

**Note:** A single Anyphone line has a maximum concurrent call capacity of 4 calls. Hardware capacity should also be taken into consideration. This means that Yealink T22P can handle up to 4 concurrent calls.

How to Configure the Yealink T22P:

The Yealink SIP-T22P desk phone supports up to 3 SIP accounts, and up to 3 Simultaneous calls.

Note: this equipment does support PoE, and does not come with a power supply. However the customer can easily obtain one. The input ratings are 5V 1.2A

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 phones 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 Nmap (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 7/15/15 the latest firmware for the YeaLink SIP-T22P is 7.73.0.50, 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.

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

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

**Yealink T22P** Log Out

**Account** Account 1

Register Status: Registered

Line Active: Enabled

Label:

Display Name: 221

Register Name: 900211484

User Name: 900211484

Password: \*\*\*\*\*

Enable Outbound Proxy Server: Disabled

Outbound Proxy Server: UC70.telecomsvc.com Port: 6060

Transport: UDP

NAT: Disabled

STUN Server: Port: 3478

**SIP Server 1**

Server Host: UC70.telecomsvc.com Port: 6060

Server Expires: 3600

Server Retry Counts: 5

**SIP Server 2**

Server Host: Port: 5060

Server Expires: 3600

Server Retry Counts: 3

**NOTE**

**Display Name**  
SIP service subscriber's name which will be used for Caller ID display.

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

**User Name**  
User account, provided by VoIP service provider.

**NAT Traversal**  
Defines the STUN server will be active or not.

You can click here to get more guides.

Confirm Cancel

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- 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 displays the Yealink T22P 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 'Advanced' section is active in the left sidebar. The main configuration area lists various parameters for the account, with 'Caller ID Source' set to 'RPID-FROM'. A 'NOTE' section on the right provides additional information for administrators.

Parameter	Value
Keep Alive Type	Default
Keep Alive Interval(Seconds)	30
Local SIP Port	6227
RPort	Disabled
SIP Session Timer T1 (0.5~10s)	0.5
SIP Session Timer T2 (2~40s)	4
SIP Session Timer T4 (2.5~60s)	5
Subscribe Period(Seconds)	1800
DTMF Type	RFC2833
DTMF Info Type	DTMF-Relay
DTMF Payload Type(96~127)	101
Retransmission	Disabled
Subscribe for MWI	Enabled
MWI Subscription Period(Seconds)	3600
Subscribe MWI To Voice Mail	Disabled
Voice Mail	4254583997
Voice Mail Display	Enabled
Caller ID Source	RPID-FROM
Session Timer	Disabled
Session Expires(30~7200s)	1800
Session Refresher	LIAC
Send user=phone	Disabled
RTP Encryption(SRTP)	Disabled
SRTP Auth-tag	AES-80&&AES-32
PTime(ms)	20

**NOTE**  
**Advanced**  
The Advanced parameters for administrator.  
You can click here to get more guides.

BLF List URI	<input type="text"/>	?
BLF List Code	<input type="text"/>	?
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 Subscrip 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 RTCP-XR Collector name	<input type="text"/>	?
VQ RTCP-XR Collector address	<input type="text"/>	?
VQ RTCP-XR Collector port	5060	?
Accept SIP Trust Server Only	Disabled	?

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- 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.

Yealink T22P Log Out

Status Account Network DSSKey Features Settings Directory Security

Basic  
PC Port  
Advanced

<b>LLDP</b> ?	Active	Enabled
	Packet Interval (1~3600s)	60
<b>VLAN</b> ?	WAN Port	Active
	VID (1-4094)	1
	Priority	0
	PC Port	Active
	VID (1-4094)	1
	Priority	0
	DHCP VLAN	Active
	Option (1-255)	132
<b>Port Link</b> ?	WAN Port Link	Auto Negotiate
	PC Port Link	Auto Negotiate
<b>Voice QoS</b> ?	Voice QoS (0~63)	46
	SIP QoS (0~63)	46
<b>Local RTP Port</b> ?	Max RTP Port (1~65535)	50000
	Min RTP Port (1~65535)	30000
<b>Web Server</b> ?	HTTP	Enabled
	HTTP Port (1~65535)	80
	HTTPS	Enabled
	HTTPS Port (1~65535)	443
<b>802.1x</b> ?		

**NOTE**

**VLAN**  
A VLAN is a logical local area network (or LAN) that extends beyond a single traditional LAN to a group of LAN segments, given specific configurations.

**QoS**  
When the network capacity is insufficient, QoS could provide priority to users by setting the value.

**Local RTP Port**  
Define the port for voice transmission.

You can click here to get more guides.

Now you are good to go! Please feel free to review the following articles for Firewall Rules and Port Requirements for the service to work correctly.

- <https://kb.intermedia.net/Article/3119>
- <https://kb.intermedia.net/Article/3042>