



Avaya Solution & Interoperability Test Lab

Application Notes for VTech 1-Line SIP Hotel and Lobby Phones with Avaya IP Office 8.1 and Voicemail Pro 8.1 – Issue 1.1

Abstract

These Application Notes describe a compliance-tested configuration consisting of Avaya IP Office 8.1, Voicemail Pro 8.1 and VTech 1-Line SIP Hotel and Lobby Phones.

VTech's hospitality product line provides a clear cost and feature advantage that is backed by decades of expertise in the corded/cordless telephony industry. These SIP endpoints register directly with Avaya IP Office 8.1.

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 a compliance-tested configuration consisting of Avaya IP Office, Voicemail Pro and VTech 1-Line SIP Hotel and Lobby Phones.

The compliance testing covers the following VTech 1-Line SIP phones:

- 1-Line S1100 (Classic Corded Lobby Telephone)
- 1-Line S1210 (Classic Corded Hotel Telephone)
- 1-Line S1410 (Classic Cordless Hotel Telephone)
- 1-Line S2100 (Contemporary Corded Lobby Telephone)
- **1-Line S2210 (Contemporary Corded Hotel Telephone)**
- 1-Line S2410 (Contemporary Cordless Hotel Telephone)
- 1-Line S2211 (Contemporary SIP Petite Telephone)

The compliance testing was performed with the VTech **S2210** SIP phone. All of the models listed above share core hardware and SIP firmware. The primary differences with these phones are either cosmetic, or corded versus cordless handsets. These variations do not impact the interoperability between the base station and the Avaya infrastructure, so use of any of these models can be expected to yield the same results as those observed in the testing described in these Application Notes.

2. General Test Approach and Test Results

The compliance test focused on the interoperability between the VTech 1-Line SIP Hotel Phone, Avaya IP Office and Voicemail Pro including the ability to make and receive calls from PSTN endpoints and Avaya SIP, H.323 and Digital phones.

As the purpose of these phones is for hotel guest rooms and hotel lobbies, certain functionality considered to be standard on Avaya endpoints is not supported and therefore was not tested. For example, the VTech phones do not support transfer and conference capabilities. More details on these limitations are described in **Sections 2.1 and 2.2**.

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

Testing consisted of typical call scenarios involving external endpoints using a simulated PSTN, and various Avaya endpoints aimed at simulating a typical hotel in which the staff uses full featured Avaya phones and guests use VTech SIP phones.

The feature testing included registration, basic calls, mute/un-mute, hold/reconnect, drop, media shuffling, G.711, G.729, G.722, codec negotiation, music on hold, DTMF, speed dial, redial, attended transfer, attended conference, call forwarding, park/unpark, MWI, and voicemail with Voicemail Pro. Hospitality features Do Not Disturb and Alarm Set with Voicemail Pro were also covered. The VTech 1-Line SIP phones are not able to initiate attended transfer and attended conference but were tested as members of these scenarios.

Serviceability testing was also performed to verify the ability for the phones to recover from loss of network connections.

The following tests were not covered because the VTech 1-Line SIP phones do not support these functions:

- Display
- Blind transfer
- Attended transfer
- Blind conference
- Attended transfer
- Hunt group member
- Call Forward Key
- Phonebook

2.2. Test Results

The objectives described in **Section 2.1** were verified. For serviceability testing, the VTech phones were able to re-register with IP Office following loss of network connections, and server reboots.

The following observations were made during the testing:

- On the VTech 1-Line SIP phone's web interface there are settings to perform call forwarding at the phone level. This function is not supported by IP Office. The workaround for this is to configure call forwarding on IP Office by using short codes.
- Default short codes in IP Office that use the “#” character at the end will not work with the VTech 1-Line SIP phone. When “#” is pressed the phone interprets this as the end of a dialed number and the “#” is not sent to IP Office. The workaround is to edit the Shortcodes in IP Office and replace the “#” with “*”.

2.3. Support

Information, documentation and technical support for VTech Hotel Phones can be obtained at:

- Phone: 1 (888) 714-7385
- <http://vtechhotelphones.com>

3. Reference Configuration

The configuration used for the compliance testing is shown below.

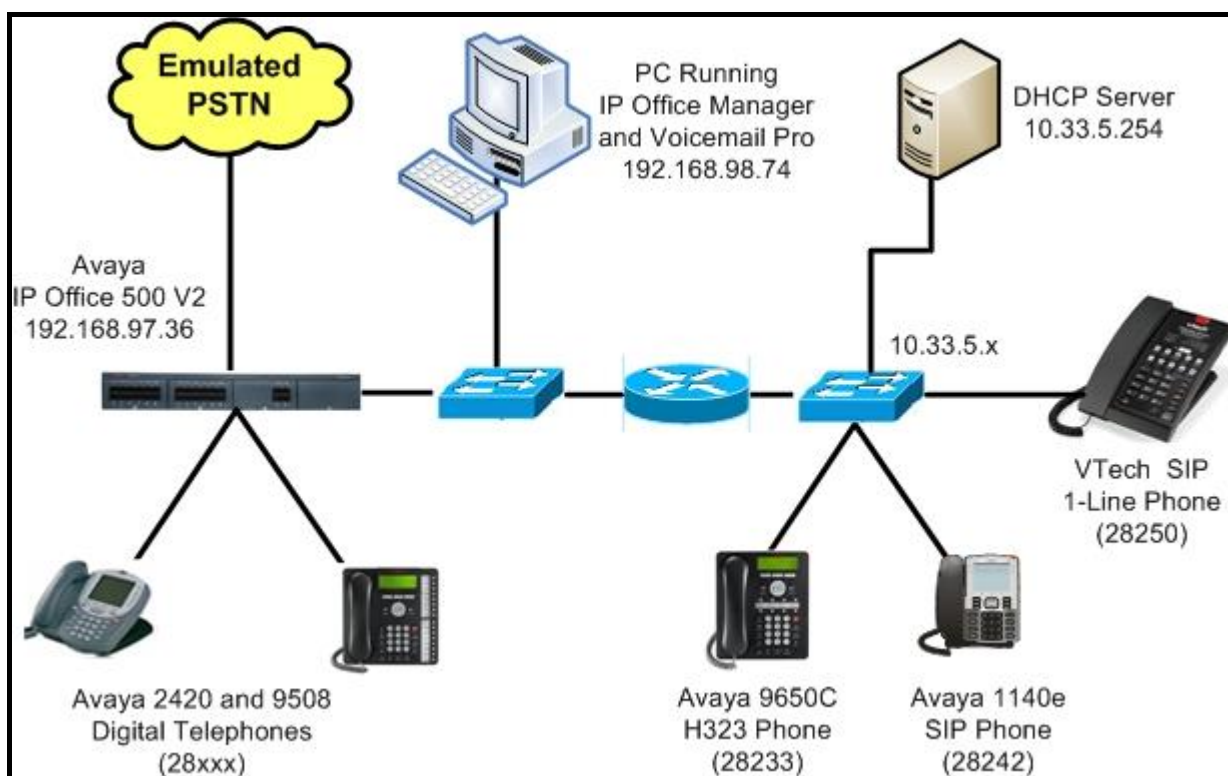


Figure 1 – VTech 1-Line SIP Hotel Phone Test Configuration

4. Equipment and Software Validated

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

Equipment	Version
Avaya IP Office 500 V2	8.1(43)
Voicemail Pro	8.1.810.0
Avaya 9650C H.323 Phone	Avaya one-X [®] Deskphone S3.104S
Avaya 1608-I H.323 Phone	I.302S
Avaya 1140E SIP Phone	04.03.12.00
Avaya 9508 Digital Phone	N/A
VTech 1-Line SIP Hotel Phone Model S2210	SIP_27.3.62.00 Aug 10 2012 21:15:23
VTech 1-Line SIP Hotel Phone Model S2210	SIP 30.3.63.05 Oct 5 2012 10:33:05 (for Long Hold testing)

Testing was performed with IP Office 500 R8.1, but it also applies to IP Office Server Edition R8.1. Note that IP Office Server Edition requires an Expansion IP Office 500 v2 R8.1 to support analog or digital endpoints or trunks.

5. Configure Avaya IP Office

This section describes the steps to configure IP Office to interoperate with 1-Line VTech SIP phones. It is assumed that IP Office has already been installed and is functioning. For additional information on IP Office installation and configuration refer to **Section 10**.

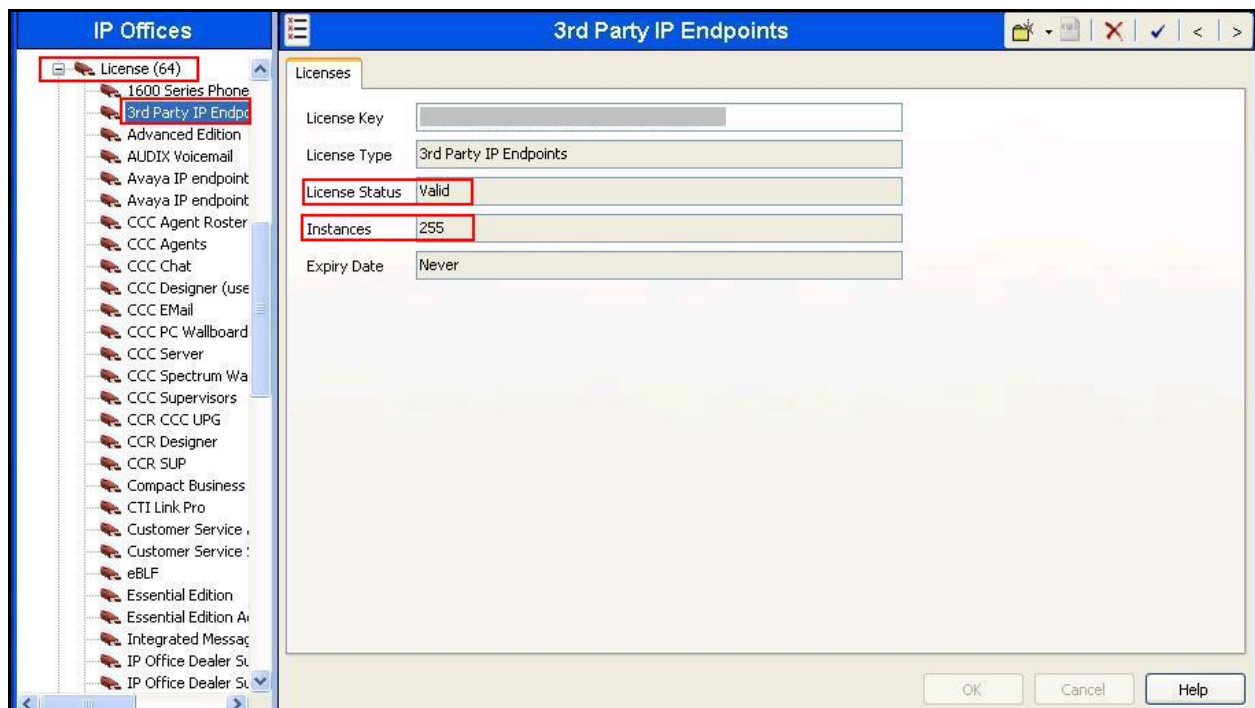
Summary of IP Office Configuration to add a SIP Endpoint:

- Verify 3rd Party IP Endpoints Licenses
- Configuring System Values
- Configuring a SIP Extension
- Configuring a SIP User
- Configuring of Short Code for Alarm Set

5.1. Verify 3rd Party IP Endpoints Licenses

This section explains the steps to verify if the license status for 3rd party IP endpoints is valid. Open the IP OFFICE Manager by navigating to **Start > Programs > IP Office > Manager** on the server IP OFFICE Manager is installed on (not shown).

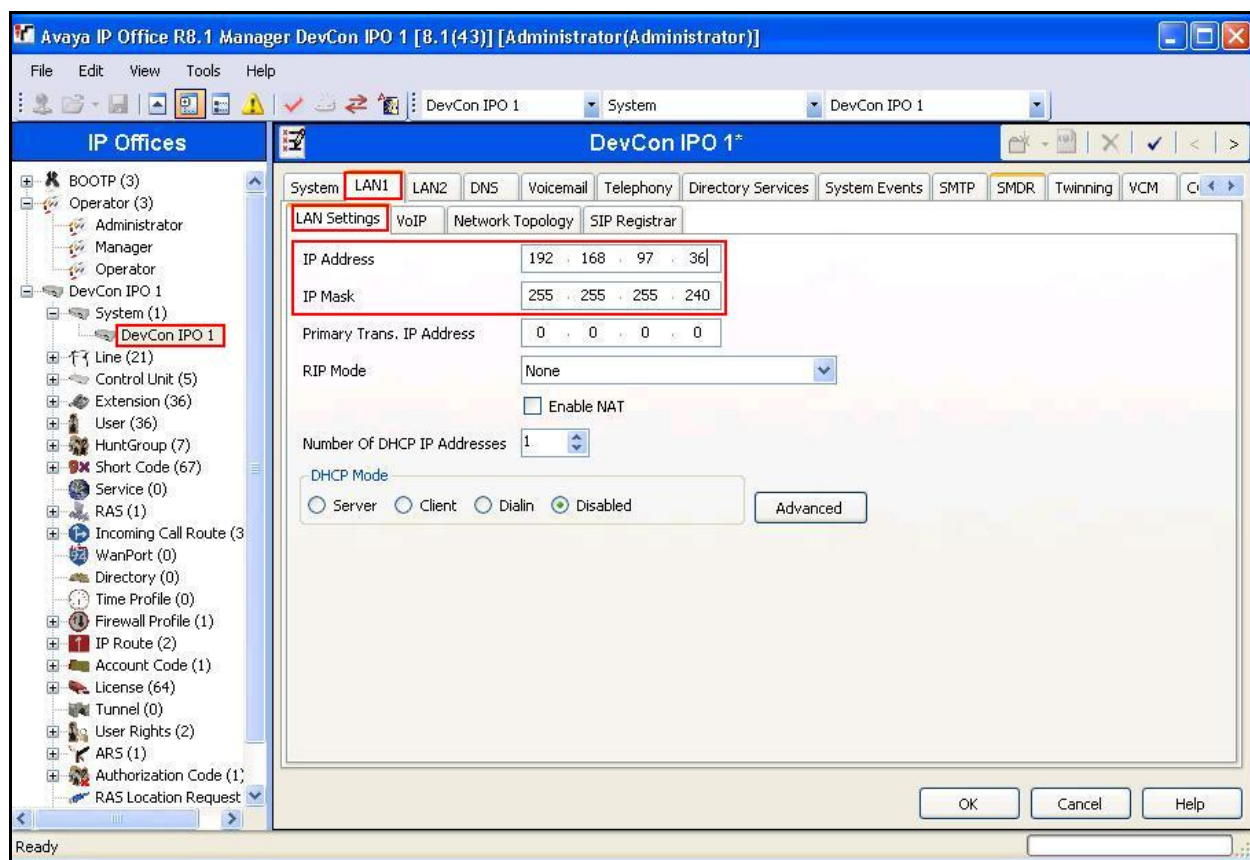
In the left panel navigate to **License** and then select **3rd party IP Endpoints**. Under the **Licenses** tab verify that the **License Status** is **Valid** and that the number of **Instances** will cover the number of phone to be added. Note the **License Key** has been masked out.



5.2. Verify LAN Settings

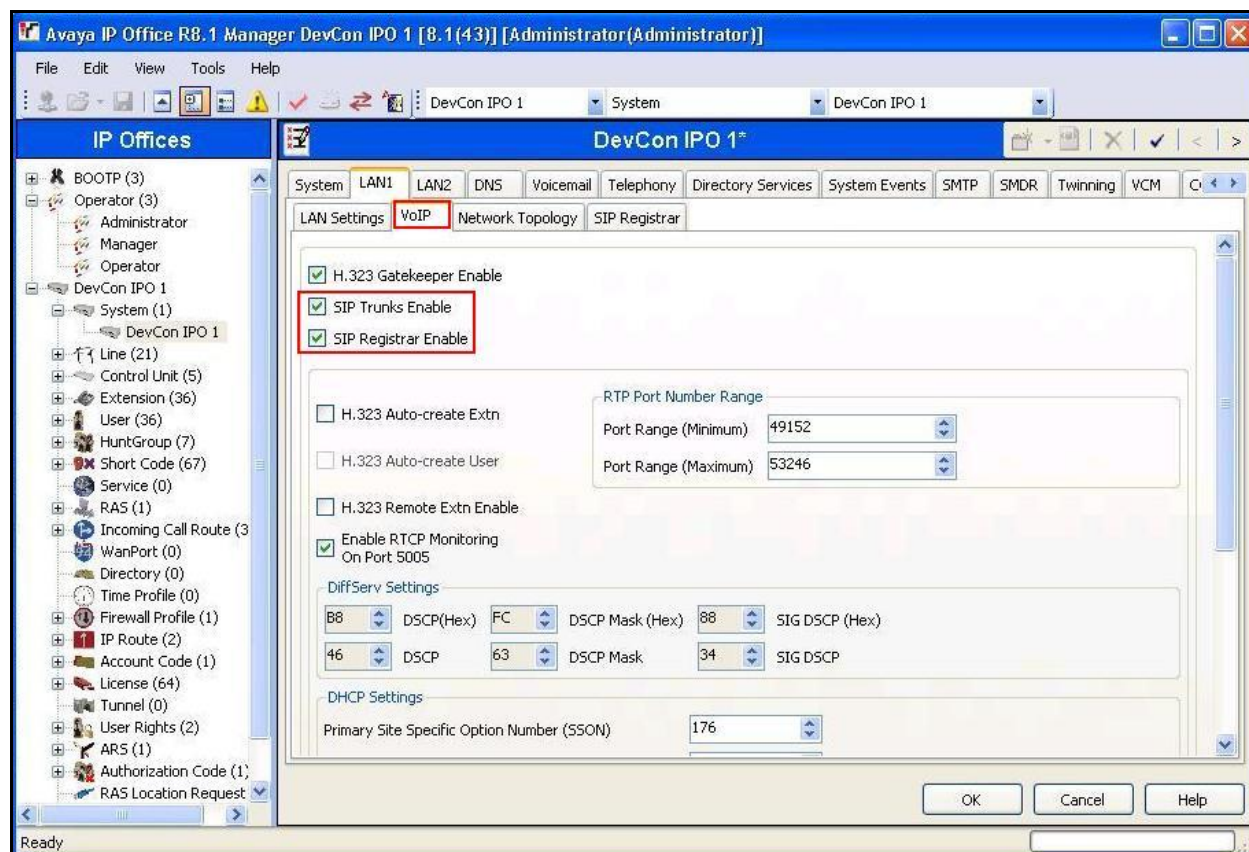
This section explains the steps to obtain the system values for **LAN1**. This information will be required in **Section 7.4** when configuring the VTech SIP phone. IP Office has two LAN interfaces; however during compliance testing only **LAN1** was used.

Navigate to the **System** in the left window as shown below. Click on the **LAN1** tab and then on the **LAN Settings** sub tab. Make note of the **IP Address** and **IP Mask**. In this example it is *192.168.97.36* and *255.255.255.240*.



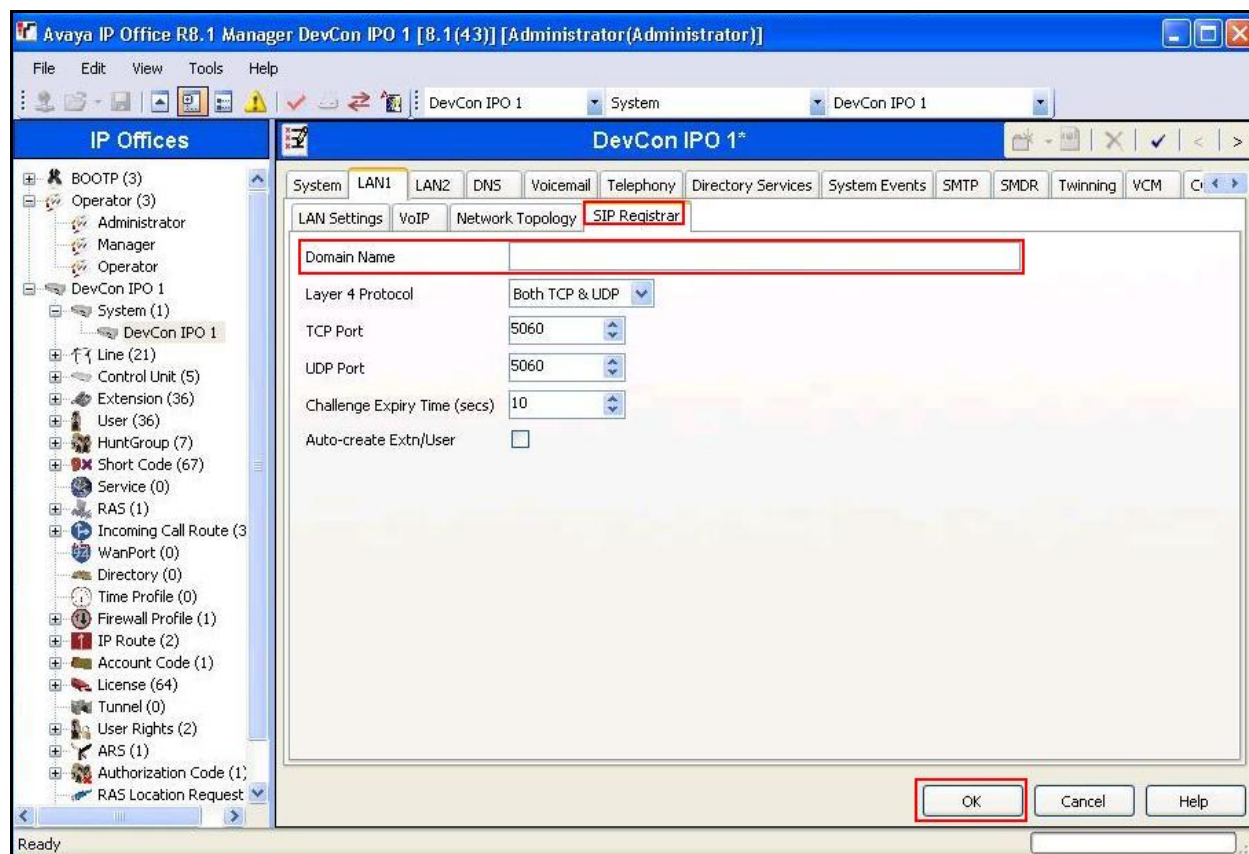
5.3. VoIP Configuration

Navigate to the **VoIP** sub tab as shown below. **SIP Trunks Enable** and **SIP Registrar Enable** boxes need to be checked. The rest of the values are left at default.



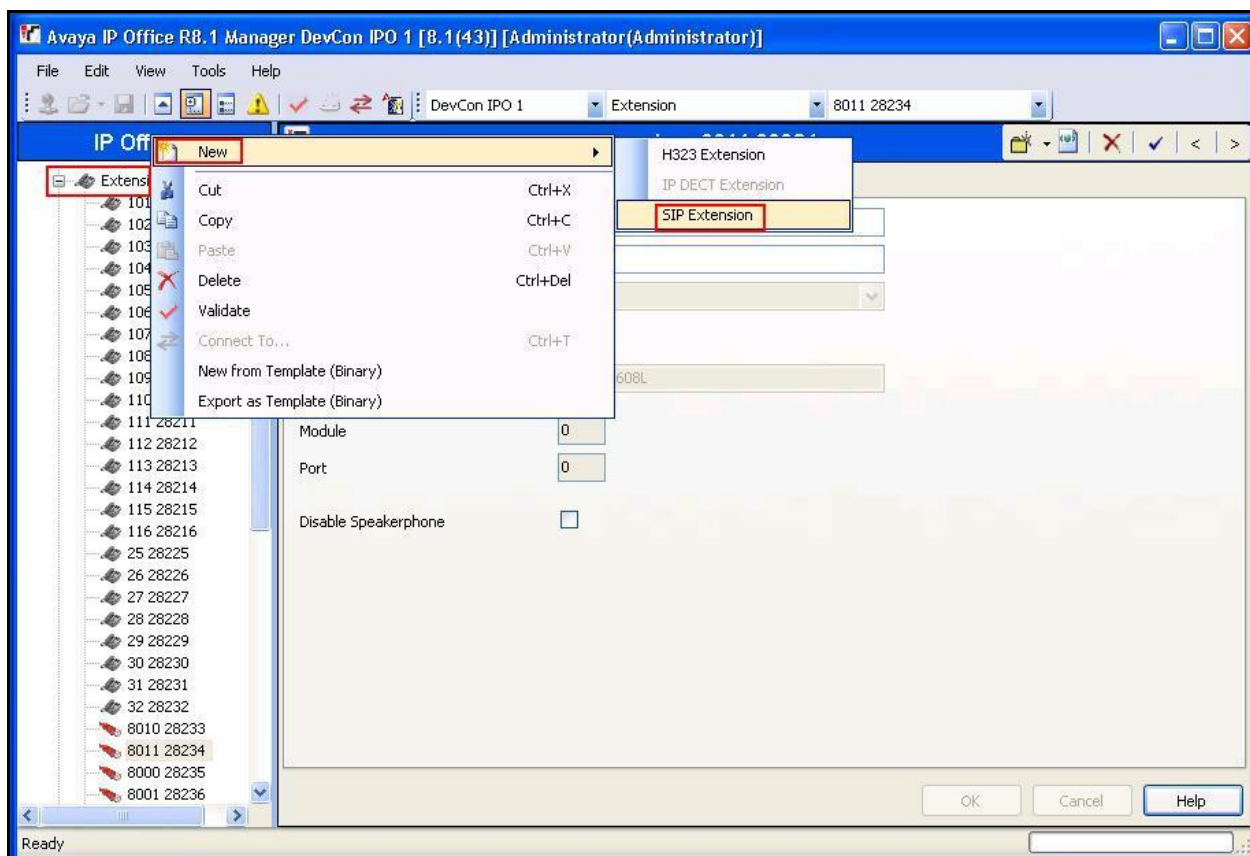
5.4. SIP Registrar Configuration

Navigate to the **SIP Registrar** sub tab as shown below. A valid **Domain Name** can be entered here for the SIP endpoints to use for registration with IP Office. During compliance testing this field was left blank and the SIP endpoints used the **LAN Settings IP Address** for registration. The rest of the values are left at default. Click on **OK** to complete the configuration.



5.5. Configuring a SIP Extension and User

This section explains the steps to add a SIP Extension and assign a user to that extension. As shown below right click on **Extension** and navigate to **New > SIP Extension**.



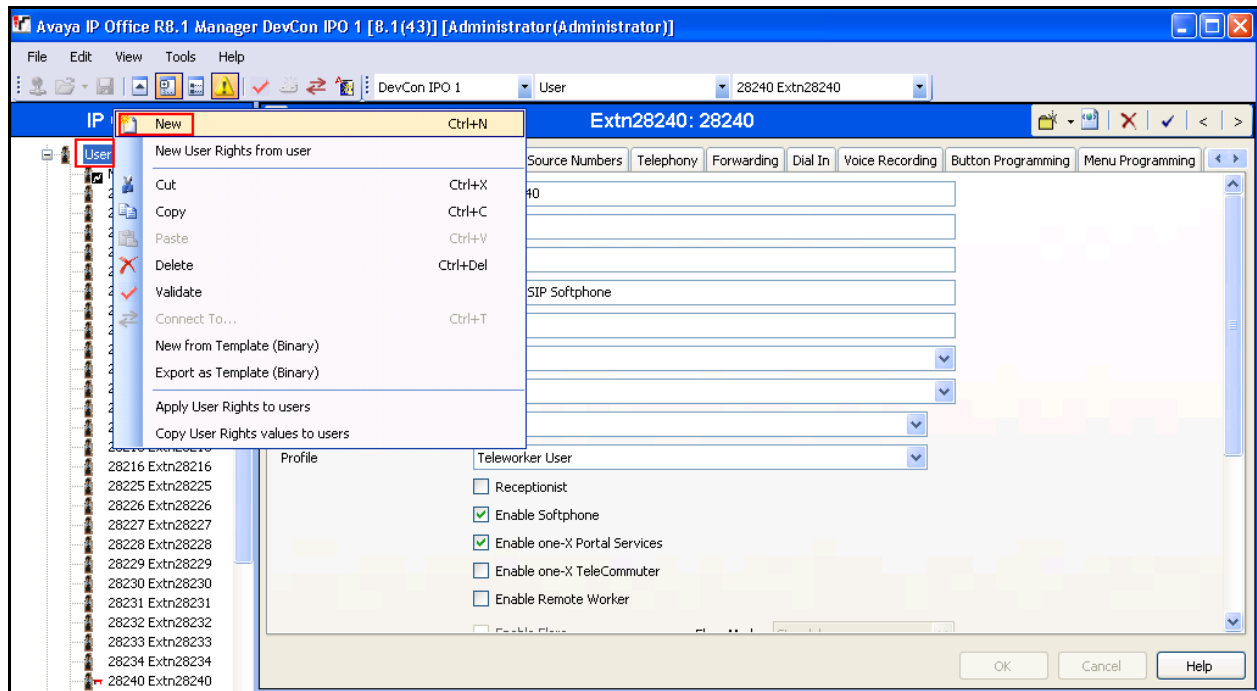
After selecting **New > SIP Extension** on the previous page, the following is displayed. The value seen in the **Extension ID** field is automatically generated by IP Office. Enter the **Base Extension** value on the **Extn** tab. During compliance testing 28250 was configured as the **Base Extension**.

The screenshot shows the 'SIP Extension: 8014 28250' configuration window in the 'Extn' tab. The 'Extension Id' field contains '8014' and the 'Base Extension' field contains '28250'. Both fields are highlighted with red rectangles. Other fields include 'Caller Display Type' set to 'On', 'Reset Volume After Calls' as an unchecked checkbox, 'Device Type' as 'Unknown SIP device', 'Module' and 'Port' both set to '0', and 'Force Authorization' as a checked checkbox. The left sidebar shows a tree view of the system configuration with 'Extension (43)' selected.

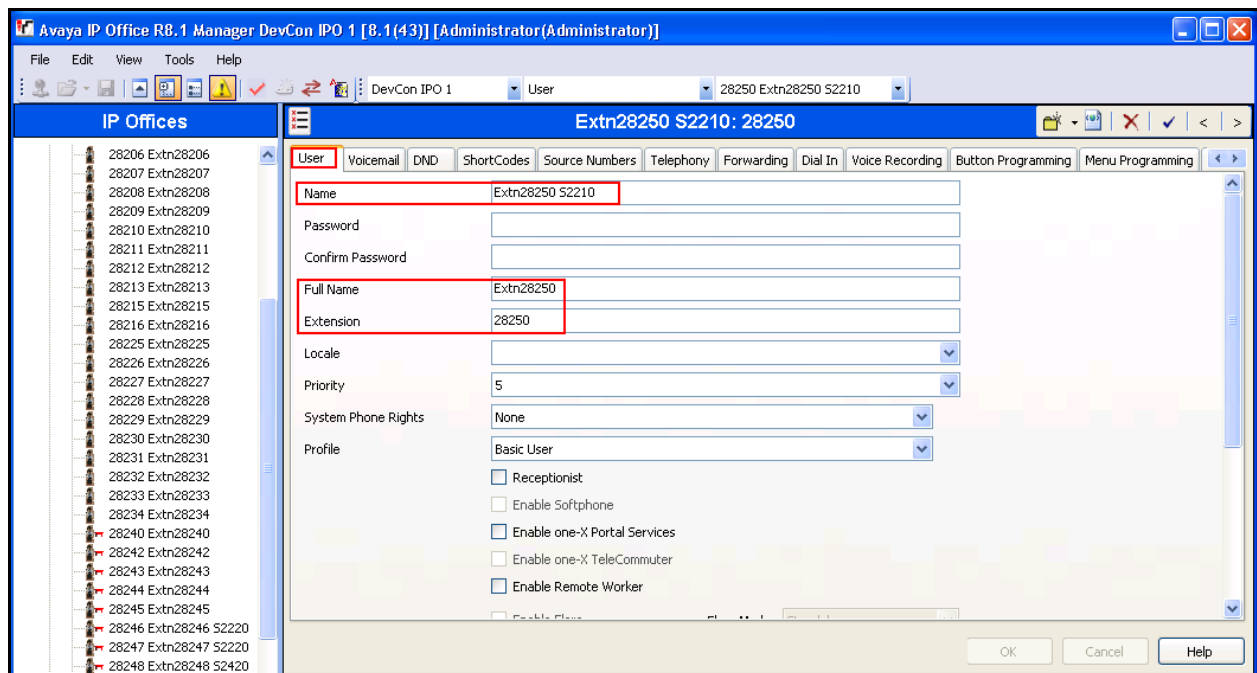
Values in the **VoIP** tab are left at default as shown below. Click on **OK** to complete the SIP Extension configuration.

The screenshot shows the 'SIP Extension: 8014 28250' configuration window in the 'VoIP' tab. The 'IP Address' field is '0 . 0 . 0 . 0'. 'Codec Selection' is 'System Default'. A list of codecs is shown: G.711 ULAW 64K, G.729(a) 8K CS-ACELP, G.722 64K, G.711 ALAW 64K, and G.723.1 6K3 MP-MLQ. On the right, several checkboxes are visible: 'VoIP Silence Suppression' (unchecked), 'Local Hold Music' (unchecked), 'Allow Direct Media Path' (checked), 'Re-invite Supported' (checked), 'Use Offerer's Preferred Codec' (unchecked), 'Reserve Avaya IP endpoint license' (unchecked), and 'Reserve 3rd party IP endpoint license' (unchecked). At the bottom, 'Fax Transport Support' is 'None', 'TDM->IP Gain' is 'Default', 'IP->TDM Gain' is 'Default', and 'DTMF Support' is 'RFC2833'. The 'OK' button is highlighted with a red rectangle.

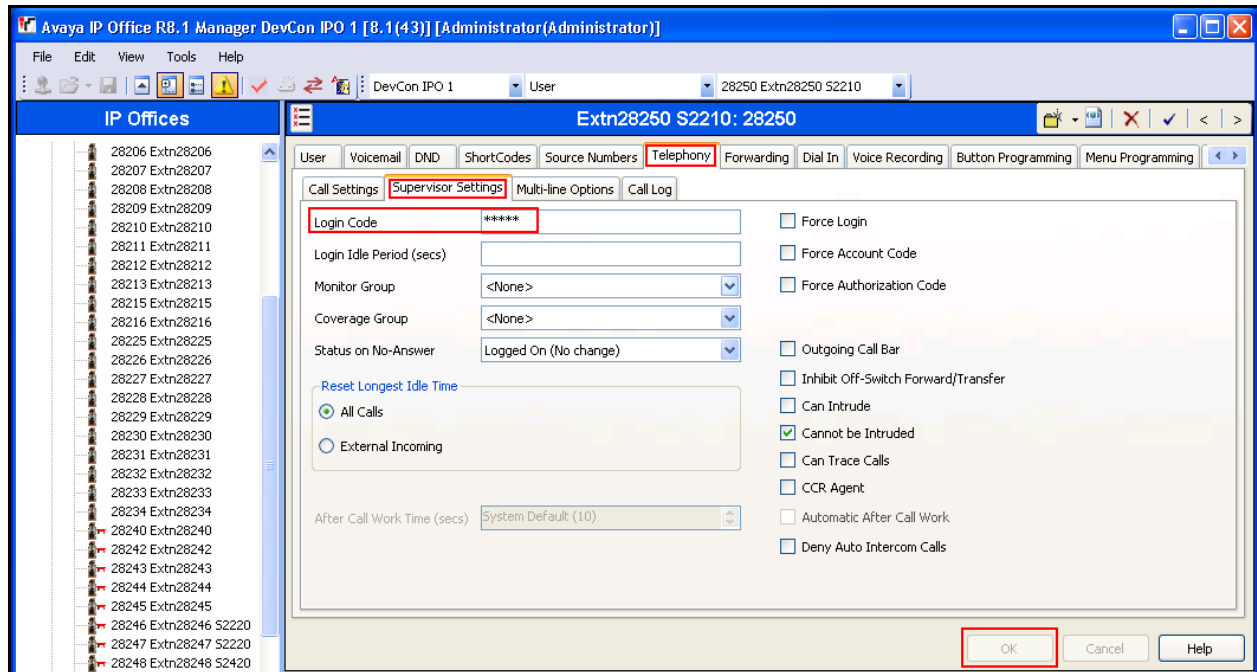
To assign a User to the above created Extension, right click on **User** seen on the left hand window pane of the IPOffice Manager as shown below and select **New**.



In the **User** tab, populate the **Name**, **Full Name** and **Extension** fields as shown below. The value of 28250 in the **Extension** field is based on the previously created extension.



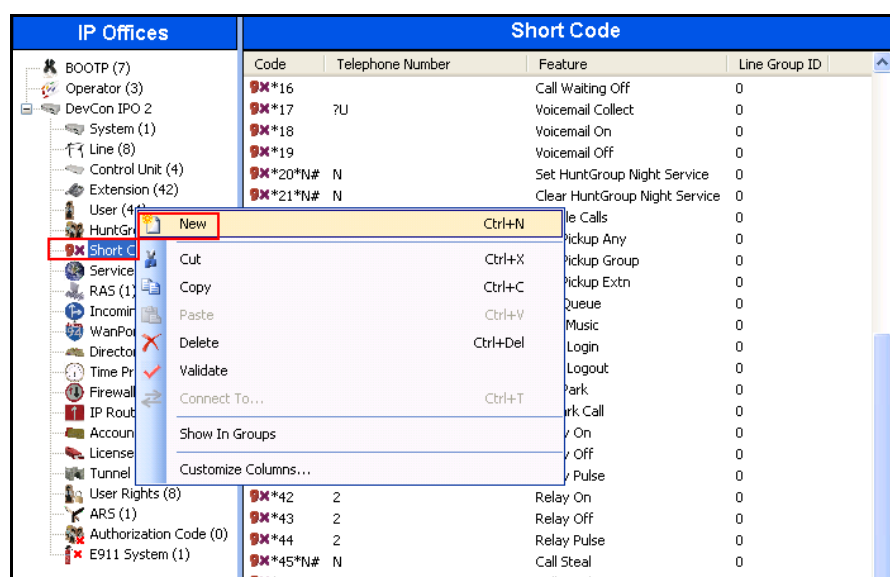
For SIP endpoints the password that the phone will register with is configured on the **Supervisor Settings** tab. Navigate to the **Telephony** tab and then select the **Supervisor Settings** sub tab. In the **Login Code** field enter a password that will be used later in the VTech phone configuration. Click on **OK** to complete the configuration.



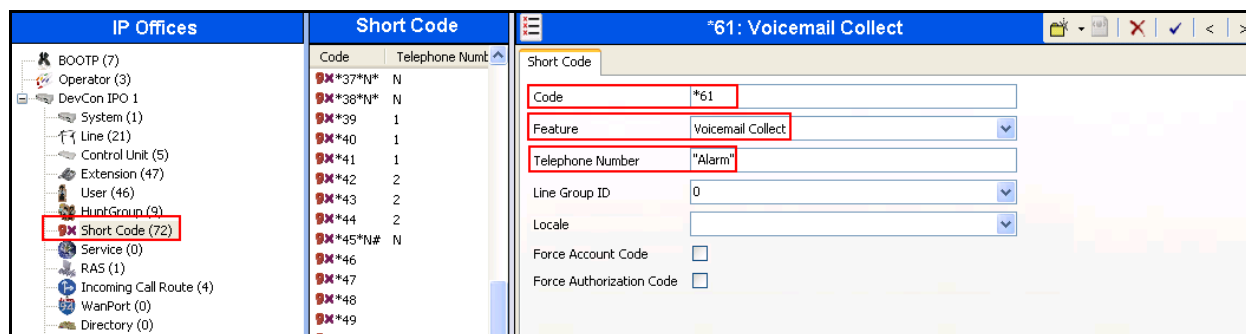
5.6. Configuring of Short Code for Alarm Set

This section describes the steps required to configure a new Short Code that can be used to access the Alarm Set feature of Voicemail Pro.

To configure a new Short Code, right click on **Short Code** as seen in the left hand window pane of IP Office Manager and select **New** as shown below.



In the right hand window pane enter a unique **Code** that will be used to access Alarm Set in Voicemail Pro. In this example configuration ***61** was used for the **Code**. For the **Feature** field select **Voicemail Collect** from the drop down menu. Now enter a unique name in double quotes in the **Telephone Number** field. In the example **"Alarm"** was used. This name needs to match the name configured in Voicemail Pro Client in **Section 6**. When finished click on **OK** (not shown).



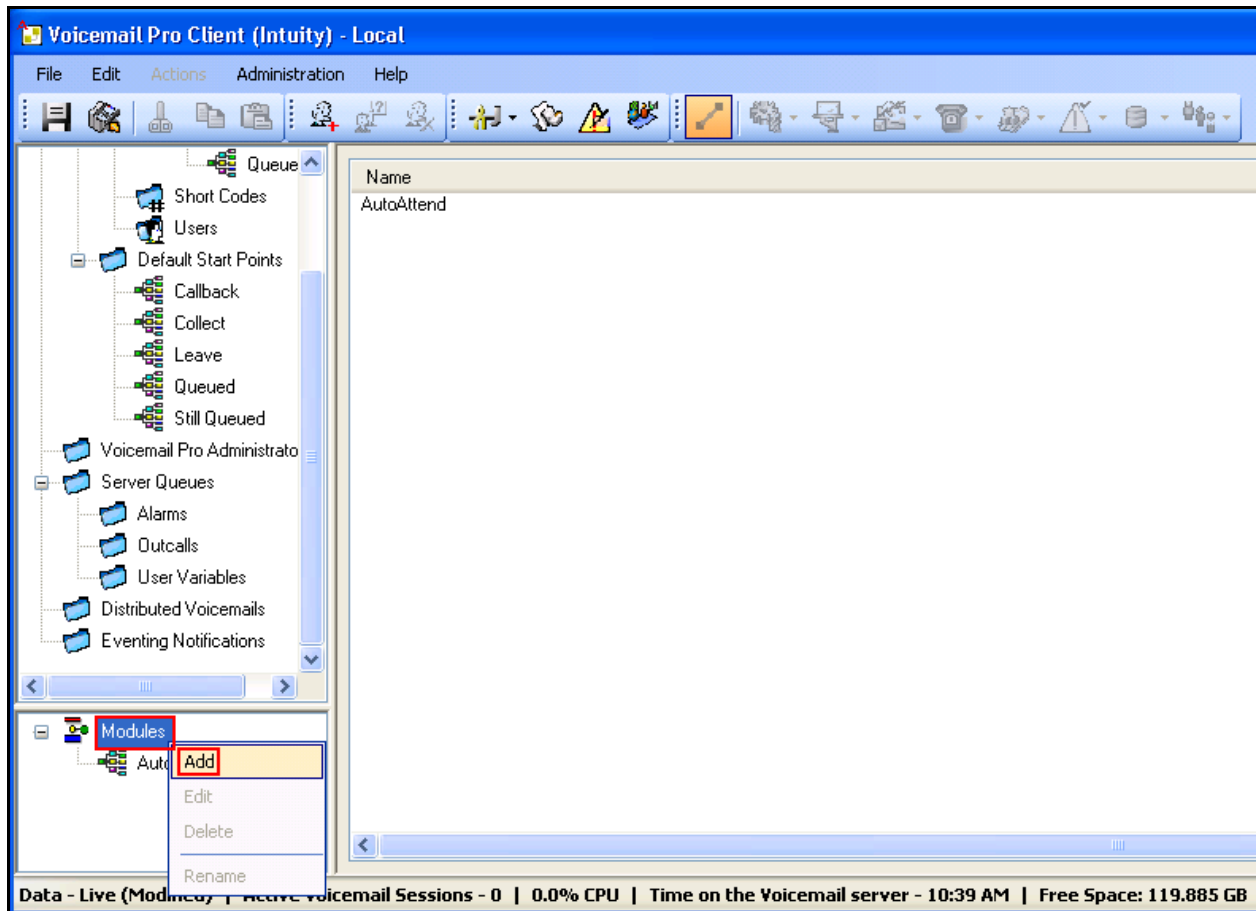
Now perform a save of the IP Office configuration (not shown).

6. Configure Voicemail Pro for Alarm Set

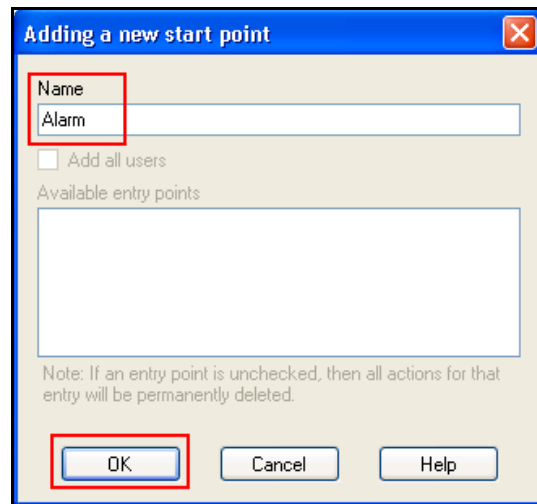
This section describes the steps required to configure Voicemail Pro for the Alarm Set feature.

Open the Voicemail Pro Client by navigating to **Start > Programs > IP Office > Voicemail Pro Client** on the server Voicemail Pro is installed on (not shown).

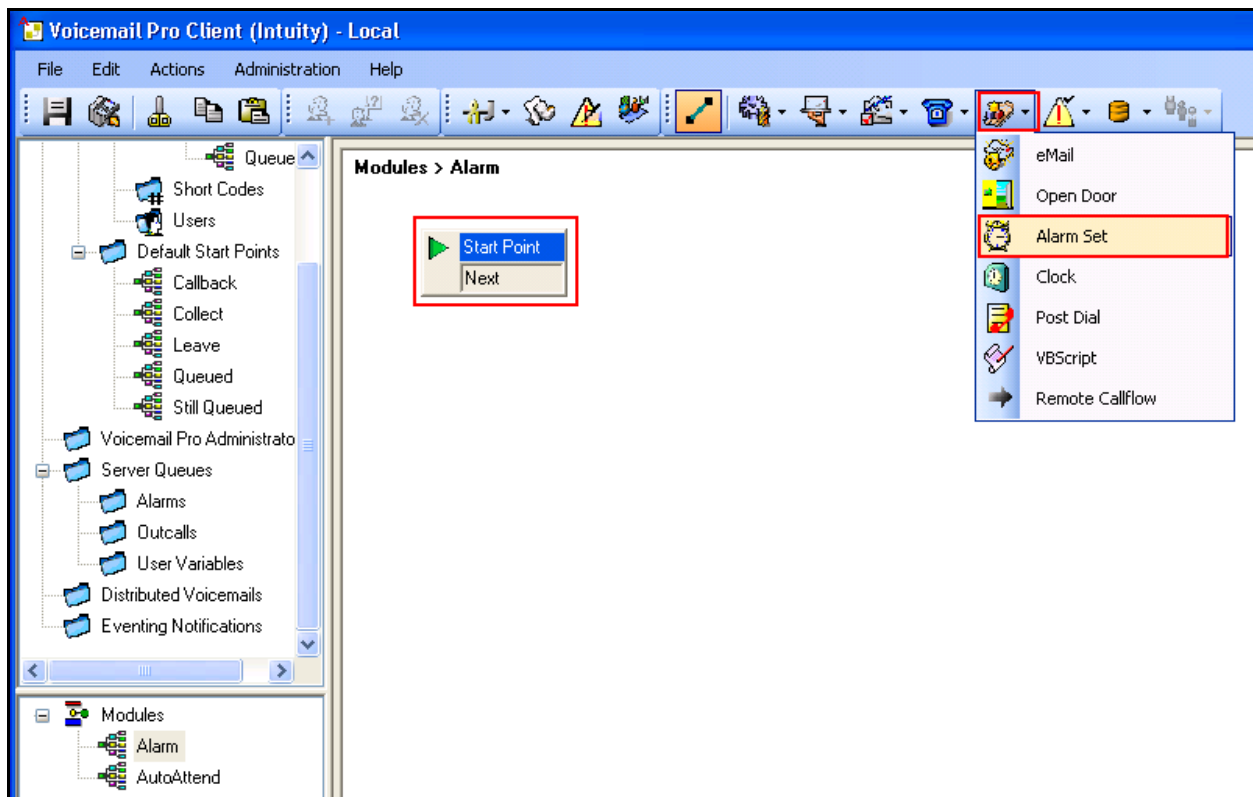
In the Voicemail Pro Client right click on **Modules** and select **Add**.



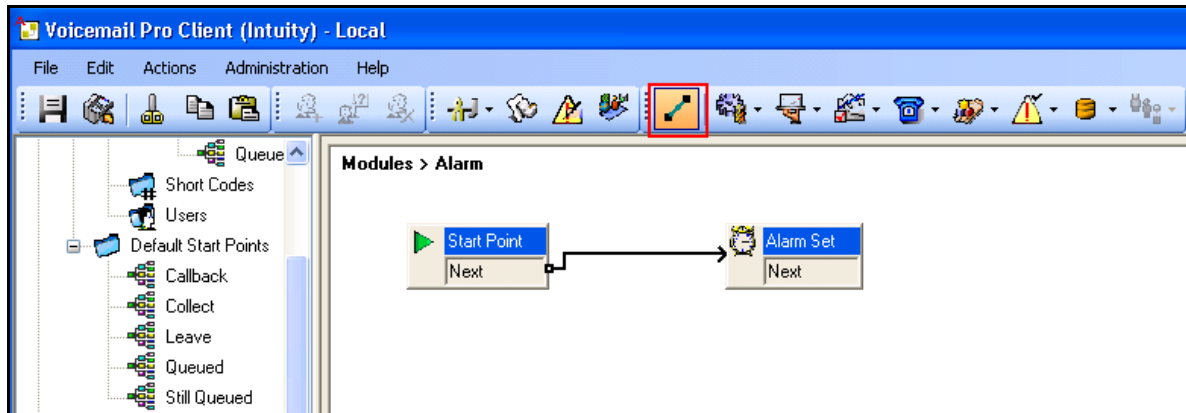
In the **Name** field of the new window that appears, enter the name that was entered in the **Telephone Number** field in **Section 5.6**. Note that the double quotes are not required here. In this example configuration **Alarm** was used for the name.



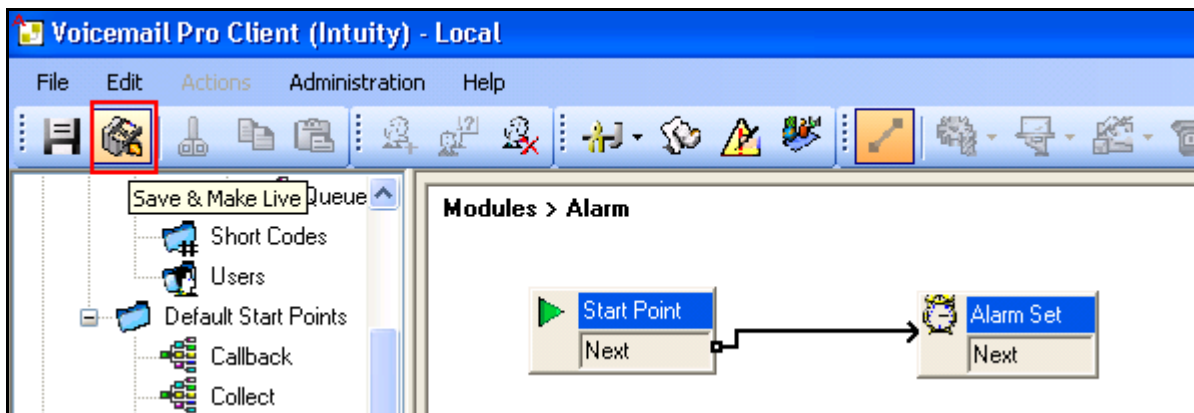
Next click on the **Start Point** object to enable the icons in the toolbar, then select the **Miscellaneous Actions** toolbar icon and select **Alarm Set**. Next, click in the **Modules > Alarm** window to add the Alarm Set object (not shown).



Now use the **Connection** tool to connect the **Start Point** object to the **Alarm Set** object as shown below.



When finished, click on the **Save & Make Live** icon.



Now when the Short Code as configured in **Section 5.6** is entered on a phone the user will be prompted to enter an alarm time using dialed digits on the phone. The Short code can also be configured as a Speed Dial Key as in **Section 7.7**

7. Configure VTech Phones

VTech SIP Hotel Phones are configured using a web browser. The phones use DHCP by default and are powered by Power over Ethernet (PoE). In the tested configuration, the phones were connected to the LAN via an Avaya Ethernet Routing Switch 5520-48T-PWR on a segment with a DHCP server.

To configure the VTech SIP phone, open a web browser and enter the URL of the phone http://<ip_address>. When prompted, login using 'root' for the user account, and the appropriate password (not shown). The initial screen is as follows; all navigation is via the navigation tree in the left panel. See the VTech documentation for more details.

vtech VTech SIP Phone Web Portal Basic Phone Information Hotel Information System Configuration Network Configuration Network Security Static IP Mapping Phone Configuration SIP Account Settings Advanced SIP Settings Audio Codec Advanced Call Features Ring Tone Speed Dial Other Phone Settings System Resources Config Update/ Backup Firmware Upgrade Reboot Phone Factory Default Inter-Op Configuration Server Options Admin Config Password	Basic Phone Information <table><tr><td>Model Number</td><td>S2210</td></tr><tr><td>MAC Address</td><td>00:12:2a:19:7d:db</td></tr><tr><td>Hardware Version</td><td>7800008_US</td></tr><tr><td>Boot Version</td><td>VTechBoot1.02.00</td></tr><tr><td>Firmware Version</td><td>SIP_30.3.63.05</td></tr><tr><td>Release Date</td><td>Oct 5 2012 - 10:33:05</td></tr><tr><td>Audio Profile Version</td><td>S2100 S2210 S2220 0007</td></tr></table>	Model Number	S2210	MAC Address	00:12:2a:19:7d:db	Hardware Version	7800008_US	Boot Version	VTechBoot1.02.00	Firmware Version	SIP_30.3.63.05	Release Date	Oct 5 2012 - 10:33:05	Audio Profile Version	S2100 S2210 S2220 0007
Model Number	S2210														
MAC Address	00:12:2a:19:7d:db														
Hardware Version	7800008_US														
Boot Version	VTechBoot1.02.00														
Firmware Version	SIP_30.3.63.05														
Release Date	Oct 5 2012 - 10:33:05														
Audio Profile Version	S2100 S2210 S2220 0007														

7.1. Confirm Network Configuration

By default, the phones are set to use DHCP. If necessary, deselect the **DHCP** box and provide necessary settings as required. The **TFTP Server Address** option was left at factory default, this is used for pushing configuration files or updated firmware files to the phones but was not tested. If changes are made click on **Save** before navigating from the screen.

The screenshot shows the VTech SIP Phone Web Portal interface. On the left is a blue sidebar with the VTech logo and a menu. The menu includes 'VTECH SIP Phone Web Portal', 'Basic Phone Information', 'Hotel Information', 'System Configuration' (with 'Network Configuration' highlighted in a red box), 'Network Security', 'Static IP Mapping', 'Phone Configuration', 'SIP Account Settings', 'Advanced SIP Settings', 'Audio Codec', 'Advanced Call Features', 'Ring Tone', and 'Speed Dial'. The main content area is titled 'Network Configuration' and contains the following settings:

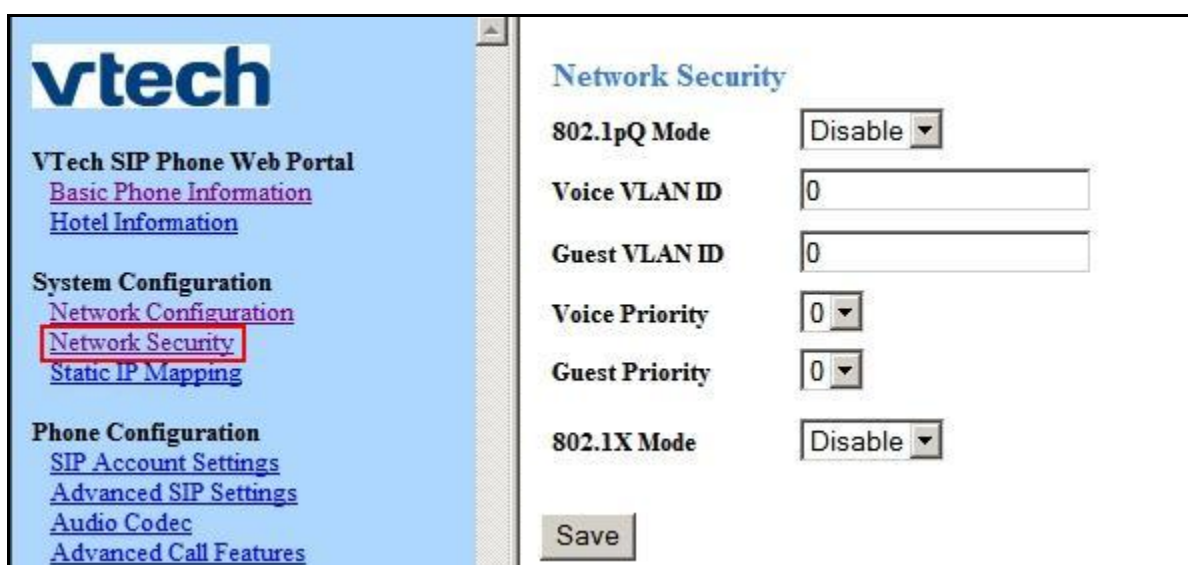
DHCP	<input checked="" type="checkbox"/>
IP Address	192.168.1.100
Subnet Mask	255.255.255.0
Gateway	192.168.1.254
DNS Address	0.0.0.0
ACS IP Address	0.0.0.0
TFTP Server Address	

At the bottom of the settings area is a 'Save' button.

7.2. Set Network Security and QoS (optional)

802.1X Authentication is supported on the VTech phones, this was not tested.

If required, enable QoS by setting the **802.1pQ Mode** selection to “**Enable**” and provide a valid **VLAN ID** and **Voice Priority**, this was not tested. By default, QoS is not enabled and Voice VLAN ID and Voice Priority are set to “0”. Click on **Save** when finished.



The screenshot shows the VTech SIP Phone Web Portal interface. On the left is a navigation menu with the VTech logo at the top. Below the logo, the menu is organized into three sections: 'VTech SIP Phone Web Portal' with links for 'Basic Phone Information' and 'Hotel Information'; 'System Configuration' with links for 'Network Configuration', 'Network Security' (which is highlighted with a red box), and 'Static IP Mapping'; and 'Phone Configuration' with links for 'SIP Account Settings', 'Advanced SIP Settings', 'Audio Codec', and 'Advanced Call Features'. The main content area on the right is titled 'Network Security'. It contains five configuration items: '802.1pQ Mode' set to 'Disable' in a dropdown menu; 'Voice VLAN ID' set to '0' in a text input field; 'Guest VLAN ID' set to '0' in a text input field; 'Voice Priority' set to '0' in a dropdown menu; and 'Guest Priority' set to '0' in a dropdown menu. At the bottom of this section is a 'Save' button.

7.3. Configure SIP Account Settings

The following illustration shows the settings used on the single line phone. For **Extension** and **Authentication Name**, enter the assigned **Extension** previously configured in **Section 5.5**. Also, enter the password configured in **Section 5.5** for this extension. Leave the **DTMF Method** set to “**RFC 2833**” to enable DTMF tones used when accessing Voicemail. Click on **Save** when finished.

The screenshot displays the VTech SIP Phone Web Portal interface. On the left, a navigation menu includes links for Basic Phone Information, Hotel Information, System Configuration (Network Configuration, Network Security, Static IP Mapping), and Phone Configuration (SIP Account Settings, which is highlighted with a red box). The main content area is titled 'Line Selection' and shows 'Line 1' as the selected line. To the right, the 'SIP Account Settings-Line 1' form is visible. It contains fields for Extension (28250), Authentication Name (28250), Password (masked with dots), DTMF Method (set to RFC 2833 via a dropdown), and External Call Prefix (empty). A 'Save' button is located at the bottom of the form.

SIP Account Settings-Line 1	
Extension	<input type="text" value="28250"/>
Authentication Name	<input type="text" value="28250"/>
Password	<input type="password" value="....."/>
DTMF Method	<input type="text" value="RFC 2833"/>
External Call Prefix	<input type="text"/>
<input type="button" value="Save"/>	

7.4. Configure Advanced SIP Setting

Navigate to the Advanced SIP Settings page. Enter the IP address of IP Office and port 5060 for **Registrar Server Address : Port** (as shown below). Also enter the IP address of IP Office for the **Message Waiting Server** field.

The **Dial Plan** field can be modified to allow immediate dialing after a number is completed without having to wait for the interdigit timeout. **282xx** was added to allow immediate dialing of 5 digit extension starting with 282 as configured on the IP Office in the sample configuration. The default entry of **.T** will dial any digits after the interdigit timeout.

The default **On Hold Timeout (minute)** value is 15. Therefore a call put on hold by the VTech phone will be dropped after 15 minutes.

All other fields were left as default. Click on **Save** when finished.

Advanced SIP Settings	
Registrar Server Address : Port	192.168.97.36 : 5060
Proxy Server Address : Port	: 5060
Message Waiting Server : Port	192.168.97.36 : 5060
Backup Registrar Server	Disable
Backup Registrar Server Address : Port	:
Backup Registrar Retry Count	2
SIP Transport	UDP
Registration Timeout (second)	300
Registration Retry Limit (attempt)	10
Message Waiting Subscribe Timeout (second)	300
PRACK	Disable
Dial Plan	282xx.T
Interdigit Timeout (second)	5
On Hold Timeout (minute)	15
Save	

7.5. Configure Audio Codecs

The VTech phones support a number of audio codecs.. Select at least one codec to match those configured in IP Office to ensure compatibility with Avaya Gateways and IP or SIP Endpoints. On completion, click the **Save** button.

The screenshot shows the VTech SIP Phone Web Portal interface. On the left is a navigation menu with sections: 'VTech SIP Phone Web Portal' (containing links for Basic Phone Information and Hotel Information), 'System Configuration' (containing links for Network Configuration, Network Security, and Static IP Mapping), and 'Phone Configuration' (containing links for SIP Account Settings, Advanced SIP Settings, Audio Codec, Advanced Call Features, Ring Tone, and Speed Dial). The 'Audio Codec' link is highlighted with a red box. The main content area is titled 'Line Selection' and shows 'Line 1'. To the right, the 'Audio Settings - Line 1' section contains four dropdown menus for Audio Codec 1 (G.711u), Audio Codec 2 (G.729), Audio Codec 3 (G.722), and Audio Codec 4 (empty). Below these is an SRTP Mode dropdown set to 'Disabled' and a 'Save' button.

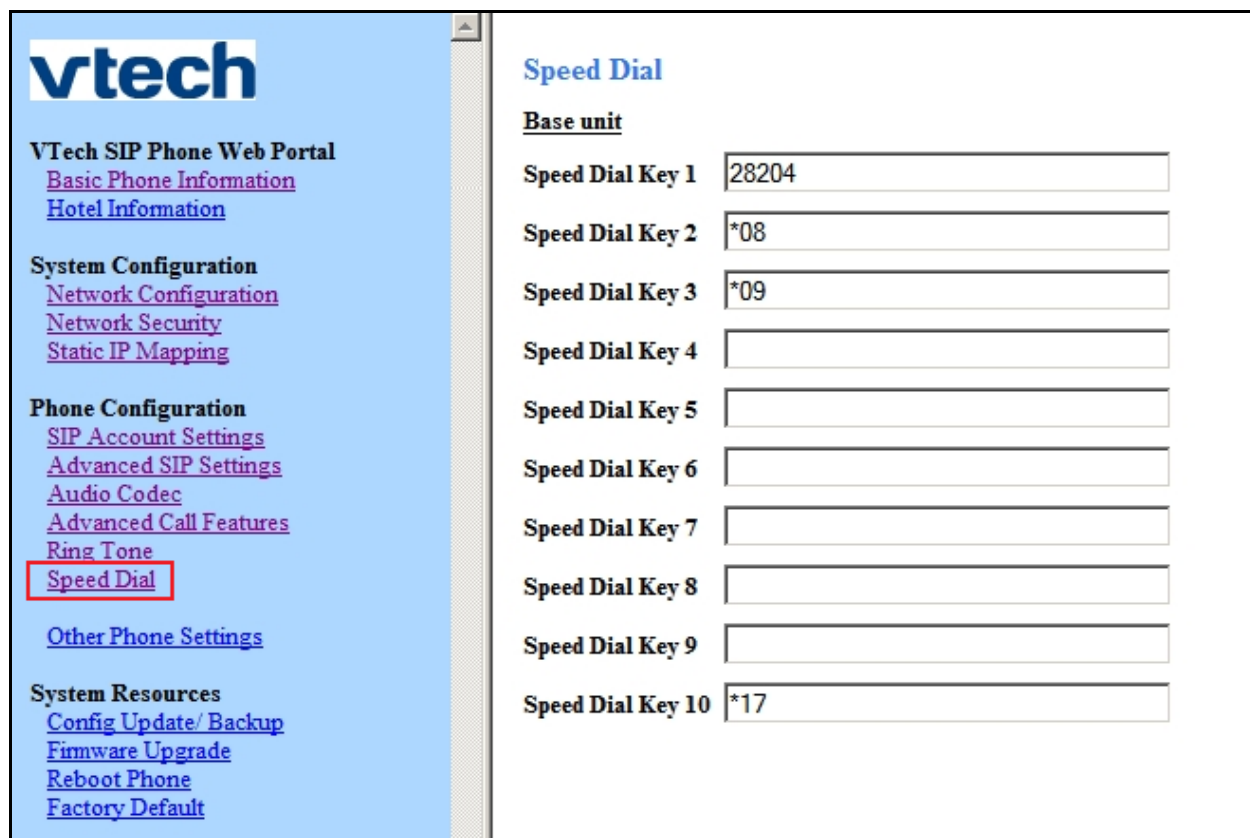
7.6. Advanced Call Features

The VTech advanced call features are not compatible with IP Office and were not configured as shown below. The feature shown in the screen below can be configured on IP Office with the use of Shortcodes if required.

The screenshot shows the VTech SIP Phone Web Portal interface, similar to the previous one. The 'Advanced Call Features' link in the 'Phone Configuration' section of the navigation menu is highlighted with a red box. The main content area is titled 'Line Selection' and shows 'Line 1'. To the right, the 'Advanced Call Features-Line 1' section contains four settings: 'Call Forward Mode-On No Answer' (checkbox), 'Call Forward Mode-On busy' (checkbox), 'Call Forward Mode-Always' (checkbox), and 'Call Forward Number' (text input field). Below these is a 'Do Not Disturb (DND)' checkbox and a 'Save' button.

7.7. Configure Speed Dial Keys

VTech phones are capable of using up to 10 Speed Dial buttons to provide one-touch access to various hotel services such as concierge, front desk, voicemail, and Do Not Disturb. This was simulated by configuring speed dial keys to dial IP, SIP and Digital phones in the test environment. IP Office Shortcodes can also be used with the Speed Dial Keys. As shown below ***08** will active **Do Not Disturb**, ***09** will disable **Do Not Disturb** and ***17** will access the phones **VoiceMail**. On completion, click the **Save** button (not shown).



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VTech SIP Phone Web Portal
[Basic Phone Information](#)
[Hotel Information](#)

System Configuration
[Network Configuration](#)
[Network Security](#)
[Static IP Mapping](#)

Phone Configuration
[SIP Account Settings](#)
[Advanced SIP Settings](#)
[Audio Codec](#)
[Advanced Call Features](#)
[Ring Tone](#)
[Speed Dial](#)
[Other Phone Settings](#)

System Resources
[Config Update/ Backup](#)
[Firmware Upgrade](#)
[Reboot Phone](#)
[Factory Default](#)

Speed Dial

Base unit

Speed Dial Key 1	28204
Speed Dial Key 2	*08
Speed Dial Key 3	*09
Speed Dial Key 4	
Speed Dial Key 5	
Speed Dial Key 6	
Speed Dial Key 7	
Speed Dial Key 8	
Speed Dial Key 9	
Speed Dial Key 10	*17

7.8. Server Options

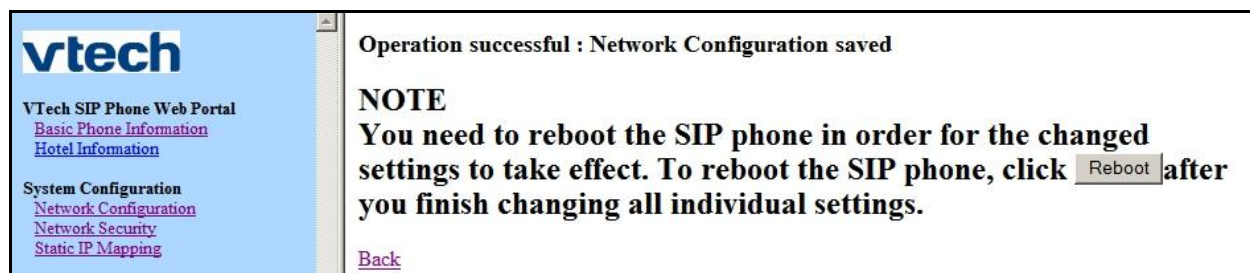
From the **Scheme** drop down box select **Avaya**. Set the **DTMF RFC2833 – Payload Type** field to **101**. On completion, click the **Save** button.

The screenshot displays the VTech SIP Phone Web Portal interface. On the left is a navigation menu with categories: VTech SIP Phone Web Portal, System Configuration, Phone Configuration, System Resources, Inter-Op Configuration, and Admin Config. The 'Server Options' link under 'Inter-Op Configuration' is highlighted with a red box. The main content area is titled 'Server Options' and contains the following settings:

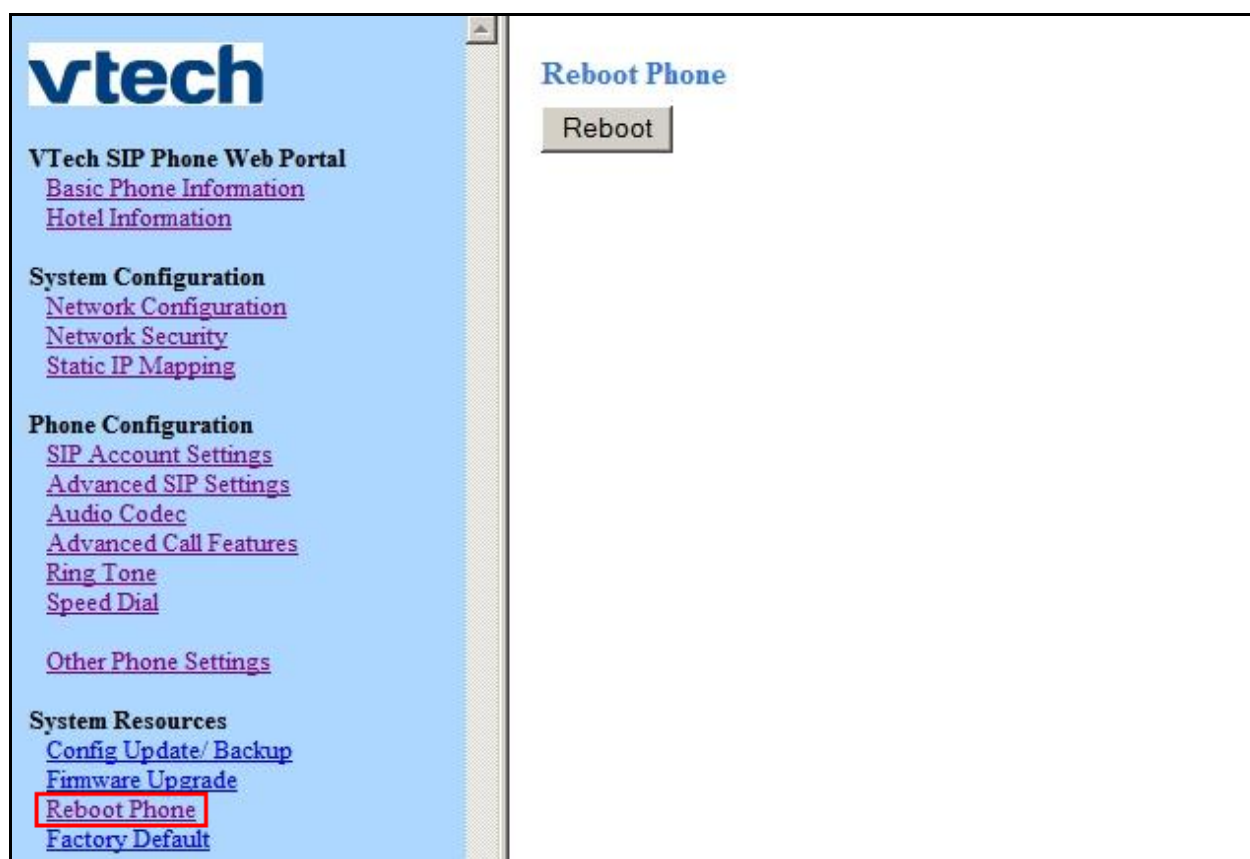
- Scheme:** Avaya (dropdown menu)
- Advance:**
 - Server Option - Blind Transfer: WITH REFER (dropdown menu)
 - Server Option - Generate MOH on Client: DISABLE (dropdown menu)
 - Server Option - Append Method to REFER message: DISABLE (dropdown menu)
 - DTMF RFC2833 - Payload Type (Leave BLANK for Dynamic Payload): 101 (text input)
 - Server Option - Server Handle Hold Resume: DISABLE (dropdown menu)
 - Server Option - Ignore Changing MWI Server IP in 200 OK Message: DISABLE (dropdown menu)
 - Server Option - Self-RingBack Tone when being transferred: DISABLE (dropdown menu)
 - Server Option - Backup Server Replace Proxy Only: DISABLE (dropdown menu)
 - Server Option - Multiple Active Call: ENABLE (dropdown menu)
 - Server Option - Accept MWI Notify Anonymous: DISABLE (dropdown menu)
- Save** (button)

7.9. Reboot to Activate Changes

After all settings are completed and saved, the phone must be rebooted for the changes to take effect. Click on the **Reboot** button as shown below after saving the last change.



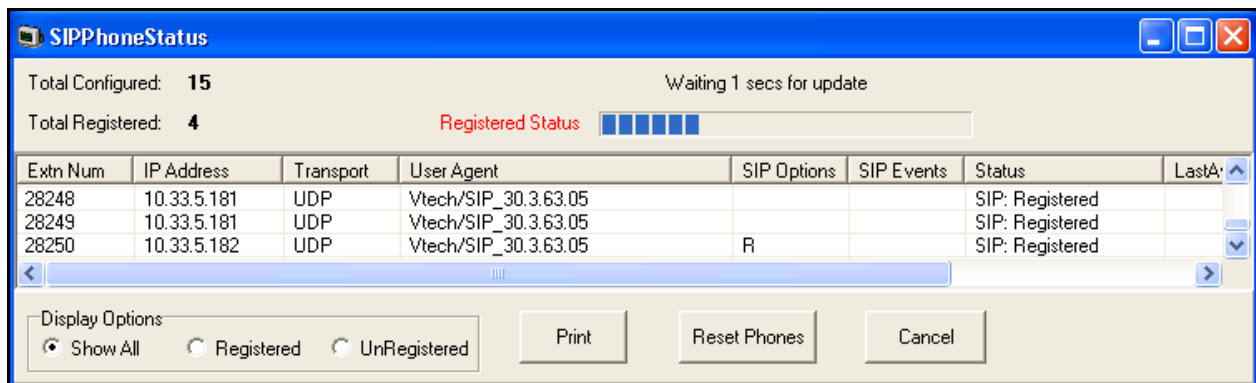
The phone can also be rebooted by navigating to the Reboot Phone page and clicking on **Reboot**.



8. Verification Steps

This section provides tests that can be performed to verify proper configuration of IP Office and VTech 1-Line SIP Phones.

From a PC running the IP Office Monitor application, select **Start > Programs > IP Office > Monitor** to launch the application. The **Avaya IP Office R8.1 SysMonitor** screen is displayed (not shown). In SysMonitor select **Status > SIP Phone Status** from the top menu. The **SIPPhoneStatus** screen is displayed. Verify that there is an entry for each SIP extension. Verify that the **User Agent** starts with “VTech”, and that the **Status** is “SIP: Registered”, as shown below.



Extn Num	IP Address	Transport	User Agent	SIP Options	SIP Events	Status	LastA
28248	10.33.5.181	UDP	Vtech/SIP_30.3.63.05			SIP: Registered	
28249	10.33.5.181	UDP	Vtech/SIP_30.3.63.05			SIP: Registered	
28250	10.33.5.182	UDP	Vtech/SIP_30.3.63.05	R		SIP: Registered	

9. Conclusion

The VTech 1-Line Hotel and Lobby SIP Phones successfully interoperated with the Avaya IP Office and Voicemail Pro as described in these notes. The observations noted in **Section 2** should be confirmed with VTech, future updates to the product might address these observed behaviors.

10. Additional References

Product documentation for Avaya products may be found at <http://support.avaya.com>.

- 1) Avaya IP Office Basic Edition - Quick Mode 8.1 Manager –Issue 05e, 25 May 2012
- 2) Avaya IP Office Technical Bulletin, Bulletin no: 145, 16 July 2012
- 3) Avaya IP Office Administering Voicemail Pro 15-601063 Issue 8b - December 11, 2012

Product information for VTech SIP Hotel and Lobby Phones may be found at <http://vttechhotelphones.com>.

11. Change History

Issue	Date	Reason
1.0	1/28/2013	Initial issue
1.1	3/6/2013	Added Voicemail Pro configuration for Alarm Set

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