

Avaya Solution & Interoperability Test Lab

Application Notes for Beijing InfoQuick SinoVoice Speech Technology jTTS 6.0 with Avaya Aura® Experience Portal 6.0 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required to integrate Beijing InfoQuick SinoVoice Speech Technology (SinoVoice) jTTS 6.0 with Avaya Aura® Experience Portal 6.0. SinoVoice jTTS uses the Media Resource Control Protocol (MRCP) version 2 for its Text-To-Speech (TTS) features to interface with VoiceXML applications running on the Avaya Aura® Experience Portal.

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 configuration steps required to integrate Beijing InfoQuick SinoVoice Speech Technology (SinoVoice) jTTS 6.0 with Avaya Aura® Experience Portal 6.0. Beijing InfoQuick SinoVoice jTTS uses the Media Resource Control Protocol (MRCP) version 2 for its Text-To-Speech (TTS) features to interface with the VoiceXML (VXML) applications running on Avaya Aura® Experience Portal.

Beijing InfoQuick SinoVoice jTTS is the core text-to-speech technology of SinoVoice which uses large scale recorded voice library and algorithm based on hierarchical prosody structure matching.

2. General Test Approach and Test Results

The general test approach is to place calls manually to Avaya Aura® Experience Portal running VXML applications that uses the TTS resources of SinoVoice jTTS.

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

2.1. Interoperability Compliance Testing

The interoperability compliance test included feature and serviceability testing. The feature testing focused on placing calls to Avaya Aura® Experience Portal 6.0 that ran VoiceXML applications that use the TTS engines on the SinoVoice jTTS solution. The compliance test focused on placing calls to verify accurate TTS synthesis.

The serviceability testing focused on verifying the ability of the SinoVoice jTTS solution to recover from adverse conditions, such as rebooting of SinoVoice jTTS and Avaya Aura® Experience Portal 6.0 and disconnecting the LAN cables to the SinoVoice jTTS server.

2.2. Test Results

All test cases passed. Avaya Aura® Experience Portal 6.0 was successful in running applications that use the TTS resources of the SinoVoice jTTS solution.

2.3. Support

For technical support on SinoVoice jTTS, contact the SinoVoice support team at:

- Phone: +86-10-82826886
- Email: <u>sinovoicesupport@sinovoice.com.cn</u>

3. Reference Configuration

Figure 1 illustrates the test configuration used to verify the SinoVoice jTTS solution. SinoVoice jTTS was installed on a Microsoft Windows 2003 R2 Server with Service Pack 2. VoiceXML applications were installed on a HTTP server. Avaya Aura® Experience Portal is connected to Avaya Aura® Session Manager and Avaya Aura® Communication Manager using SIP VoIP Connections. Avaya IP telephones were used to place calls to Avaya Aura® Experience Portal, which would run the VoiceXML applications. The applications would use the SinoVoice jTTS Server for speech synthesis.

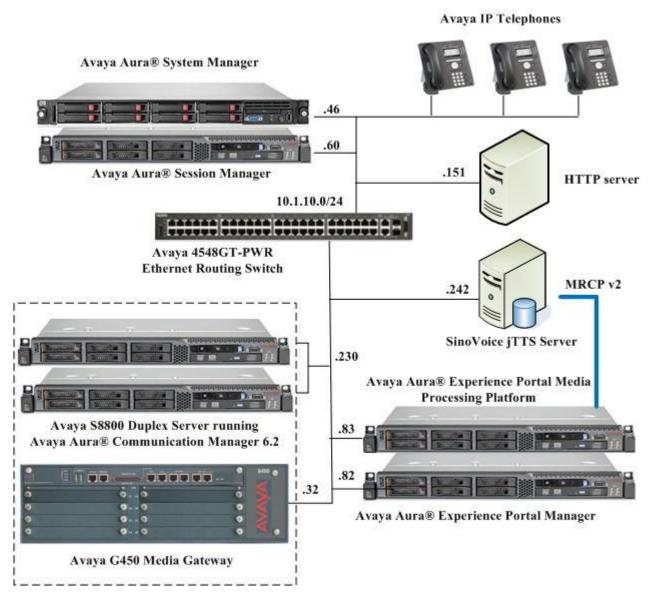


Figure 1: Test Configuration of SinoVoice jTTS with Avaya Aura® Experience Portal 6.0

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya Aura® Experience Portal 6.0 on	R6.0 SP1
Avaya S8800 Server	
Avaya Aura® Communication Manager	R6.2 SP2.01
on Avaya S8800 Server (Duplex)	
Avaya G450 Media Gateway	31.22.0
Avaya Aura® System Manager on	6.2 SP 3
HP DL360 G7	
Avaya Aura® Session Manager on	6.1 SP 3
Avaya S8800 Server	
Avaya 9621 IP Telephones	6.2 SP2 (H.323)
Avaya 4548GT-PWR Ethernet Routing	V6.2.4.010
Switch	
SinoVoice jTTS on Microsoft Windows	6.0
Server 2003 R2 Standard Edition SP2	
HTTP server on Windows Vista Business	Service Pack 2
Edition	

5. Configure Avaya Aura® Communication Manager

The configuration of the SIP Trunks between Communication Manager and Session Manager, and the routing of calls to Experience Portal are assumed to be in place and will not be discussed here. This section provides the additional procedures to configure Communication Manager for the purpose of administering SinoVoice jTTS. The configuration is performed via the System Access Terminal (SAT).

Step	Description							
1.	Enter the chang	e ip-codec-set n	command	l where n is a	valid IP code	c-set ass	ociated	with
	the IP network r	region that is use	d by Expe	rience Portal,	typically the	IP netwo	ork regio	n
	Ū.	Session Manager		0 00	1			
	appropriate valu	e supported by A	Avaya Exp	erience Porta	l and SinoVoi	ce jTTS	. In this	
	configuration, th	ne G.711Mu cod	ec was us	ed.				
	change ip-code	c-set 6				Page	1 of	2
		IP	Codec Set					
	Codec Set:	6						
	Audio	Silence	Frames	Packet				
	Codec	Suppression						
	1: G.711MU 2:	n	2	20				
	3:							
	4:							
	5: 6:							
	7:							

6. Configure Avaya Aura® Experience Portal

The initial administration of Avaya Aura® Experience Portal and the configuration of the SIP VoIP Connection to Session Manager are assumed to be in place and will not be discussed here. This section covers the additional procedures of Avaya Aura® Experience Portal that is required for the purpose of administering SinoVoice jTTS. The following steps will be covered:

- Configuring the VoIP audio format
- Adding SinoVoice jTTS as a TTS server
- Adding applications

ep	Description		
1.	web interface. T internet browser	xperience Portal is configured via the Experi o access the web interface, enter https://<ip< b="">- , where <ip-addr></ip-addr> is the IP address of the E stration role to display the main page.</ip<>	-addr> as the URL in an
		22/V: 𝒫 - ♥ C ■ C × @ Experience Portal Manager ×	(미만) ^ 슈 ☆ 영
	AVAYA		Welcome, admin Last logged in today at 1:55:20 PM SG
		Portal 6.0 (ExperiencePortal)	📅 Home 📪 Help 🛛 Logoff
	Expand All Collapse All Vuser Management Roles Users Login Options Real-Time Monitoring System Monitor Active Calls Port Distribution System Maintenance Audit Log Viewer Trace Viewer Alarm Manager Software Upgrade Software Upgrade System Backup System Configuration Alarm Ocdes	You are here: Home Avaya Aura® Experience Portal Manager Avaya Aura® Experience Portal Manager (EPM) is the consolidated web-based a Through the EPM interface, you can configure Experience Portal, check the state reports related to system operation. Installed Components Media Processing Platform Modia Processing Platform VoiceXML or CCXML application on an application server and communicates wit call.	us of a Experience Portal component, and generate an MPP receives a call from a PBX, it invokes a
	Alarm/Log Options Applications EPM Servers Report Data SIMP Speech Servers VoIP Connections Security Certificates Licensing Reports Standard Custom Scheduled	Legal Notice © 2005 - 2012 Avaya Inc. All Rights Reserved. Notice While reasonable efforts were made to ensure that the in: this document was complete and accurate at the time of p Avaya Inc. can assume no liability for any errors. Chang corrections to the information in this document might be incorporated in future releases.	rinting, es and

o configura th		
	ne codec used by the Media Processing Platform (MPP) server, c	
ystem Config	guration → MPP Servers in the left pane and click VoIP Settin	igs.
AVAYA	Last logged in to	Weld
	Portal 6.0 (ExperiencePortal) ff Home	?• Help
Expand All Collapse All	You are here: Home > System Configuration > MPP Servers	
Roles Users Login Options	MPP Servers This page displays the list of Media Processing Platform (MPP) servers in the Experience Portal system. When an MPP receives a call	from a
 Real-Time Monitoring System Monitor Active Calls 	invokes a VoiceXML application on an application server and communicates with ASR and TTS servers as necessary to process the c	
Port Distribution System Maintenance Audit Log Viewer	Name Host Address Network Network Maximum	
Trace Viewer Log Viewer Alarm Manager	Name Host Address Address Address Address Address Simultaneous Trace MRCP Address Address Address Address Calls Trace	e Level
System Management MPP Manager Software Upgrade	MPP1 10.1.10.83 <default> <default> <default> 10 Use MPI</default></default></default>	P Setting
System Backup System Configuration Alarm Codes	Add Delete	
Alarm/Log Options Applications EPM Servers	MPP Settings Browser Settings Event Handlers Video Settings VoIP Settin	igs
MPP Servers Report Data SNMP		
Speech Servers VoIP Connections Security		
Certificates Licensing Reports		
Standard Custom Scheduled		
age and click	onfiguration on Communication Manager in Section 5 . Scroll do Save .	1 mi wn f
	6	
age and click	Save.	wn T
age and click	Save.	WN We
AVAYA	Save.	wn we oday at
Avaya Aura® Experience Po Avaya Aura® Experience Po Apand All Collapse All User Management Roles Users	Save. Last logged in tr ortal 6.0 (ExperiencePortal)	WN We
Age and click of the second se	Save. Lest logged in to ortal 6.0 (ExperiencePortal) You are here: Home > System Configuration > MPP Servers > VoIP Settings VoIP Settings Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP). Use this page to configure parameters that affect how voice data is transferred through the network	WN Wa oday at ?+ Hel
Avaya Av	Save. Last logged in to ortal 6.0 (ExperiencePortal) Vou are here: Home > System Configuration > MPP Serverg > VoIP Settings Vour Settings Voire over Internet Protocol (NOIP) is the process of sending voice data through a network using one or more standard protocols st Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network wake any changes to this page, you must restart all MPPs.	WN We oday at ?+ Hel
age and click of a second clic	Save. Last logged in to ortal 6.0 (ExperiencePortal) You are here: Home > System Configuration > MPP Servers > VoIP Settings Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols si Real-Port Ranges > Low High	WN oday at ?• Hel
age and click of a second clic	Save. Last logged in to ortal 6.0 (ExperiencePortal) f* Home You are here: Home > System Configuration > MPP Servers > VoIP Settings VoIP Settings Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network make any changes to this page, you must restart all MPPs. Port Ranges VUDP: 23000 30999	WN oday at ?• Hel
Age and click of a second clic	Save. Last logged in to react lo	WN Wa oday at ?+ Hel
age and click of a second clic	Save. Last logged in to total 6.0 (ExperiencePortal) A Home You are here: Home > System Configuration > MPP Servers > VoIP Settings VoIP Settings Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols so Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the networ make any changes to this page, you must restart all MPPs. Port Ranges VIDP: 23000 30999 TOP: 31000 31999 VIDP: 31000 31999 VIDP: VIDP: VIDP: VIDP VIDP VIDP VIDP VIDP VIDP VIDP VIDP	WN Wa oday at ?+ Hel
Age and click of the second se	Save. Last logged in to text logged in text logged in to text logged in text logged	WN Wa oday at ?+ Hel
Age and click of a second clic	Save. Last logged in to to the form of the set logged in to to use here: Home > System Configuration > MPP Serverg > VoIP Settings VoIP Settings VoIP Settings Voice over Internet Protocol (KTP). Use this page to configure parameters that affect how voice data is transferred through the network area on changes to this page, you must restart all MPPs. Port Ranges • UDP: 12000 10999 TCP: 12000 10999 TCP: 13000 19999 H.323 35000 50000 RICP Monitor Settings • Host Address:	WN We oday at ?+ Hel
Age and click of a second clic	Save. Last logged in to the set logged in the set logg	WN We oday at ?+ Hel
Age and click of a second clic	Save. Last logged in to the set logged in the set logg	WN We oday at ?+ Hel
Age and click of a second seco	Save. Let logged in te to the control of the contro	WN We oday at ?+ Hel
Age and click of a second clic	Save. Let logged in te to the format to the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network and changes to this page, you must restart all MPPs. Port Ranges Port Ranges Port Ranges Port Ranges Port: P	WN We oday at ?+ Hel
Age and click of a second seco	Save. Let logged in to the set logged in to the set logged of the set logged in to the set logged of t	WN We oday at ?• Help
Age and click of a second seco	Save. Let logged in te to the format to the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols as Real-time Transfer Protocol (RTP). Use this page to configure parameters that affect how voice data is transferred through the network and changes to this page, you must restart all MPPs. Port Ranges Port Ranges Port Ranges Port Ranges Port: P	WN We oday at ?• Help
Age and click of a second seco	Save. Last logged in to rotal 6.0 (ExperiencePortal) Prove are have: Home > System Configuration > MPP Serverg > VoIP Settings VoIP Settings Voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols of Real-voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols of Real-voice over Internet Protocol (VoIP) is the process of sending voice data through a network using one or more standard protocols of Real-voice over Internet Protocol (VoIP) use this page to configure parameters that affect how voice data is transferred through the network make any changes to this page, you must restart all MPPs. Port Ranges Port Port	Wn 1 We oday at 2 ?• Help

Step	Description
4.	SinoVoice jTTS is not natively included in the set of TTS engines supported by Avaya
	Aura® Experience Portal and will not initially appear in the TTS configuration screen. To
	add SinoVoice jTTS to the list of supported engines, log into the EPM server, either
	locally or remotely through Secure Shell (SSH), and locate the languages.properties file
	found in /opt/Tomcat/apache-tomcat-6.0.32/webapps/VoicePortal/WEB-
	INF/classes/messages/. Edit the file and add the lines shown below to the appropriate
	section.
	< Some lines removed for brevity >
	#>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>>
	#{{START:PROPERTIES:EXPERIENCEPORTAL_6.0
	######################################
	# 1. ADD ANY NEW PROPERTIES FOR EXPERIENCE 6.0 TO ADDITIONS SECTION WITHIN 6.0
	SECTION.
	# 2. TO MODIFY A PRE-EXISTING PROPERTY, MOVE IT TO MODIFICATIONS SECTION WITHIN 6.0 AND THEN MODIFY IT.
	# 3. TO DELETE A PRE-EXISTING PROPERTY, MOVE IT TO DELETIONS SECTION WITHIN 6.0 AND THEN COMMENT IT OUT.
	#{{START:PROPERTIES:EXPERIENCEPORTAL_6.0:ADDITIONS
	# Specify any new properties for Experience Portal 6.0 here.
	SinoVoiceMRCPLabels=MRCP V1,MRCP V2
	SinoVoiceTransportLabels=TCP
	<pre>#}}END:PROPERTIES:EXPERIENCEPORTAL_6.0:ADDITIONS</pre>
	< remaining lines removed for brevity >

```
Step
      Description
  5.
      Locate the languages.properties file found in /opt/Tomcat/apache-tomcat-
      6.0.32/webapps/VoicePortal/WEB-INF/classes/config. Edit the file and add the fields
      and lines shown below to the appropriate section.
      #
      # Engine Type options displayed on the page
      #
      asrEngines=IBM WVS, Loquendo, Nuance
      ttsEngines=IBM WVS,Loquendo,Nuance,SinoVoice
      asrEnginesAmsOnly=Nuance
      ttsEnginesAmsOnly=Nuance
      # Engine Type conversion from display to internal data in the databas
      < Some lines removed for brevity >
      SinoVoiceTTS=sinovoice
      # Engine Type conversion from internal data in the database to display
      < Some lines removed for brevity >
      sinovoice=SinoVoice
      # TTS LANGUAGE
      < Some lines removed for brevity >
      SinoVoiceTTSlanguages=zh-cn XiaoKun F,zh-cn Liang M,en-us Julie F,en-us Paul M
      # Language Default
      < Some lines removed for brevity >
      SinoVoiceTTSlanguagesDefault=zh-cn XiaoKun F
      # default base port
      < Some lines removed for brevity >
      SinoVoiceBasePort=5060
      # default New Connection per Session
      < Some lines removed for brevity >
      SinoVoicePerPort=Yes
      # default URL
      < Some lines removed for brevity >
      SinoVoiceRtspUrlTts=/media/sinovoicesynthesizer
```

Step	Description
	< Some lines removed for brevity > #
	# MRCP Protocol #
	< Some lines removed for brevity >
	SinoVoiceMRCPValues=mrcpv1,mrcpv2
	# # Transport #
	< Some lines removed for brevity >
	SinoVoiceTransportValues=tcp
	< remaining lines removed for brevity >
6.	Reboot the EPM server for the above changes to take effect.
7.	To configure the SinoVoice jTTS server, click System Configuration \rightarrow Speech Servers . Click the TTS tab and click Add .
	Αναγα
	Avaya Aura® Experience Portal 6.0 (ExperiencePortal) Expand All Collapse All You are here: Home > System Configuration > Speech Servers
	Vuser Management Speech Servers Voies Speech Servers
	Login Options This page displays the list of Automated Speech Recognition (ASR) and Text-to-Speech (TTS) servers that Experience Portal communicates with System Monitor Active Calls
	Port Distribution
	Log Viewer No TTS Servers are configured. Alarm Manager * System Management Add Delete
	MPP Manager Detect Software Upgrade System Backup Customize Help
	Alarm Codes Alarm Log Options Applications EPM Servers
	MPP Servers Report Data SIMP Speech Servers
	VoIP Connections Security Cartificates Learning
	Licensing Reports Standard Custom Scheduled

Step	Description			
8.	In the Add TTS	S Server page, select	t SinoVoice as the Engine	Type . This engine type
	option was add	ed by modifying the	languages.properties file	es in Steps 4 and 5. In the
	-			e, select Yes to Enable , set
			or Full FQDN of the Since	
			-	Number of Licensed TTS
				ble on the SinoVoice jTTS
	Server. All othe	er fields were left at t	heir default values. Click	Save.
	Αναγα			Welcome, admin
	AVALYA			Last logged in today at 2:43:27 PM SGT
	Avaya Aura® Experience Po Expand All Collapse All			📅 Home 📪 Help 😡 Logoff
	▼ User Management Roles	You are here: <u>Home</u> > System Configur Add TTS Server	ation > <u>Speech Servers</u> > Add TTS Server	
	Users Login Options		Postal to communicate with a new TTC conver	
	 Real-Time Monitoring System Monitor 	Use this page to configure Experience i	Portal to communicate with a new TTS server.	
	Active Calls Port Distribution	Name:	SinoVoice jTTS	
	 System Maintenance Audit Log Viewer 	Enable:	🖲 Yes 🔘 No	
	Trace Viewer Log Viewer	Engine Type:	SinoVoice 🔻	
	Alarm Manager • System Management	Network Address:	sinovoice.sglab.com	
	MPP Manager Software Upgrade	Base Port:	5060	
	System Backup	Total Number of Licensed TTS Resource	es: 10	
	 System Configuration Alarm Codes 	New Connection per Session:	• Yes No	
	Alarm/Log Options Applications		Chinese(Simplified) zh-cn XiaoKun F	
	EPM Servers MPP Servers		Chinese(Simplified) zh-ch Liang M	
	Report Data SNMP	Voices:	English(USA) en-us Julie F	
	Speech Servers VoIP Connections		English(USA) en-us Paul M	
	▼ Security Certificates			
	Licensing Reports	MRCP		
	Standard	Ping Interval: 15 second(s)		
	Scheduled	Response Timeout: 4 second(s)		
		Protocol: MRCP V2 -		
		Transport Protocol: TCP -		
		Listener Port: 5060		
		Save Cancel Help		

Step	Description		
9.	To assign Sino	Voice jTTS to an Avaya Experience Portal application, clic	k System
	0	\rightarrow Applications and then click Add on the Applications p	•
	0	gure the Add Application page as shown below. This config	0
	, · · ·		
		aya Aura® Experience Portal test application deployed on the	1
		ber 10399. Specify the Name, select Yes to Enable, set MI	• •
	VoiceXML and	d set VoiceXML URL to HTTP server address location of	the VoiceXML
	script. Select S	inoVoice for TTS and then select the appropriate Voices to	o use. Click
	Save (not show	vn).	
	,	,	
	Depent this pro	cedure to assign SinoVoice jTTS to other Experience Porta	applications
	Repeat this pro	cedure to assign sind voice ji is to other Experience i orta	applications.
	AVAYA	Last	Welcome, admin logged in today at 2:43:27 PM SGT
		Portal 6.0 (ExperiencePortal)	n Home ?, Help 🛛 Logoff
	Expand All Collapse All • User Management	You are here: Home > System Configuration > Applications > Add Application	-
	Roles Users	Add Application	
	Login Options Real-Time Monitoring System Monitor	Use this page to deploy and configure a new application on the Experience Portal system.	
	Active Calls Port Distribution	Name: TestAppCN	
	 System Maintenance Audit Log Viewer 	Enable: 💿 Yes 🔘 No	
	Trace Viewer Log Viewer	Type: VoiceXML 🔻	
	Alarm Manager System Management MPP Manager	URI	
	Software Upgrade System Backup	Single Single Load Balance	
	 System Configuration Alarm Codes Alarm/Log Options 	VoiceXML URL: http://pcl.sglab.com/VXMLCN/intro1.vxml	
	Applications EPM Servers		
	MPP Servers Report Data SNMP	Mutual Certificate Authentication: 🔘 Yes 🖲 No	
	Speech Servers VoIP Connections	Basic Authentication: O Yes No	
	 Security Certificates Licensing 		
	▼ Reports Standard	ASR: No ASR V TTS: SinoVoice V	
	Custom Scheduled	Chinese(Simplified) zh-cn XiaoKun F	
		Voices: Chinese(Simplified) zh-cn Liang M □ English(USA) en-us Julie F →	
		Application Launch	
		Inbound □ Inbound Default ○ Outbound □	
		Number Number Range URI	
		Called Number: Add	
		10399	
		Remove	
		Speech Parameters >	
		Reporting Parameters >	
			-

7. Configure SinoVoice jTTS

The following components are required to run the SinoVoice jTTS MRCP Server. In this test configuration, both the jTTS Platform Engine and jMRCP Server are installed on the same server.

- jTTS Platform Engine
- jTTS Voice Library
- jMRCP Server

Step	Description		
1.	On the SinoVoice	jTTS server, click Start → All Programs → jTTS 6	5.0 Professional \rightarrow
	jTTS System Inf	ormation. On the jTTS SysInfo (系统信息)window,	click the Voice
	(音库) tab. Verify	that the desired voices are installed. Verify also that t	he value for Lines
	(授权线数) shows	s sufficient number of license required. Click OK to c	lose the window.
		-	
	● 系统信息		
	系统模块 授	図信息 日志 音库 音技DLL	
	系统路径:	C:\Program Files\SinoVoice\jTTS 6.0 Pro\Bin	刷新
	音库:	XiaoKun (中文 女声) XiaoKun (中文 女声)	
	标题	HaoBo(中文 男声) Julie(美国英语 女声)	
	名称 唯一标识	Paul (美国英语 男声) 84316E85-143E-4410-B00B-9DF681684C6C	
	支持语言音色	中文 青年的 女声	
	领域 提供商	通用领域 InfoQuick SinoVoice	
	引擎名 版本号	C:\Program Files\SinoVoice\jTTS 6.0 Pro\Bin\C 5.0.7da.33e	
	授权状态授权线数	<u> </u>	
	提示音组号		
	道授权数 过期日期	0 2013-01-01	
		OK Cancel	Help

Step	Description		
2.	Click Start → All	Programs → jTTS 6.0 Professional → Mrcp Config	g Tool. On the
	jMrcpConfig wind	low, configure as shown below.	
	Mrcp Ver	sion: Select 2.	
	Rtp Time	Slice: Enter 20.	
	Sip Listen	Port: Enter 5060, which is the default port value for S	IP.
	Local IP: 1	Enter the SinoVoice jTTS Server IP address as shown i	n Figure 1 .
		-	-
	Click Apply , and	then OK to complete the configuration.	
	JMrcpConfig		×
	Mrcp Version :	2	
		240 (0~240)	1
	Max Session Number :	240 (0~240)	Log
	Session Max Life Time :	10 min 🗖 Auto Destroy Invalid Session	TTS Server
			IPC
	Rtp Port Scope :	6000 - 60000 (6000~60000)	
	Rtp Time Slice :	20 (10~500) ms Rtp IP: 0 , 0 , 0 , 0	
	Mrcp Listen Port :	2550 (0~65565)	
		CARACTERISTIC CORPORTING AND	
	TTS Plugin :	C:\Program Files\SinoVoice\jTTS 6.0 Pro\Bin\jTTSPlugin.dll	Browse
	Sip Listen Port :	5060 (0~65535)	
	Local IP :	10 , 1 , 10 , 242	
	Audio Buffer Size :	(64~5120) KB	
	Audio burrer bize :	(04-0120) ND	
		OK Cancel Apply	

8. Verification Steps

This section provides the verification steps that may be performed to verify that Avaya Aura® Experience Portal can run VoiceXML applications that use the SinoVoice jTTS for TTS speech synthesis.

8.1. Verify Avaya Aura® Experience Portal

Step	Description	
1.	From the VPMS	S web interface, click MPP Manager on the left pane. On the MPP
	Manager page,	verify that the MPP server is Online and Running .
	Αναγα	Last logged in
	Avaya Aura® Experience Portal 6.0 (Ex Expand All Collapse All	xperiencePortal) from the term of term
	Legin Options Real-Tune Returning Arthy Calis Port GathyLion Arthy Calis Port GathyLion Arthy Calis Port GathyLion Arthy Calis Port GathyLion Arthy Calis Port GathyLion Arthy Calis Port GathyLion Port GathyLi	Commands rt Stop Restart Reboot Hall Cancel Commands Com

AVAYA				
Avaya Aura® Experience	Portal 6.0 (Experien	cePortal)		
Expand All Collapse All			Notice & Best Distribution	
▼ User Management	You are here: I	lome > Real-Time Mor	itoring > Port Distribution	
Roles Users	Port Dist	ribution (Nov 2	, 2012 3:26:57 PM S	GT)
Login Options Real-Time Monitoring				
System Monitor		ays information about ne VoIP Connections p	how the telephony resources age.	have been distributed to the
Active Calls Port Distribution			-3	
 System Maintenance Audit Log Viewer 	Total Ports: 1			Nov 2, 2012 3:26:46 PM SG
Trace Viewer Log Viewer		<u> </u>	up \$ Protocol \$ Current Al SIP_Trunk MPP1	location Base Allocation
Alarm Manager	<u>1</u> Online 2 Online	In service SM1	SIP_Trunk MPP1	
 System Management MPP Manager 	<u>3</u> Online		SIP_Trunk MPP1	
Software Upgrade System Backup	_	In service SM1	SIP_Trunk MPP1	
▼ System Configuration	5 Online		SIP_Trunk MPP1	
Alarm Codes Alarm/Log Options	<u>6</u> Online <u>7</u> Online	In service SM1 In service SM1	SIP_Trunk MPP1 SIP_Trunk MPP1	
Applications EPM Servers		In service SM1	SIP_Trunk MPP1	
MPP Servers Report Data	<u>9</u> Online		SIP_Trunk MPP1	
SNMP	<u>10</u> Online	In service SM1	SIP_Trunk MPP1	
Speech Servers VoIP Connections				
 Security Certificates 	Help			
Licensing Reports				
Standard Custom				
Scheduled				
11 11	Avava Aurau	9 Experience		-
uses the SinoVoice calls and that the ap caller. From the VI	b jTTS for spec pplication is a PMS web inter	ech synthesis. ble to annound rface, click A	ce the TTS synthe	sized prompts to t
Avaya Aurof Superference Portal 6.0 (Experimental Superference Portal	e jTTS for spec pplication is a PMS web inter use is SinoVo	ech synthesis. ble to annound rface, click A bice jTTS.	ce the TTS synthe	sized prompts to t
Avaya Aura® Experience Portal 6.0 (Experience Portal 6.1 (Clapse All Voul are here Vou	e jTTS for spee pplication is a PMS web inter use is SinoVo (mcePortal) at Jona > Real-Time Montholog > Ad Calls (Oct 29, 2012 3:56:4 isplays the status of all the active call re calls: 4	ech synthesis. ble to annound fface, click A bice jTTS. we calls 2 PM 5GT) s being handled by the Experience P	ce the TTS synthes ctive Calls on the	sized prompts to t left pane and verit
Active of Total Active of Tota	e jTTS for spee pplication is a PMS web inter use is SinoVo (encePortal) a: Home > Real-Time Monitoring > Act Calls (Oct 29, 2012 3:56:4 isplays the status of all the active call the calls : 4 forcup (Protocol Call type - SIP_Trank Inbound SIP_Trank Inbound SIP_Trank Inbound	ech synthesis. ble to annound fface, click A bice jTTS. we calls 2 PM SGT) a being handled by the Experience P MPP1 Oct 29, 2012 315311 MPP1 Oct 29, 2012 315311 MPP1 Oct 29, 2012 315313	rtal system.	Est Poli: Oct 29, Last Poli: Oct 29, Internet Content of Content

8.2. Verify SinoVoice jTTS

On the SinoVoice jTTS server, click **Start** \rightarrow **Administrative Tools** \rightarrow **Services**. In the Services window, verify that the **jMrcpWndServer** is started.

Services							
<u>File A</u> ction ⊻iew	Help						
🖏 Services (Local)	🍇 Services (Local)						
	Select an item to view its description.	Name A	Description	Status	Startup Type	Log On As	
		w Intersite Messaging	Enables me		Disabled	Local System	
		PSEC Services	Provides e	Started	Automatic	Local System	
		🖓 jMrcpWndServer		Started	Manual	Local System	
		iTTSService4_Pro		Started	Automatic	Local System	
		Kerberos Key Distribution Center	On domain		Disabled	Local System	
		Cicense Logging	Monitors a		Disabled	Network S	
		Cogical Disk Manager	Detects an	Started	Automatic	Local System	
		Cogical Disk Manager Administrative Service	Configures		Manual	Local System	
		Messenger .	Transmits		Disabled	Local System	
		Microsoft .NET Framework NGEN v4.0.30319_X86	Microsoft		Automatic	Local System	
		Microsoft Software Shadow Copy Provider	Manages s		Manual	Local System	
		Ret Logon	Maintains a		Manual	Local System	
		Net. Tcp Port Sharing Service	Provides a		Disabled	Local Service	
		NetMeeting Remote Desktop Sharing	Enables an		Disabled	Local System	
		Retwork Connections	Manages o	Started	Manual	Local System	
		Network DDE	Provides n		Disabled	Local System	
		Network DDE DSDM	Manages D		Disabled	Local System	
		Network Location Awareness (NLA)	Collects an	Started	Manual	Local System	
		Network Provisioning Service	Manages X		Manual	Local System	
		NT LM Security Support Provider	Provides s		Manual	Local System	
		Performance Logs and Alerts	Collects pe		Automatic	Network S	
		Plug and Play	Enables a c	Started	Automatic	Local System	
		Portable Media Serial Number Service	Retrieves t		Manual	Local System	
		Print Spooler	Manages al	Started	Automatic	Local System	
		Protected Storage	Protects st	Started	Automatic	Local System	
		Remote Access Auto Connection Manager	Creates a		Manual	Local System	
		Remote Access Connection Manager	Creates a	Started	Manual	Local System	
		Remote Desktop Help Session Manager	Manages a		Manual	Local System	
		Remote Packet Capture Protocol v.0 (experimental)	Allows to c		Manual	Local System	

9. Conclusion

These Application Notes describe the compliance-tested configuration used to validate Avaya Aura® Experience Portal 6.0 with Beijing InfoQuick SinoVoice jTTS 6.0. All test cases were completed successfully.

10. Additional References

The following documents are available at <u>http://support.avaya.com</u>.

[1] Administering Avaya Aura® Experience Portal, Aug 2011.

Product information on Beijing InfoQuick SinoVoice jTTS 6.0 can be found at <u>http://www.sinovoice.com/english/jtts.html</u>.

©2012 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by [®] and TM are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at <u>devconnect@avaya.com</u>.