



Configuration Guide For Yealink IP Phone & AVAYA

1. Requirements and Preparation

- Avaya C363T-PWR Converged Stackable Switch 4.3.12 or higher running on AVAYA system.
- Software image x.43.x.x or higher running on the Yealink IP Phone.

2. Reference Configuration

Figure 1 illustrates a sample configuration that was used to compliance test the interoperability of Yealink SIP Phones and Avaya IP Office. The configuration consists of an Avaya IP Office 500 connected to an Avaya switch to which the Yealink T-28 phone is connected. This system has connections to the following: Avaya 1600 Series IP Phones, Avaya Digital Phones and a PRI trunk to the PSTN. The phones connected to the system will be used to generate call traffic to the IP Office. These phones will be used to generate intra-switch calls and outbound/inbound calls to/from the PSTN.

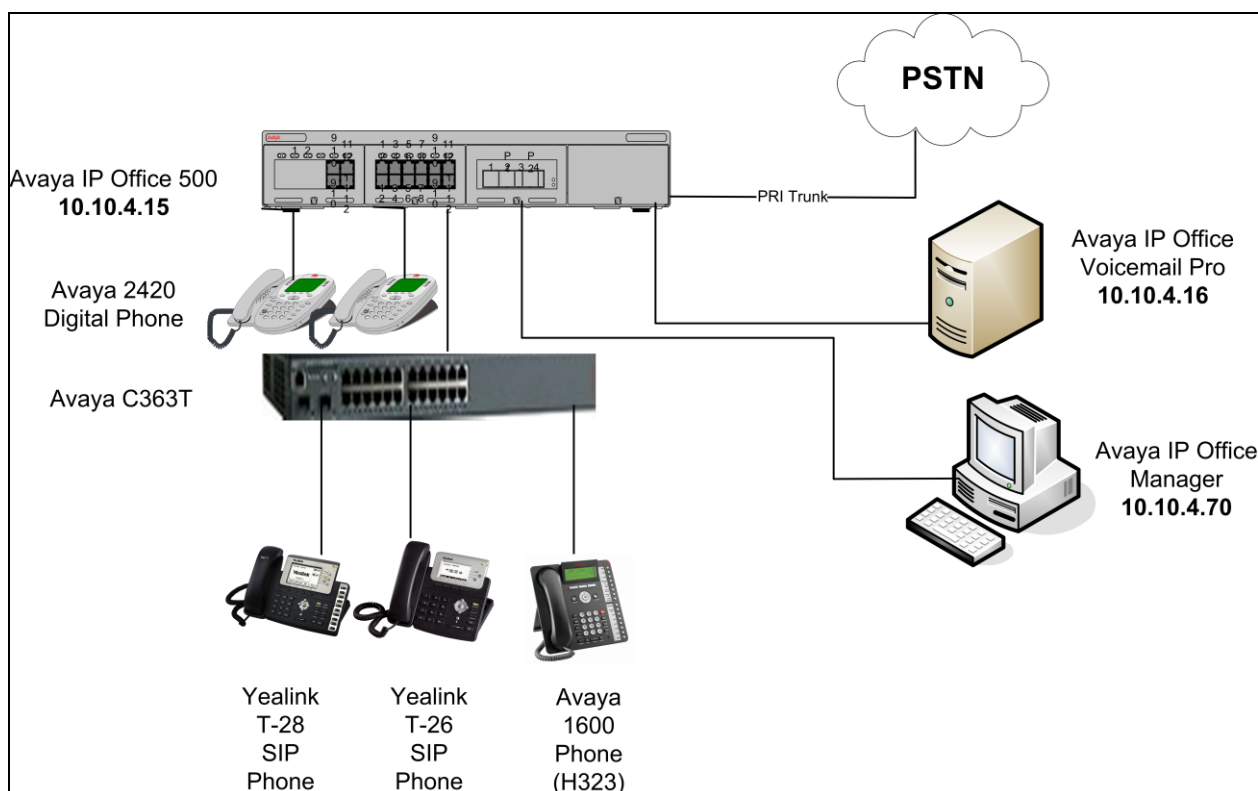


Figure 1: Network Configuration of Yealink SIP Phones with Avaya IP Office

3. Configure the Avaya IP Office

All the configuration changes in this section for IP Office are performed through the IP Office Manager. For more information on configuring IP Office, refer to the Avaya product documentation.

This section provides the procedures for configuring IP Office. The procedures fall into the following areas:

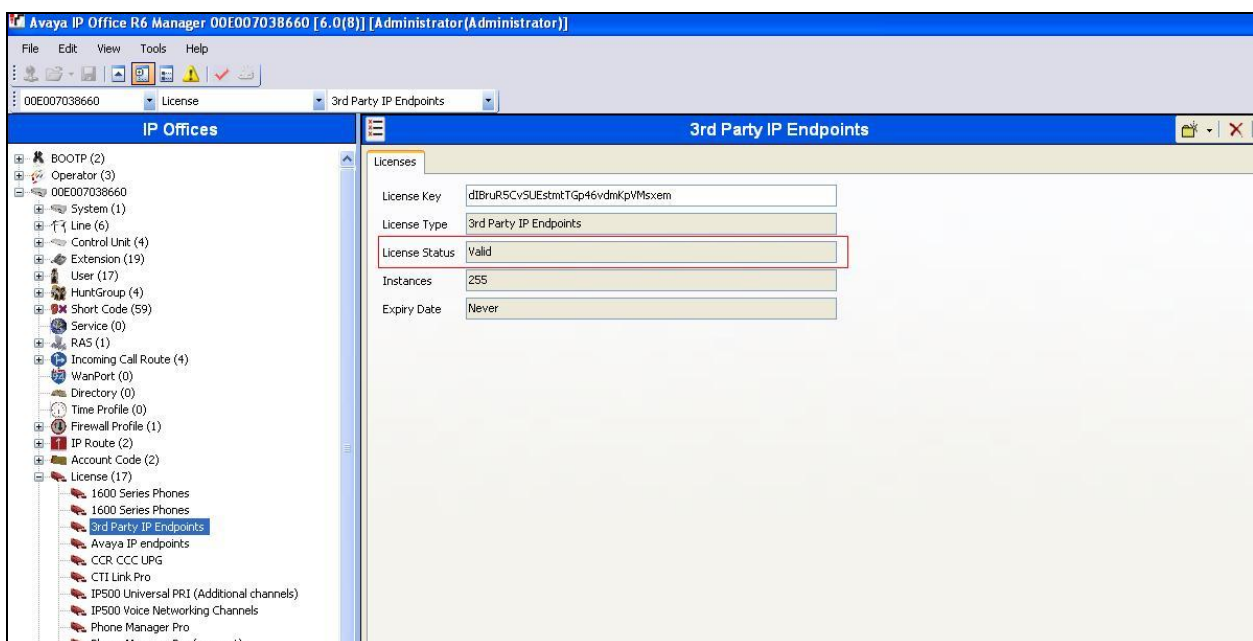
- Verify IP Office Licensing
- Administer SIP Registrar
- Add SIP Extensions
- Configure SIP User Names
- Save Configuration

The configuration of the PRI interface to the PSTN is outside the scope of these Application Notes.

3.1 Verify IP Office Licensing

From a PC running the Avaya IP Office Manager application, select **Start -> Programs -> IP Office -> Manager** to launch the Manager application. Select the IP Office system, and log in with the appropriate credentials. The **Avaya IP Office R6 Manager** screen is displayed.

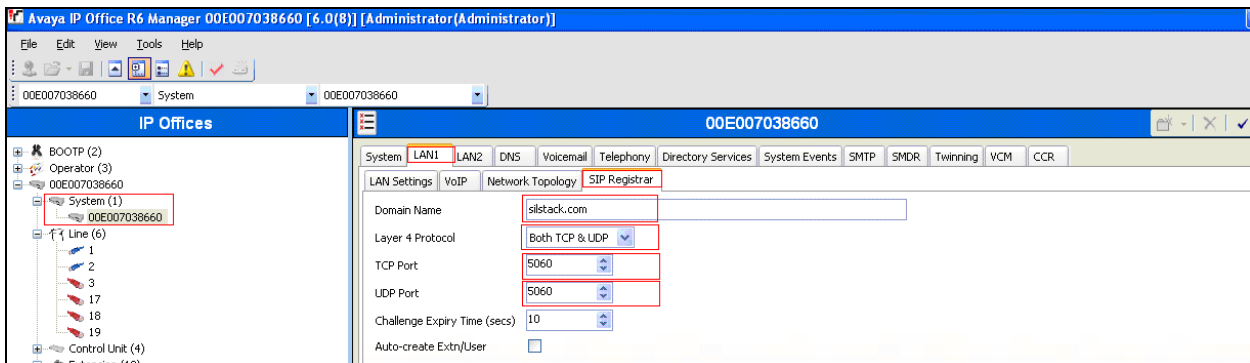
From the configuration tree in the left pane, select **License -> 3rd Party IP Endpoints** to display the **3rd Party IP Endpoints** screen in the right pane. Verify that the **License Status** is **Valid**.



3.2 Administer SIP Registrar

Select **SIP Registrar** sub-tab in the right pane and enter following values:

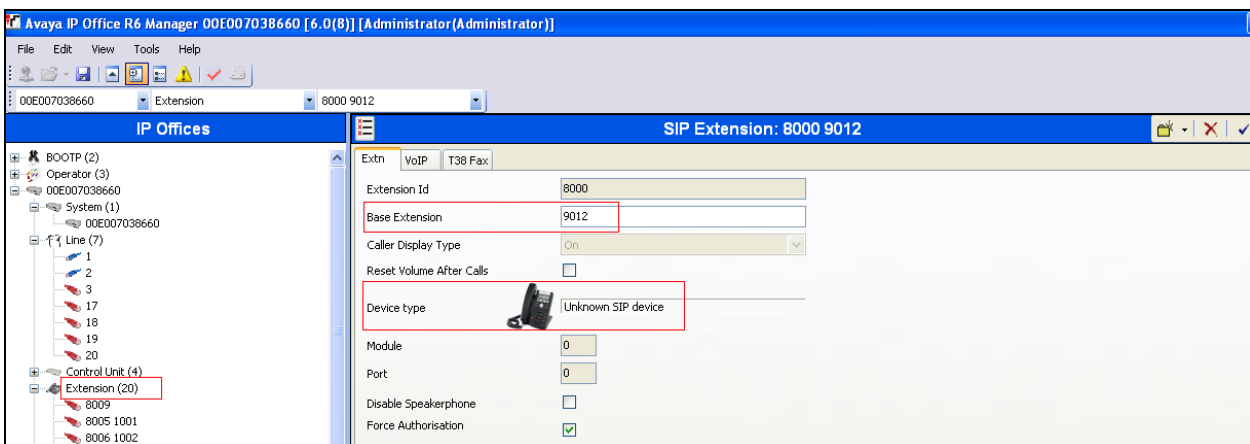
- **Domain Name** Enter a valid Domain Name, in this case **silstack.com** is used.
- **Layer 4 Protocol** Select **Both TCP & UDP**.
- **TCP Port** Select **5060**
- **UDP Port** Select **5060**



Click **OK** (not shown).

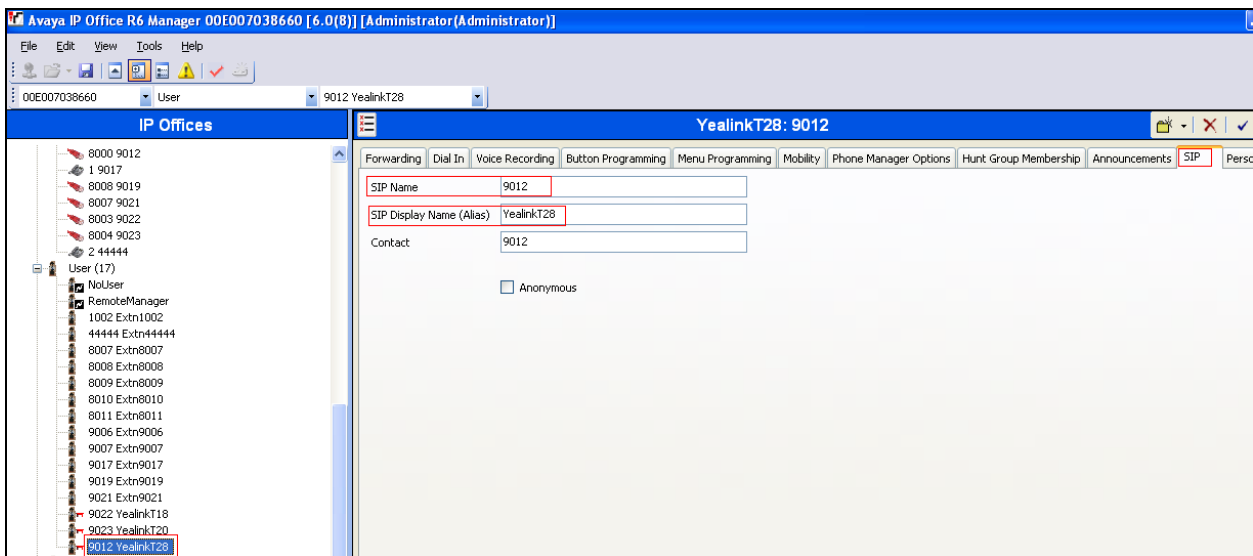
3.3 Add SIP Extensions

Add a SIP Extension by selecting **Extension** from the left pane. Right-click and choose **New** and **SIP Extension** (not shown). The **Extension Id** is automatically created i.e., **8000** in this case. Set the **Base Extension** to **9012**. Note that the **Device type** is **Unknown SIP device**. Click **OK**.

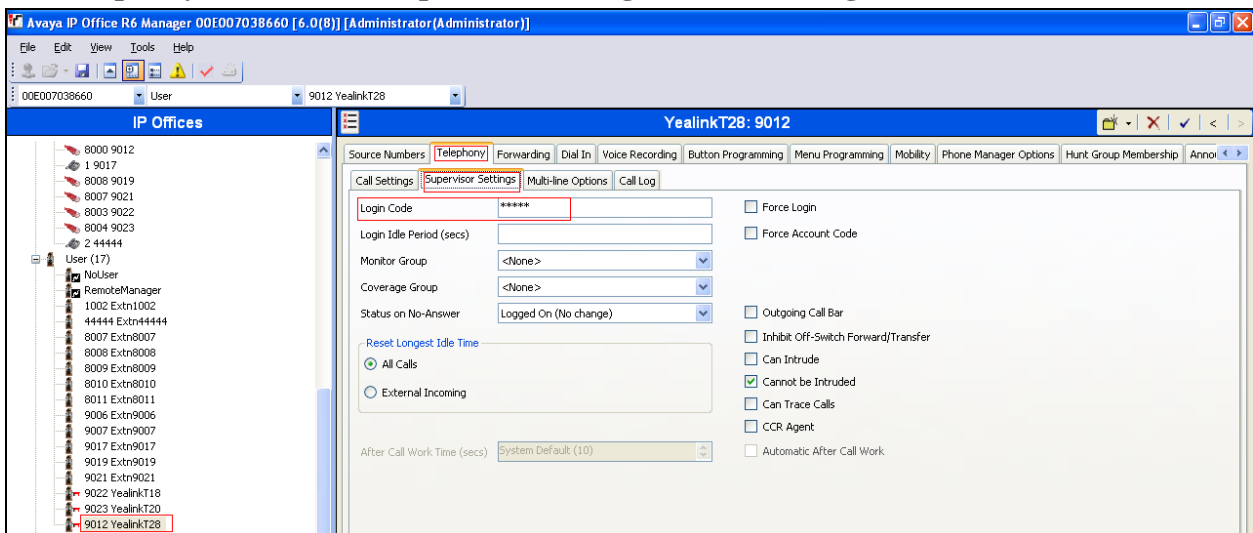


3.4 Configure SIP User Names

From the left pane, select a **User** and in the right-hand pane, select **SIP** tab. Modify the **SIP Name** to be the same as the user's extension number, in this case, **9022**. Set the **SIP Display Name** as required, in this case **YealinkT28**. The other fields can be left as default. Click **OK** (not shown). The completed user should be displayed as shown below. Repeat this for all users.



On the **Telephony** tab select the **Supervisor Settings** tab. Set the **Login Code**.

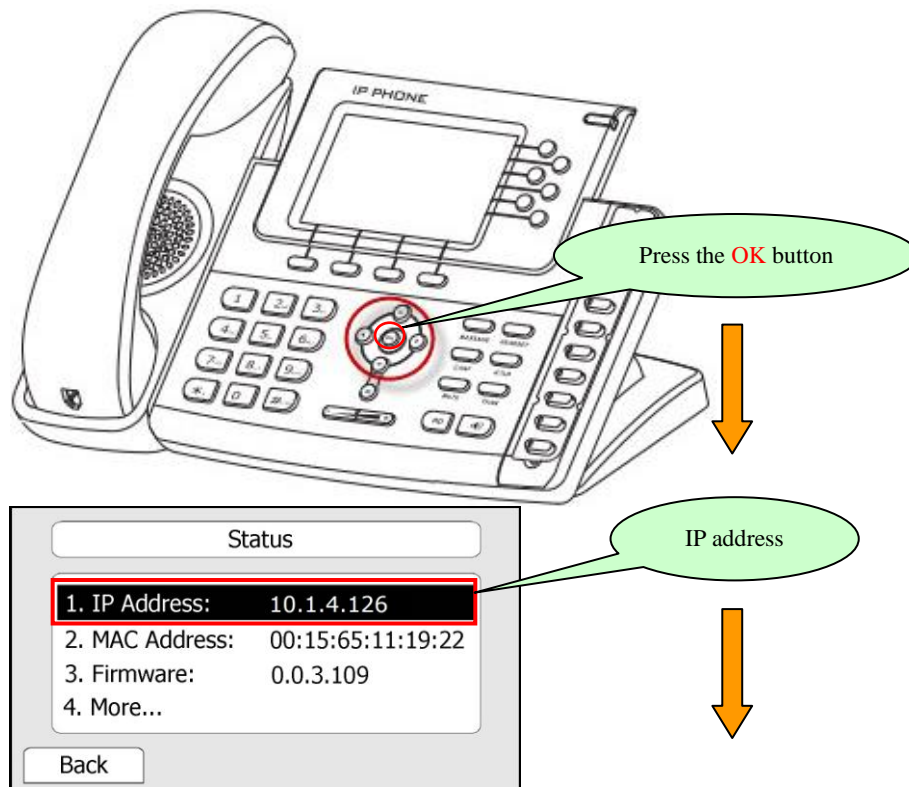


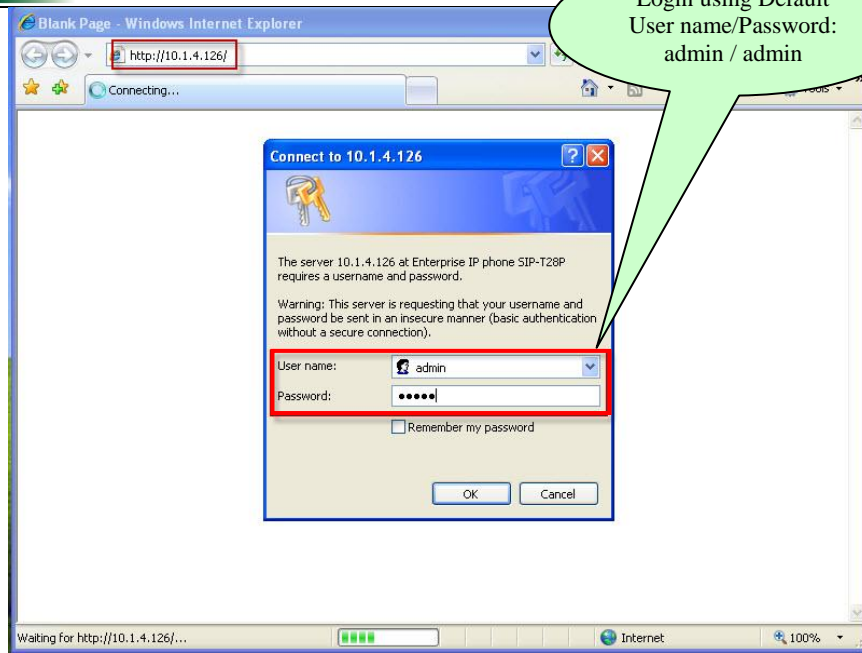
3.5 Save Configuration

Select **File** → **Save Configuration** to save and send the configuration to the IP Office server.

4. Configure Yealink IP Phone

4.1 Login to WEB management





4.2 Configure the Account Settings

The screenshot shows the 'Account' configuration page in the Yealink Easy VoIP interface. The 'Account' tab is selected. A dropdown menu shows 'Account 1' selected. The 'Basic >>' section contains various settings. A red box highlights the 'Account Active' checkbox (checked), 'Label' (100), 'Display Name' (100), 'Register Name' (100), 'User Name' (100), 'Password' (masked), and 'SIP Server' (192.168.1.199). Another red box highlights the 'Voice Mail' field with the value '*4'. A green bubble labeled 'check Note' points to the 'NAT Traversal' dropdown, which is set to 'STUN'. A 'NOTE' section on the right explains the 'Register Name' and 'Proxy Require' fields. A 'Voicemail code' callout points to the 'Voice Mail' field. The 'Advanced >>' section is also visible at the bottom.

① Select "Account"

② Select one Account

③ Active Account

④ Fill in these fields

⑤ Voicemail code

check Note

NOTE

Register Name
SIP service subscriber's ID used for authentication.

Proxy Require
A special parameter just for Nortel server. If you login to Nortel server, the

Defines the STUN server will be active or not.

Advanced
The Advanced parameters for administrator.

After entering the above settings, Account1 will be available to make calls.

Note : If the SIP server is behind a NAT, you should enable "NAT Traversal" as "STUN" and then specify a STUN Server. For more details about STUN, please refer to <http://www.voip-info.org/wiki/view/STUN>. To learn about NAT, you should refer to <http://www.voip-info.org/wiki/view/NAT+and+VOIP>

5. Auto-configure Yealink Phone

Please refer to Yealink Document [Auto Provision Manual](http://www.yealink.com/fae/Auto Provision Manual.rar):
<http://www.yealink.com/fae/Auto Provision Manual.rar>

Appendix

1. Default Basic Dial Code on AVAYA System

Voice Mail (VMail)	*17	
Call Park	*37*<park ref number>#	
Callpark Retrieve	*38*<park ref number>#	
Call pickup	*30	Pick up the longest ringing call on the system
	*31	Pick up longest ringing hunt group call (to which you are a member)
	32<ext>#	Pick up a particular extension
	53<group ext>#	Pick up a call ringing in a pickup group