

How to Configure the Yealink T40P:

The Yealink SIP-T40P desk phone supports up to 3 SIP accounts, and up to 3 Simultaneous calls.

Note: this equipment support PoE. And when using power supply, the input ratings are 5V 0.6A.

This guide will cover the following steps

1. Logging into the phone
2. Upgrading the firmware
3. Entering in all provisioning information
4. Configuring device to access its voice mailbox.

1. Accessing the phone's Web User Interface

- Connect the phone to a DHCP enabled network, and allow the phone to pull an IP address.
- You can either pull the IP address of the phone by using network scanning software like Netscan (windows), Angry Ip scanner (Mac), or you can view the IP address of the phone by navigating the menus Eg: Menu > Status.
- Navigate to the phones IP address in your web browser.
- The default username and password are admin.

2. Upgrading the firmware

- As of 4/27/16 the latest firmware for the Yealink SIP-T40P is 54.80.25.1.rom, and can be located [here](#).
- Within the phones web interface go to Settings > Upgrade. Click on Choose file, and navigate to the directory which the update was downloaded to, and then click "Upgrade".

WARNING Do not refresh the web browser, or power off the phone while the firmware upgrade is in progress.

3. Configuring Phone with Generic SIP device information provided from the Host Pilot portal.

- Pull up HostPilot, and locate the specific Generic SIP device you are looking to configure.
- In the web interface of the of Yealink desk phone you will want to click on "Account".
- Make sure the drop down for "Line Active" is set to Enabled, and not Disabled.
- Below is a key for all of the information needed, and provided by HostPilot, in relation to the terminology used within the Yealink web interface.

SIP Username/Username	Unique 9 digit ID located in Host Pilot
SIP Authorization ID/Register Name	Unique 9 digit ID located in Host Pilot
SIP Password/Password	Unique alpha-numeric password located in Host Pilot.
Outbound Proxy/Outbound Proxy Server	usbc.telecomsvc.com or UC70.telecomsvc.com
SIP Domain/Server Host	usbc.telecomsvc.com or UC70.telecomsvc.com

- Once all of the above information listed has been entered in, and looks similar to this click confirm.

The screenshot shows the Yealink T40P web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'DSSKey', 'Features', 'Settings', 'Directory', and 'Security'. The 'Account' tab is selected, showing configuration for 'Account 1'. The left sidebar has 'Register' selected, with sub-tabs for 'Basic', 'Codec', and 'Advanced'. The main content area is divided into sections: 'Register Status' (Registered), 'Line Active' (Enabled), 'Label' (101), 'Display Name' (2066864846), 'Register Name' (900283927), 'User Name' (900283927), and 'Password' (masked). Below this are 'SIP Server 1' and 'SIP Server 2' settings, each with 'Server Host', 'Transport' (UDP), 'Server Expires' (3600), and 'Server Retry Counts' (3). 'SIP Server 1' has a port of 6060, while 'SIP Server 2' has a port of 5060. At the bottom, there are 'Enable Outbound Proxy Server' (Enabled), 'Outbound Proxy Server 1' (uc70.telecomsvc.com, Port 5060), 'Outbound Proxy Server 2' (Port 5060), 'Proxy Fallback Interval' (3600), and 'NAT' (Disabled). A 'NOTE' section on the right contains information about 'Account Registration', 'Server Redundancy', and 'NAT Traversal'. At the bottom of the page, there are 'Confirm' and 'Cancel' buttons, and a copyright notice: 'Copyright © 1998-2016 **Inc. All Rights Reserved'.

- Next select “Advanced” under the “Account” tab, and make sure that the following fields are configured as documented below.

Keep Alive Type	Default
Keep Alive Interval (Seconds)	30
Local SIP Port	6xxx (xxx = 3 digit Ext, ie. Ext 133 = 6133)
DTMF Type	RFC2833
DTMF Payload Type	101
Voice Mail	DID of phone
SIP Registration Retry Timer	30
Caller ID Source	RPID-FROM

- Once all of the above information has been entered in, it should look similar to this.

The screenshot shows the Yealink T40P web interface with the 'Account' tab selected. The settings for 'Account 1' are as follows:

- Keep Alive Type: Default
- Keep Alive Interval(Seconds): 30
- RPort: Disabled
- Subscribe Period(Seconds): 1800
- DTMF Type: RFC2833
- DTMF Info Type: DTMF-Relay
- DTMF Payload Type(96~127): 101
- Retransmission: Disabled
- Subscribe Register: Disabled
- Subscribe for MWI: Disabled
- MWI Subscription Period(Seconds): 3600
- Subscribe MWI To Voice Mail: Disabled
- Voice Mail: 2066864846
- Voice Mail Display: Enabled
- Caller ID Source: RPID-FROM
- Session Timer: Disabled
- Session Expires(30~7200s): 1800
- Session Refresher: UAC
- Send user=phone: Disabled
- RTP Encryption(SRTP): Disabled
- PTime(ms): 20
- BLF List URI:
- BLF List Pickup Code:
- BLF List Barge In Code:
- BLF List Retrieve Call Parked Code:

NOTE

DTMF
It is the signal sent from the IP phone to the network, which is generated when pressing the IP phone's keypad during a call.

Session Timer
It allows a periodic refresh of SIP sessions through a re-INVITE request, to determine whether a SIP session is still active.

Busy Lamp Field/BLF List
Monitors a specific extension/a list of extensions for status changes on IP phones.

Shared Call Appearance (SCA)/ Bridge Line Appearance (BLA)
It allows users to share a SIP line on several IP phones. Any IP phone can be used to originate or receive calls on the shared line.

Network Conference
It allows multiple participants (more than three) to join in a call.

VQ-RTCPXR
The VQ-RTCPXR mechanism, compliant with RFC 6035, sends the service quality metric reports contained in SIP PUBLISH messages to the central report collector.

[You can click here to get more guides.](#)

Shared Line	Disabled
Dialog Info Call Pickup	Disabled
BLA Number	
BLA Subscription Period	300
SIP Send MAC	Disabled
SIP Send Line	Disabled
SIP Registration Retry Timer(0~1800s)	30
Conference Type	Local Conference
Conference URI	
ACD Subscribe Period(120~3600s)	3600
Early Media	Disabled
SIP Server Type	Default
Music Server URI	
Directed Call Pickup Code	
Group Call Pickup Code	
Distinctive Ring Tones	Enabled
Unregister When Reboot	Disabled
Out Dialog BLF	Disabled
VQ RTP-XR Collector name	
VQ RTP-XR Collector address	
VQ RTP-XR Collector port	5060

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- Next select the “Network” tab, then the “Advanced” sub tab, and make the following changes.

Voice QoS	46
SIP QoS	46
Maximum RTP Port	50000
Minimum RTP Port	30000

- When done you should have something that looks similar to this.

Yealink T40P Log Out

Status Account **Network** DSSKey Features Settings Directory Security

Basic
PC Port
Advanced

LLDP	Active	Enabled
	Packet Interval (1~3600s)	60
CDP	Active	Disabled
	Packet Interval (1~3600s)	60
VLAN		
WAN Port	Active	Disabled
	VID (1-4094)	1
	Priority	0
PC Port	Active	Disabled
	VID (1-4094)	1
	Priority	0
DHCP VLAN	Active	Enabled
	Option (1-255)	132
NAT	Active	Disabled
	STUN Server	
	STUN Port(1024~65000)	3478
Port Link		
	WAN Port Link	Auto Negotiate
	PC Port Link	Auto Negotiate
Voice QoS		
	Voice QoS (0~63)	46
	SIP QoS (0~63)	46
Local RTP Port		
	Max RTP Port (1~65535)	50000
	Min RTP Port (1~65535)	30000

NOTE

VLAN
It is used to logically divide a physical network into several broadcast domains. VLAN membership can be configured through software instead of physically relocating devices or connections.

The priority of VLAN assignment method (from highest to lowest)
:LLDP/CDP->manual configuration->DHCP VLAN

NAT Traversal
It is a general term for techniques that establish and maintain IP connections traversing NAT gateways. STUN is one of the NAT traversal techniques.

You can configure NAT traversal for the IP phone.

Quality of Service (QoS)
It is the ability to provide different priorities for different packets in the network, allowing the transport of traffic with special requirements.

Web Server Type
It determines access protocol and port of the IP phone's web user interface.

802.1X Authentication
It offers an authentication mechanism for the IP phone to connect/link to a LAN or WLAN.

VPN
It provides remote offices or individual users with secure access to their organization's network.

You can click here to get more guides.

- Lastly select the "Settings" tab, then the "SIP" sub tab, and make the following change:

Local SIP Port	6xxx (xxx = 3 digit Ext, ie. Ext 133 = 6133)
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Now you are good to go! Please feel free to review KB article for Firewall Rules and Port Requirements for the service to work correctly.

<https://kb.intermedia.net/Article/3119>