

How to Configure the Yealink T48G:

The Yealink SIP-T48G 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-T48G is 35.80.0.79.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.

Yealink T48G Log Out

Account Account 1

Register

Basic

Codec

Advanced

Account

Account 1

Register Status: Registered

Line Active: Enabled

Label: 100

Display Name: 100

Register Name: 900283926

User Name: 900283926

Password:

SIP Server 1

Server Host: uc70.telecomsvc.com Port: 6060

Transport: UDP

Server Expires: 3600

Server Retry Counts: 3

SIP Server 2

Server Host: Port: 5060

Transport: UDP

Server Expires: 3600

Server Retry Counts: 3

Enable Outbound Proxy Server: Enabled

Outbound Proxy Server 1: uc70.telecomsvc.com Port: 6060

Outbound Proxy Server 2: Port: 5060

Proxy Fallback Interval: 3600

NAT: Disabled

NOTE

Account Registration
Registers account(s) for the IP phone.

Server Redundancy
It is often required in VoIP deployments to ensure continuity of phone service, for events where the server needs to be taken offline for maintenance, the server fails, or the connection between the IP phone and the server fails.

NAT Traversal
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 this account.

You can click here to get more guides.

Confirm Cancel

- 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

Register

Basic

Codec

Advanced

Account	Account 1	?
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	900283926	?
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		?

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.

BLF List Barge In Code	<input type="text"/>	?
BLF List Retrieve Call Parked Code	<input type="text"/>	?
Shared Line	Disabled	?
Dialog Info Call Pickup	Disabled	?
BLA Number	<input type="text"/>	?
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	<input type="text"/>	?
ACD Subscribe Period(120~3600s)	3600	?
Early Media	Disabled	?
SIP Server Type	Default	?
Music Server URI	<input type="text"/>	?
Directed Call Pickup Code	<input type="text"/>	?
Group Call Pickup Code	<input type="text"/>	?
Distinctive Ring Tones	Enabled	?
Unregister When Reboot	Disabled	?
Out Dialog BLF	Disabled	?
VQ RTPC-XR Collector name	<input type="text"/>	?
VQ RTPC-XR Collector address	<input type="text"/>	?
VQ RTPC-XR Collector port	5060	?

- Lastly 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.

Basic

PC Port

Advanced

Wi-Fi

LLDP ?

Active

Packet Interval (1~3600s)

CDP ?

Active

Packet Interval (1~3600s)

VLAN ?

WAN Port Active

VID (1-4094)

Priority

PC Port Active

VID (1-4094)

Priority

DHCP VLAN Active

Option (1-255)

NAT ?

Active

STUN Server

STUN Port(1024~65000)

Port Link ?

WAN Port Link

PC Port Link

Voice QoS ?

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 corporate networks.

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
Web Server ?	
HTTP	Enabled
HTTP Port (1~65535)	80
HTTPS	Enabled
HTTPS Port (1~65535)	443
802.1x ?	
802.1x Mode	Disabled
Identity	
MD5 Password	*****
CA Certificates	<input type="text"/> Browse...
	Upload
Device Certificates	<input type="text"/> Browse...
	Upload
Span to PC ?	
Span to PC Port	Disabled
Registration Random ?	
Registration Random (0~60s)	0
ICMPv6 Status ?	
Active	Enabled

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>