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".

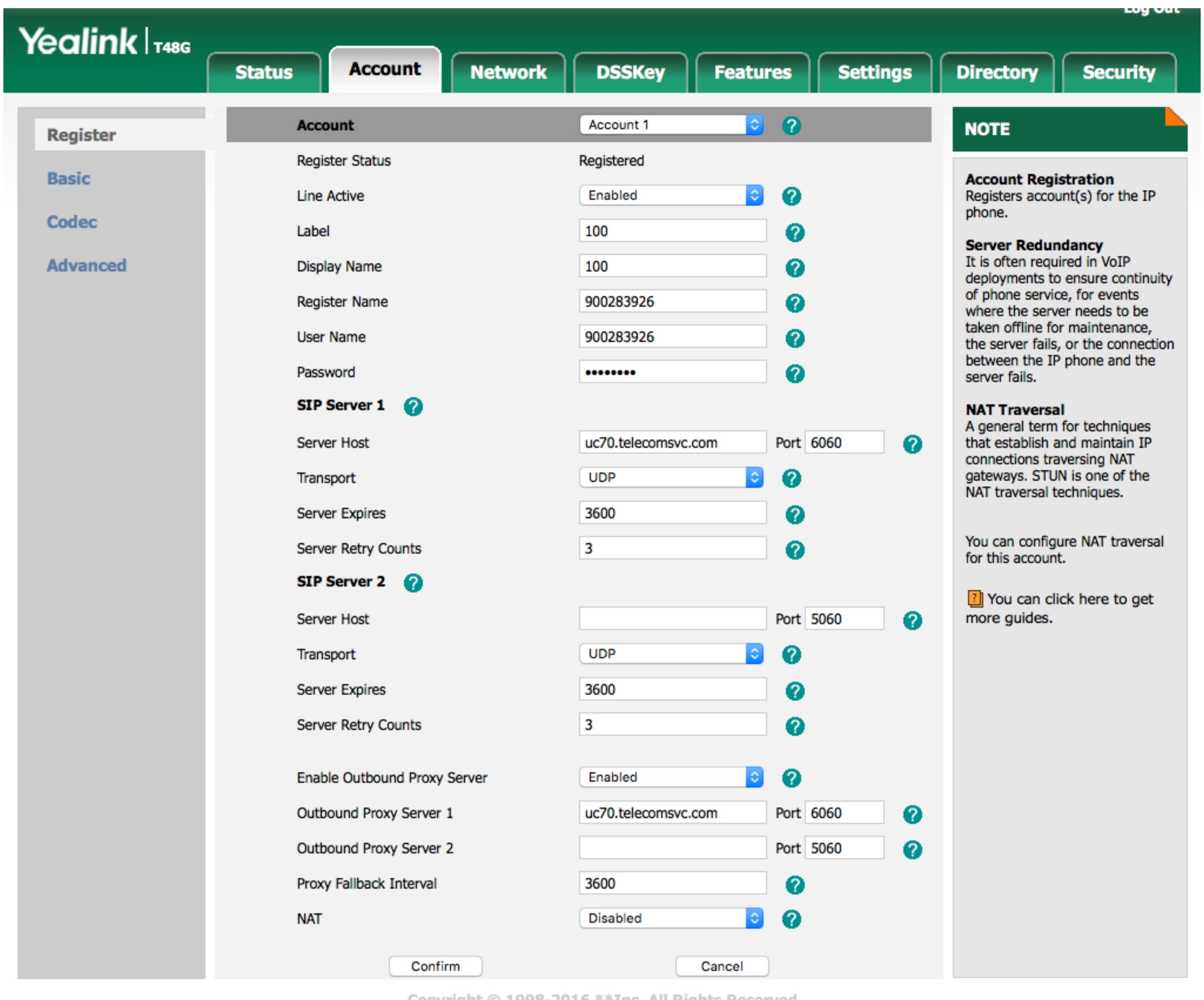
***WARNING*** Do not refresh the web browser, or power off the phone while the firmware upgrade is in progress.

**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** |

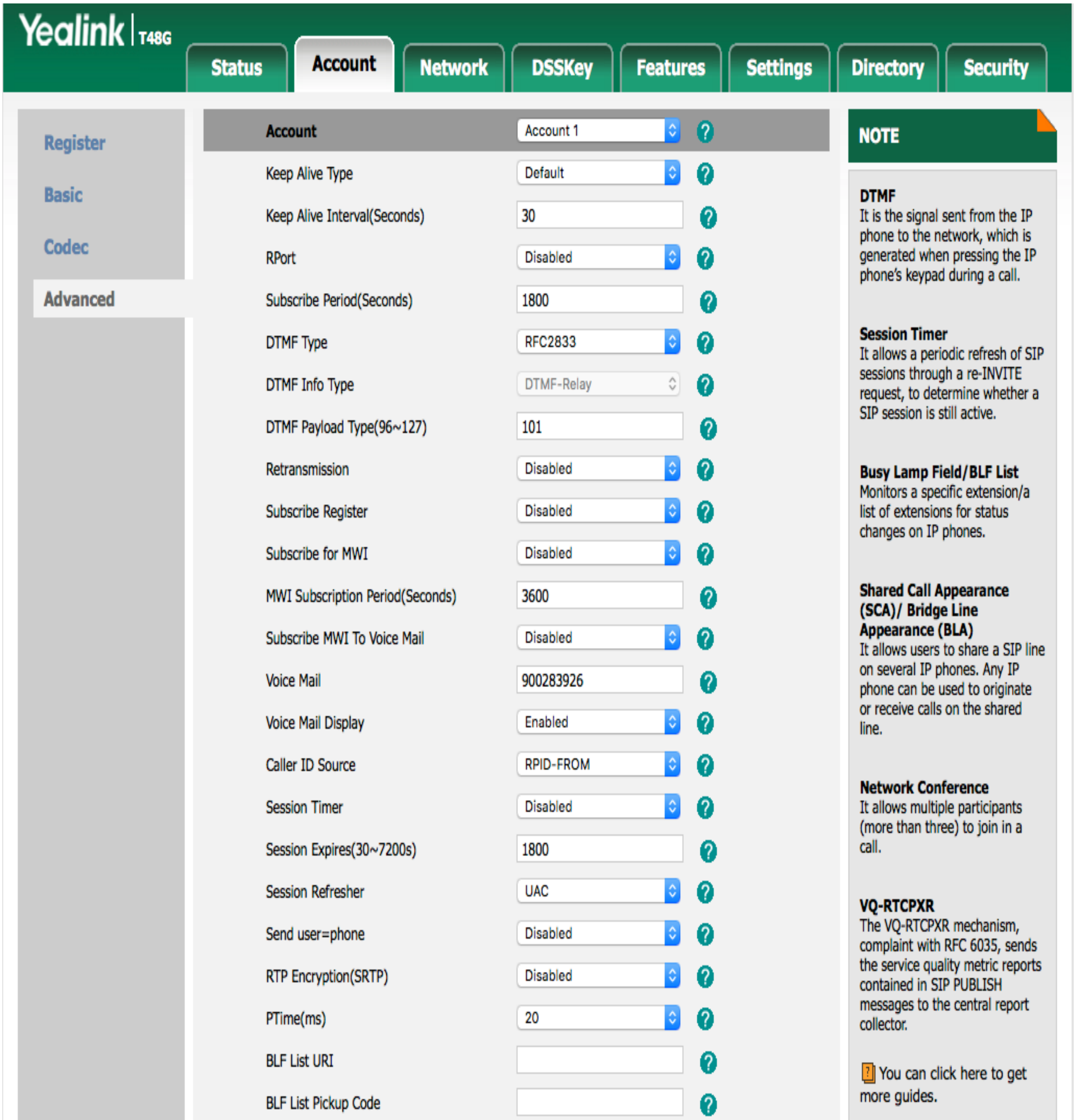
● Once all of the above information listed has been entered in, and looks similar to this click confirm.

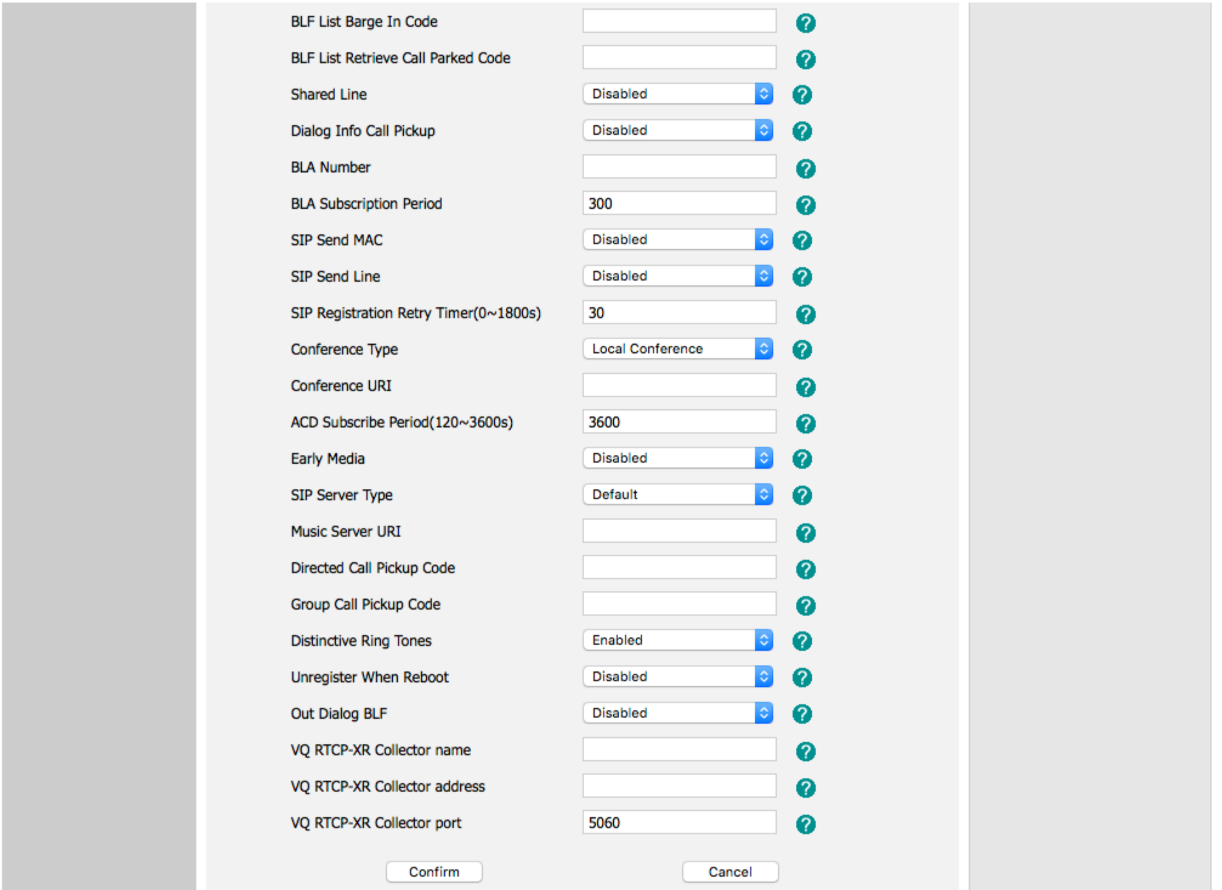


● Next select “Advanced” under the “Account” tab, and make sure that the following fields are configured as documented below.

| | |
|---|---|
| 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

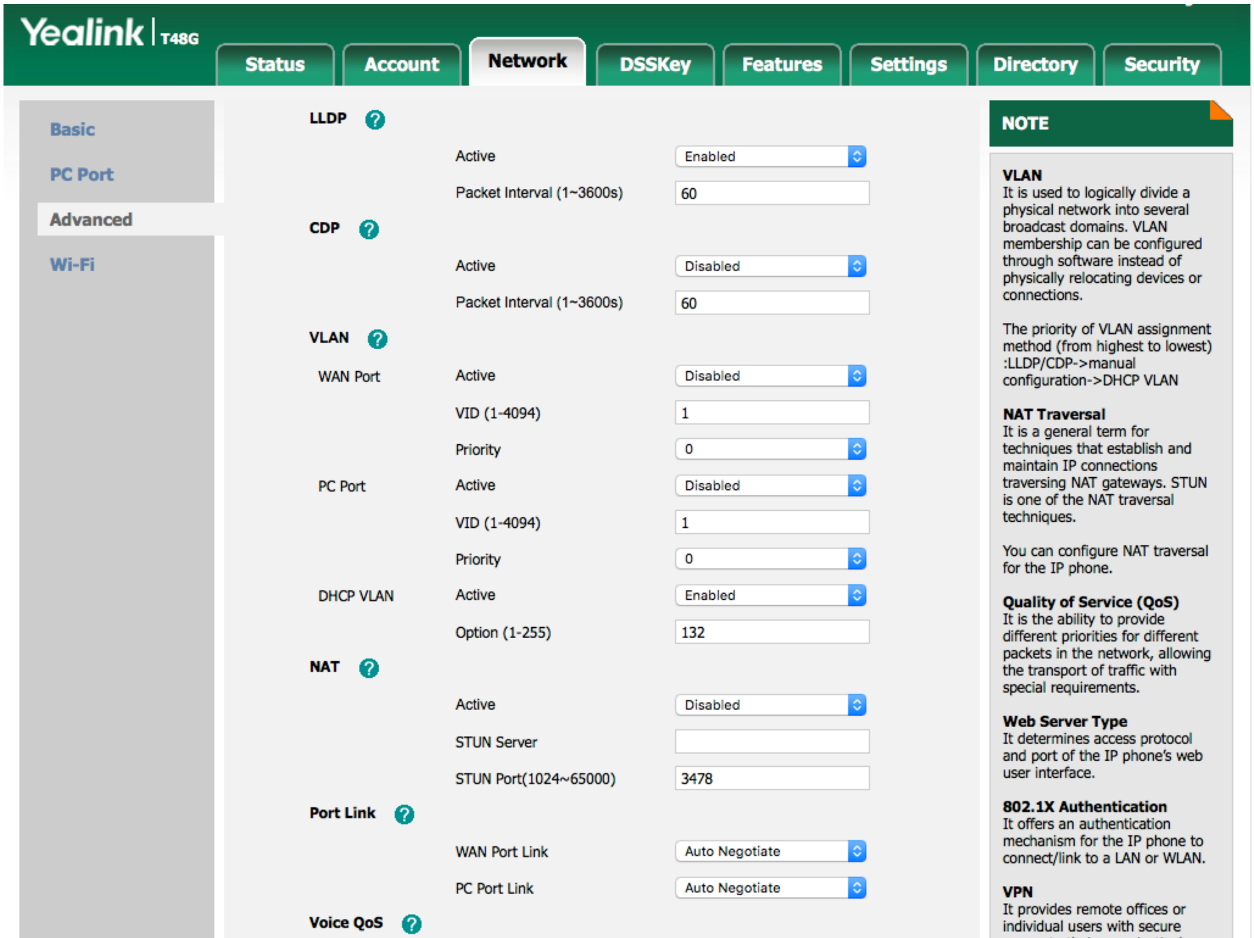

Yealink T48G
Status
Account
Network
DSSKey
Features
Settings
Directory
Security
Register
Basic
Codec
Advanced
Account
Account 1
Keep Alive Type
Default
Keep Alive Interval(Seconds)
30
RPort
Disabled
Subscribe Period(Seconds)
1800
DTMF Type
RFC2833
DTMF Info Type
DTMF-Relay
DTMF Payload Type(96~127)
101
Retransmission
Disabled
Subscribe Register
Disabled
Subscribe for MWI
Disabled
MWI Subscription Period(Seconds)
3600
Subscribe MWI To Voice Mail
Disabled
Voice Mail
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, 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.

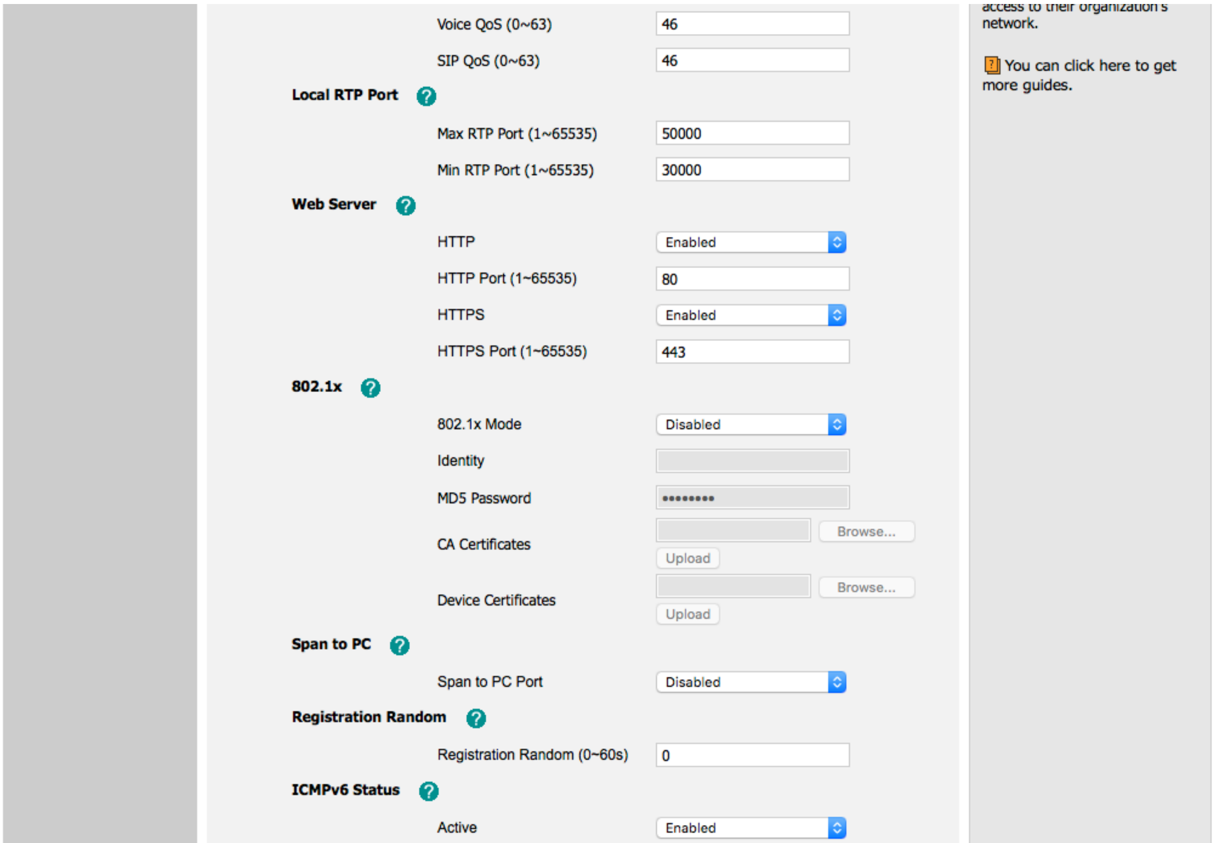



- Lastly Select the “Network” tab, then the “Advanced” sub tab, and make the following changes.

| | |
|---|---|
| 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 T48G
Status
Account
Network
DSSKey
Features
Settings
Directory
Security
Basic
PC Port
Advanced
Wi-Fi
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
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




● 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