How to Configure the Yealink T46G:

The Yealink SIP-T46G desk phone supports up to 16 SIP accounts, and up to 16 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 3/29/16 the latest firmware for the YeaLink SIP-T46G is 28.80.0.79.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.

● 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
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

BLF List Barge In Code	
BLF List Retrieve Call Parked Code	
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 RTCP-XR Collector name	
VQ RTCP-XR Collector address	
VQ RTCP-XR Collector port	5060

Confirm   Cancel

● 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.

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 ⇕	

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

● 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)

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