How to Configure the Yealink T41P:

The Yealink SIP-T41P desk phone supports up to 6 SIP accounts, and up to 6 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-T41P is 36.80.0.79.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".

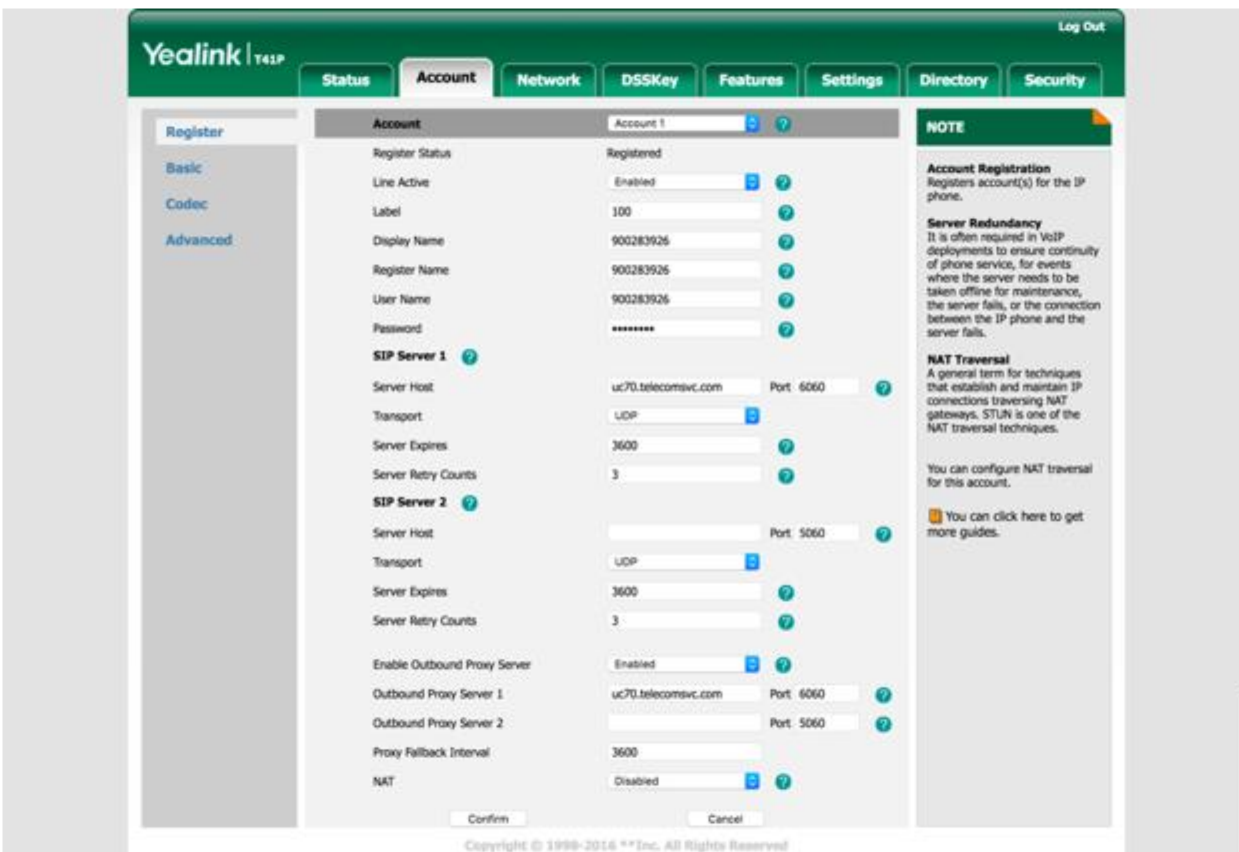
WARNING Do not refresh the web browser, or power off the phone while the firmware upgrade is in progress.

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.

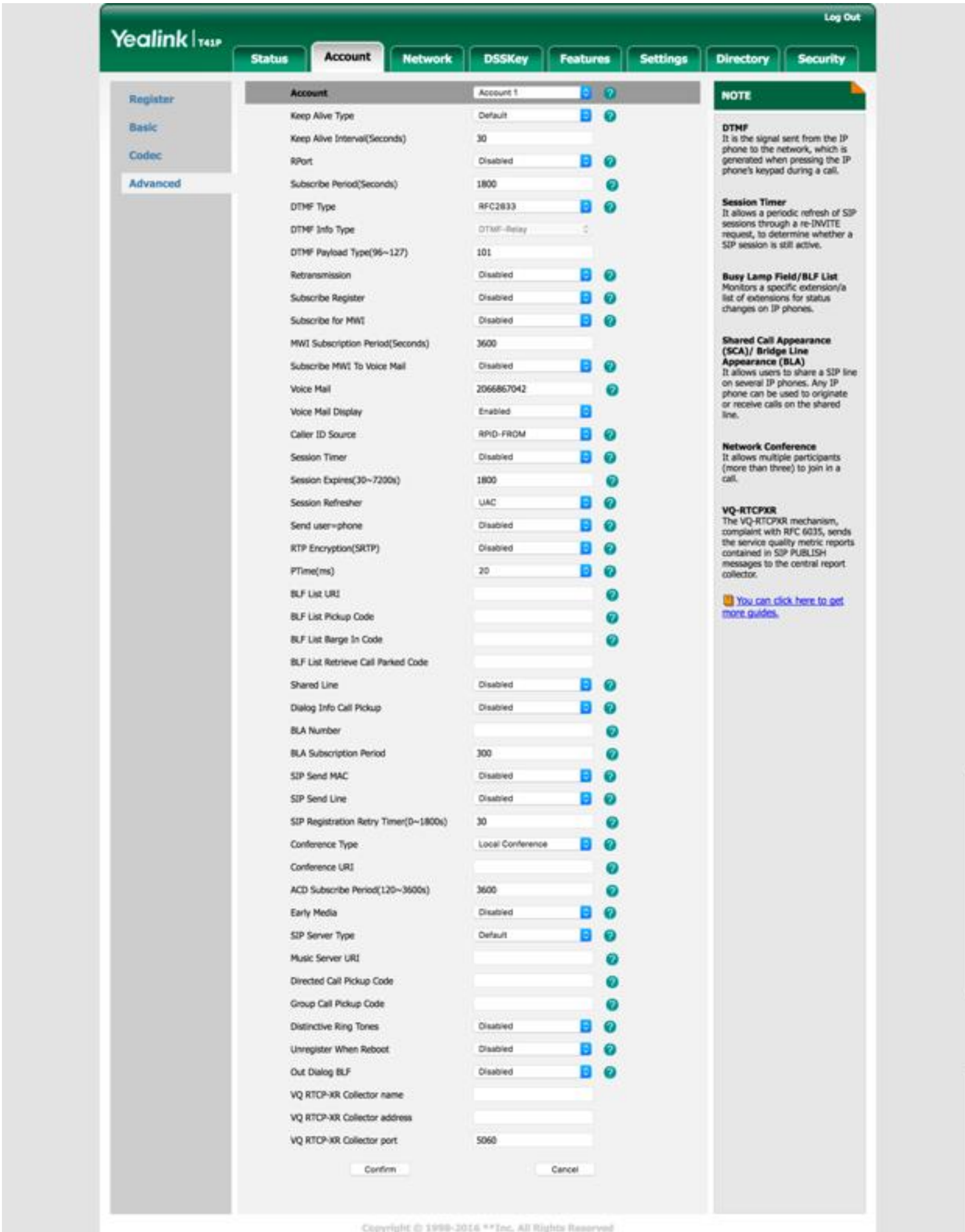


● Next select “Advanced” under the “Account” tab, and make sure that the following fields are configured as documented below.

Keep Alive Type	Default

Keep Alive Interval (Seconds)	30
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.



Yealink T41P
Log Out
Status
Account
Network
DSSKey
Features
Settings
Directory
Security
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 | 2066867042
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
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 | Disabled
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
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, complaint 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.
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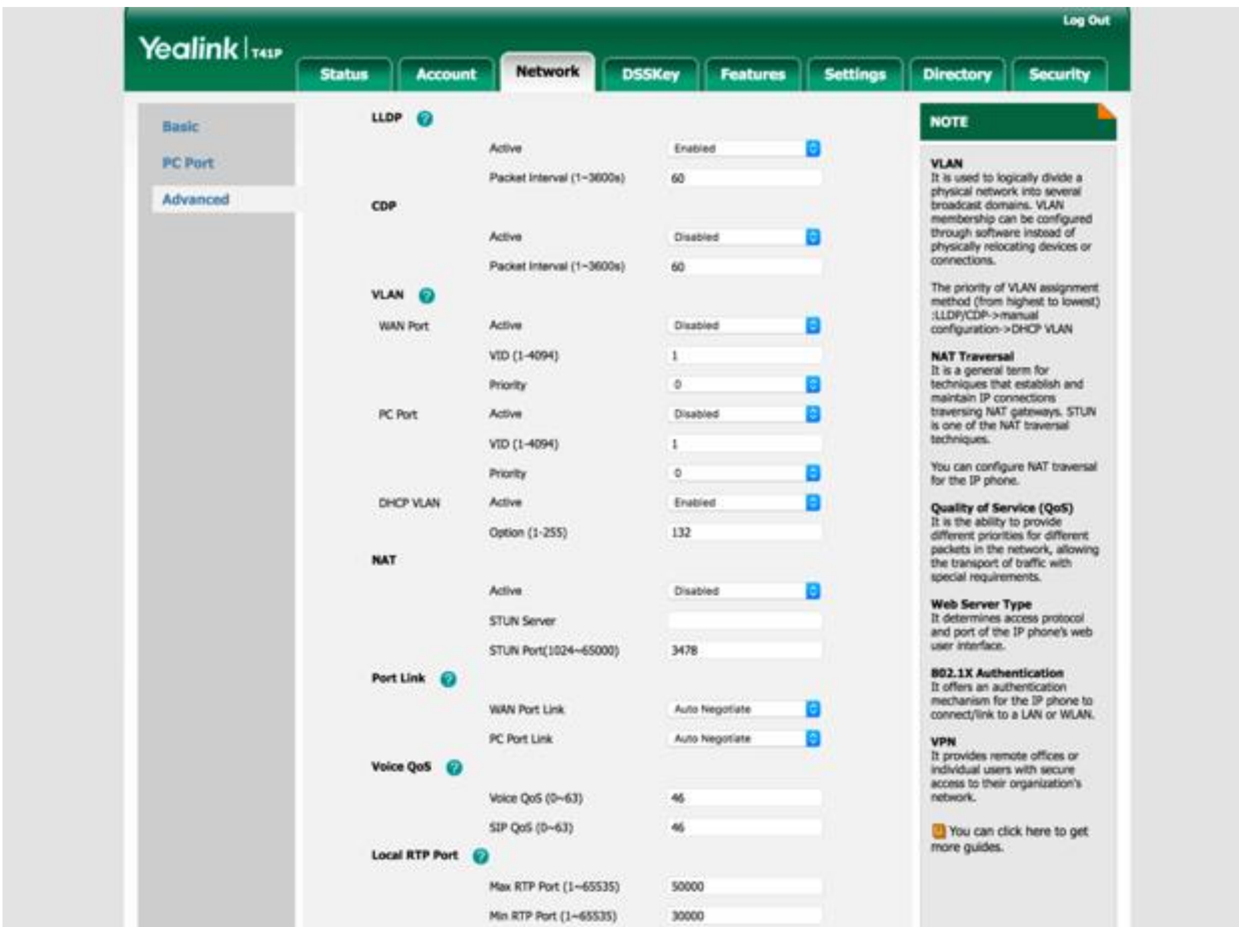
● Lastly Select the “Network” tab, then the “Advanced” sub tab, and make the following changes.

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.



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