



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

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# **Application Notes for Configuring SIP IP Telephony Using Avaya 4600 Series IP Telephones, Avaya one-X™ Desktop Edition, and Asterisk Business Edition™ PBX – Issue 1.0**

## **Abstract**

These Application Notes describe the configuration steps required to configure Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition with the Asterisk Business Edition™ PBX. The Asterisk Business Edition PBX supports PBX telephony features, standard SIP features, and some supplementary SIP features. The Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition support the standard SIP features and some supplementary SIP features. The Avaya one-X™ Desktop Edition also supports presence and instant messaging. Testing was conducted at the Avaya Solution and Interoperability Test Lab.

# 1. Introduction

These Application Notes describe the configuration steps for using Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition<sup>1</sup> with the Asterisk Business Edition PBX. Only those configuration steps pertinent to interoperability of the Asterisk and Avaya equipment are covered. General administration information can be found in the product documentation as well as the specific references listed in **Section 9**.

The configuration used in the test is shown in **Figure 1**. All components are physically connected to a single Avaya C363T-PWR Converged Stackable Switch and are administered as a single subnet. A PC provides HTTP/HTTPS, DHCP, and TFTP server support. The Avaya one-X™ Desktop Edition clients run on separate PCs running Microsoft Windows XP Professional operating system.

The main difference among the four Avaya 4600 Series IP Telephones (4602SW, 4610SW, 4620SW, & 4621SW), for SIP functionality, is the number of line appearances supported by each phone (two line appearances for the 4602SW and five for the 4610SW, 4620SW, and 4621SW). The other major differences are that the 4610SW, 4620SW, and 4621SW telephones provide call log and speed dial applications and a variety of options to view and/or modify the current settings for the phone. The 4602SW does not support these applications and options. Avaya one-X™ Desktop Edition supports up to ten line appearances and provides a Windows-based graphical user interface for access to call log, speed dials, and settings.

The configuration steps described in these Application Notes apply to all four models of the Avaya 4600 Series IP Telephones and to Avaya one-X™ Desktop Edition. **Table 1** profiles the network management capabilities of the phones.

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<sup>1</sup> “Avaya one-X™ Desktop Edition” is the new brand for “Avaya SIP Softphone”. Therefore, that name is used in these Application Notes to document the product. However, the version of the product that was used for testing still has internal references to “Avaya SIP Softphone” or “SIP Softphone”.



Figure 1: Network Configuration Diagram

	Avaya 4600 Series IP Telephones	Avaya one-X™ Desktop Edition
Administration mechanisms	Configuration files	Manual, Configuration files
File transfer server	TFTP, HTTP, HTTPS	HTTP

Table 1: Network Management Capabilities of Avaya 4600 Series IP Telephones & Avaya one-X™ Desktop Edition

Avaya  
4602SW

Avaya  
4610SW

Avaya  
4620SW

## 2. Equipment and Software Validated

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

Equipment	Software
Avaya C363T-PWR Converged Stackable Switch	4.5.14
Avaya 4602SW, 4610SW, 4620SW, 4621SW IP Telephones	2.2.2 (SIP)
Avaya one-X™ Desktop Edition	2.1 (Load 43)
Web License Manager (WebLM)	3.1.1
Asterisk Business Edition™ PBX	A.1-3
Cisco 3745 Router	12.2(8r)T2
Windows PCs for Avaya one-X™ Desktop Edition	Microsoft Windows XP 2002 SP2
Windows Server PC for DHCP, HTTP/HTTPS, TFTP	Microsoft Windows 2000 Server SP4
Microsoft Internet Explorer web browser	6.0.2900.2180.xpsp_sp2_gdr.050301-1519
Microsoft Internet Information Services (IIS)	5.0
PumpKIN TFTP Server	2.00

**Table 2: Equipment and Version**

## 3. Supported SIP Features

In addition to standard calling capabilities (i.e., make a call, answer a call, drop a call), the Internet Engineering Task Force (IETF) has defined a supplementary set of calling features, often referred to as the SIPING-19 [10]. This reference provides a useful framework to describe product capabilities and compare features supported by various equipment vendors.

**Table 3** provides a summary of SIP features supported on the Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition when connected to the Asterisk Business Edition PBX.

Asterisk Business Edition PBX is a SIP registrar and acts as a "back-to-back user agent" (B2BUA). The Asterisk server can be configured to support most of the SIP features. Based on information from Reference [6], it is possible to configure the server to support some of the remaining more advanced SIP features (e.g., single line extension, find-me). However, the details in configuring these advanced SIP features are not readily available. For that reason, these features are not tested. In **Table 3**, the features that were not tested due to lack of configuration are identified in the *Comments* column.

**Section 4**, **Section 5**, and **Section 6** of these Application Notes describe the steps for configuring the Asterisk server and the Avaya clients to support the SIP features.

**NOTE:** The features that were tested for these Applications Notes are listed as “**Yes**” in the “**Supported**” column in **Table 3**. “**Yes**” denotes that the feature is implemented by the Asterisk server and by the Avaya phone device. “**Yes**” does not denote that the feature works. Features that did not work are identified in the “**Comments**” column and in **Section 7**. “**N/A**” denotes that this feature is implemented by the Asterisk server but is not implemented by the Avaya phone device. “**N/C**” denotes that this feature is implemented by the Asterisk server but was not configured for the test. “**No**” denotes that this feature is not implemented by the Asterisk server.

#	Feature	Supported		Comments
		Avaya 4600 Series IP Telephones	Avaya one-X™ Desktop Edition	
Supplementary Features				
1.	Call Hold	Yes	Yes	
2.	Consultation Hold	Yes	Yes	
3.	Music On Hold	Yes	Yes	
4.	Unattended Transfer – via the phone	Yes	Yes	
	Unattended Transfer – via the server	Yes	Yes	
5.	Attended Transfer	Yes	Yes	
6.	Transfer – Instant Messaging	No	No	Not applicable for Avaya 4600 Series IP Telephones; not supported by Avaya one-X™ Desktop Edition.
7.	Call Forward Unconditional – via the server	Yes	Yes	
	Call Forward Unconditional – via the phone	Yes	N/A	Not supported by the Avaya 4602SW or Avaya one-X™ Desktop Edition..
8.	Call Forward Busy – via the server	Yes	Yes	
	Call Forward Busy – via the phone	Yes	N/A	Not supported by the Avaya 4602SW or Avaya one-X™ Desktop Edition.
9.	Call Forward No Answer – via the server	Yes	Yes	
	Call Forward No Answer – via the phone	Yes	N/A	Not supported by the Avaya 4602SW or Avaya one-X™ Desktop Edition.
10.	3-way conference – 3 <sup>rd</sup> party added by user	Yes	Yes	
11.	3-way conference – 3 <sup>rd</sup> party calls and is joined by user	Yes	Yes	
12.	Single Line Extension (forking)	N/C	N/C	Requires advanced configuration of the server.
13.	Find-me	N/C	N/C	Requires advanced configuration of the server.
14.	Incoming Call Screening	N/C	N/C	Requires advanced configuration of the server.
15.	Outgoing Call Screening	N/C	N/C	Requires advanced configuration of the server.
16.	Call Park	Yes	Yes	
17.	Call Pick-up	Yes	Yes	
18.	Automatic Redial	N/C	N/C	Requires advanced configuration of the server.
19.	Click to Dial – via the server	N/C	N/C	Requires advanced configuration of the server.
	Click to Dial – via the phone	N/A	Yes	Not supported by the Avaya 4600 Series IP Telephones.
Other Features				
20.	Message Waiting Indicator (MWI)	Yes	Yes	Asterisk supports unsolicited MWI notification. This is incompatible with the Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition. The Avaya devices support Subscribe/Notify.
21.	Presence Tracking	N/A	Yes	Limited support provided by Asterisk server; not supported by the Avaya 4600 Series IP Telephones.
22.	Instant Messaging	No	No	Not supported by the Avaya 4600 Series IP Telephones.
23.	DTMF support	Yes	Yes	

**Table 3: SIP Feature Support with Asterisk Business Edition PBX**

### 3.1. Call Forwarding

In addition to the call forwarding features provided by the Asterisk server, the Avaya 4600 Series IP Telephones, except for the 4602SW, support local call forwarding. (Avaya one-X™ Desktop Edition does not support local call forwarding.) These features can be enabled via the `46xxsettings.txt` configuration file (see **Section 5**).

### 3.2. Message Waiting Indicator (MWI)

Message Waiting Indicator (MWI), as defined in IETF RFC 3265 [11] and IETF RFC 3842 (Subscribe/Notify method) [12], will illuminate or extinguish a MWI lamp when new voice messages are left or cleared for an extension, respectively. This feature was tested using the built-in voice mail system that comes with the Asterisk server. The method supported by Asterisk for MWI is to notify the client via an unsolicited message. As the Avaya devices support the Subscribe/Notify method, the MWI lamps were never illuminated or extinguished.

### 3.3. Click to Dial

While there is a click to dial application supported with Avaya one-X™ Desktop Edition, this implementation uses a direct link from Microsoft Internet Explorer (using an Internet Explorer Browser Helper Object) to Avaya one-X™ Desktop Edition (see **Section 6.4**, step #17 for the configuration of this feature). This implementation is not the same as the Click to Dial feature defined for SIP.

### 3.4. DTMF

For these Application Notes, SIP DTMF signaling (defined in IETF RFC 2833 [13]) is tested via interaction with the internal voice mail system. The server-based unattended transfer, call park, and call pickup features all use DTMF for feature activation. DTMF was also tested as part of the testing of these features.

### 3.5. Presence Tracking and Instant Messaging

SIP Presence and Instant Messaging (IETF RFC 3856 [14], RFC 2779 [15], and SIMPLE [SIP for Instant Messaging and Presence Leveraging Extensions] [16]) are supported by Avaya one-X™ Desktop Edition but not by the Avaya 4600 Series IP Telephones. The Asterisk server does not support instant messaging but does support presence tracking.

As stated in **Table 3**, the Asterisk server has limited support for presence tracking. Presence can be tracked for all stations (not just limited to Softphone stations) but the server can only send one of three states:

- the station is on-line and idle (registered) [*Available*]
- the station is off-line (un-registered) [*Offline*]
- the station is on-line and “on the phone” [*On the Phone*]

Avaya one-X™ Desktop Edition is able to set and track the following states:

- **the user's station is on-line and idle (registered)** [*Available*]
- **the user's station is off-line (un-registered)** [*Offline*]
- the user's station is on-line but does not want to be tracked [*Invisible/Offline*]
- **the user's station is on-line and "on the phone"** [*On the Phone*]
- the user's station is on-line and the user is "away from the phone" [*Away*]
- the user's station is on-line and the user is "busy" [*Busy*]

The states that both the Asterisk server and Avaya one-X™ Desktop Edition support are *Available*, *Offline*, and *On the Phone*.

## 4. Configure the Asterisk Business Edition PBX

### 4.1. Install Asterisk Business Edition PBX Software

Asterisk Business Edition PBX must be installed on Red Hat Enterprise 3 or Fedora Core 3. For these Application Notes, Red Hat Enterprise 3 was used. The installation of the software is covered in Reference [4] and Reference [5]. These Application Notes do not cover the installation of the software.

### 4.2. Administer the Asterisk Server

The following steps describe configuration of the Asterisk Business Edition PBX for use with Avaya SIP clients. The administration of the server is covered in Reference [4]. The installation of the server is covered in References [4] and [5]. For additional information and examples on configuring of the server, see Reference [6].

The Asterisk Business Edition PBX is configured by editing the following configuration files on the Linux system.

- `/etc/asterisk/sip.conf`
- `/etc/asterisk/extensions.conf`
- `/etc/asterisk/features.conf`
- `/etc/asterisk/voicemail.conf`

Steps	Description
1.	<p>First, log into the Linux system using the administration account (<b>admin</b>) that was created when Linux was installed. Enter the password for the administration account.</p> <pre>login as: admin admin@10.2.2.60's password: [admin@interop-asterisk-be admin]</pre> <p>Log in as the <b>root</b> account by entering “<b>su -</b>”. Enter the password for the root account.</p> <pre>[admin@interop-asterisk-be admin]\$ su - Password: [root@interop-asterisk-be root]#</pre> <p>Change the directory to “<b>/etc/asterisk</b>”.</p> <pre>[root@interop-asterisk-be root]# cd /etc/asterisk [root@interop-asterisk-be asterisk]#</pre> <p>Continue to configure the SIP domain.</p>



Steps	Description
2.	<p><i>Configure the SIP domain.</i></p> <p>Edit the <b>sip.conf</b> file with the vi editor (or other editor).</p> <p>All of the configuration files are organized into different “<i>contexts</i>”. A context is created by placing the context name in brackets (e.g., [<b>general</b>]).</p> <p>A parameter in this file, like all of the <b>.conf</b> files, will be in the following format:</p> <pre>[context1] parameter1=value parameter2=value</pre> <p>To configure the SIP domain, place the following parameters in the <b>general</b> context.</p> <pre>domain=&lt;domain-name&gt; fromdomain=&lt;domain-name&gt;</pre> <p>The following is an excerpt from the <b>sip.conf</b> file used for the testing documented for these Application Notes.</p> <pre><b>sip.conf</b>  [general] domain=asterisk.com fromdomain=asterisk.com</pre> <p>Save the file and exit the editor.</p>

Steps	Description
3.	<p><i>Configure users on the Asterisk server.</i></p> <p>The configuration of the users is done in two files:</p> <ul style="list-style-type: none"> <li>• <code>/etc/asterisk/sip.conf</code></li> <li>• <code>/etc/asterisk/extensions.conf</code></li> </ul> <p>The majority of the configuration is in the <code>sip.conf</code> file. The following parameters must be defined to configure a user.</p> <pre>[&lt;extension-number&gt;] ; comment - text after a semi-colon are comments type=friend host=dynamic username=&lt;extension-number&gt; secret=&lt;password&gt; callerid=&lt;caller-ID-string&gt; in the form of "caller-ID-name" &lt;caller-ID-#&gt;</pre> <p>Edit the <code>sip.conf</code> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <code>sip.conf</code> file used for the testing documented for these Application Notes.</p> <pre><i>sip.conf</i></pre> <div style="background-color: #e0ffff; padding: 10px;"> <pre>[60001] type=friend host=dynamic username=60001 secret=60001 ; password callerid="Avaya 4602SW" &lt;60001&gt;</pre> </div> <p>Save the file and exit the editor.</p> <p>The other configuration is in the <code>extensions.conf</code> file. The following parameters must be set to configure a user.</p> <pre>[default] exten =&gt; someexten,priority,application(arg1,arg2,...)</pre> <p>The following is an excerpt from the <code>extensions.conf</code> file used for the testing documented for these Application Notes (for the same extension listed above in the <code>sip.conf</code> file).</p> <pre><i>extensions.conf</i></pre> <div style="background-color: #e0ffff; padding: 10px;"> <pre>[default] exten =&gt; 60001,1,Macro(stdexten,60001,SIP/60001)</pre> </div> <p>Save the file and exit the editor.</p>

Steps	Description
4.	<p><i>Configure the users for voice mail.</i></p> <p>The configuration of the voice mail users is done in two files:</p> <ul style="list-style-type: none"> <li>• <code>/etc/asterisk/sip.conf</code></li> <li>• <code>/etc/asterisk/voicemail.conf</code></li> </ul> <p>The following parameters must be set in <code>sip.conf</code> to configure a user for voice mail.</p> <pre>[&lt;extension-number&gt;] mailbox=&lt;extension-number&gt;</pre> <p>Edit the <code>sip.conf</code> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <code>sip.conf</code> file used for the testing documented for these Application Notes.</p> <pre>sip.conf  [60001] ; This is a comment type=friend host=dynamic username=60001      ; This is another comment secret=60001 callerid="Avaya 4602SW" &lt;60001&gt; <b>mailbox=60001</b></pre> <p>The following parameters must be defined in <code>voicemail.conf</code> to configure a user for voice mail.</p> <pre>[default] &lt;extension-number&gt; =&gt; &lt;extension-number&gt;,&lt;caller-ID-name&gt;</pre> <p>Edit the <code>voicemail.conf</code> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <code>voicemail.conf</code> file used for the testing documented for these Application Notes.</p> <pre>voicemail.conf  [default] <b>60001 =&gt; 60001,Avaya 4602SW</b></pre> <p>Save the file and exit the editor.</p>

Steps	Description
5.	<p>Verify users configured on the Asterisk server.</p> <p>Log into the Linux system using the administration account (<b>admin</b>) that was created when Linux was installed. Log in as the root account by entering “<b>su -</b>”. Start the Asterisk command line interface (CLI) by entering “<b>asterisk -c</b>”</p> <pre>login as: admin admin@10.2.2.60's password: [admin@interop-asterisk-be admin]\$ su - Password: [root@interop-asterisk-be root]# asterisk -c</pre> <p>The Asterisk CLI window is displayed.</p> <pre>Asterisk ABE-A.1, Copyright (C) 1999 - 2005 Digium. Written by Mark Spencer &lt;markster@digium.com&gt; =====  Asterisk Ready. *CLI&gt;</pre> <p>Enter “<b>sip show peers</b>”. An entry will be displayed for all of the users that were defined. If the user is registered, the user’s IP address is listed under the <i>Host</i> column.</p> <pre>*CLI&gt; sip show peers Name/username      Host           Dyn Nat ACL Mask           Port      Status HT286FXS/60014     (Unspecified) D   D   255.255.255.255 0      Unmonitored 60013/60013        (Unspecified) D   D   255.255.255.255 0      Unmonitored HT488FXS/60012     (Unspecified) D   D   255.255.255.255 0      Unmonitored 60011/60011        10.2.2.112    D   D   255.255.255.255 5060   Unmonitored 60010/60010        (Unspecified) D   D   255.255.255.255 0      Unmonitored 60009/60009        (Unspecified) D   D   255.255.255.255 0      Unmonitored 60008/60008        (Unspecified) D   D   255.255.255.255 0      Unmonitored 60007/60007        (Unspecified) D   D   255.255.255.255 0      Unmonitored 60006/60006        (Unspecified) D   D   255.255.255.255 0      Unmonitored 60005/60005        (Unspecified) D   D   255.255.255.255 0      Unmonitored 60004/60004        (Unspecified) D   D   255.255.255.255 0      Unmonitored 60003/60003        10.2.2.106    D   D   255.255.255.255 5060   Unmonitored 60002/60002        10.2.2.103    D   D   255.255.255.255 5060   Unmonitored 60001/60001        10.2.2.105    D   D   255.255.255.255 5060   Unmonitored 14 sip peers [4 online , 10 offline] *CLI&gt;</pre>
6.	<p>The Asterisk server is ready to support SIP clients. <b>NOTE:</b> If the <b>asterisk</b> program is not running, clients will not be able to register with the Asterisk server.</p>

### 4.3. Administer the Supplementary Features on the Asterisk Server

Additional administration is needed to support the following features:

- Unattended transfer (via the server)
- Call park
- Call pickup
- Call forwarding
- Shuffling (Reinvite)<sup>2</sup> (See Reference [8])
- Presence tracking

**NOTE:** Some of the features listed above can be configured in one sequence. However, there are some features (e.g., Unattended Transfer & Shuffling, Call Forward Unconditional & Call Forward Busy) that are mutually exclusive. The steps listed below document how to configure all of the features. At the end, there is the step to restart the Asterisk server. In some cases, enabling one feature will disable another feature. Therefore, when it is time to test such features, the change should be made as described in the appropriate steps below followed by restarting the server.

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<sup>2</sup> Similar to Communications Manager (CM), Asterisk is involved in the media path on every call. Also similar to CM, Asterisk can be optioned to *shuffle* calls so that the media goes directly from one client to the other. This is referenced within Asterisk as *reinvite*.

Steps	Description
1.	<p><i>Configure the server for Unattended Transfer (via the server).</i></p> <p>The Asterisk server must be configured to support Unattended Transfer. When the feature is enabled, a user that is active on a call can press the “#” sign and a prompt from the Asterisk IVR system is played. The user is asked to enter the extension to which the call should be transferred. Upon entering a valid extension, the call is transferred.</p> <p>To enable this feature, the macro used for standard extensions (<b>macro-stdexten</b>) must be modified to add references to “t” or “T” in the <b>Dial()</b> command. This command is further described in Reference [7]. <b>NOTE:</b> This will disable shuffling/reinvites per the explanation in Reference [8]. The explanation of the above options for the <b>Dial()</b> command is as follows:</p> <ul style="list-style-type: none"> <li>• <b>t</b>: Allow the <i>called</i> user to transfer the call by hitting #</li> <li>• <b>T</b>: Allow the <i>calling</i> user to transfer the call by hitting #</li> </ul> <p>Edit the <b>extensions.conf</b> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <b>extensions.conf</b> file used for the testing documented for these Application Notes. <b>NOTE:</b> Anything that follows a semi-colon is a comment.</p> <pre><b>extensions.conf</b>  ; Original entry without the additional parameters needed to turn on transfer ;exten =&gt; s,1,Dial(\${ARG2},20)           ; Ring the interface, 20 seconds maximum  ; Dial() command change to allow the calling or called party to transfer <b>exten =&gt; s,1,Dial(\${ARG2},20,tT) ; Ring the interface, 20 seconds maximum</b></pre> <p>Save the file and exit the editor.</p> <p><b>NOTES:</b></p> <ol style="list-style-type: none"> <li><i>1. The Call Park feature is implemented as an Unattended Transfer to a known extension. Therefore, the above configuration is needed also for the Call Park feature.</i></li> <li><i>2. Enabling Unattended Transfer will disable shuffling. To test shuffling, use the instructions in step #7. Restart the Asterisk server after making these changes.</i></li> <li><i>3. Enabling the Unattended Transfer feature may interfere with external voice mail systems, external IVR systems, or any external system where the use of “#” is important. The Asterisk system will automatically start an Unattended Transfer when # is pressed.</i></li> </ol>

Steps	Description
2.	<p><i>Configure the server for Call Pickup.</i></p> <p>Normally, no configuration is needed to support the Call Pickup feature. However, the default dial string to invoke the feature is “*8”. The version of the Avaya SIP clients under test cannot dial “*” as part of the dial string. Therefore, the call pickup code was changed. For these Application Notes, Call Pickup dial string was changed to “<b>200</b>”.</p> <p>Edit the <b>features.conf</b> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <b>features.conf</b> file used for the testing documented for these Application Notes. <b>NOTE:</b> Anything that follows a semi-colon is a comment.</p> <pre><b>features.conf</b> ;pickupexten = *8           ; Configure the pickup extension. Default is *8 pickupexten = 200          ; Change the pickup extension to 200</pre> <p>Save the file and exit the editor.</p>
3.	<p><i>Configure the user for Call Pickup.</i></p> <p>In addition to configuring the server to support the Call Pickup feature, the users have to be placed in a pickup group. This is done in the <b>sip.conf</b> file.</p> <p>The following parameters must be set in <b>sip.conf</b>.</p> <pre>[&lt;extension&gt;] pickupgroup=&lt;pickup-group-#&gt;</pre> <p>The following is an excerpt from the <b>sip.conf</b> file used for the testing documented for these Application Notes.</p> <pre><b>sip.conf</b> [60001] type=friend username=60001 secret=60001 mailbox=60001@default callerid="Avaya 4602SW" &lt;60001&gt; host=dynamic <b>pickupgroup=2</b></pre> <p>Save the file and exit the editor.</p>

Steps	Description
4.	<p><i>Configure the server for Call Forwarding Unconditional.</i></p> <p>Call forwarding is supported by modifying the existing scripts or creating custom scripts in the <b>extensions.conf</b> file.</p> <p>The method used for these Application Notes to enable the call forwarding feature was to change the dial plan so that all calls are forwarded. This method was expedient for this testing. However, this is not the method that would be used in a production system as there is no easy way for the user to control how calls are forwarded. A user-friendly method is to invoke the local call forwarding feature on the telephone (see <b>Section 5</b>, step #3). Other examples for call forwarding are available in Reference [9].</p> <p>Call forwarding was enabled by modifying the macro used for standard extensions (<b>macro-stdexten</b>). In this example, all calls (to all extensions) are forwarded to extension 60007.</p> <p>Edit the <b>extensions.conf</b> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <b>extensions.conf</b> file used for the testing documented for these Application Notes. <b>NOTE:</b> Anything that follows a semi-colon is a comment.</p> <div> <p><b>extensions.conf</b></p> <pre> ; Comment out the standard dial macro for extensions ;exten =&gt; s,1,Dial(\${ARG2},20,tT)      ; Ring the interface, 20 seconds maximum  ; Unconditional forward to 60007 exten =&gt; s,1,Dial(SIP/60007,20,tT)    ; Ring the interface, 20 seconds maximum </pre> </div> <p>Save the file and exit the editor.</p> <p><i>Alternatively</i>, a single extension can be forwarded by modifying the definition of the extension in the “<i>default</i>” context. In this example, only calls to extension 60001 are forwarded unconditionally to extension 60007.</p> <p>Edit the <b>extensions.conf</b> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <b>extensions.conf</b> file used for the testing documented for these Application Notes. <b>NOTE:</b> Anything that follows a semi-colon is a comment.</p> <div> <p><b>extensions.conf</b></p> <pre> [default] ;exten =&gt; 60001,1,Macro(stdexten,60001,SIP/60001) exten =&gt; 60001,1,Dial(SIP/60007,20) </pre> </div> <p>Save the file and exit the editor.</p> <p><b>NOTE:</b> <i>It is not possible to test regular calls or the other forwarding features with unconditional forwarding enabled. This feature must be disabled before testing the other call features.</i></p>



Steps	Description
5.	<p><i>Configure the server for Call Forwarding No Answer.</i></p> <p>In this example, all calls are forwarded to extension 60007. Normally, calls that are not answered would be forwarded to voice mail. Similar to the Call Forwarding Unconditional feature, the method used to turn on this feature was expedient for this testing. The user can invoke this feature from the telephone (see <b>Section 5</b>, step #3).</p> <p>Edit the <b>extensions.conf</b> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <b>extensions.conf</b> file used for the testing documented for these Application Notes. <b>NOTE:</b> Anything that follows a semi-colon is a comment.</p> <pre><b>extensions.conf</b>  ; Comment out the standard dial macro for no answer ;exten =&gt; s-NOANSWER,1,VoiceMail(u\${ARG1})           ; If unavailable, send to voicemail w/ unavail announce ; Forward on no answer to 60007 <b>exten =&gt; s-NOANSWER,1,Dial(SIP/60007,20,tT) ; If unavailable, send to 60007</b></pre> <p>Save the file and exit the editor.</p> <p><b>NOTE:</b> <i>To test interaction with the voice mail system, this feature must be disabled.</i></p>
6.	<p><i>Configure the server for Call Forwarding Busy.</i></p> <p>In this example, all calls are forwarded to extension 60007. Normally, calls would be forwarded to voice mail. Similar to the Call Forwarding Unconditional feature, the method used to turn on this feature was expedient for this testing. The user can invoke this feature from the telephone (see <b>Section 5</b>, step #3).</p> <p>Edit the <b>extensions.conf</b> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <b>extensions.conf</b> file used for the testing documented for these Application Notes. <b>NOTE:</b> Anything that follows a semi-colon is a comment.</p> <pre><b>extensions.conf</b>  ; Comment out the standard dial macro for busy ;exten =&gt; s-BUSY,1,VoiceMail(b\${ARG1})           ; If busy, send to voicemail w/ busy announce ; Forward on busy to 60007 <b>exten =&gt; s-BUSY,1,Dial(SIP/60007,20,tT) ; If busy, send to 60007</b></pre> <p>Save the file and exit the editor.</p> <p><b>NOTE:</b> <i>To test interaction with the voice mail system, this feature must be disabled.</i></p>

Steps	Description
7.	<p><i>Configure the server for Shuffling/Reinvite.</i></p> <p>By default, all voice goes through the server. Shuffling can be turned on by modifying the <b>sip.conf</b> file.</p> <p>The following parameters must be set in <b>sip.conf</b> to configure a user for shuffling.</p> <pre>[&lt;extension&gt;] canreinvite=yes/no                                [shuffle or don't shuffle]</pre> <p>Edit the <b>sip.conf</b> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <b>sip.conf</b> file used for the testing documented for these Application Notes. <b>NOTE:</b> Anything that follows a semi-colon is a comment.</p> <pre><b>sip.conf</b>  [60001] type=friend username=60001 secret=60001 mailbox=60001@default callerid="Avaya 4602SW" &lt;60001&gt; host=dynamic pickupgroup=2 <b>canreinvite=yes</b>                                ; turn shuffling on</pre> <p>Save the file and exit the editor.</p> <p>In addition, to turn shuffling on, the macro used for standard extensions (<b>macro-stdexten</b>), in the <b>extensions.conf</b> file, must be modified to remove any references to “t” or “T” in the <b>Dial()</b> command (see Reference [7]) <b>NOTE:</b> This will disable unattended transfer as supported by the server. Also, shuffling must be turned on for both extensions that are involved in a conversation.</p> <p>The following is an excerpt from the <b>extensions.conf</b> file used for the testing documented for these Application Notes. <b>NOTE:</b> Anything that follows a semi-colon is a comment.</p> <pre><b>extensions.conf</b>  ; Entry without the additional parameters needed to turn off transfer <b>exten =&gt; s,1,Dial(\${ARG2},20)</b>                ; Ring the interface, 20 seconds maximum  ; Entry with the additional parameters needed to turn on transfer ;exten =&gt; s,1,Dial(\${ARG2},20,tT); Ring the interface, 20 seconds maximum</pre> <p>Save the file and exit the editor.</p> <p><b>NOTE:</b> Enabling Shuffling will disable Unattended Transfer and Call Park. To test Unattended Transfer and Call Park, use the instructions in step #1. Restart the Asterisk server after making these changes.</p>

Steps	Description
8.	<p><i>Configure the system to support Presence Tracking.</i></p> <p>Presence tracking is supported but only in a limited fashion. Asterisk reports presence status based on the actual status of the phone. That is, whether the phone is registered &amp; idle [<i>Available</i>], active on the phone [<i>On the Phone</i>], or un-registered [<i>Offline</i>]. The configuration of this feature is in the <b>extensions.conf</b> file.</p> <p>The following parameters must be set in <b>extensions.conf</b>.</p> <pre>[default] exten =&gt; &lt;extension-#&gt;,hint,SIP/&lt;extension-#&gt;</pre> <p>Edit the <b>extensions.conf</b> file with the vi editor (or other editor).</p> <p>The following is an excerpt from the <b>extensions.conf</b> file.</p> <pre>extensions.conf  exten =&gt; 60001,1,Macro(stdexten,60001,SIP/60001) exten =&gt; 60001,hint,SIP/60001</pre> <p>Save the file and exit the editor.</p>
9.	<p><i>Restart the Asterisk server.</i></p> <p>Open the Asterisk CLI (as documented in <b>Section 4.2</b>, step # 5). Enter “<b>restart gracefully</b>”.</p> <pre>*CLI&gt; restart gracefully Preparing for Asterisk restart... Restarting Asterisk NOW... Asterisk ABE-A.1, Copyright (C) 1999 - 2005 Digium. Written by Mark Spencer &lt;markster@digium.com&gt; ===== Asterisk Ready. *CLI&gt;</pre> <p>All of the SIP features are configured and ready to use.</p>

## 5. Configure the Avaya 4600 IP Telephones

The SIP software should be installed in the Avaya 4600 Series IP Telephones using the procedures described in Reference [1]. The SIP specific software can be downloaded from the Avaya Support Center site (<http://www.avaya.com/support>). Download the **46xxSIP\*.zip** file (where \* is the date) and install the files per the instructions in Reference [1]. Any Avaya 4600 Series IP Telephone that has the H.323 software loaded must be optioned to install the SIP software by modifying the **SIG** value to **SIP** (press <MUTE>-**S-I-G** on the telephone keypad) and restarting the telephone.

The installation of the telephone software and the configuration of the telephone require access to a TFTP server or to an HTTP/HTTPS server.<sup>3</sup> These Application Notes will not provide the details of the administration of Avaya 4600 Series IP Telephones (see Reference [1]). These Application Notes will provide details of the specific SIP related configuration items.

Steps	Description
1.	<p>Locate the <b>46xxsettings.txt</b> file to administer the SIP specific settings for the Avaya 4600 Series IP Telephones.</p> <p>This file is stored on the PC that provides TFTP or HTTP/HTTPS services to the telephones. The location of this file depends on the software that is used to provide these services. For these Application Notes, HTTP services were used, provided by Windows Internet Information Services (IIS), and the file was stored in the <code>Inetpub\wwwroot</code> folder on the Windows server PC.</p> <p>The <b>46xxsettings.txt</b> file is a text file that can be edited with Windows Notepad, Windows WordPad, or other text editor. Refer to <b>Figure 2</b> and <b>Table 4</b> in the following steps for information on the changes that are required.</p>

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<sup>3</sup> If the telephone is running software prior to 2.2, HTTP is not supported and access to a TFTP server will be needed. It is possible to need both a TFTP server and an HTTP/HTTPS server until all phones have been upgraded to 2.2.

Steps	Description
2.	<p>An excerpt of the <b>46xxsettings.txt</b> file with the SIP specific parameters is shown in <b>Figure 2</b>. A description of the important parameters is in <b>Table 4</b>. Edit these parameters to configure the SIP settings for the Avaya 4600 Series IP Telephones. The sample in <b>Figure 2</b> shows the values used in the compliance test.</p> <p><b>NOTE:</b> Lines that start with “##” are comments.</p> <pre> ##### SIP-Specific Settings ##### ## Use the following setting to configure SIP-specific parameters ## ## Examples: ## SET CALLFWDSTAT  "3" ## SET CALLFWDDELAY  "5" ## SET CALLFWDADDR  "cover@avaya.com" ## SET COVERAGEADDR "cover@avaya.com" ## SET DIALPLAN      "[23]xxxx 91xxxxxxxxxxx 9[2-9]xxxxxxxxxx" ## SET DIALWAIT      "5" ## SET MUSICSRVR     "" ## SET MWISVR        "192.168.0.7" ## SET PHNUMOFSA     "3" ## SET REGISTERWAIT  "3600" ## SET SIPDOMAIN     "avaya.com" ## SET SIPPROXYSRVR  "192.168.0.8" ## SET SIPPORT       "5060" ## SET SIPREGISTRAR  "192.168.0.9" ## SET SPEAKERSTAT   "2" ## SET RTCPMON       "192.168.0.10" ## SET RTCPMONPORT   "5005"  SET DIALPLAN      "6xxxx" SET SIPDOMAIN     "asterisk.com" SET SIPPROXYSRVR  "10.2.2.60" SET SIPREGISTRAR  "10.2.2.60" SET CALLFWDSTAT   "7" SET CALLFWDADDR   "60007@asterisk.com" </pre> <p><b>Figure 2: Excerpt from Sample Avaya 4600 Series IP Telephone 46xxsettings.txt File</b></p>

Steps	Description																																																						
3.	<p><b>Table 4</b> shows the SIP specific parameters that can be configured.</p> <p>The parameters that are critical to configure are <b>DIALPLAN</b>, <b>SIPDOMAIN</b>, <b>SIPROXYSRVR</b>, and <b>SIPREGISTRAR</b>. In addition, for testing local call forwarding, also configure <b>CALLFWDSTAT</b> and <b>CALLFWDADDR</b>.</p> <ul style="list-style-type: none"><li>▪ <b>CALLFWDSTAT</b>: This parameter defines which call forwarding buttons are configured on the phone. In this example, all three buttons (unconditional call forward [<b>Call Forward</b>], call forward on busy [<b>CFwrd-Busy</b>], and call forward on no answer [<b>CFwrd-DA</b>] are configured on the phone. On the phones, except the 4602SW, the buttons <b>Call Forward</b>, <b>CFwrd-Busy</b>, and <b>CFwrd-DA</b> will appear.</li><li>▪ <b>CALLFWDADDR</b>: This parameter defines the address to which calls will be forwarded when one of the call forwarding features are invoked. <b>NOTE</b>: The user can change this value using the interface on the phone.</li><li>▪ <b>DIALPLAN</b>: This parameter defines the dial plan used by the phone. In this example, “<b>6xxxx</b>” defines that extensions will start with the number 5 and will have a length of 5 digits. The phone will automatically dial the number once the 5<sup>th</sup> digit is entered.</li><li>▪ <b>SIPDOMAIN</b>: This parameter defines the SIP domain. In this example, “<b>asterisk.com</b>” is used as the SIP domain.)</li><li>▪ <b>SIPROXYSRVR</b>: This parameter is for the IP address of the proxy.</li><li>▪ <b>SIPREGISTRAR</b>: This parameter takes the same value as the proxy server (<b>SIPROXYSRVR</b>).</li></ul> <p><b>NOTE</b>: The <b>PHNUMOFSA</b> and <b>SPEAKERSTAT</b> default values do not apply to the 4602SW. The 4602SW telephone has a maximum of two (2) line appearances and the speakerphone works as a monitor or one-way speaker. The <b>CALLFWDSTAT</b> and <b>CALLFWDADDR</b> values also do not apply to the 4602SW.</p> <table><tr><th>SIP Setting</th><th>Parameter Name</th><th>Example Value</th></tr><tr><td>Call Forwarding Status</td><td>CALLFWDSTAT</td><td>7</td></tr><tr><td>Call Forwarding Delay</td><td>CALLFWDDELAY</td><td>5 (Default)</td></tr><tr><td>Call Forwarding Address</td><td>CALLFWDADDR</td><td>60007@asterisk.com</td></tr><tr><td>Forwarding Coverage Address</td><td>COVERAGEADDR</td><td>NULL [“”] (Default)</td></tr><tr><td>Dial Plan</td><td>DIALPLAN</td><td>“6xxxx”</td></tr><tr><td>Dial Wait</td><td>DIALWAIT</td><td>5 (Default)</td></tr><tr><td>Music On Hold Server</td><td>MUSICSRVR</td><td>NULL [“”] (Default)</td></tr><tr><td>Message Waiting Server</td><td>MWISVR</td><td>NULL [“”] (Default)</td></tr><tr><td>No of Line Appearances</td><td>PHNUMOFSA</td><td>3 (Default)</td></tr><tr><td>Registration Expiration Timer</td><td>REGISTERWAIT</td><td>3600 (Default)</td></tr><tr><td>SIP Domain</td><td>SIPDOMAIN</td><td>asterisk.com</td></tr><tr><td>SIP Proxy Server</td><td>SIPPROXYSRVR</td><td>“10.2.2.60”</td></tr><tr><td>SIP Proxy Server Port</td><td>SIPPORT</td><td>5060 (Default)</td></tr><tr><td>SIP Registrar</td><td>SIPREGISTRAR</td><td>“10.2.2.60”</td></tr><tr><td>Speakerphone Status</td><td>SPEAKERSTAT</td><td>2 (Default)</td></tr><tr><td>RTCP Monitor</td><td>RTCPMON</td><td>NULL [“”] (Default)</td></tr><tr><td>RTCP Monitor Port</td><td>RTCPMON</td><td>5005 (Default)</td></tr></table> <p><b>Table 4: Avaya 4600 Series IP Telephones SIP Parameters</b></p>	SIP Setting	Parameter Name	Example Value	Call Forwarding Status	CALLFWDSTAT	7	Call Forwarding Delay	CALLFWDDELAY	5 (Default)	Call Forwarding Address	CALLFWDADDR	60007@asterisk.com	Forwarding Coverage Address	COVERAGEADDR	NULL [“”] (Default)	Dial Plan	DIALPLAN	“6xxxx”	Dial Wait	DIALWAIT	5 (Default)	Music On Hold Server	MUSICSRVR	NULL [“”] (Default)	Message Waiting Server	MWISVR	NULL [“”] (Default)	No of Line Appearances	PHNUMOFSA	3 (Default)	Registration Expiration Timer	REGISTERWAIT	3600 (Default)	SIP Domain	SIPDOMAIN	asterisk.com	SIP Proxy Server	SIPPROXYSRVR	“10.2.2.60”	SIP Proxy Server Port	SIPPORT	5060 (Default)	SIP Registrar	SIPREGISTRAR	“10.2.2.60”	Speakerphone Status	SPEAKERSTAT	2 (Default)	RTCP Monitor	RTCPMON	NULL [“”] (Default)	RTCP Monitor Port	RTCPMON	5005 (Default)
SIP Setting	Parameter Name	Example Value																																																					
Call Forwarding Status	CALLFWDSTAT	7																																																					
Call Forwarding Delay	CALLFWDDELAY	5 (Default)																																																					
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Forwarding Coverage Address	COVERAGEADDR	NULL [“”] (Default)																																																					
Dial Plan	DIALPLAN	“6xxxx”																																																					
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Music On Hold Server	MUSICSRVR	NULL [“”] (Default)																																																					
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SIP Proxy Server Port	SIPPORT	5060 (Default)																																																					
SIP Registrar	SIPREGISTRAR	“10.2.2.60”																																																					
Speakerphone Status	SPEAKERSTAT	2 (Default)																																																					
RTCP Monitor	RTCPMON	NULL [“”] (Default)																																																					
RTCP Monitor Port	RTCPMON	5005 (Default)																																																					

Steps	Description
4.	<p>Connect the Avaya 4600 Series IP Telephone to the network and reboot the phone. If TFTP or HTTP/HTTPS support have been properly configured, the phone will download the software and configuration files and prompt the user for the extension and password. Enter the extension followed by the # key. For example, 60000#.</p> <div> <pre>Ext. = #=OK New=60000</pre> </div> <p>Enter the password followed by the # key. For example, 123456#.</p> <div> <pre>Password= #=OK</pre> </div> <p>On successful registration, the telephone's display will display something similar to the following:</p> <div> <pre>9:50pm 4/2/06 60000</pre> </div> <p>The first line displays the date and the second line displays the extension number.</p>
5.	Repeat step #4 for each of the Avaya 4600 Series IP Telephones.

## 6. Configure the Avaya one-X™ Desktop Edition Software

The Avaya one-X™ Desktop Edition R2.1 software is available on the Avaya Support Center site (<http://www.avaya.com/support>). The installation and usage instructions for Avaya one-X™ Desktop Edition are documented in Reference [2]. The Avaya one-X™ Desktop Edition can be configured manually using the graphical user interface. As an option, there are some configurations items that can be configured via the `46xxsettings.txt` file. The steps for this configuration are documented in Reference [3] and further described in these Application Notes.

The configuration of Avaya one-X™ Desktop Edition via the `46xxsettings.txt` file requires the use of a DHCP server and an HTTP server. Avaya one-X™ Desktop Edition uses the same DHCP and HTTP mechanism used by the Avaya 4600 Series IP Telephones (see **Section 5**). These Application Notes will document only the steps to configure the DHCP server to support Avaya one-X™ Desktop Edition and the modifications to the `46xxsettings.txt` file in support of Avaya one-X™ Desktop Edition. **NOTE:** Using a DHCP and HTTP server to configure Avaya one-X™ Desktop Edition is optional. The information can be entered manually using the Avaya one-X™ Desktop Edition configuration wizard.

### 6.1. WebLM License Server

Avaya one-X™ Desktop Edition must connect to a WebLM license server and acquire a license from the license server. Without this license, Avaya one-X™ Desktop Edition will not support the full set of SIP features. The license server IP address can be entered manually using the Avaya one-X™ Desktop Edition configuration wizard. Alternatively, the IP address of the license server can be entered as one of the downloadable options in the `46xxsettings.txt` file.

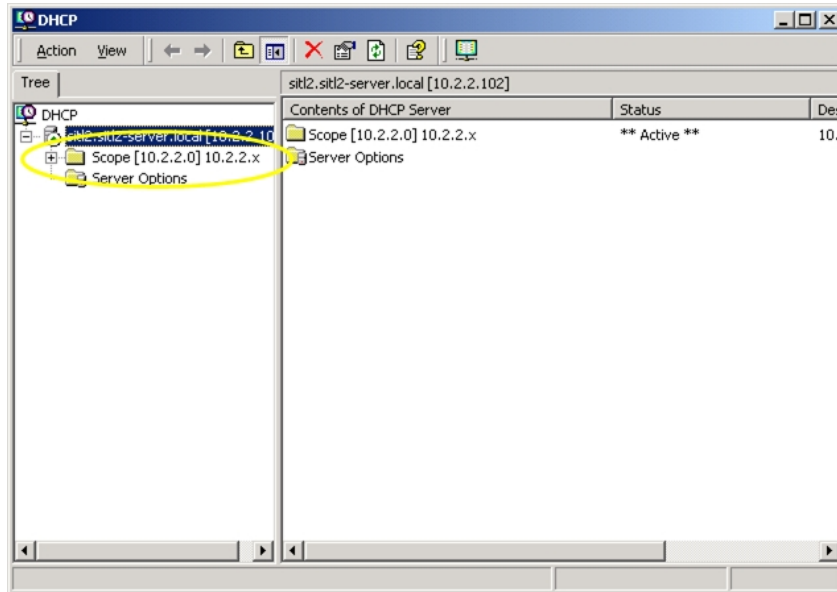
The WebLM server software can be installed on several operating systems (including Microsoft Windows and Linux). The WebLM software for these operating systems is available on the Avaya Support Center site (<http://www.avaya.com/support>). The installation of the WebLM software is described in Reference [3] but is not covered in these Application Notes.

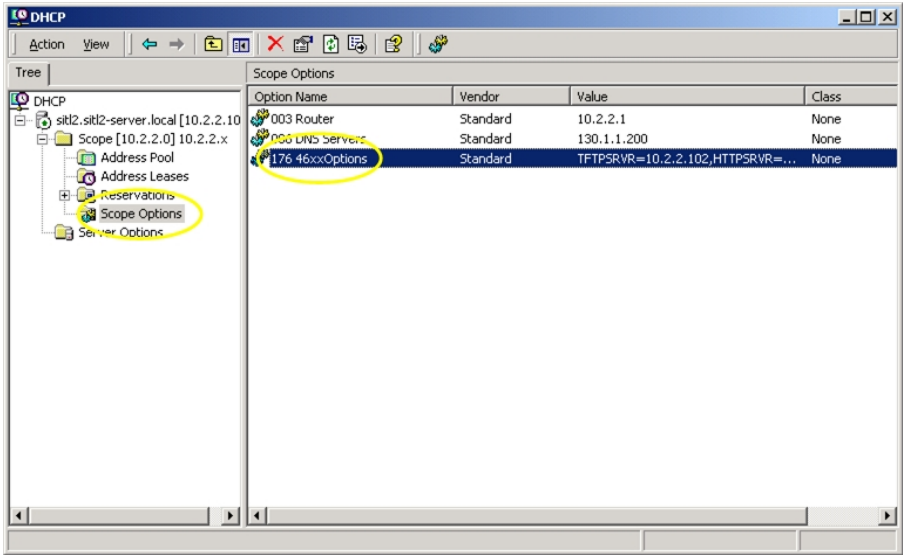
WebLM is also installed as part of Avaya SIP Enablement Services (SES). As these Application Notes describe testing the Avaya phones on a third party server, it is not expected that an SES would be part of the configuration. However, for simplicity, Avaya one-X™ Desktop Edition acquires its license by connecting to the WebLM server installed on an SES server (see **Figure 1**).

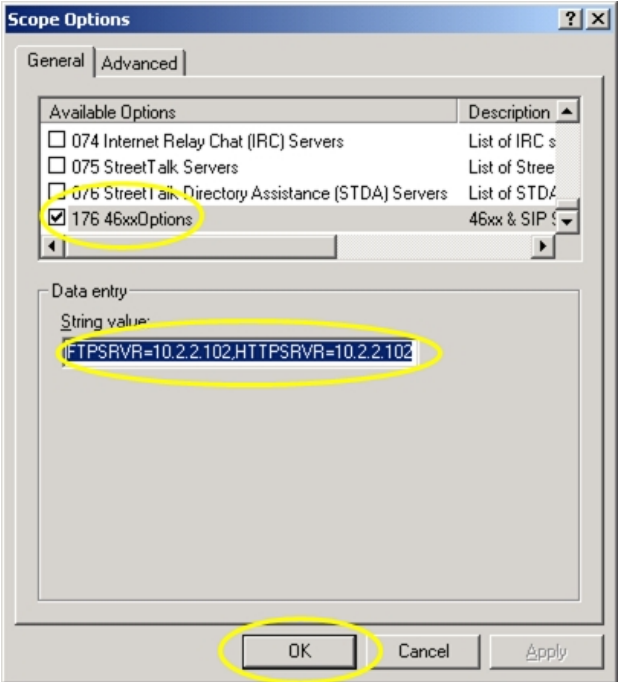
### 6.2. Configure the DHCP Server for Avaya one-X™ Desktop Edition (OPTIONAL)

The DHCP server that supports the Avaya 4600 Series IP Telephones can also be used to support Avaya one-X™ Desktop Edition as documented in Reference [1]. The DHCP server used in this testing is provided with Microsoft Windows 2000 Server. The examples shown in these Applications Notes are specific to this DHCP server implementation but also apply to other DHCP servers.



Steps	Description						
1.	<p>Edit the DHCP option configured for the Avaya 4600 Series IP Telephones.</p> <p>Start the DHCP server client from the Windows Control Panel (<b>Start → Settings → Control Panel</b>). Click on <b>Administrative Tools</b>. Click on <b>DHCP</b>.</p>						
2.	<p>The DHCP main dialogue is displayed.</p> <p>Click on the “+” next to the DHCP scope that was created for the Avaya 4600 Series IP Telephones. In this example, select the scope for the <i>10.2.2.x</i> subnet.</p> <div><p>The screenshot shows the DHCP console window. The tree view on the left is expanded to show the contents of the DHCP server. The 'Scope [10.2.2.0] 10.2.2.x' is highlighted with a yellow oval. The right pane shows the 'Contents of DHCP Server' table with columns for Name, Status, and Description. The table contains one entry: 'Scope [10.2.2.0] 10.2.2.x' with status '** Active **' and description '10.2.2.x'.</p><table><thead><tr><th>Name</th><th>Status</th><th>Description</th></tr></thead><tbody><tr><td>Scope [10.2.2.0] 10.2.2.x</td><td>** Active **</td><td>10.2.2.x</td></tr></tbody></table></div>	Name	Status	Description	Scope [10.2.2.0] 10.2.2.x	** Active **	10.2.2.x
Name	Status	Description					
Scope [10.2.2.0] 10.2.2.x	** Active **	10.2.2.x					

Steps	Description
3.	<p>Additional parameters are shown for the <i>10.2.2.x</i> DHCP scope on the left hand side of the DHCP window. Click on <b>Scope Options</b>.</p> <p>Select the scope option that was created for the Avaya 4600 Series IP Telephones on the right hand side of the DHCP window. In this example, select the scope option <i>176 46xxOptions</i>.</p> <p>From the menu, select <b>Action → Properties</b>.</p> 

Steps	Description
4.	<p>The <i>Scope Options</i> dialogue is displayed.</p> <p>The value for the <i>176 46xxOptions</i> option needs to be modified. The default string used by Avaya one-X™ Desktop Edition is in the form of:</p> <p><b>HTTPSRVR=nnn.nnn.nnn.nnn</b></p> <p><b>HTTPSRVR</b> is the IP address of the server that holds the configuration script for Avaya one-X™ Desktop Edition. The <b>TFTPSRV</b> parameter is also set for the Avaya 4600 Series IP Telephones. For this example, the option is set to the following.</p> <p><b>TFTPSRV=10.2.2.102,HTTPSRVR=10.2.2.102</b></p> <p>Click the <b>OK</b> button to save the change.</p> 
5.	<p>Refresh the DHCP server to use the updated scope options.</p> <p>To refresh the options, select the scope option (<i>176 46xxOptions</i>). Then from the menu, select <b>Action → Refresh</b>.</p>

### 6.3. Configure 46xxsettings.txt for Avaya one-X™ Desktop Edition (OPTIONAL)


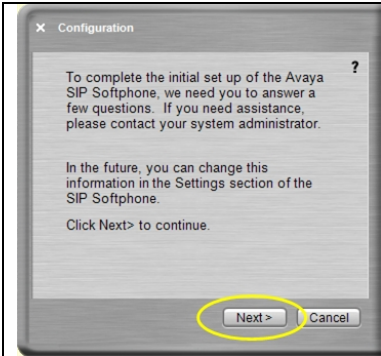
Assuming that the DHCP server has been configured to support Avaya one-X™ Desktop Edition (see **Section 6.2**), the **46xxsettings.txt** configuration file must be modified to provide the Avaya one-X™ Desktop Edition configuration information.

Steps	Description
1.	<p>Locate the <b>46xxsettings.txt</b> file to administer the SIP specific settings for Avaya one-X™ Desktop Edition.</p> <p>This file is stored on the PC that provides HTTP services to the telephones. The location of this file depends on the software that is used to provide these services. For these Application Notes, the HTTP services were provided by Windows Internet Information Services (IIS) and the <b>46xxsettings.txt</b> file was stored on the <code>Inetpub\wwwroot</code> folder on the Windows server PC.</p> <p>The <b>46xxsettings.txt</b> file is a text file that can be edited with Windows Notepad, Windows WordPad, or other text editor. Refer to <b>Figure 3</b> and <b>Table 5</b> in the following steps for information on the changes that are required.</p>
2.	<p>An excerpt of the <b>46xxsettings.txt</b> file with the SIP specific parameters is shown in <b>Figure 3</b>. A description of the important parameters is in <b>Table 5</b>. Edit these parameters to configure the SIP settings for Avaya one-X™ Desktop Edition. The sample in <b>Figure 3</b> shows the values used in the compliance test.</p> <p><b>NOTE:</b> Lines that start with “##” are comments.</p> <div style="border: 1px solid black; background-color: #e0f7fa; padding: 10px; margin: 10px 0;"> <pre>##### ## Avaya one-X™ Desktop Edition Variables ##### ## ## Examples: ## SET SIPPROXYSRVR 192.168.0.10 ## SET WEBLMSRVR 192.168.0.11 ## SET SP_DIRSRVR ldap-east.post.avaya.com ## SET SP_DIRSRVRPORT 389 ## SET SP_DIRTOPDN ou=People,o=avaya.com ## SET SP_AC 212 ## SET LOCAL_CALL_PREFIX AC ## SET SIPPROXYSRVR ## SET PHNCC ## SET PHNDPLENGTH ## SET PHNIC ## SET PHNLD ## SET PHNLDLENGTH ## SET PHNOL  SET SIPPROXYSRVR 10.2.2.60 SET WEBLMSRVR 10.2.2.70</pre> </div> <p><b>Figure 3: Excerpt from Sample Avaya one-X™ Desktop Edition 46xxsettings.txt File</b></p>



Steps	Description																																										
3.	<p><b>Table 5</b> shows the SIP specific parameters that can be configured.</p> <p>The parameters that are critical to configure are <b>SIPPROXYSRVR</b> and <b>WEBLMSRVR</b>.</p> <ul style="list-style-type: none"><li>▪ <b>SIPPROXYSRVR</b>: Enter the IP address of the Asterisk server.</li><li>▪ <b>WEBLMSRVR</b>: Enter the address of the WebLM server.</li></ul> <table><tr><th>SIP Setting</th><th>Parameter Name</th><th>Example Value</th></tr><tr><td>SIP Proxy Server</td><td>SIPPROXYSRVR</td><td>"10.2.2.60"</td></tr><tr><td>WebLM Server</td><td>WEBLMSRVR</td><td>"10.2.2.70"</td></tr><tr><td>LDAP Directory Server</td><td>SP_DIRSRVR</td><td>5 (Default)</td></tr><tr><td>LDAP Directory Server Port</td><td>SP_DIRSRVRPORT</td><td>NULL ["" ] (Default)</td></tr><tr><td>LDAP Directory Topmost Distinguished Name</td><td>SP_DIRTOPDN</td><td>NULL ["" ] (Default)</td></tr><tr><td>Area Code</td><td>SP_AC</td><td>NULL ["" ] (Default)</td></tr><tr><td>Local Call Prefix</td><td>LOCAL_CALL_PREFIX</td><td>"DIAL_AS_IS" (Default)</td></tr><tr><td>Country Code</td><td>PHNCC</td><td>1 (Default)</td></tr><tr><td>Dial Plan Length</td><td>PHNDPLENGTH</td><td>NULL ["" ] (Default)</td></tr><tr><td>International Access Code</td><td>PHNIC</td><td>011 (Default)</td></tr><tr><td>Long Distance Access Code</td><td>PHNLD</td><td>1 (Default)</td></tr><tr><td>National Telephone Length</td><td>PHNLDLENGTH</td><td>10 (Default)</td></tr><tr><td>Outside Line Access Code</td><td>PHNOL</td><td>9 (Default)</td></tr></table> <p><b>Table 5: Avaya one-X™ Desktop Edition SIP Parameters</b></p>	SIP Setting	Parameter Name	Example Value	SIP Proxy Server	SIPPROXYSRVR	"10.2.2.60"	WebLM Server	WEBLMSRVR	"10.2.2.70"	LDAP Directory Server	SP_DIRSRVR	5 (Default)	LDAP Directory Server Port	SP_DIRSRVRPORT	NULL ["" ] (Default)	LDAP Directory Topmost Distinguished Name	SP_DIRTOPDN	NULL ["" ] (Default)	Area Code	SP_AC	NULL ["" ] (Default)	Local Call Prefix	LOCAL_CALL_PREFIX	"DIAL_AS_IS" (Default)	Country Code	PHNCC	1 (Default)	Dial Plan Length	PHNDPLENGTH	NULL ["" ] (Default)	International Access Code	PHNIC	011 (Default)	Long Distance Access Code	PHNLD	1 (Default)	National Telephone Length	PHNLDLENGTH	10 (Default)	Outside Line Access Code	PHNOL	9 (Default)
SIP Setting	Parameter Name	Example Value																																									
SIP Proxy Server	SIPPROXYSRVR	"10.2.2.60"																																									
WebLM Server	WEBLMSRVR	"10.2.2.70"																																									
LDAP Directory Server	SP_DIRSRVR	5 (Default)																																									
LDAP Directory Server Port	SP_DIRSRVRPORT	NULL ["" ] (Default)																																									
LDAP Directory Topmost Distinguished Name	SP_DIRTOPDN	NULL ["" ] (Default)																																									
Area Code	SP_AC	NULL ["" ] (Default)																																									
Local Call Prefix	LOCAL_CALL_PREFIX	"DIAL_AS_IS" (Default)																																									
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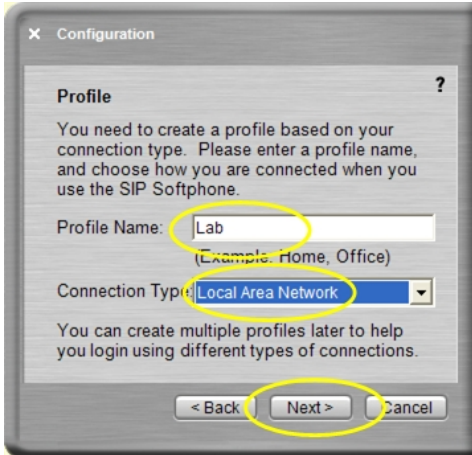
## 6.4. Configure Avaya one-X™ Desktop Edition

After Avaya one-X™ Desktop Edition is installed, it must be configured.


Steps	Description
1.	<p>Start Avaya one-X™ Desktop Edition via the menu <b>Start → Programs → Avaya SIP Softphone → Avaya SIP Softphone</b>.<sup>4</sup> The first time the program is executed, the Configuration Wizard will be displayed. Edit the <b>46xxsettings.txt</b> file to administer the Avaya one-X™ Desktop Edition specific settings. The main <i>Avaya one-X™ Desktop Edition</i> dialogue and the <i>Configuration</i> dialogue are displayed.</p> <p>Click the <b>Next</b> button to continue.</p> <div></div>

<sup>4</sup> While the name of the product has changed to “Avaya one-X™ Desktop Edition”, the software used for testing still refers to “Avaya SIP Softphone”. This includes the installation folder and Windows program group.

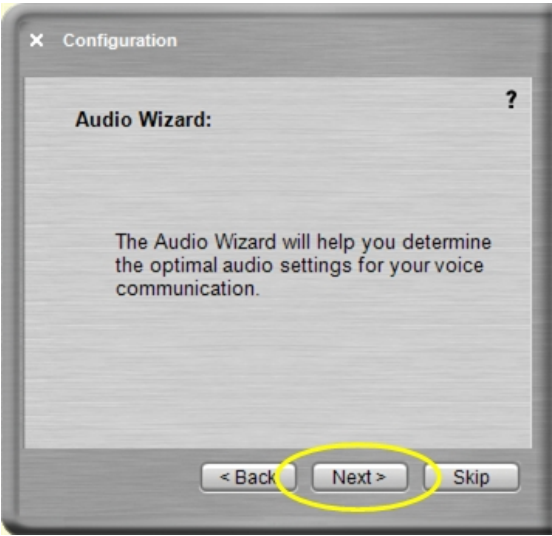
Steps	Description
2.	<p>The <i>Account</i> dialogue is displayed.</p> <p>Enter the name for the Avaya one-X™ Desktop Edition station. This name will be used for caller identification. Enter the user name that will be used to authenticate with the SIP registrar. <b>NOTE:</b> It is recommended to use a domain name instead of an IP address for the user name. For this example, the name is <b>60006@asterisk.com</b> instead of <b>60006@10.2.2.60</b>. Choose the appropriate option for the password.</p> <p>Click the <b>Next</b> button to continue.</p> 
3.	<p>The <i>SIP Server/License Server</i> dialogue is displayed.</p> <p>If the DHCP server was configured properly in <b>Section 6.2</b> and the <b>46xxsettings.txt</b> file has the proper information as described in <b>Section 6.3</b>, Avaya one-X™ Desktop Edition will get the IP address of the proxy server and the license server.</p> <p>Click the <b>Next</b> button to continue.</p> 


Steps	Description
4.	<p>An updated <i>SIP Server/License Server</i> dialogue is displayed.</p> <p>If the information was not gathered from the DHCP &amp; HTTP servers, the fields would be blank and have to be entered them manually.</p> <p>Click the <b>Next</b> button to continue.</p> 
5.	<p>The <i>Profile</i> dialogue is displayed.</p> <p>For this example, <b>Lab</b> was entered for the <i>Profile Name</i>. Any name can be used. Select <b>Local Area Network</b> for the <i>Connection Type</i>. The other options are <b>Cable, xDSL, or IDSN and Modem (28800 bps or faster)</b>.</p> <p>Click the <b>Next</b> button to continue.</p> 

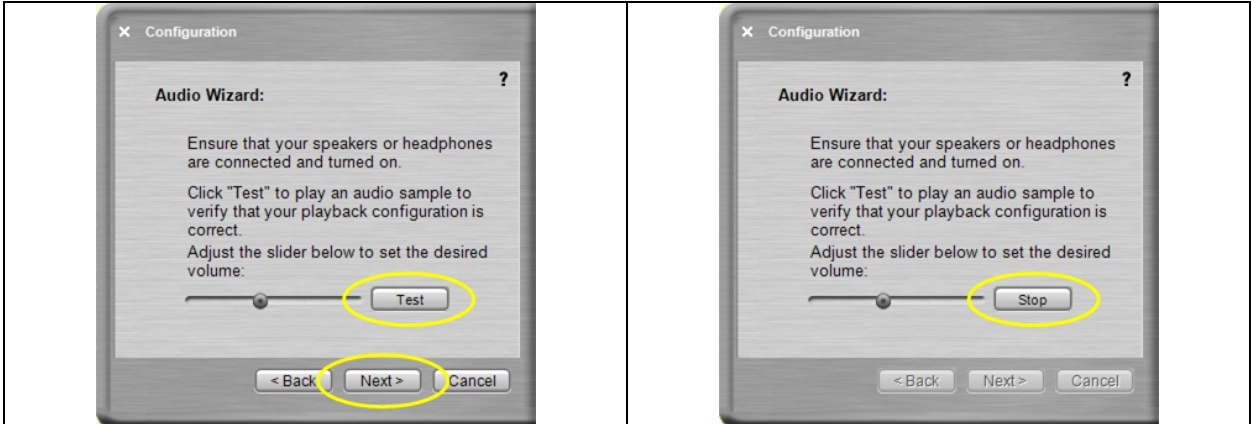



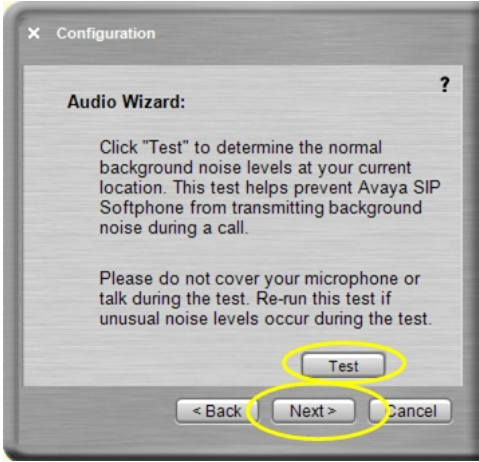
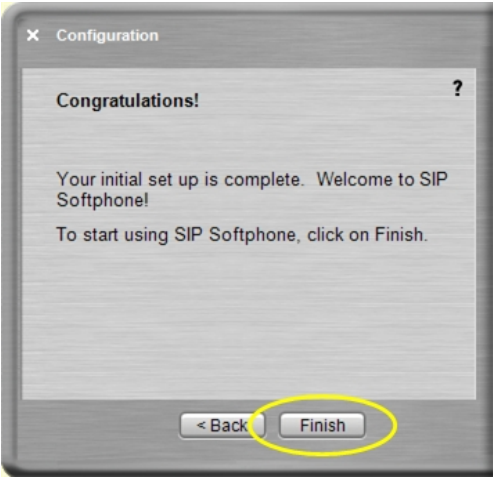
Steps	Description
6.	<p>The <i>Dialing Rules</i> dialogue is displayed.</p> <p>As there are only internal calls made, the default values are used for these Application Notes. To configure the software for calls outside of the Asterisk server (local, long distance, and international), enter the values that correspond to the external dialing rules for the server.</p> <p>Click the <b>Next</b> button to continue.</p> 


Steps	Description
7.	<p>The <i>Voicemail Integration</i> dialogue is displayed.</p> <p>Avaya one-X™ Desktop Edition can be optioned to dial the voice mail extension. In this example, the default voice mail extension for the Asterisk server is <b>8500</b>. When this option is configured and the Avaya client is registered with the server, clicking on the voice mail button (see <b>Figure 4</b>) will dial the voice mail system.</p> <p>Click the <b>Next</b> button to continue.</p> <div> </div> <div> </div> <p><b>Figure 4: Registered Avaya one-X™ Desktop Edition (with Voice Mail Button)</b></p>

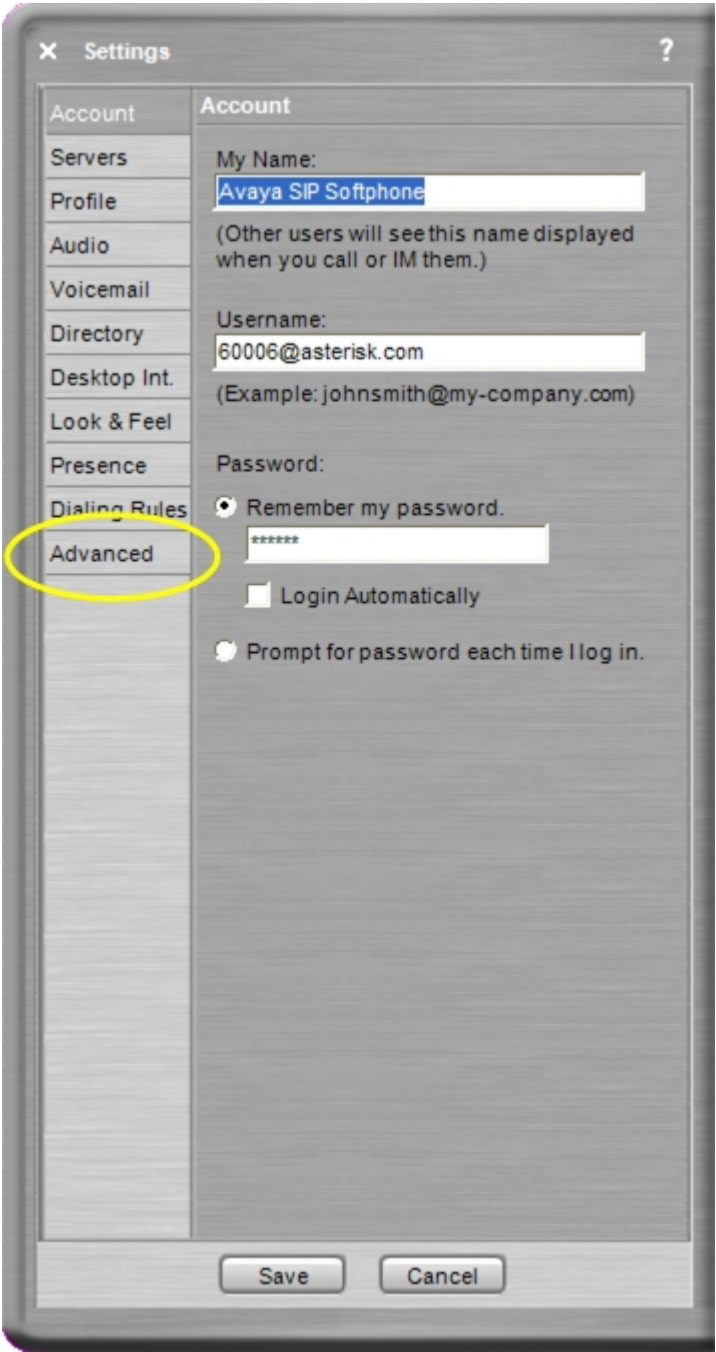
Steps	Description
8.	<p>The <i>Audio Wizard</i> dialogue is displayed.</p> <p>Click the <b>Next</b> button to continue.</p> 

Steps	Description
9.	<p>The <i>Audio Wizard: Select Sound Device</i> dialogue is displayed.</p> <p>The recommendation is to use a USB headset with Avaya one-X™ Desktop Edition. A list of recommended headsets is available on the Avaya Support Center (<a href="http://www.avaya.com/support">http://www.avaya.com/support</a>).</p> <p>Depending on how many sound devices are installed or connected to the PC (e.g., built-in sound device, USB headset), there may be one or more audio output (speaker) and audio input (microphone) devices available. Select a sound device from the pull-down list to be used for the audio output. Select a microphone from the pull-down list to be used for the audio input.</p> <p>Click the <b>Next</b> button to continue.</p> 

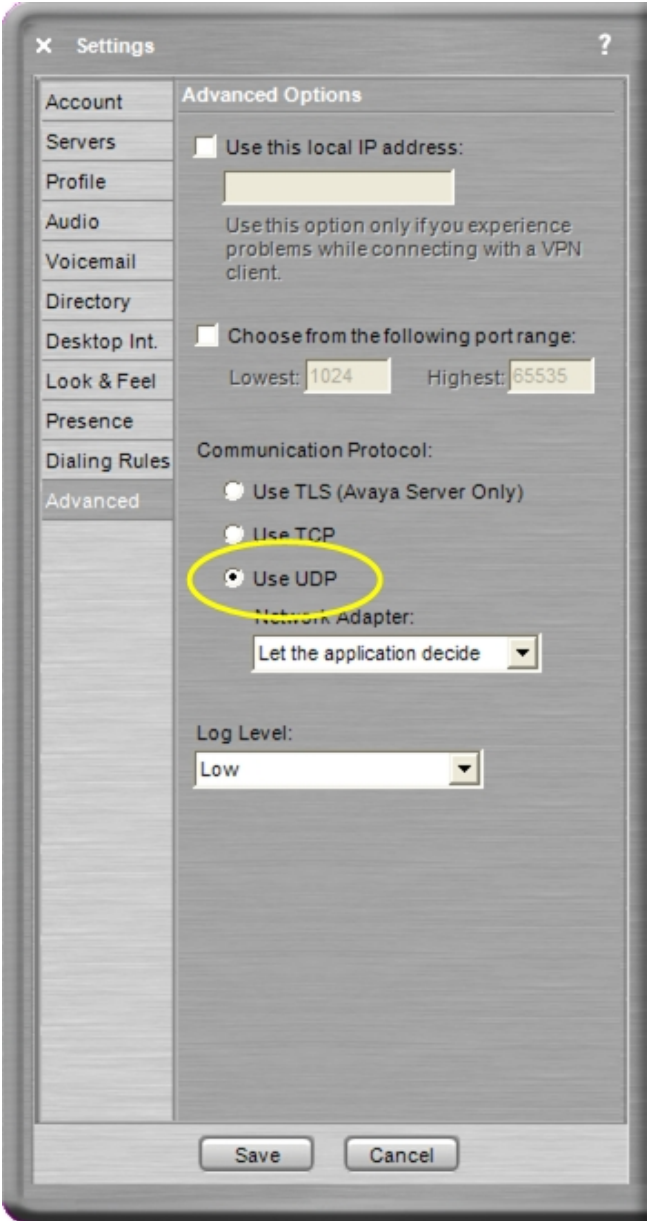
Steps	Description
10.	<p>The <i>Audio Wizard: Test Speaker</i> dialogue is displayed.</p> <p>Click the <b>Test</b> button and adjust the volume with the slider.</p> <p>The <b>Test</b> button changes to a <b>Stop</b> button. Click the <b>Stop</b> button once the proper volume for playback is achieved.</p> <p>Click the <b>Next</b> button to continue.</p> <div>  </div>
11.	<p>The <i>Audio Wizard: Tune Microphone</i> dialogue is displayed.</p> <p>Talk into the microphone and adjust the volume with the slider.</p> <p>Click the <b>Next</b> button to continue.</p> <div>  </div>

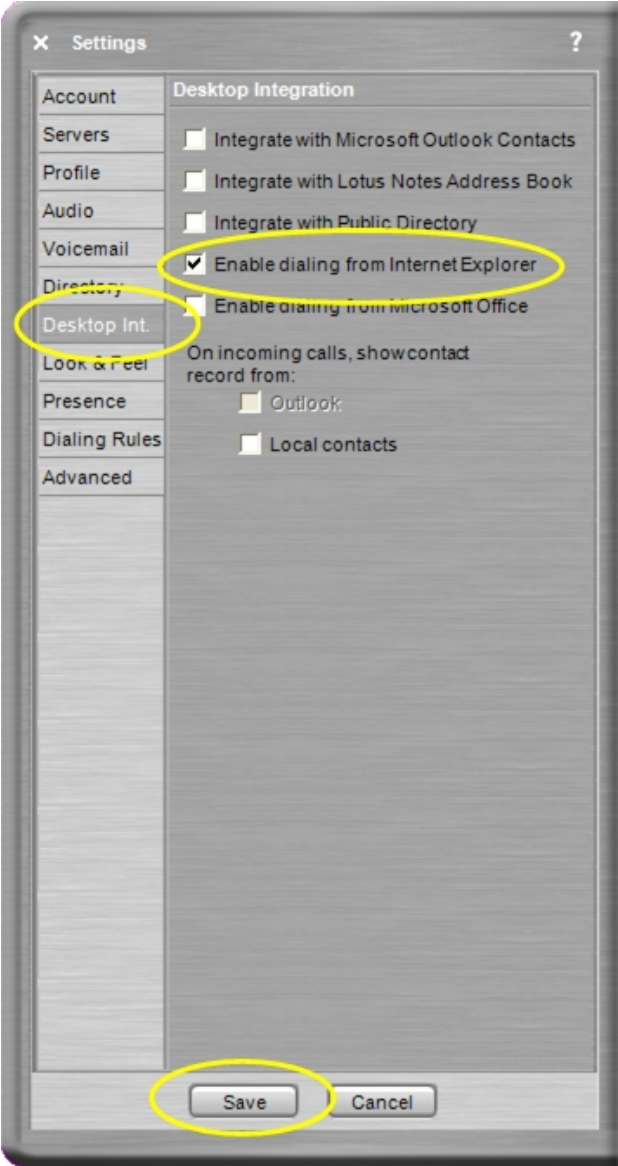
Steps	Description
12.	<p>The <i>Audio Wizard: Test Background Noise</i> dialogue is displayed.</p> <p>Click on the <b>Test</b> button. Do not talk during this test.</p> <p>Click the <b>Next</b> button to continue.</p> 
13.	<p>The <i>Congratulations</i> dialogue is displayed. The configuration is complete.</p> <p>Click the <b>Finish</b> button to continue.</p> 


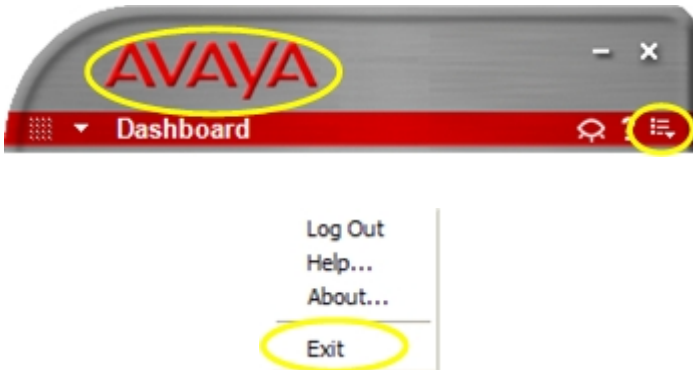
Steps	Description
14.	<p>The <i>Avaya one-X™ Desktop Edition</i> dialogue is displayed again, with its configuration information populated.</p> <p>Before Avaya one-X™ Desktop Edition can be used with the Asterisk server, it must be optioned to use UDP to communicate with the server.</p> <p>Click the <b>Settings</b> button.</p> 

Steps	Description
15.	<p>The <i>Settings: Account</i> dialogue is displayed.</p> <p>Click on <b>Advanced</b>.</p> 



Steps	Description
16.	<p>The <i>Settings: Advanced Options</i> dialogue is displayed.</p> <p>Select <b>Use UDP</b> for the <b>Communications Protocol</b>.</p> 

Steps	Description
17.	<p>Avaya one-X™ Desktop Edition can be configured to support Click to Dial using Microsoft Internet Explorer.</p> <p>Click <b>Desktop Int.</b> from the <i>Settings</i> dialogue.</p> <p>Enable the option <b>Enable dialing from Internet Explorer</b>.</p> <p>Click the <b>Save</b> button to continue.</p> 

Steps	Description
18.	<p>A warning message is displayed indicating that Avaya one-X™ Desktop Edition must be re-started before the changes can take place.</p> <p>Click the <b>OK</b> button.</p> 
19.	<p>Click on the Avaya logo on the <i>Avaya one-X™ Desktop Edition</i> dialogue and a pop-up menu is displayed. Alternatively, right-click on the far right of the Dashboard to bring up the pop-up menu.</p> <p>Click <b>Exit</b> to exit Avaya one-X™ Desktop Edition.</p> 

## 7. Verification Steps

All features shown in **Table 3** that have a “Yes” in the *Supported* column were tested. Two problems were found.

1. The MWI feature, as implemented by the Asterisk Business Edition PBX, is not compatible with the Avaya 4600 Series IP Telephones nor with Avaya one-X™ Desktop Edition. Thus, MWI does not function in this combined solution.
2. On completion of an attended transfer, the caller ID information for the transferred call is not sent along with the call.

The following steps can be used to verify and/or troubleshoot installations in the field.

- Avaya 4600 Series IP Telephones

1. Verify that the **SIPDOMAIN** parameter is configured in the **46xxsettings.txt** configuration file as this may affect ability to complete transfers.
2. If there are problems with receiving calls, verify that call forwarding has not been locally configured. This may impact receiving calls depending on which forwarding feature is configured. This configuration is done in the **46xxsettings.txt** configuration file (**CALLFWDADDR** and **COVERAGEADDR** parameters).
3. As the Avaya 4600 Series IP Telephones (running the SIP software) cannot dial “\*” properly, verify that the dial plan does not include dialing a “\*”. For example, the default number to dial for call pickup for an Asterisk server is “\*8”. This must be changed to a different number.

- Avaya one-X™ Desktop Edition

1. Verify that the Username field for Avaya one-X™ Desktop Edition is in the format of **userID@domain-name** instead of **userID@IP-address** (e.g., [john.doe@company.org](mailto:john.doe@company.org) instead of [john.doe@123.22.33.444](mailto:john.doe@123.22.33.444)). If an IP address is used, this may affect the ability to complete transfers.
2. As the Avaya one-X™ Desktop Edition cannot dial “\*” properly, verify that the dial plan does not include dialing a “\*”. For example, the default number to dial for call pickup for an Asterisk server is “\*8”. This must be changed to a different number.
3. Verify that Avaya one-X™ Desktop Edition has been configured for UDP (**Settings → Advanced → Communications Protocol**). Otherwise, it will not register with the server.
4. Verify that Avaya one-X™ Desktop Edition is able to connect to a WebLM license server and acquire a license. Otherwise, only basic telephony (make & receive calls) will work. From the local PC, open a Command Prompt window and ping the license server. From the license server, verify that there are available SIP Softphone licenses installed.

5. Verify that the Dialing Rules are properly configured (**Settings → Dialing Rules**). When improperly configured, external calls may be dialed incorrectly. From the same dialogue (**Settings → Dialing Rules**), enable the **Display confirmation window before dialing a number** to view the number that will be dialed before it is actually dialed.

- Asterisk Business Edition PBX

1. Verify that the users have been entered into the system properly using the “**sip show peers**” in the Asterisk CLI (see **Section 4.2**, step #5).
2. Verify the dial plan configured in the **sip.conf** file is configured properly to support attended transfers, call park, shuffling, and call forwarding. Refer to **Section 4.3** for example configurations. Also, view the dial plan using the “**show dialplan**” command in the Asterisk CLI.
3. If there are problems with receiving calls, verify that call forwarding has not been configured and enabled on the server. This may impact receiving calls depending on which forwarding feature is configured (see **Section 4.3**).
4. Verify that the Asterisk server is running if the telephones are unable to register. Use the instructions in see **Section 4.2** (step #5) to start the server.
5. If there are problems dialing “#” to interact with external voice mail systems, external IVR systems, or any other system that used “#”, verify whether the **Dial()** command has been configured to allow “#” to initiate an internal unattended transfer (see **Section 4.3**, step #1).

## 8. Conclusion

These Application Notes have described the administration steps required to use Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition with the Asterisk Business Edition PBX. Both standard and supplementary features were covered for all of the Avaya 4600 Series IP Telephones and Avaya one-X™ Desktop Edition and presence tracking was covered for Avaya one-X™ Desktop Edition.

## 9. Additional References

The following are additional references.

- [1] *4600 Series IP Telephone R2.3 LAN Administrator Guide*, Issue 2.3, Doc ID 555-233-507, April 2006, available at <http://www.avaya.com/support>.
- [2] *Avaya one-X™ Desktop Edition R2.1 Getting Started Guide*, February 2006. *Avaya one-X™ Desktop Edition Overview*, April 2006, available at <http://www.avaya.com/support>.
- [3] *Avaya one-X™ Desktop Edition Administration*, April 2006, available at <http://www.avaya.com/support>.
- [4] *Asterisk Business Edition Technical Reference*, Version A. Available at <http://www.digium.com/en/products/software/abe.php>.
- [5] *Asterisk Business Edition QuickStart Guide*. Available at <http://www.digium.com/en/products/software/abe.php>.
- [6] Asterisk WiKi page on voip-info.org. <http://www.voip-info.org/wiki-Asterisk>.
- [7] *Asterisk cmd Dial*. Available at <http://www.voip-info.org/wiki/index.php?page=Asterisk+cmd+Dial>.
- [8] *Asterisk SIP media path*. Available at <http://www.voip-info.org/wiki/view/Asterisk+SIP+media+path>.
- [9] *Asterisk call forwarding*. Available at <http://www.voip-info.org/wiki/view/Asterisk+call+forwarding>.
- [10] *Session Initiation Protocol Service Examples - draft-ietf-sipping-service-examples-10*, SIPPING Working Group, Internet-Draft, March 5, 2006 (expires September 6, 2006), available at <http://tools.ietf.org/wg/sipping/draft-ietf-sipping-service-examples/draft-ietf-sipping-service-examples-10.txt>.
- [11] *Session Initiation Protocol (SIP)-Specific Event Notification*. RFC 3265. Available at <http://www.ietf.org/rfc/rfc3265.txt>.
- [12] *A Message Summary and Message Waiting Indication Event Package for the Session Initiation Protocol (SIP)*. RFC 3842. Available at <http://www.ietf.org/rfc/rfc3842.txt>.
- [13] *RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals*. RFC 2833. Available at <http://www.ietf.org/rfc/rfc2833.txt>.
- [14] *A Presence Event Package for the Session Initiation Protocol (SIP)*. RFC 3856. Available at <http://www.ietf.org/rfc/rfc3856.txt>.
- [15] *Instant Messaging / Presence Protocol Requirements*. RFC 2779. Available at <http://www.ietf.org/rfc/rfc2779.txt>.
- [16] *SIP for Instant Messaging and Presence Leveraging Extensions (simple)*. Available at <http://www.ietf.org/html.charters/simple-charter.html>.

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