                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
```

- 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
                                               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 x**, where **x** 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
1: 1    0
2:
3:
4:
5:
6:
                                           n user
                                           n user
                                           n user
                                           n user
                                           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          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 String	Total Min Max		Route Pattern	Call Type	Node Num	ANI 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.

**AVAYA** Welcome, vpadmin  
Last logged in today at 10:10:49 GMT

**Voice Portal 4.1 (VP41)** Home Help Logoff

Expand All | Collapse All

- ▼ User Management
  - Users
- ▼ System Maintenance
  - System Monitor
  - MPP Manager
  - Active Calls
  - Port Distribution
  - Audit Log Viewer
  - Log Viewer
  - Alarm Manager
- ▼ System Configuration
  - Applications
  - Certificates
  - Licensing
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - Viewer Settings
  - VoIP Connections
  - VPMS Servers
- ▼ Reports
  - Application Summary
  - Application Detail
  - Call Summary
  - Call Detail
  - Performance
  - Session Summary
  - Session Detail

You are here: Home

### Voice Portal Management System Version 4.1.0.3.0111

Voice Portal Management System (VPMS) is the consolidated web-based application for administering Voice Portal. Through the VPMS interface, you can configure Voice Portal, check the status of a Voice Portal component, and generate reports related to system operation.

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Last Login: 11/06/09 10:10:49 GMT

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 **System Configuration** → **Licensing**.

**AVAYA** Welcome, vpadmin  
Last logged in today at 10:10:49 GMT

**Voice Portal 4.1 (VP41)** Home Help Logoff

Expand All | Collapse All

- ▼ **User Management**
  - Users
- ▼ **System Maintenance**
  - System Monitor
  - MPP Manager
  - Active Calls
  - Port Distribution
  - Audit Log Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Configuration**
  - Applications
  - Certificates
  - Licensing**
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - Viewer Settings
  - VoIP Connections
  - VPMS Servers
- ▼ **Reports**
  - Application Summary
  - Application Detail
  - Call Summary
  - Call Detail
  - Performance
  - Session Summary
  - Session Detail

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

Telephony Ports:	200
ASR Connections:	200
TTS Connections:	200
Version:	4
Last Successful Poll:	11/06/09 10:24:00 GMT

**License Server**

License Server URL:  **Verify**

**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 today at 10:24:07 GMT

**Voice Portal 4.1 (VP41)** Home Help Logoff

Expand All | Collapse All

- ▼ **User Management**
  - Users
- ▼ **System Maintenance**
  - System Monitor
  - MPP Manager
  - Active Calls
  - Port Distribution
  - Audit Log Viewer
  - Log Viewer
  - Alarm Manager
- ▼ **System Configuration**
  - Applications
  - Certificates
  - Licensing
  - MPP Servers
  - Report Data
  - SNMP
  - Speech Servers
  - Viewer Settings
  - VoIP Connections**
  - VPMS Servers
- ▼ **Reports**
  - Application Summary
  - Application Detail
  - Call Summary
  - Call Detail
  - Performance
  - Session Summary
  - Session Detail

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	20	20	20

**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 4.1 (VP41) interface. At the top right, it says "Welcome, vpadmin" and "Last logged in today at 10:57:27 GMT". The main header is "Voice Portal 4.1 (VP41)" with navigation links for Home, Help, and Logoff. A left-hand navigation menu is expanded to show "System Configuration" > "VoIP Connections" > "Add SIP Connection". The main content area has a breadcrumb trail: "You are here: Home > System Configuration > VoIP Connections > Add SIP Connection". The title is "Add SIP Connection". Below the title is the instruction: "Use this page to add a new SIP connection." The form contains the following fields:

- Name:
- Proxy Transport:
- Proxy Server Address:
- Proxy Server Port:

At the bottom of the form are three buttons: "Continue", "Cancel", and "Help". A dark "Administration" button is also visible to the right of the Proxy Server Address field.

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 **20** (The maximum number of calls that this SIP trunk can handle at one time)Click on the **Save** button.

The screenshot shows the Avaya Voice Portal 4.1 (VP41) interface. The top navigation bar includes the Avaya logo, the user name 'Welcome, vpadmin', and the last login time 'Last logged in today at 10:57:27 GMT'. The main header shows 'Voice Portal 4.1 (VP41)' and navigation links for Home, Help, and Logoff. A left sidebar contains a tree view of system management options, including User Management, System Maintenance, System Configuration, and Reports. The main content area is titled 'Add SIP Connection' and contains the following configuration fields:

- Name:** ses-srv
- Enable:**  Yes  No
- Proxy Transport:** TCP
- Proxy Server Address:** 10.10.15.20
- Proxy Server Port:** 5060
- Listener Port:** 5060
- SIP Domain:** du.rnd.avaya.com
- P-Asserted-Identity:** (empty field)

Below the fields is a section for **Call Capacity** with the following options:

- Maximum Simultaneous Calls:** 20
- All Calls can be either inbound or outbound
- Configure number of inbound and outbound calls allowed

At the bottom of the form are three buttons: **Save**, **Cancel**, and **Help**. A dark grey button labeled **Administration** is also visible next to the Proxy Server Address field.

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

The screenshot shows the Avaya Voice Portal 4.1 (VP41) interface. The top navigation bar includes the Avaya logo, the user name 'Welcome, vpadmin', and the last login time 'Last logged in today at 10:57:27 GMT'. The main header shows 'Voice Portal 4.1 (VP41)' and navigation links for Home, Help, and Logoff. The left sidebar contains a menu with categories: User Management, System Maintenance, System Configuration, and Reports. The main content area is titled 'VoIP Connections' and includes a breadcrumb trail: 'You are here: Home > System Configuration > VoIP Connections'. Below the title, there is a message: '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.' A confirmation message states: 'The information that you entered has been saved.' The main content area features a tabbed interface with 'H.323' and 'SIP' tabs. Below the tabs is a table with the following columns: Name, Enable, Proxy Transport, Proxy Server Address, Proxy Server Port, Listener Port, SIP Domain, Maximum Simultaneous Calls, Inbound Calls Allowed, and Outbound Calls Allowed. The table contains one entry for 'sss-srv' with the following values: Enable: Yes, Proxy Transport: TCP, Proxy Server Address: 10.10.15.20, Proxy Server Port: 5060, Listener Port: 5060, SIP Domain: du.rnd.avaya.com, Maximum Simultaneous Calls: 20, Inbound Calls Allowed: 20, and Outbound Calls Allowed: 20. Below the table are three buttons: Add, Delete, and Help.

	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"/>	sss-srv	Yes	TCP	10.10.15.20	5060	5060	du.rnd.avaya.com	20	20	20

### 5.3. Configure an Application for ImmediateVoice

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 ARS server
- Adding a TTS server

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

The screenshot shows the Avaya Voice Portal 4.1 (VP41) web interface. The top navigation bar includes the Avaya logo, the user name 'Welcome, vpadmin', and the last login time 'Last logged in today at 10:57:27 GMT'. The main navigation menu on the left lists various system management and configuration options, with 'System Configuration' expanded to show 'Applications'. The main content area displays the 'Applications' page, which includes a table of currently deployed applications. The table has columns for Name, Enable, MIME Type, URL, and Launch. Three applications are listed: 'Test', 'playprompts', and 'OutboundBlast'. Below the table are buttons for 'Add', 'Delete', and 'Help'.

<input type="checkbox"/>	Name	Enable	MIME Type	URL	Launch
<input type="checkbox"/>	<a href="#">Test</a>	Yes	VoiceXML	http://10.10.15.80/mpp/misc/avptestapp/intro.vxml	2221, 2225, 5000, 2220, 2229
<input type="checkbox"/>	<a href="#">playprompts</a>	Yes	CCXML	http://10.10.15.80/mpp/misc/avptestapp/root.ccxml	5001
<input type="checkbox"/>	<a href="#">OutboundBlast</a>	Yes	CCXML/VoiceXML	http://127.0.0.1:8080/OutboundCallBlastCC/ccxml/OutboundCall.jsp, http://127.0.0.1:8080/OutboundCallBlastDialog/Start	Outbound

2. Enter the following fields to add a new application:

- **Name:** set to **ImmediateVoice** (A descriptive name for the test VXML (CCXML) 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:** 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.

The screenshot displays the 'Add Application' configuration page in the Avaya Voice Portal 4.1 (VP41) interface. The page is titled 'Add Application' and provides instructions for deploying and configuring a new VoiceXML or CCXML application. The configuration fields are as follows:

- Name:** ImmediateVoice
- Enable:** Yes (selected)
- MIME Type:** VoiceXML
- VoiceXML URL:** http://10.10.15.90:3188/ImmediateVoice/iv.ccxml.jsp
- Speech Servers:**
  - ASR:** Nuance
  - Languages:** Dutch(Netherlands) nl-nl, English(GreatBritain) en-gb, English(USA) en-us
  - TTS:** Nuance
  - Voices:** Dutch(Netherlands) nl-NL Claire F, English(British) en-GB Daniel M, English(Great Britain) en-GB Serena
- Application Launch:**
  - Type:** Inbound (selected), Inbound Default, Outbound
  - Number** (selected), Number Range, URI
  - Called Number:** 2225

The page also includes sections for Speech Parameters, Reporting Parameters, and Advanced Parameters, each with a right-pointing arrow. At the bottom, there are buttons for Save, Cancel, and Help.

## 6. Configure SIP Enablement Services

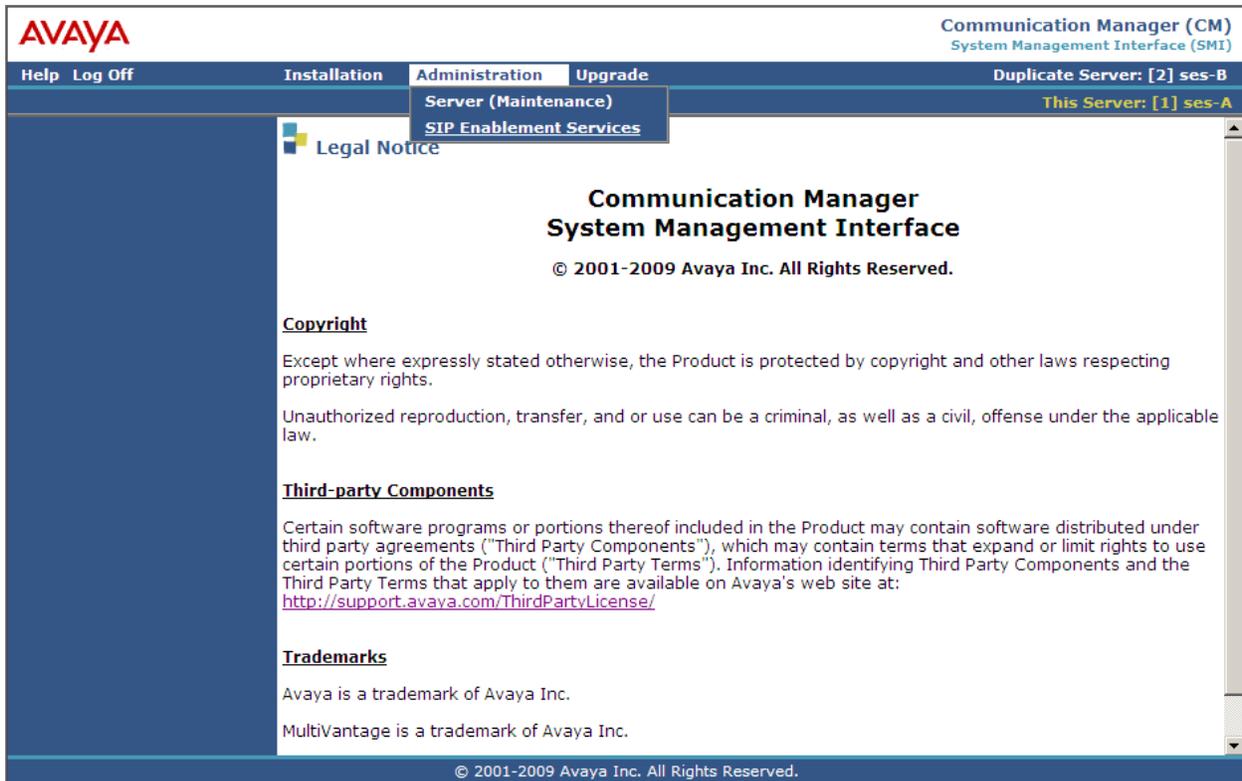
This section only details the configuration required on SIP Enablement Services to set up SIP connectivity to Communication Manager and Avaya Voice Portal and assumes the Avaya SES service is already in place. For additional administration information on Avaya SES, please consult reference [7].

### 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:

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 server information: 'Primary Server: [1] ses-A Duplicate Server: [2] ses-B'. A navigation menu on the left lists various system components. The main content area features a 'Top' section with a list of management tasks and their descriptions.

Top	
<b>Manage Users</b>	Add and delete Users.
<b>Manage Address Map Priorities</b>	Adjust Address Map Priorities.
<b>Manage Adjunct Systems</b>	Add and delete Adjunct Systems.
<b>Manage Event Aggregators</b>	Add/Delete Event Aggregators.
<b>Certificate Management</b>	Manage Certificates.
<b>Manage Conferencing</b>	Add and delete Conference Extensions.
<b>Manage Emergency Contacts</b>	Add and delete Emergency Contacts.
<b>Export Import to ProVision</b>	Export and import data using ProVision on this host.
<b>Manage Hosts</b>	Add and delete Hosts.
<b>IM logs</b>	Download IM Logs.
<b>Manage Communication Manager Servers</b>	Add and delete Communication Manager Servers.
<b>Manage Communication Manager Extensions</b>	Add and delete Communication Manager Extensions.
<b>Server Configuration</b>	View Properties of the system.
<b>Manage SIP Phone Settings</b>	Add/Delete Phone Settings
<b>Manage Survivable Call Processors</b>	Add and delete Survivable Call Processors.
<b>System Status</b>	View System Status.
<b>Trace Logger</b>	Manage SIP Trace Logs.
<b>Manage Trusted Hosts</b>	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 page title is "Add Communication Manager Server Interface". The interface includes a navigation menu on the left 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 contains the following configuration fields:

- Communication Manager Server Interface Name\***: CMIV
- Host**: 10.10.15.20
- SIP Trunk Link Type**:  TCP  TLS
- SIP Trunk IP Address\***: 10.10.15.15
- Communication Manager Server Admin Address\***: 10.10.15.10 (see Help)
- 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 of the form 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." Below the note, it says "Fields marked \* are required." and there is an "Add" button.

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

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  
 Add  
**List**  
 Communication

List Communication Manager Servers

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

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

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  
 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

List Communication Manager Server Address Map

No address map entries.

Add Map In New Group

6. Add **Name** field and **Pattern** field 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 main heading is "Add Communication Manager Server Address Map". There are two input fields: "Name\*" with the value "71xx" and "Pattern\*" with the value "^sip:71[0-9]{2}". Below the 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 displaying a list of Communication Manager Server Address Maps. The table has columns for "Commands", "Name", "Commands", and "Contact".

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

Below the table, there are buttons for "Add Another Map", "Add Another Contact", and "Delete Group". There is also a link "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 **voiceportalsystem** 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 displays the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes the Avaya logo, 'Help Exit', and server status: 'Primary Server: [1] ses-A Duplicate Server: [2] ses-B'. A left-hand menu lists various system management options, with 'Adjunct Systems' expanded to show 'Add' and 'List'. The main content area is titled 'Add Adjunct System' and contains the following form fields:

- System Name\***: A text input field containing 'voiceportalsystem'.
- Host**: A dropdown menu with '10.10.15.20' selected.
- Replace URI**: An unchecked checkbox.

Below the form fields, a note states: 'Fields marked \* are required.' A blue 'Add' button is positioned at the bottom of the form area.

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

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

**List Adjunct Systems**

Commands			System	Host	
Edit	Delete	List Adjunct Servers(1)	List Application IDs(2)	silmm	10.10.15.20
Edit	Delete	List Adjunct Servers(1)	List Application IDs(3)	voiceportal50	10.10.15.20
Edit	Delete	List Adjunct Servers(0)	<b>List Application IDs(1)</b>	voiceportalsystem	10.10.15.20

[Add Another Adjunct System](#)

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

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

**List Application IDs for system voiceportalsystem**

Commands	Application IDs	
Edit	Delete	voiceportalsystem

[Add an Application ID](#)

- Enter the extension **2225** for the **Application ID** field (refer to **Step 2** 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.

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes 'Help' and 'Exit' on the left, and 'Primary Server: [1] ses-A Duplicate Server: [2] ses-B' on the right. A left-hand menu lists various system management options, with 'Adjunct Systems' expanded to show 'Add' and 'List'. The main content area is titled 'Add Application ID for system voiceportalsystem' and contains a form with the following fields:

- Application ID\*:
- Host: 10.10.15.20

Below the form is an 'Add' button and a note: 'Fields marked \* are required.'

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

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top navigation bar is the same as in the previous screenshot. The left-hand menu has 'Adjunct Systems' expanded to 'List'. The main content area is titled 'List Adjunct Systems' and displays a table with the following data:

		Commands	System	Host	
Edit	Delete	List Adjunct Servers(1)	List Application IDs(2)	silmm	10.10.15.20
Edit	Delete	List Adjunct Servers(1)	List Application IDs(3)	voiceportal50	10.10.15.20
Edit	Delete	List Adjunct Servers(0)	List Application IDs(2)	voiceportalsystem	10.10.15.20

Below the table is an 'Add Another Adjunct System' button.

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

The screenshot shows the Avaya Integrated Management SIP Server Management interface. The top navigation bar includes the Avaya logo, 'Help Exit', and server status: 'Primary Server: [1] ses-A Duplicate Server: [2] ses-B'. The left sidebar contains a menu with categories like 'Top', 'Users', 'Adjunct Systems', 'Aggregator', 'Certificate Management', 'Conferences', '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', and 'Trusted Hosts'. The main content area is titled 'List Adjunct Servers' and contains the text: 'There are no Adjunct Servers to show for Adjunct System voiceportalsystem.' Below this text is a blue button labeled 'Add Another Adjunct Server to System voiceportalsystem'.

- Enter **voiceportal** (a unique name for the MPP adjunct server) in the **Server Name** field. Enter the extension **2229** 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 **10.10.15.80** for the **Server IP Address** field. Click the **Add** button to submit the change. Click on **Continue** button (not shown) to continue.

The screenshot shows the 'Add Adjunct Server' form in the Avaya Integrated Management SIP Server Management interface. The form fields are: Host (10.10.15.20), System (voiceportalsystem), Server Name (voiceportal), Server ID (2229), Link Type (radio buttons for TCP and TLS, with TCP selected), and Server IP Address (10.10.15.80). An 'Add' button is visible at the bottom left of the form. The text 'Fields marked \* are required.' is displayed below the form fields.

## 7. Configuring ImmediateVoice Speech Application Engine

1. Access the ImmediateVoice Application Server by typing the following URL on a web browser and then pressing enter: “http://<hostname or IP address of ImmediateVoice server>/ImmediateVoice”, Click on **Login**.

The screenshot shows the ImmediateVoice website homepage. At the top left is the ImmediateVoice logo with the tagline "Multi-Channel Self Service Applications". To the right is a navigation menu with links for HOME, LOGIN, SERVICES, SUPPORT, and CONTACTS. The LOGIN link is highlighted. Below the navigation is a main content area with three columns. The left column features a headline "Turn Your E-Commerce Website into an All-round Communication Center!" and a paragraph about using ImmediateVoice to grow a customer audience. The middle column is titled "ImmediateVoice™ - KEY FEATURES" and describes the platform's capabilities, including Speech Application Creation and Service Monitoring. The right column is titled "Dialogs Unlimited" and highlights their expertise in multi-channel solutions. At the bottom right, there is an email subscription form with the text "Enter your e-mail to receive our latest news and proposals" and a "Go" button.

2. Input **Account Id**, **User Id** and **Password**, then Click on **submit** button.

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HOME LOGIN SERVICES SUPPORT CONTACTS

**NOT YET REGISTERED?**  
Registration is free and easy and it takes just a couple of minutes to complete. Please proceed to our [registration](#) page.

**WISH TO AMEND YOUR REGISTRATION DETAILS?**  
Please login first by providing your current registration details on the right.

**LOGIN :** To access **ImmediateVoice™ Speech Application** or to amend your registration details please provide your login information below

Account Id

User Id

Password

Forgotten your password? You can obtain your new password by calling us at +31 (0) 76 572 4789 9am-5pm during working days. You will need to be the actual account holder and be able to provide the following information: your name, your address with postcode and your answer to your security question. Please note that we require all the information for security purposes only.

Certain areas of our web site are password protected. To prevent unauthorised access, we follow appropriate security procedures.

You can help to preserve your privacy by never sharing your password with anyone else.

**Dialogs Unlimited** respects the privacy of its customers and business partners and is committed to complying with all laws relating to the protection of personal data. Please review our [Privacy Policy](#).

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- Select speech application for testing from the list of **Existing Speech Applications** on the right hand side of the screen and click on it to enter Application Editing page. In this sample application, speech application **Schiphol\_GB** is used.

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**WELCOME TO ImmediateVoice™**  
**TO START USING APPLICATION**  
please follow instructions on the right

**TO AMEND YOUR REGISTRATION DETAILS**  
please click [here](#)

**TO OBTAIN ADVISE AND SUPPORT**  
please visit our [support](#) page where you can find useful advise and information with regard to ImmediateVoice usage

**STEP 1 :** To start a New Speech Application please provide application details below  
To edit an Existing Speech Application please select from the list on the right

Start URL

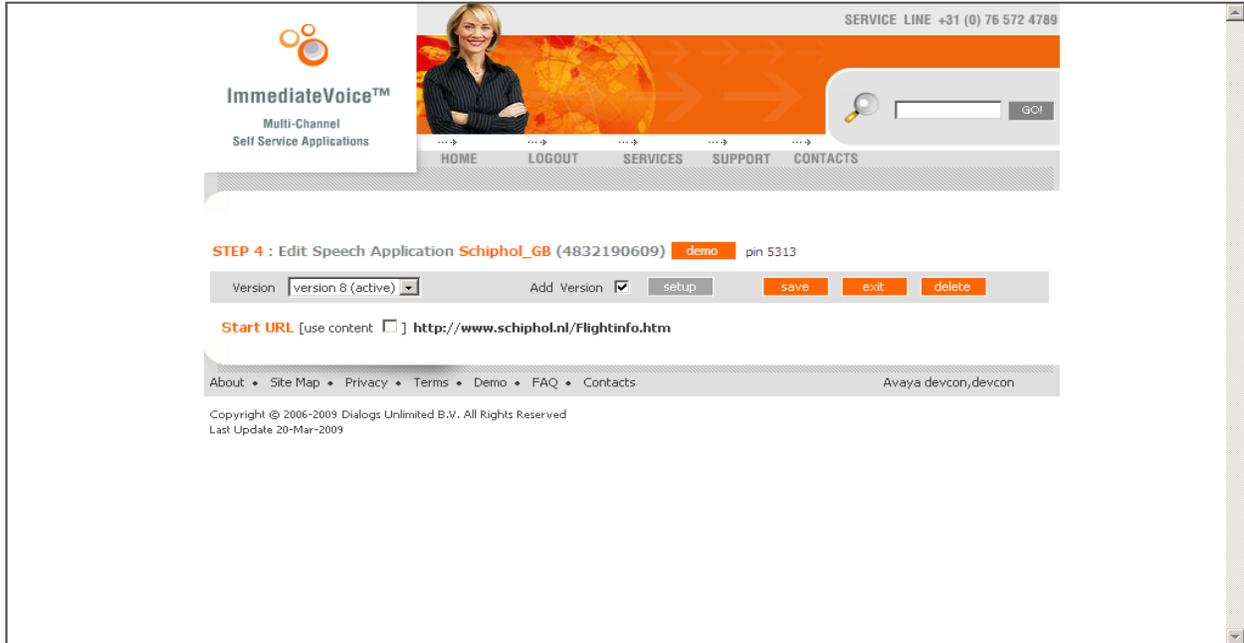
Application Name

[submit](#) [setup](#) [cancel](#)

**Existing Speech Applications**  
[POS Schiphol\\_GB](#)

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Vladimir devcon,devcon

4. Click on **setup** button in the Edit Speech Application Page



- Click on **TELEPHONY** tab; fill in **transfer #** and speech engine credentials (**Server, Server Name**). Use the same settings for Alt Server and NVP. Click **Save** button to save updated settings.

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**STEP 4 :** Edit Speech Application **Schiphol\_GB** (4832190609) **demo** pin 5313  
Application resources are protected against modifications. To unprotect change **protected** status in **SETUP**

Version  Add Ver

Start URL [use content ] <http://www.schiphol.nl/Flightinfo.htm>

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Last Update 20-Mar-2009

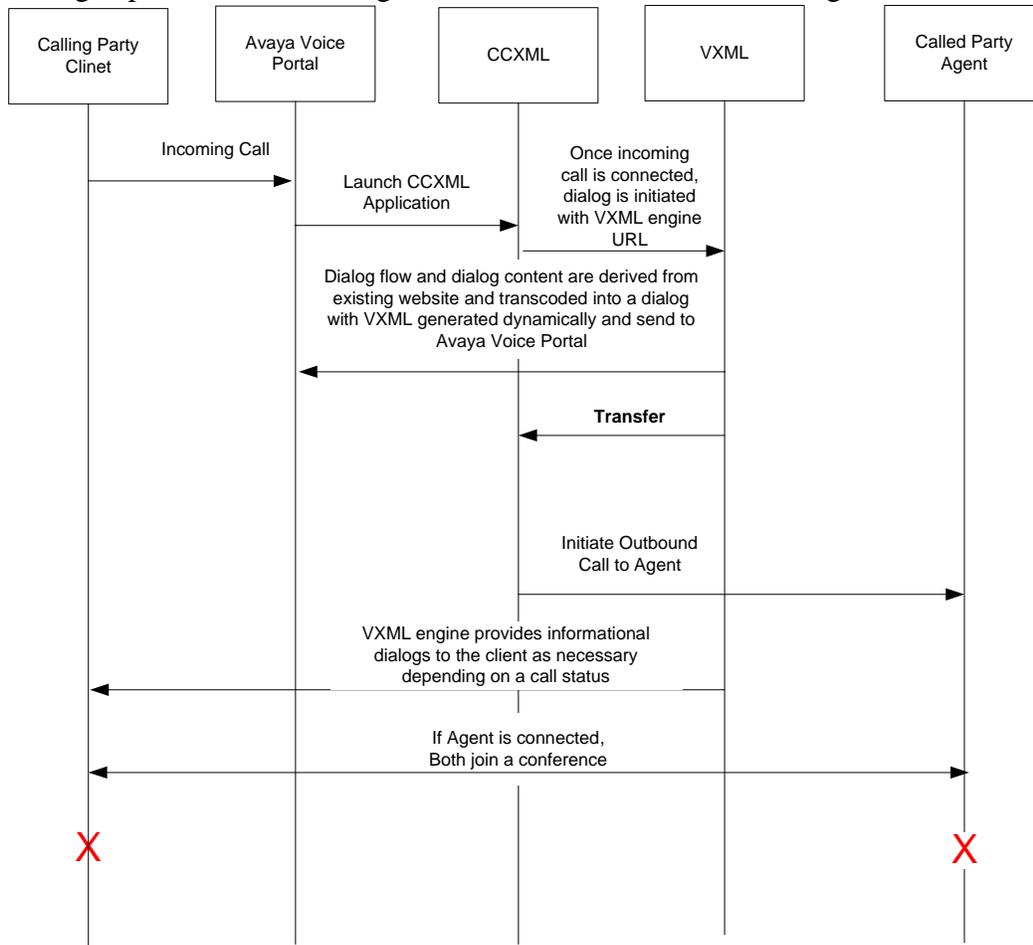
**SETUP**

click on section name to show/hide

GENERAL	protected <input checked="" type="checkbox"/>	TELEPHONY
		Transfer # (e.g. 0765724777) <input type="text" value="7110"/>
		Server <input type="text" value="10.10.15.90:3184"/>
		Server Name <input type="text" value="10.10.15.90"/>
		Alt Server <input type="text" value="10.10.15.90:3184"/>
		Alt Server Name <input type="text" value="10.10.15.90"/>
		NVP <input type="text" value="10.10.15.90:3184"/>
<b>TECHNOLOGY</b>		<b>ANALYSIS</b>
<b>SPEECH STYLES</b>		
WAVS & TTSS <input type="button" value="upload"/> - once uploaded re-enter setup		
RESOURCES - Import from. Use <Ctrl> for multiple selection		

## 8. General Test Approach and Test Results

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



**Figure 2: ImmediateVoice Call flow**

The focus of testing is VXML(CCXML) interoperability and support of major call functions like call pickup, making inputs in web forms by voice and web browsing by voice as per selected Immediate Voice Speech Application scope (in this sample configuration, Schiphol airport website is used), call transfers etc. 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 ImmediateVoice Speech Application Engine or Avaya Voice Portal.

All feature and serviceability test cases were performed manually. Tests are performed by calling Avaya Voice Portal and monitoring inputs made into website form and browsing which conversation is in progress. The monitoring function is supported by ImmediateVoice speech application engine. Monitoring can be performed on a Windows PC over the web. No special software is required, just Internet Explorer browser version 6, 7 or 8.

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. Get demo pin in the Edit Speech Application Schiphol\_GB webpage. Click on demo button.

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**STEP 4 : Edit Speech Application Schiphol\_GB (4832190609) demo pin 5313**

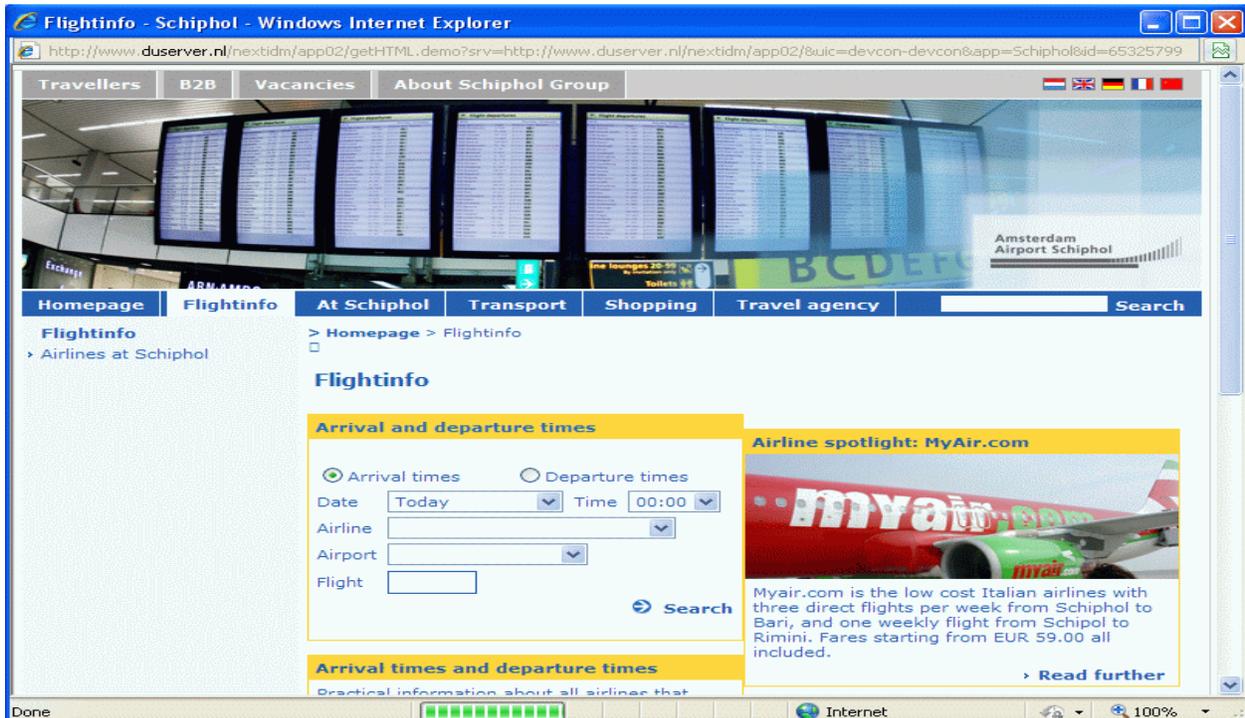
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2. Place an inbound call to the Schiphol airport speech application, and input demo pin through voice, verify if the Schiphol airport website will be open.



3. Verify the call will be transferred to the number configured in **Section 7 Step 5** after providing wrong input a few times through voice.
4. Verify the transferred phone ringing for several times and disconnect properly.

## 10. Conclusion

These Application Notes describe the steps for configuring Avaya Voice Portal 4.1 with Dialogs Unlimited ImmediateVoice 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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