Avaya Solution & Interoperability Test Lab

Application Notes for InfoTalk-Vbrowser 3.0 with Avaya Aura® Communication Manager and Avaya Aura® Session Manager 6.3 – Issue 1.0

Abstract

These Application Notes describe the configuration steps required for InfoTalk-Vbrowser 3.0 to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager 6.3. InfoTalk-Vbrowser is a VoiceXML Interpreter which runs on Windows PCs or Servers.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as any observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

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 for InfoTalk-Vbrowser 3.0 to successfully interoperate with Avaya Aura® Communication Manager and Avaya Aura® Session Manager 6.3. InfoTalk-Vbrowser is a VoiceXML Interpreter which runs on Windows 7/8 or Windows Server 2008/2012. Incoming calls over SIP Trunks routed from Avaya Aura® Communication Manager through Avaya Aura® Session Manager to the InfoTalk-Vbrowser application will be detected and answered with an initial application-specific VoiceXML document retrieved from a Web Server.

2. General Test Approach

The general test approach is to place calls manually to InfoTalk-Vbrowser running VXML applications to test the capability of the system.

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

This Interoperability Compliance Test included feature and serviceability testing. The feature testing focused on placing calls to InfoTalk-Vbrowser that ran Voice XML scripts in English. A copy of the VoiceXML application (index_test.vxml) that was used in the test configuration is shown in **Appendix A**. The compliance test focused on placing calls of different companding mode, shuffled and non-shuffled call as well as from different type of phones including digital and IP Phones. Also, call transfer, play back of speech recording and verification of Caller ID received were tested.

The serviceability testing focused on verifying the ability of InfoTalk-Vbrowser solution to recover from adverse conditions such as rebooting of InfoTalk server, Communication Manager and Session Manger; disconnecting the LAN cables to the InfoTalk server.

2.2. Test Results

All test cases passed. InfoTalk-Vbrowser was successful in running applications from various types of phones over SIP trunk.

2.3. Support

For technical support on InfoTalk-Vbrowser contact:

- Telephone : +852 2190 9600
- Fax : +852 2788 2306
- Email : support@infotalkcorp.com

3. Reference Configuration

Figure 1 illustrates the configuration used to verify the InfoTalk-Vbrowser 3.0 solution. The InfoTalk-Vbrowser 3.0 software was installed on Microsoft Windows 2008 server. VoiceXML scripts are located within the local server. Calls routed from Avaya Aura® Communication Manager through Avaya Aura® Session Manager were routed to the InfoTalk-Vbrowser. Avaya IP (H.323 and SIP) and Digital telephones were used to place calls to the InfoTalk-Vbrowser, which runs the VoiceXML applications.

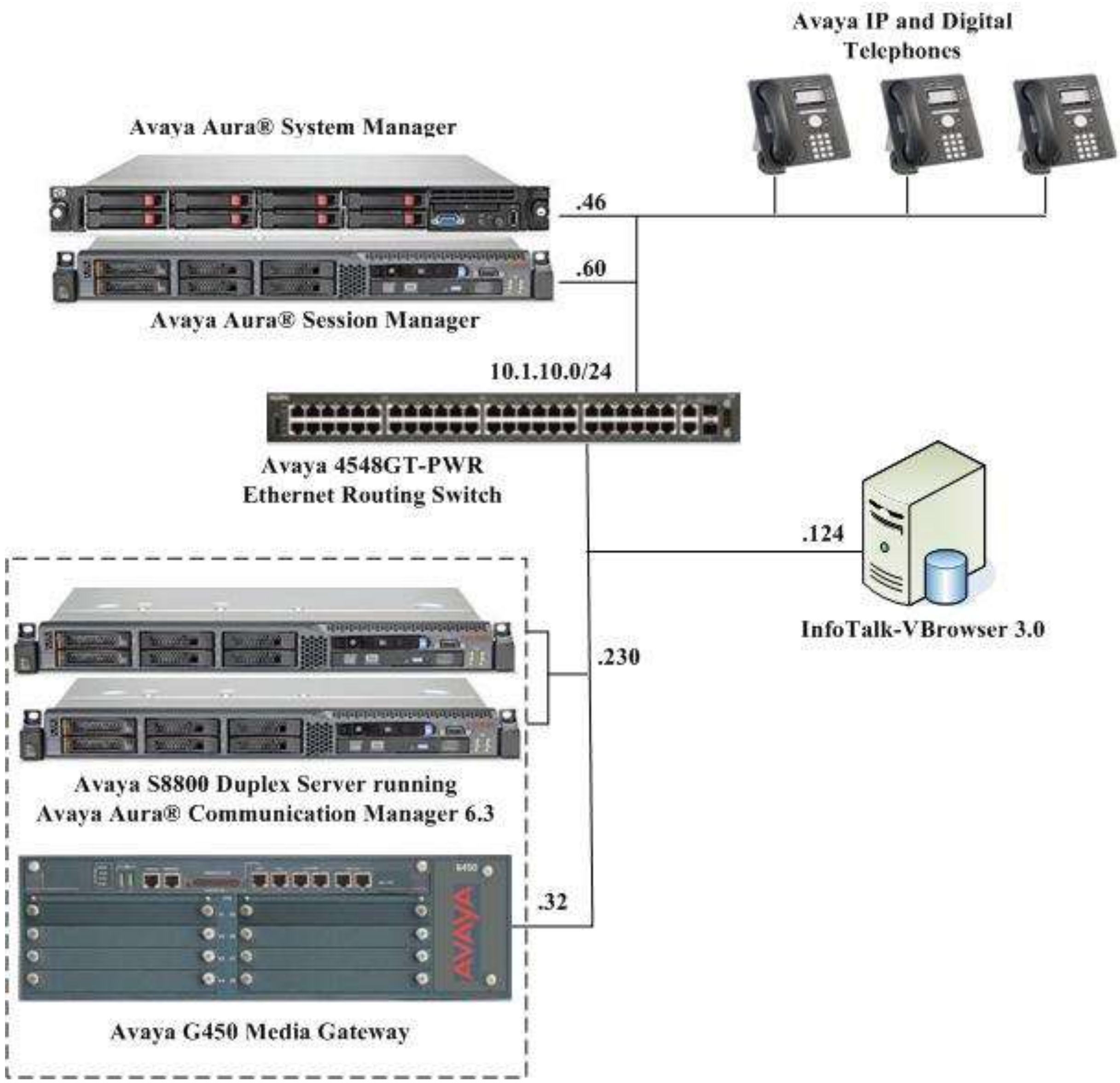


Figure 1: Test Configuration

4. Equipment and Software Validated

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

Equipment	Software
Avaya Aura® Communication Manager on Avaya S8800 Server (Duplex)	6.3.9 (Build R016x.03.0.124.0-21971)
Avaya G450 Media Gateway	36.7.0
Avaya Aura® System Manager on HP DL360 G7	6.3.11.8.2933
Avaya Aura® Session Manager on Avaya S8800 Server	6.3.11.0.631103
Avaya 96x1 Series Deskphones (H.323 IP)	6.4014
Avaya 96x1 Series Deskphones (SIP)	6.5.0.17
Avaya 94xx Series Digital Deskphones	FW 12
Avaya 4548GT-PWR Ethernet Routing Switch	FW: 5.3.0.3 SW: v5.6.1.052
InfoTalk-Vbrowser on Microsoft Windows Standard Server 2008 R2 running on VMware 4.1u1	3.0

Note: Avaya Aura® Communication Manager, Avaya Aura® System Manager and Avaya Aura® Session Manager 6.3 are part of Avaya Aura® Feature Pack 4 solution.

5. Configure Avaya Aura® Communication Manager

The configuration of the SIP Trunks between Communication Manager and Session Manager, and the routing of calls to InfoTalk-Vbrowser as well as the necessary SIP Trunking license 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 InfoTalk-Vbrowser. The configuration is performed via the System Access Terminal (SAT).

Step	Description
1.	Enter the **change ip-codec-set n** command where **n** is a valid IP codec-set associated with the IP network region that is used by InfoTalk-Vbrowser, typically the IP network region assigned to the Session Manager SIP Trunk signaling group. Set **Audio Codec** to an appropriate value supported by InfoTalk-Vbrowser. Only G.711 is supported for both companding mode i.e. MuLaw and ALaw. In this configuration, the **G.711Mu** codec was used.

```
change ip-codec-set 6                                            Page   1 of   2

                          IP Codec Set

    Codec Set: 6

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

6. Configure Avaya Aura® Session Manager

This section describes the procedures for configuring Avaya Aura® Session Manager to support the routing of calls to InfoTalk server.

These instructions assume other administration activities have already been completed such as defining SIP entities for Session Manager, defining the network connection between Communication Manager and Session Manager, and defining Communication Manager as a Managed Element. The domain and location parameters were also assumed to be defined on Session Manager during setup.

The following administration activities will be described:

- Define SIP Entity for InfoTalk Server
- Define Entity Links, which describe the SIP trunk parameters used by Session Manager when routing calls between SIP Entities
- Define Routing Policies and Dial Patterns which control routing between SIP Entities

Configuration is accomplished by accessing the browser-based GUI of Avaya Aura® System Manager, using the URL "**http://<ip-address>/SMGR**", where "**<ip-address>**" is the IP address of System Manager. Log in with the appropriate credentials.

6.1. Define SIP Entities

A SIP Entity must be added for InfoTalk Server. To add a SIP Entity, expand **Elements →Routing** and select **SIP Entities** from the left navigation menu.

Click **New (**not shown**).** In the **General** section, enter the following values and use default values for remaining fields**.**

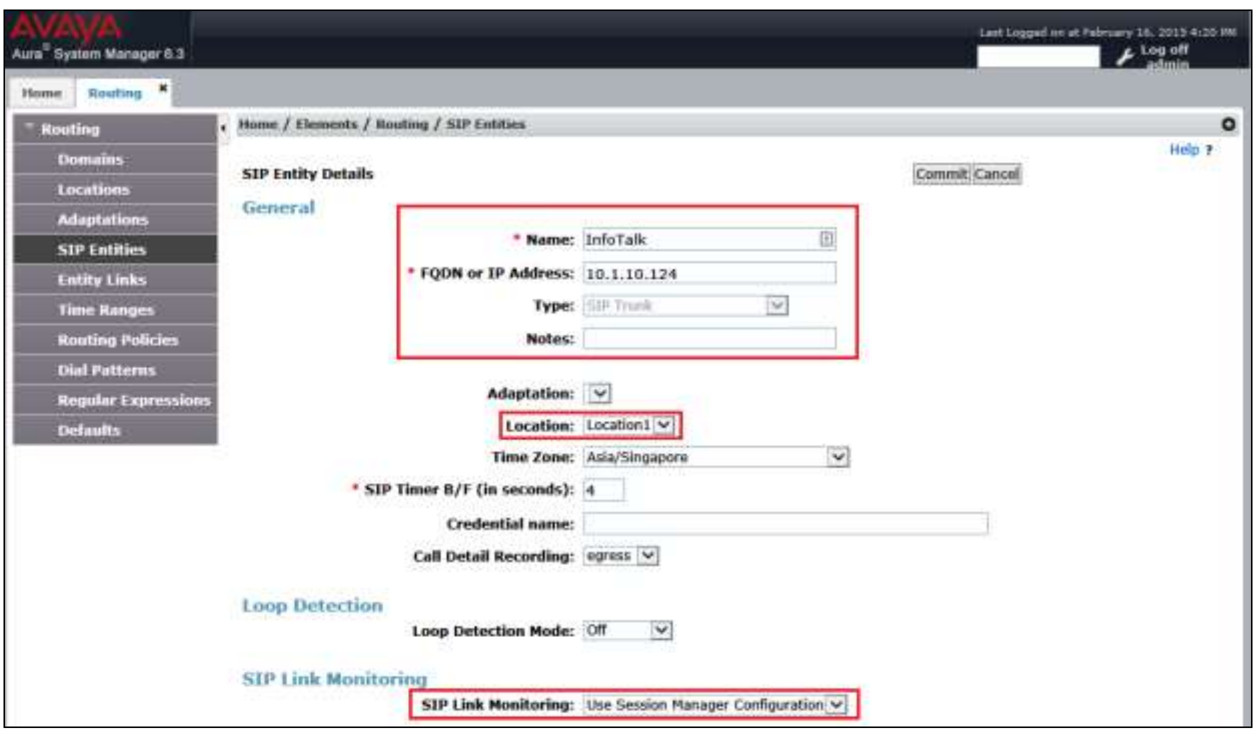
- **Name:** Enter an identifier for new SIP Entity.
  In the sample configuration, "**InfoTalk**" was used.
- **FQDN or IP Address:** Enter ip address **10.1.10.124**
- **Type:** Select "**SIP Trunk**".
- **Notes:** Enter a brief description. [Optional].
- **Location:** Select appropriate location defined for InfoTalk Server.

In the **SIP Link Monitoring** section:

- **SIP Link Monitoring:** Select "**Use Session Manager Configuration**". The default for Session Manager Configuration is that Link Monitoring is enabled.

Click **Commit** to save SIP Entity definition.

The following screen shows the SIP Entity defined for InfoTalk Server.



6.2. Define Entity Links

A SIP trunk between InfoTalk Server and Session Manager is described by an Entity Link. In the sample configuration, a SIP Entity Link was added between Session Manager and InfoTalk Server.

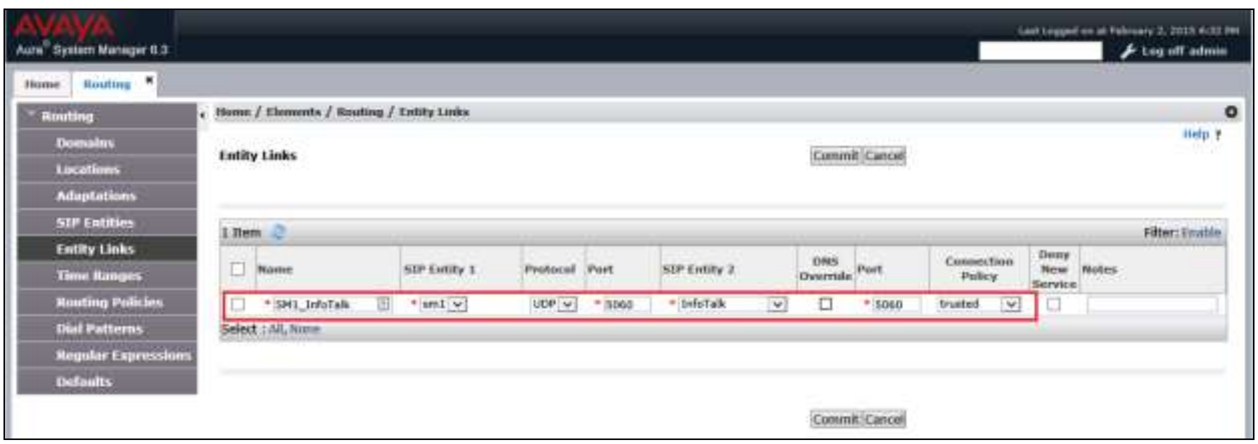
To add an Entity Link, expand **Elements** →**Routing** and select **Entity Links** from the left navigation menu.

Click **New** (not shown). Enter the following values**.**

- **Name** Enter an identifier for the link to InfoTalk Server.
- **SIP Entity 1** Select the Session Manager already defined.
- **SIP Entity 2** Select the SIP Entity added for **InfoTalk** defined in **Section 6.1** from drop-down menu.
- **Protocol** After selecting both SIP Entities, verify "**UDP**" is selected as the required Protocol.
- **Port** Verify **Port** for both SIP entities is "**5060**".
- **Connection Policy** Select **Trusted** from drop-down menu.

Click **Commit** to save Entity Link definition.

The following screen shows the Entity Link defined between InfoTalk and Session Manager.



6.3. Define Routing Policy

Routing policies describe the conditions under which calls will be routed.

To add a routing policy, expand **Elements →Routing** and select **Routing Policies.**

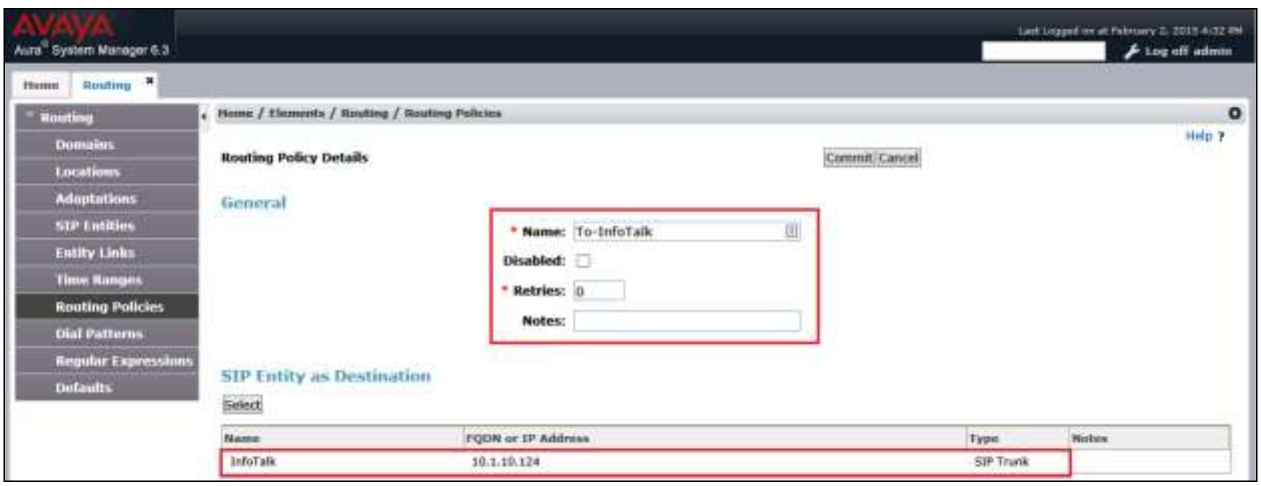
Click **New** (not shown). In the **General** section, enter the following values.

- **Name:** Enter an identifier for routing to InfoTalk Server.
- **Disabled:** Leave unchecked.
- **Retries:** Retain default value of "**0**".
- **Notes:** Enter a brief description. [Optional].

In the **SIP Entity as Destination** section, click **Select.** The **SIP Entity List** page opens (not shown). Select the SIP Entity defined for InfoTalk in **Section 6.1** and click **Select.**

The selected SIP Entity displays on the **Routing Policy Details** page. Use default values for remaining fields. Click **Commit** to save Routing Policy definition.

The following screen shows the Routing Policy for InfoTalk Server.



6.4. Define Dial Pattern

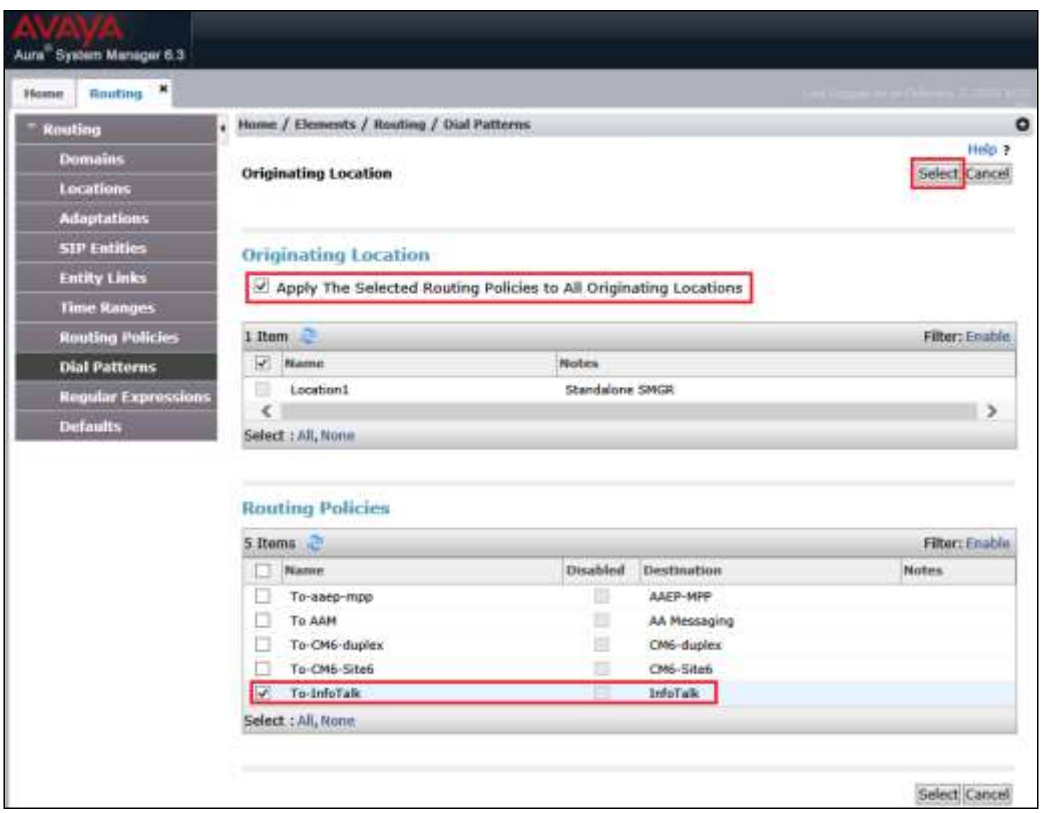
This section describes the steps to define a dial pattern to route calls to InfoTalk Server. In the test configuration, 5-digit extensions "**70000**" are assigned to calls routed to InfoTalk Server.

To define a dial pattern, expand **Elements →Routing** and select **Dial Patterns.** Click **New** (not shown). In the **General** section, enter the following values and use default values for remaining fields.

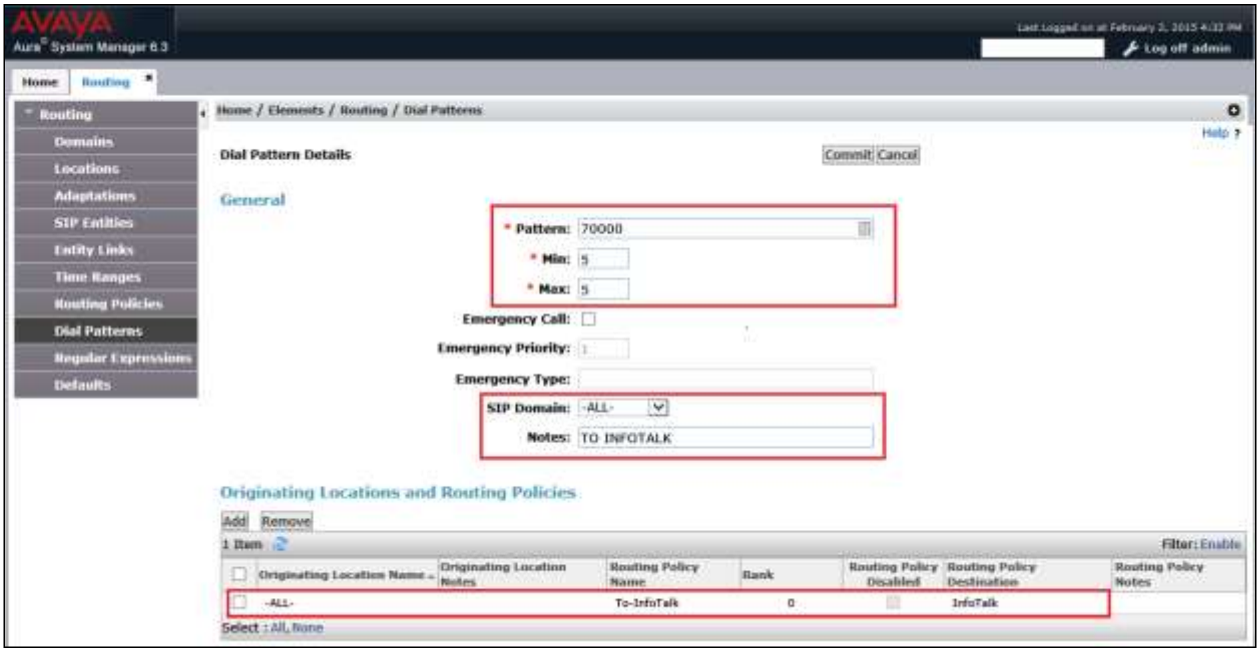
- **Pattern:** Enter the fixed number **70000** for calls to InfoTalk Server.
- **Min:** Enter the minimum number digits that must be dialed.
- **Max:** Enter the maximum number digits that may be dialed.
- **SIP Domain:** Select the SIP Domain from drop-down menu or select "**ALL**" if Session Manager should accept incoming calls from all SIP domains.
- **Notes:** Enter a brief description. [Optional].

In the **Originating Locations** and **Routing Policies** section, click **Add.**
The **Originating Location** and **Routing Policies** list page opens (not shown).

- In **Originating Location**, select "**Apply The Selected Routing Policies to All Originating Locations**".
- In **Routing Policies** table, select **To-InfoTalk** for routing to InfoTalk Server defined in **Section 6.3.**
- Click **Select** to save these changes and return to **Dial Patterns Details** page.



Click **Commit** to save the new definition. The following screen shows the Dial Pattern defined for routing calls to InfoTalk Server.

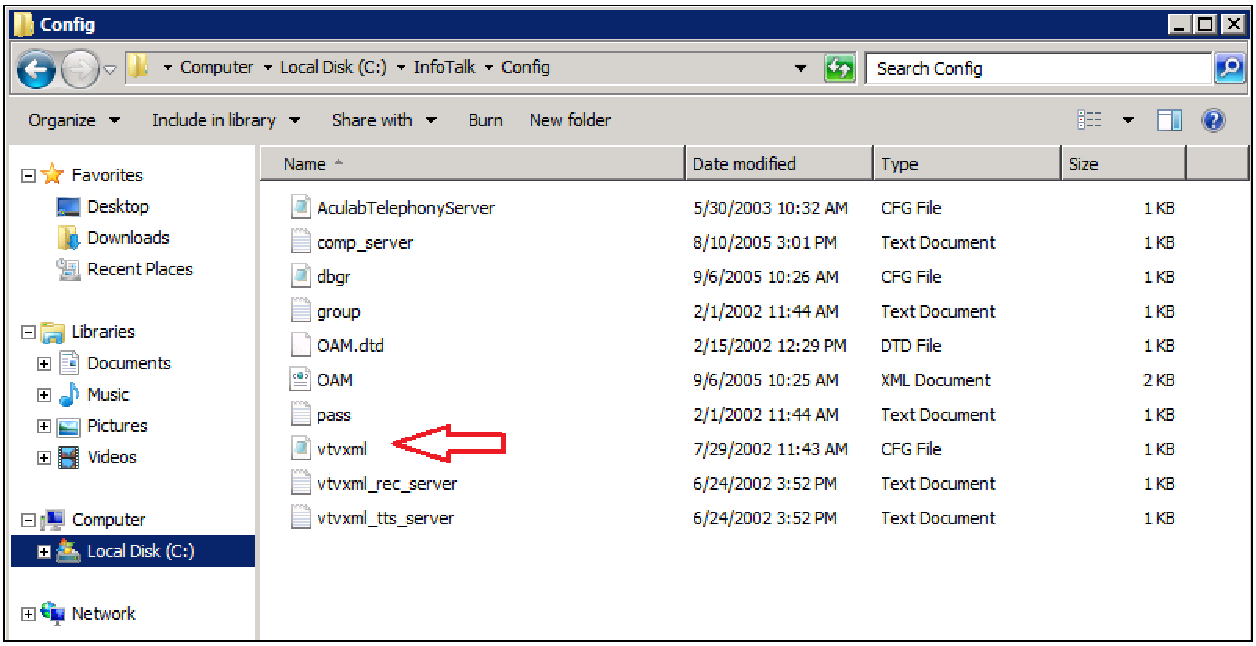


7. Configure InfoTalk-Vbrowser

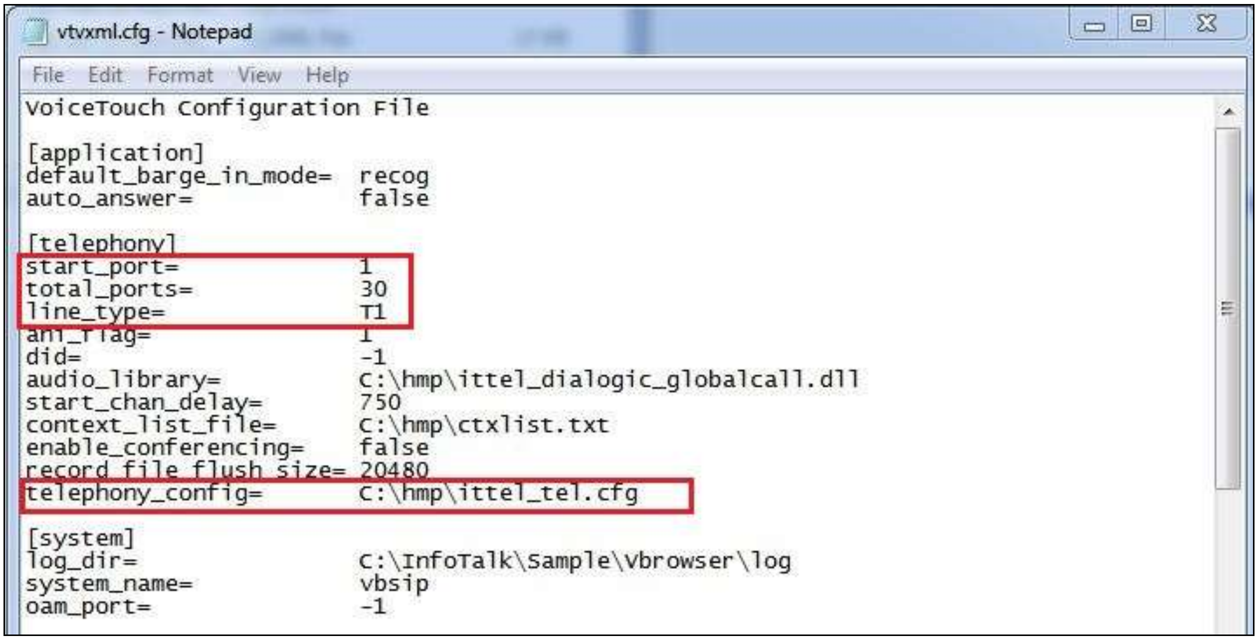
InfoTalk-Vbrowser is installed on a Windows Standard Server 2008 R2. The following sections describe the InfoTalk-Vbrowser configurations. The appropriate software is assumed to be installed by staff from InfoTalk Corporation.

7.1. Configuration

Locate the configuration file in the directory **..\InfoTalk\Config** under the name **vtxml.cfg.**



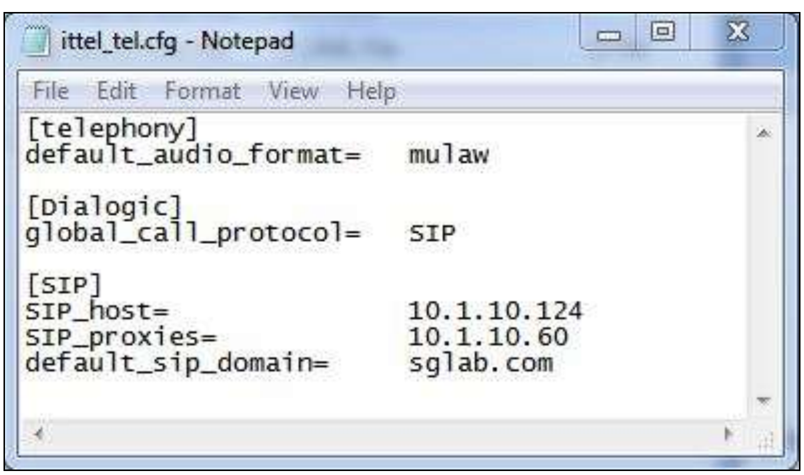
Below is a capture of the parameters in vtvxml.cfg config file. The important part will be the interface to Session Manager via the entity link and this is specified by the **telephony_config** parameter. The path and file name **"c:\hmp\ittel_tel.cfg"** specify the interface to Session Manager. This will override telephony settings **line_type**. The **start_port** and **total_ports** control the maximum number of simultaneous (SIP) calls.




```
VoiceTouch Configuration File

[application]
default_barge_in_mode=  recog
auto_answer=            false

[telephony]
start_port=             1
total_ports=            30
line_type=              T1
ani_flag=               1
did=                    -1
audio_library=          c:\hmp\ittel_dialogic_globalcall.dll
start_chan_delay=       750
context_list_file=      c:\hmp\ctxlist.txt
enable_conferencing=    false
record_file_flush_size= 20480
telephony_config=       c:\hmp\ittel_tel.cfg

[system]
log_dir=                C:\InfoTalk\Sample\vbrowser\log
system_name=            vbsip
oam_port=               -1
```

In the **ittel_tel.cfg** configuration file, the relevant interface is specified.

Default_audio_format:	**mulaw** specifies the codec used.
global_call_protocol:	**SIP** specifies the protocol for communication.
SIP_host:	**10.1.10.124** is IP address of InfoTalk-Vbrowser server.
SIP_proxies:	**10.1.10.60** specifies IP address of Session Manager which functions as the SIP proxy server.
default_sip_domain:	**sglab.com** specifies the domain use in the lab setup.



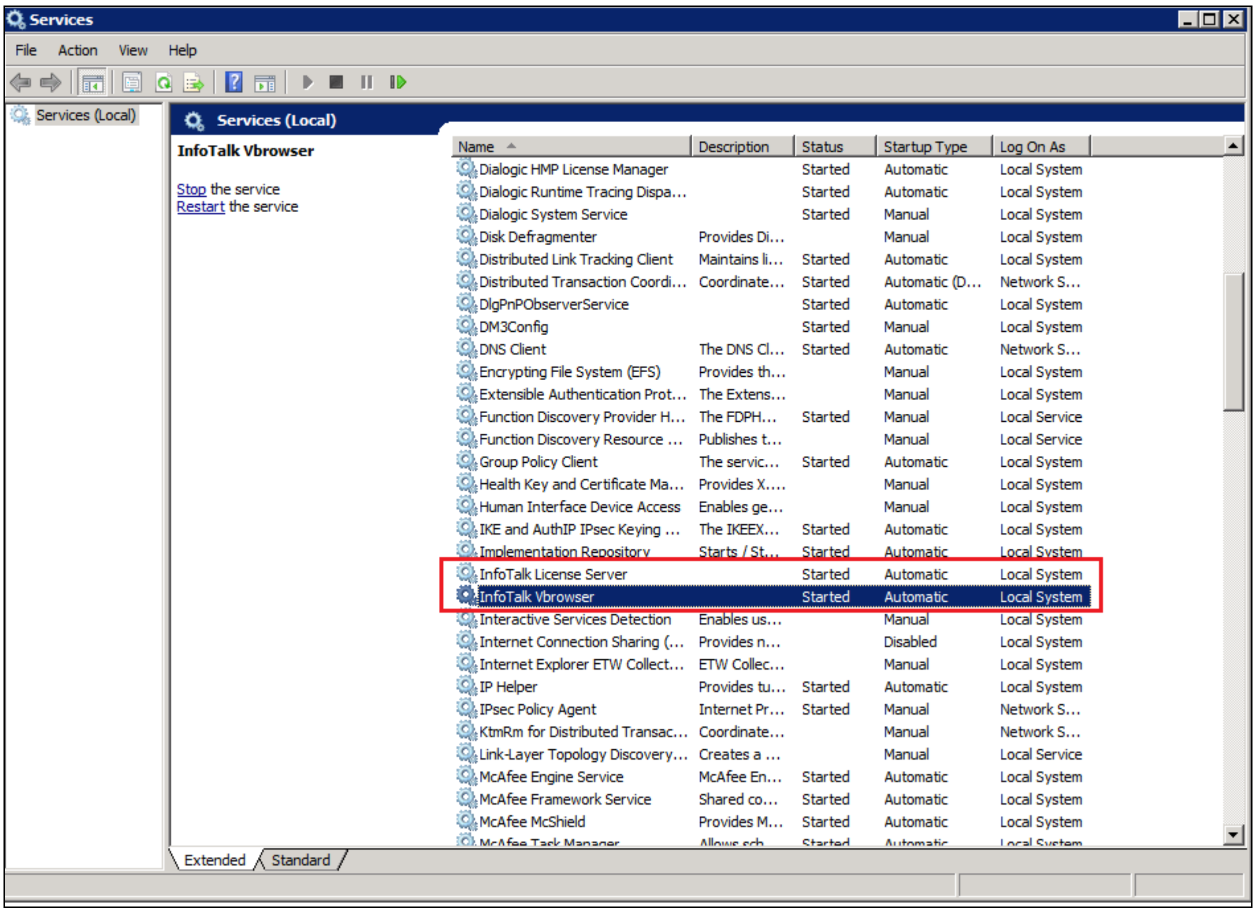

```
[telephony]
default_audio_format=   mulaw

[Dialogic]
global_call_protocol=   SIP

[SIP]
SIP_host=               10.1.10.124
SIP_proxies=            10.1.10.60
default_sip_domain=     sglab.com
```

Please refer to the InfoTalk-Vbrowser Developer's Guide specified in **Section 9** reference **[3]** for more details on the other parameters.

7.2. Starting and Stopping the InfoTalk-Vbrowser service.

Below are the procedures to manually start or stop the service:

1. Click **Start →Settings → Control Panel** then choose **Administrative Tools → Services**.



2. On the panel list, check that the **InfoTalk License Server** service is **Started**.

3. Click the **InfoTalk Vbrowser.** If the status displays "**Started**", the service is already started. Otherwise, the service is stopped at the moment.

4. Right Click on **InfoTalk-Vbrowser** and select **Start** or **Stop** to start or stop the service. Once InfoTalk-Vbrowser is started, it will run continuously and will stop only when the process is terminated explicitly.

8. Verification Steps

Make inbound calls to the administered number that is routed to the InfoTalk-Vbrowser and check that the voice announcement is heard and response to the test script menu works. Also, verify on the SIP Trunk status on Communication Manager from the SAT Terminal that InfoTalk-Vbrowser is connected in the **Audio Connection Far-end** ip address and the appropriate **Codec Type** is used.

```
status trunk 7/4                                                Page   2 of   4
                                CALL CONTROL SIGNALING

Near-end Signaling Loc: PROCR
  Signaling    IP Address                                    Port
   Near-end:   10.1.10.230                                 : 5061
    Far-end:   10.1.10.60                                  : 5061
 H.245 Near:
  H.245 Far:
   H.245 Signaling Loc:          H.245 Tunneled in Q.931? no

 Audio Connection Type: ip-tdm       Authentication Type: None
    Near-end Audio Loc: 02A0701                 Codec Type: G.711MU
   Audio       IP Address                                    Port
   Near-end:   10.1.50.23                                  : 3176
    Far-end:   10.1.10.124                                 : 49192

 Video Near:
  Video Far:
 Video Port:
  Video Near-end Codec:                  Video Far-end Codec:
```

9. Conclusion

These Application Notes describe the compliance-tested configuration used to validate Avaya Aura® Communication Manager and Avaya Aura® Session Manager 6.3 with InfoTalk-Vbrowser 3.0. All test cases were completed successfully.

10. Additional References

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

[1] *Administering Avaya Aura® Communication Manager*, Release 6.3, Issue 10, June 2014, Document ID 03-300509.

[2] *Administering Avaya Aura® Session Manager*, Release 6.3, Issue 5, June 2014.

The following documents are available from InfoTalk:

[3] *InfoTalk-Vbrowser Developer's Guide*, Version 3.0

Appendix A

```
<?xml version="1.0" encoding="UTF-8"?>
<vxml version="1.0" xml:lang="english">
	<!-- Global vxml property -->
	<property name="bargein" value="true"/>

	<form id="menu">
		<!-- recognition start here -->
		<field name="menu">
			<prompt>
			 <audio src="welcome.vox"/>
			</prompt>
			<dtmf> 1 | 2 | 3</dtmf>
			<filled>
				<if cond="menu == '1'">
					<goto next="#voice_message"/>
				<elseif cond="menu == '2'"/>
					<goto next="#transfer"/>
				<elseif cond="menu == '3'"/>
					<goto next="#playcli"/>
				</if>
			</filled>
		</field>
	</form>

	<form id="voice_message">
	 <record name="recording" beep="true" sd="false" gaintype="agc" dtmfterm="true"
maxtime="300s" type="audio/basic">
			<prompt bargein="false">
			 <audio src="recording.vox"/>
			</prompt>
	 </record>

		<block>
			<prompt bargein="false">
			 <audio src="record_echo.vox"/>
				<value expr="recording"/>
			</prompt>
			<goto next="#menu"/>
		</block>
	</form>

	<form id="transfer">
	 <field name="ask_phone_num">
		<prompt>
		     <audio src="ask_number.vox"/>
		 </prompt>
			<dtmf>
%%%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%[%]]]]]]]]]]]]]]]]]]]]]]]]]]]]]]
</dtmf>
	     <filled>
		 <assign name="ask_phone_num" expr="ask_phone_num.replace(/[ #]/g, '')"/>
				<script>
					Print('ask_phone_num :' + ask_phone_num);
				</script>
				<prompt bargein="false">
				 <audio src="transfer.vox"/>
				 <value expr="ask_phone_num" class="digits"/>
				</prompt>
	     </filled>
	 </field>

	 <transfer name="transfer" bridge="false" destexpr="ask_phone_num" connecttimeout="0s" >
	 </transfer>
	</form>

	<form id="playcli">
		<block>
```

```
                        <prompt bargein="false">
                                <value expr="session.connection.ccxml.values.ani" class="digits"/>
                        </prompt>
                        <goto next="#menu"/>
                </block>
        </form>

    <catch event="noinput nomatch">
        <prompt>
                        <audio expr="_event+'.vox'"/>
         </prompt>
         <reprompt/>
    </catch>

    <catch event="error">
        <prompt bargein="false">
                        <audio src="error.vox"/>
         </prompt>
         <exit/>
    </catch>

        <catch event="telephone.disconnect.hangup">
                <exit/>
        </catch>
</vxml>
```

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