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 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 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"

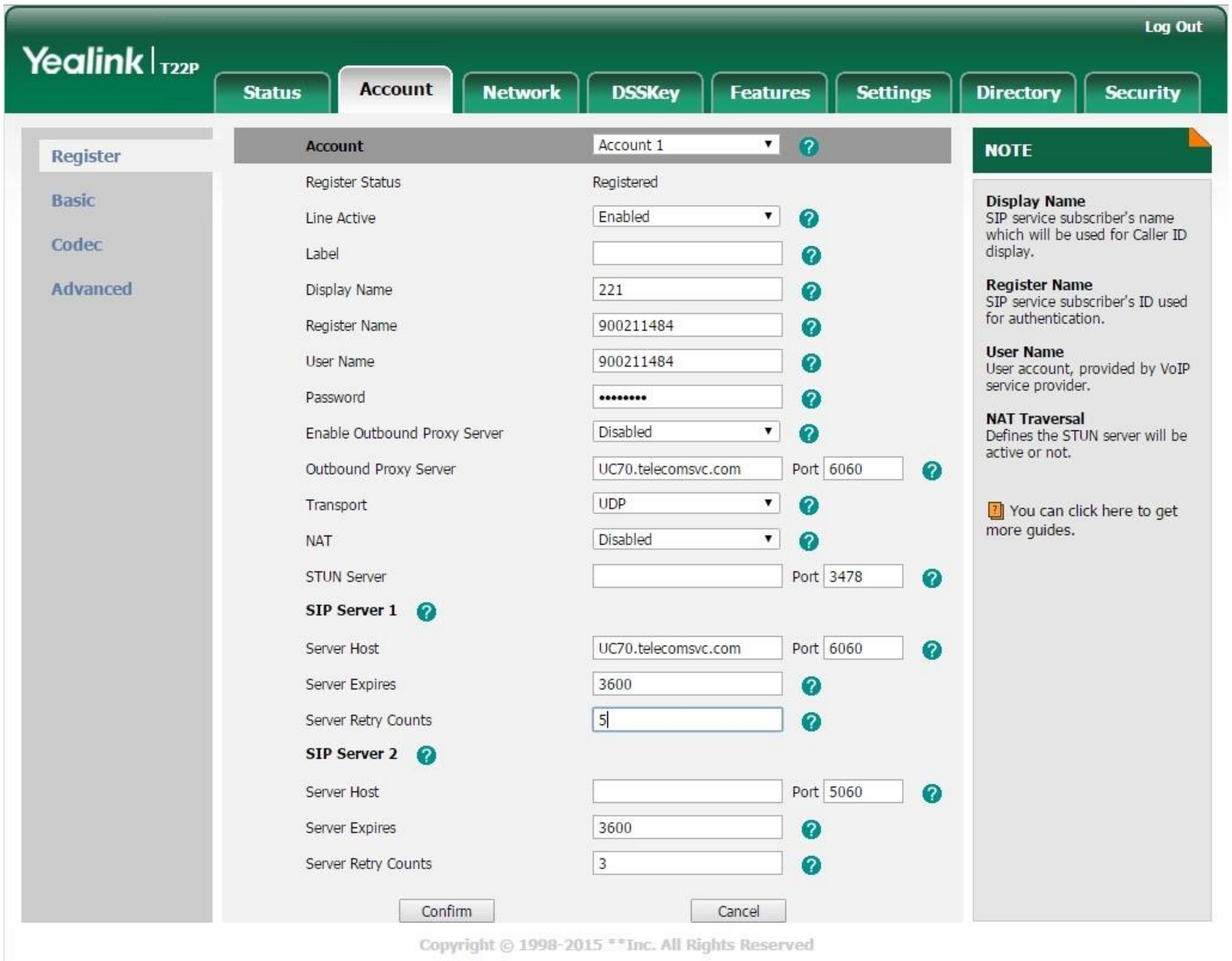
WARNING Do not refresh the web browser, or power off the phone while the firmware upgrade is in progress.

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.

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.

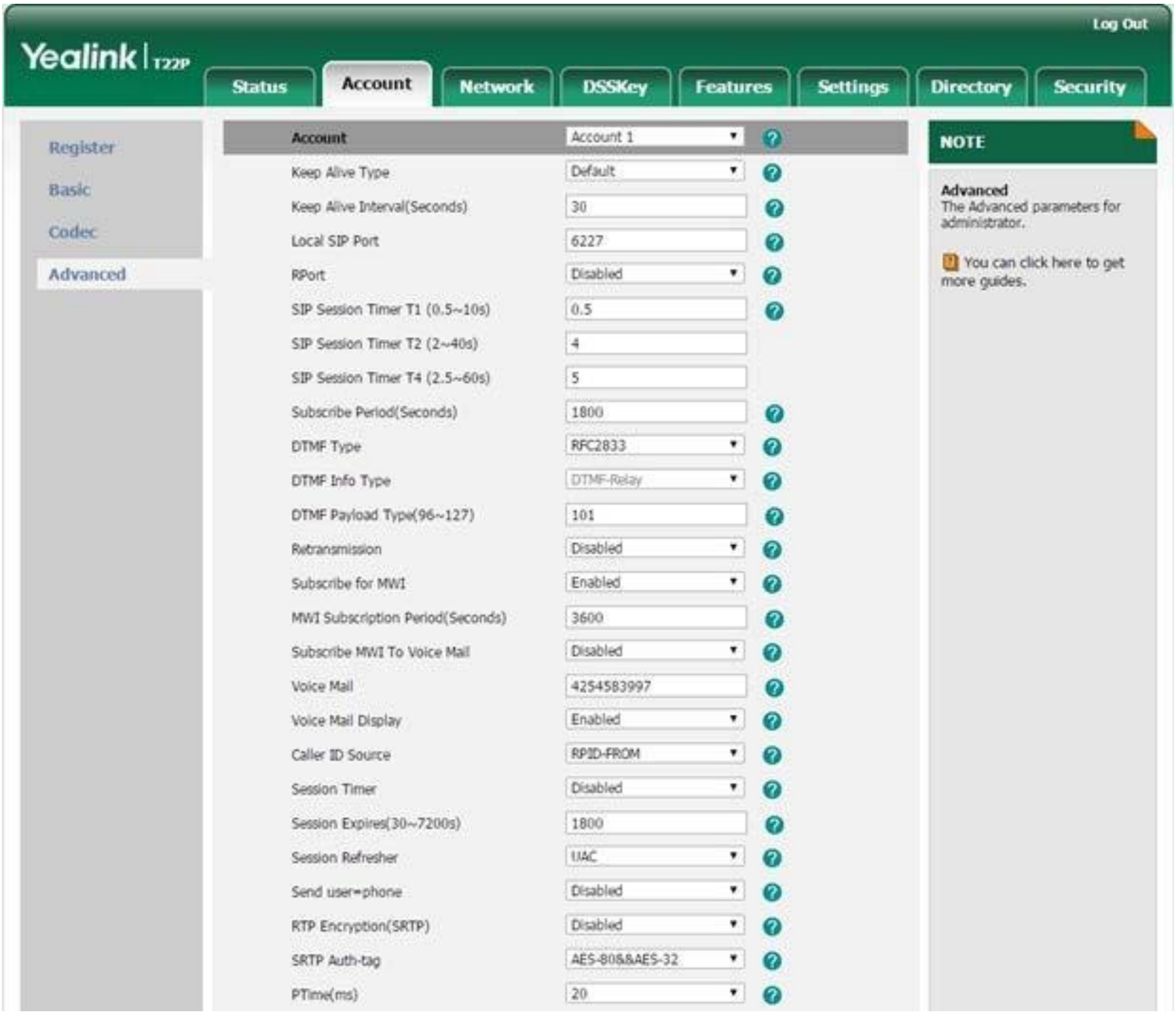


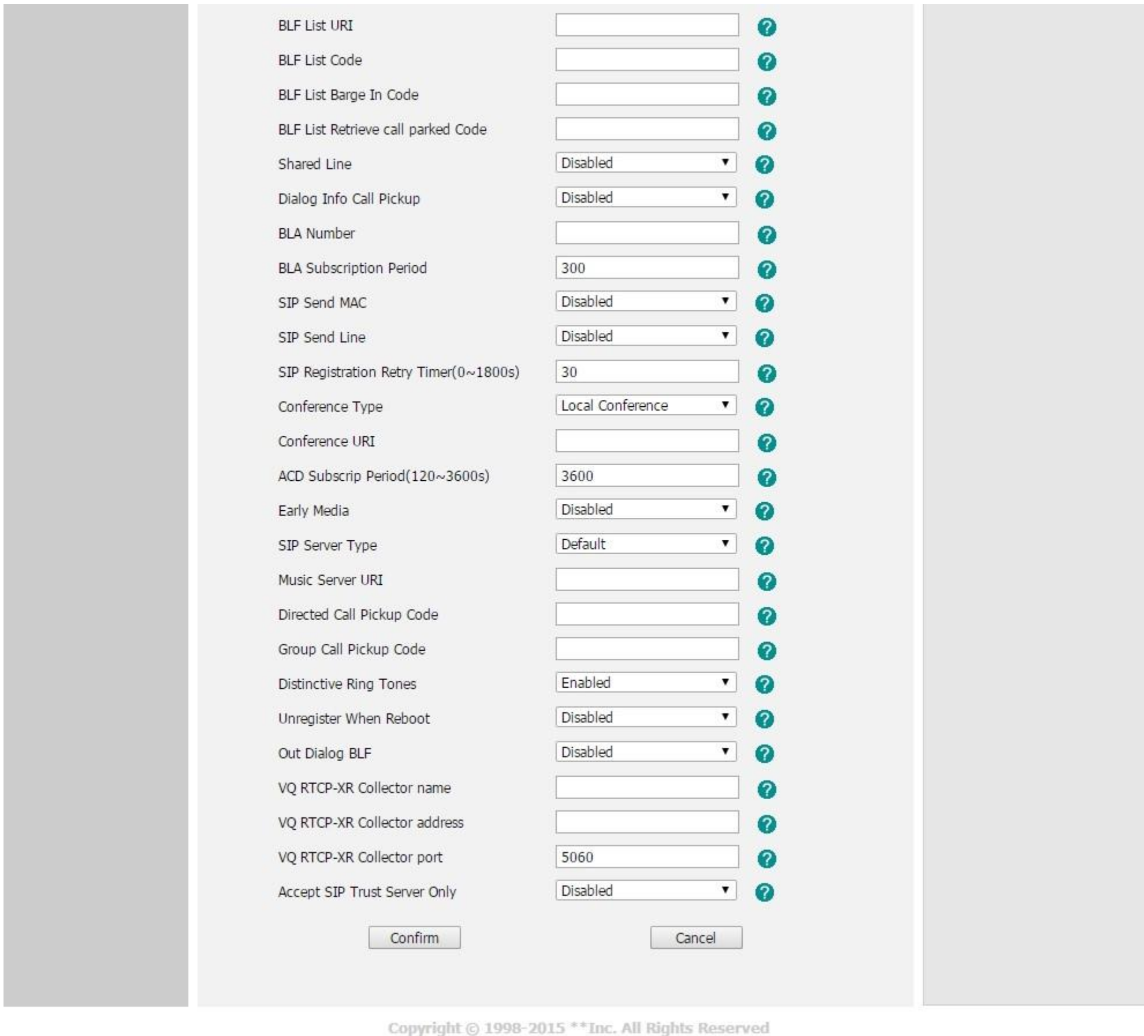
- Next select "Advanced" under the "Account" tab, and make sure that the following fields are configured as documented below.

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

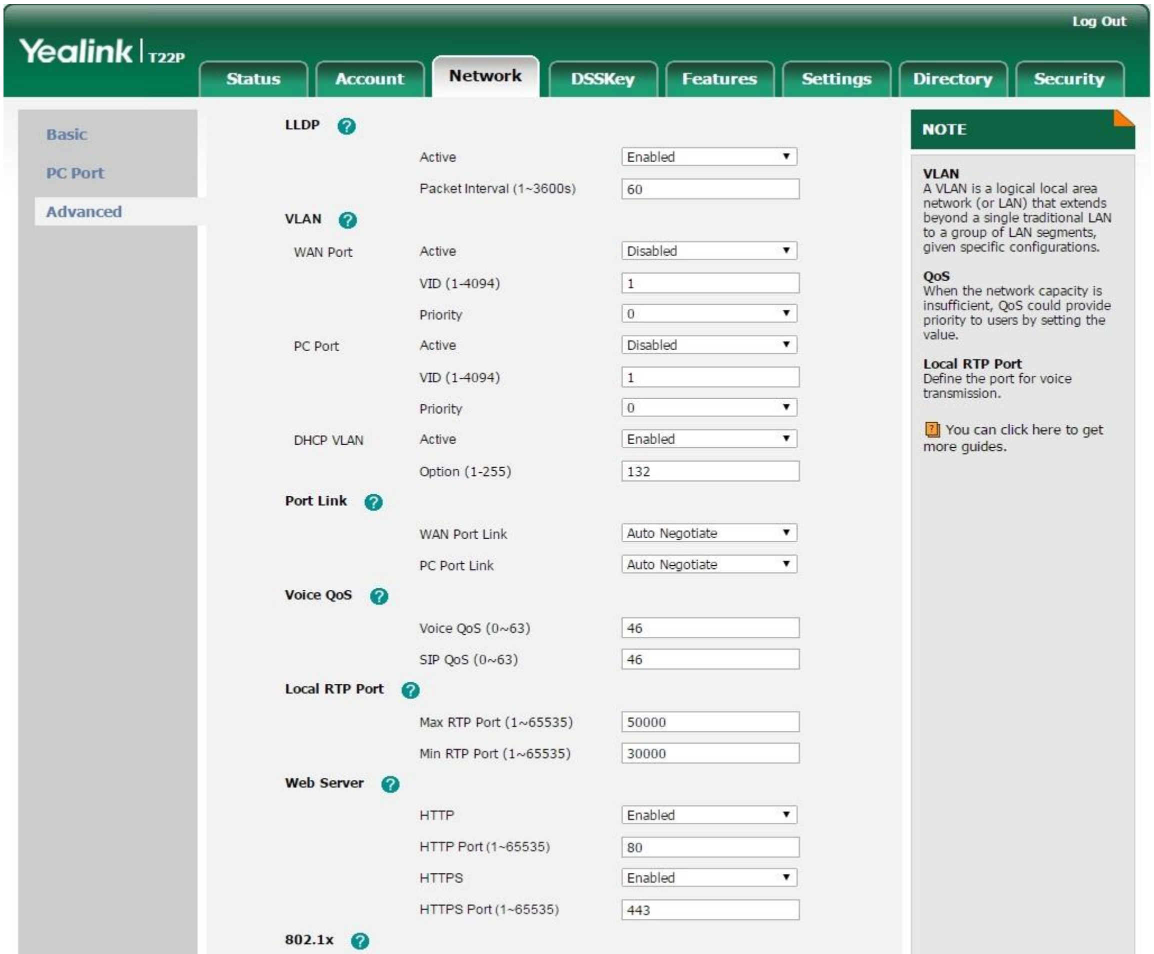




- Lastly Select the “Network” tab, then the “Advanced” sub tab, and make the following changes.

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 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