

# IP Phone

## User Manual



# Key Feature

## New Feature

- Six lines: The phone supports up to 6 VoIP lines, each with a stand-alone configuration.
- HD Voice: The phone supports advanced wideband codec G.722 & G.722.2 and full-duplex hands-free speakerphone with enhanced AEC, delivering voice clarity.
- Excellent Level phone: 16 programmable function keys with LCD, 4 soft keys, BLF which increase working efficiency.
- Intuitive User Interface: Featuring a backlit 320×160 graphical LCD display, the phone delivers all of its capability through an intuitive interface.
- Value-Added services: online advertisement, SMS, and voice mail etc.

**Note:** these functions are available if service provider supports them.

## Network Feature

- Supports SIP 2.0 (RFC3261) protocol.
- Supports NAT transverse: STUN mode.
- IP Assignment: Static IP/ DHCP/PPPoE.
- Supports in-band DTMF and out-of band RFC2833 DTMF.
- Supports Proxy mode and peer-to-peer SIP link mode.
- Supports standard encryption and authentication (MD5 and MD5-sess).

## **Voice Feature**

- GIPS voice engine embedded to generate stable and clear voice quality.
- Voice Codec: G.722, G.711, G.729AB, G.726, G.723.1.
- Supports VAD, CNG, AEC, AGC and Volume adjustment.

## **Phone Feature**

- Large graphic LCD with blue backlight supports multi-language.
- Call hold, call waiting, call forward, call transfer, 3-way conference, auto answer and Hotline settings.
- Supports Caller ID/Name display and DND.
- Supports phone book, speed dial, call list, volume adjustment and rings selection.

## **Management Feature**

- The phone can be configured via keypad, web browser or remote.
- Firmware can be upgraded through HTTP, FTP or TFTP.
- Automated provisioning and software upgrading even through firewall/NAT.
- Phone can be diagnosed and configured by remote.

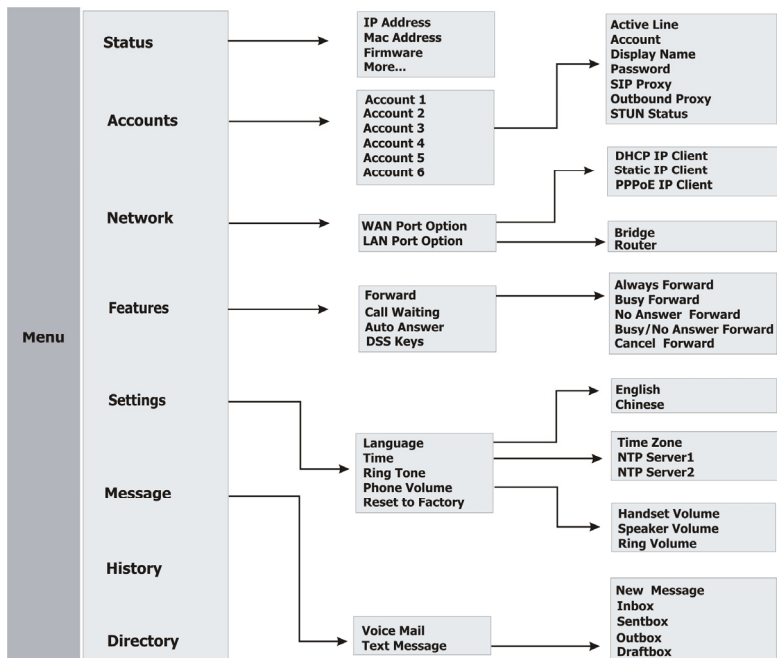
## **Physical Feature**

- Two RJ45 ports: Dual 10M/100M auto-sensing, with router built-in, one for internet, the other for PC.
- LCD: 320 x 160 dot matrix graphic LCD with white backlight, supports multi-language.
- Power adaptor: Input: AC 100~240V, output: DC 5V/1.2A.
- Operating Temperature: 0℃~40℃.
- Power over Ethernet (Optional).

## **Package Content**

- One SIP phone Main Body
- One Stand of the Phone
- One Handset
- One Handset Cable
- One Universal Power Adaptor
- One Ethernet Cable
- One User Manual

# Menu Guide



## Check IP Address

- Check WAN (Internet) IP address: press **Menu** and **Enter** to check the IP address.
- Check LAN IP address: press **Menu** and then press **▼** key twice to go to **Network**, select **LAN Port Option**, go to **Router** to check the IP address. The default is 10.0.0.1.

## Default Account and Password

### User:

Account: user



Password: user

### Administrator:

Account: admin

Password: admin

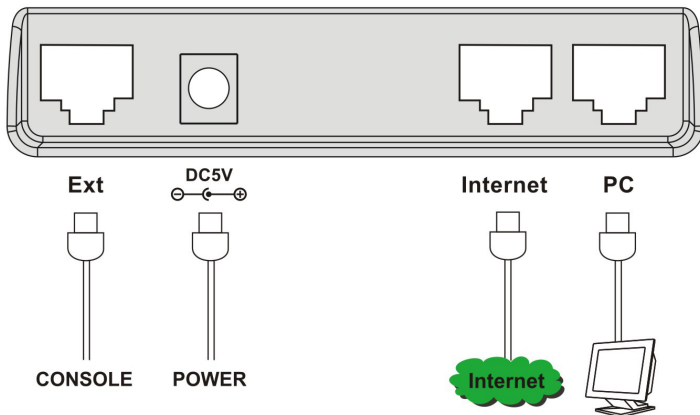
# Keypads

Key	Description
0 ~ 9	Input number and alphabet
*.	Input * and other special characters
#SEND	Start dialing process
-  +	Adjust the volume
◀/▶/▼/▲	Navigation key
OK	Enter submenu Confirm
X	Exit
MESSAGE	Connect to the voice mail or text message
HEADSET	Switch to headset mode
CONF	Start 3-way conference
HOLD	Hold the call during a conversation
MUTE	Mute the microphone during a conversation
TRAN	Transfer a call during a conversation
RD	Redial
	Speaker (Hand free) key

# Quick Install

## Connecting Your Phone

Please install the phone as the connection chart below:

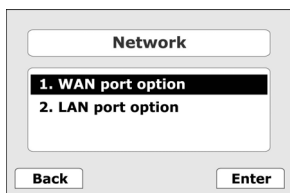


## **Method 1:**

### Configuring by Phone Keypad

#### 1. Configure Network

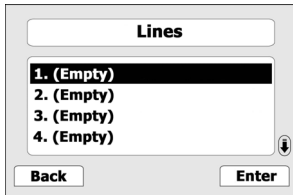
- Press **Menu** and then press ▼ key twice to go to **Network**, then press **Enter** or **OK** key to select, the LCD will show as below:



- Choose your **WAN port option** (INTERNET) connection type. The default is DHCP.

## 2. Register Account

Press **Menu** and **▼** key once to go to **Accounts**, then press **Enter** or **OK** key to select.



- a) Select line 1 and press **Enter**.
- b) Press **◀/▶** key to select **Disable** or **Enable** to set the status of this line.
- c) Press **▼** key to **Account** and enter the phone number to set account.
- d) Follow step c) to set the **Display name** and **Password**.
- e) Configure **SIP Proxy**, enter the address of your registrar SIP phone Server.
- f) Configure **Outbound Proxy**, if the service provider supports Outbound, please configure outbound server information. otherwise, leave it blank.
- g) Configure **STUN Status**, press **◀▶** key to enable or disable STUN, Normally configure the STUN as Disable.
- h) Press **Save** to return.

If you have another account, you can repeat the steps, this phone can support 6 accounts at most.

**Note:** You can get the information from your service provider. If you do not have a display name, you can use phone number as your display name.

## 3. Make Calls

Press **Speaker** key or Pick up the handset, dial the number, then press the **Send/#send** key to dial out.



## Method 2:

### Configuring by Webpage

#### 1. Login the Webpage

- Login via LAN port
  - a) Connect the PC to LAN port of the phone. The default IP address of LAN port is 10.0.0.1.
  - b) Open web browser and input **http://10.0.0.1**.
  - c) Enter the account and password (default account and password are admin).
- Login via WAN (Internet) port
  - a) Connect the PC and SIP phone with router.
  - b) Check WAN (Internet) port IP address.
  - c) Open web browser and input **http://WAN-ip-address**.
  - d) Enter the account and password (default account and password are admin).

#### 2. Network Configuration

Select Network to configure WAN port connection type.

The screenshot shows the Yealink web interface for network configuration. The 'Network' tab is active, and the 'Internet Port (WAN)' connection type is selected. The interface is divided into three main sections for IP address configuration:

- Obtain an IP Address Automatically:** A radio button option that is currently selected.
- Use the Following IP Address:** A radio button option with input fields for IP Address, Subnet Mask, Default Gateway, Primary DNS, and Secondary DNS.
- Behind xDSL Modem (PPPoE):** A radio button option with input fields for User and Password.

At the bottom, there are 'Confirm' and 'Cancel' buttons. On the right side, a 'NOTE' section provides additional information:

- Obtain an IP Address Automatically:** The unit will acquire its IP address from the DHCP server.
- Use the Following IP Address:** Configure the IP address, Subnet Mask, Default Router IP address, Primary DNS, Secondary DNS fields by hand.
- Behind xDSL Modem (PPPoE):** This setting provide by DSL.

### 3. Register Account

The screenshot shows the Yealink web interface with the 'Account' tab selected. The interface includes a top navigation bar with tabs for Status, Account, Network, Phone, Contacts, Upgrade, and Security. Below the navigation bar, there are links for Account 1 through Account 6. The main content area is divided into sections: 'Basic >>', 'Codecs >>', and 'Advanced >>'. The 'Basic >>' section contains the following settings:

Register Status	REGISTERED
Line Active	<input checked="" type="checkbox"/> On <input type="checkbox"/> Off
Display Name	<input type="text"/>
User Name	<input type="text"/>
Register Name	<input type="text"/>
Password	<input type="text"/>
SIP Server	192.168.1.199 Port:5060
Enable Outbound Proxy Server	Disabled
Outbound Proxy Server	<input type="text"/> Port:5060
NAT Traversal	STUN
STUN Server	stun.xten.com Port:3478
Voice Mail	*97
Proxy Require	<input type="text"/>

Below the 'Basic >>' section are 'Codecs >>' and 'Advanced >>' sections. At the bottom of the form are 'Confirm' and 'Cancel' buttons. On the right side, there is a 'NOTE' section with the following information:

**NOTE**

**Display Name:** SIP service subscribers name which will be used for Caller ID display.

















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

**Register Name:** SIP service subscribers Authenticate ID used for authentication.

You may get account information from your service provider. Press **Confirm** button to save the settings.


Wait a moment for registering to the server, then return to Account page to check the register status. If it displays "Registered", you can make calls now.

## ICON on the LCD

	Network status icon: Flash in the case of Ethernet linking failure
	Register status icon: fail to register to the server.
	Registering status icon
	Registered status icon
	Missed calls
	Call in
	Call out
	All kinds of characters input mode icon, press this key to select input methods
	Digital input
	Small letter input
	Capital letter input
	Mute microphone
	Call hold
	Voice mail
	SMS
	Call forward
	DND (Don't disturb) (图标错误)
	Auto Answer
	Handset
	Headset

# Basic Functions


## Using the Handset or Speaker

Using speaker: to place and answer calls using the speaker, press the **Answer**/ key.


Using the handset: to place and answer calls using the handset, simply lift the handset.

Using the headset: press **Headset** key to switch to headset mode during the call.


## Making Calls

- Direct dial telephone number: pick up the handset or press the , enter the phone numbers and press the **Send** key.
- Redial: press the **RD** to redial the last number.
- Dial from phone book: press the **Directory** to review the phone book, press **Send** to dial out the desired number.
- Dial from call list: press the **History** to review call list, press **Send** to dial out the desired number.
- Dial from DSS key: press the DSS key to dial the number you have set out.

## Receiving Calls

When the phone rings, pick up the handset or press **Answer**/ to answer the call.

## Call Hold

During a call, press **Hold** key to hold, and the hold icon  will be shown. Press **Resume** key to return to the call.

## **Call Waiting**

- Press **Answer** key during a call, the first call is put on hold. To reject the new call, press the **Reject**. To forward the new call, press **Forward** key, dial the forward number then press **Send** key.
- Switch between the two calls, press ▼/▲ and Resume key. To End the active call, just hang up the phone or press **End Call**.

## **Call Transfer**

The phone supports both Blind **Transfer** and Attended **Transfer**.

- **Blind Transfer**

During a call, press **Transfer/TRAN** key, dial the phone number and then hang up to complete the Transfer.

- **Attended Transfer**

During a call, press **Transfer/TRAN** key, dial the second person's phone number and press **Send**, you will talk to the second person. Then hang up to connect this call to the first person and complete the transfer.

## **3-Way Conference**

During a call, press **Conference/CONF** key to hold on the call, and dial the second person's number. When the second person accepts the call, the three parties will be participating in a conference call automatically.

**Note:** When you hang up, the other two parties will be disconnected.

## **Call Forward**

The phone supports automatic forward and manual forward.


Automatic forward:

- a) Press **Menu** and **▼** key to Features, then press **Enter** twice.
- b) Press **▼/▲** key to switch between 1.Always forward/ 2.Busy forward/3. No answer forward/4. Busy/No Answer forward/5. Cancel forward, choose one of these options and press **Enter/ok** key.
- c) Input the forward number, then press **Save/ok** key to save your configuration.


Manual forward:

When a call comes in, press **Forward** key and dial your forward number, then the call will be forward to the designed number.

### **Do Not Disturb**


- Press **DND** to start this function.
- Press **DND**/ or pick up the handset to cancel this function.

### **Voice Mail**

When the phone is idle, the icon  shows you have a missed message on the server. Press **Connect** key on the keyboard, you can enter voice mail system, then follow the prompt voice, you can listen to your messages.

**Note:** You should get the voice mail number form your service provider.

### **Missed Call**


When you have missed calls, the screen will show the symbol  and how many missed calls you have. Press **View** to see the details or press **Exit** to omit this information.

## **BLF**


The buttons can be configured for Asterisk Busy Line Field function with specified account. Choose the key mode as BLF, fill in the extension's number and select the line ID, then you can see the status of the extension at any moment.

**Note:** This function must be supported by the ISP.


## **Volume Adjustment**

Press -  + key to adjust the volume.

## **Ring selection**

- a) Press **Menu** and **▼** key to **Settings**, then press **Enter/ok**.
- b) Press **▼** key twice to **Ring Tone**, then press **Enter/ok**.
- c) Press **▼/▲** key to choose a ring tone, then press **Save/ok** to save your configuration and return to the Phone Setting interface.
- d) To adjust the ring volume, please press -  + key on the phone when the phone is idle.

## **SMS**

The icon  shows you have a text message, press **MESSAGE** key on the keyboard to read it.

**Note:** This function must be supported by the ISP.

## **Auto Provision**

The phone can auto update and configure settings from the server. Make sure that you have enabled this function and entered the auto provision server and selected the method of the auto provision in the webpage.

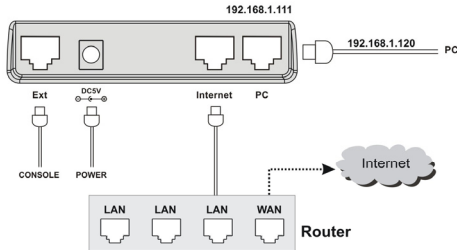
**Note:** you should get the auto provision server from your service provider.

# Configure with Web Browser

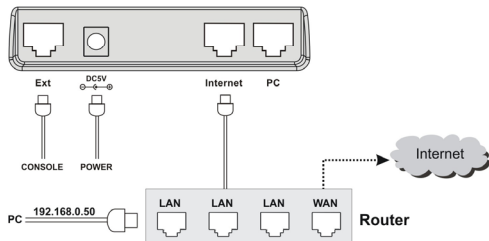
The phone has an embedded Web server that will respond to HTTP requests.

## Login the Webpage

### Method A



### Method B



Choose your LAN port connect type:

Environment	LAN port type
Office Mode	Bridge
	Router
Home Mode	Router



## 1. Have No Router

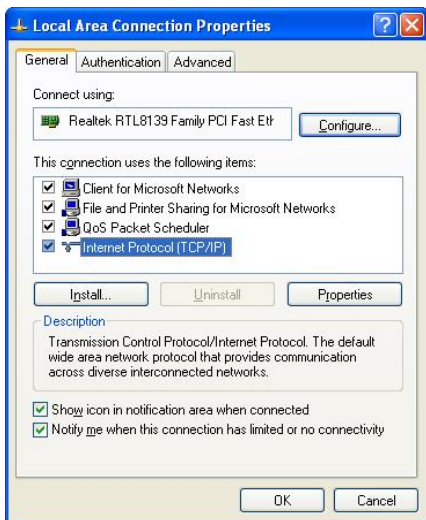
### • Configure as router

a) Connect the phone with PC in method A, and set your LAN port as a router.

b) Configure your PC's IP address as below:

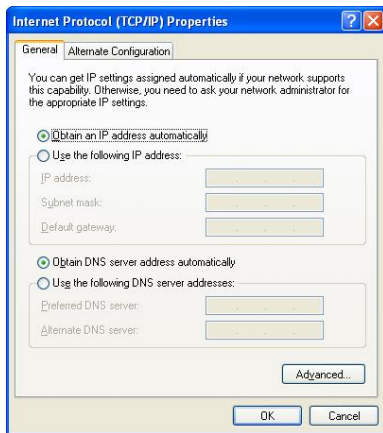
Right click the *Network* of your PC and select property, then right click *local area connection* icon and select property.

Access to *Local Area Connection Properties* window:



Select *Internet Protocol (TCP/IP)* and click Properties.

Configure your PC as below:



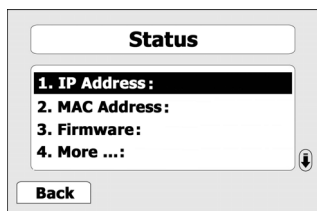
After all the settings, you could access the webpage of the phone by the URL: <http://10.0.0.1>.

## • Configure as bridge

- Connect the phone with PC in method A and set your LAN port as bridge.
- Press **Menu** and **Enter** to check the IP address of Internet (WAN) port.
- Open web browser and input **<http://WAN-ip-address>** to access the webpage.

## 2. Have Router

Connect the phone with PC in method B. Please do not configure the LAN port. Press **Menu** and **Enter** to check the IP address.



Open web browser and input http://WAN-ip-address to access the webpage. For example: http://192.168.0.24. You will see the login screen as below.



Default account and password:

**User:**

Account: user

Password: user

**Administrator:**

Account: admin

Password: admin

## Check the Status

This page shows the status of the phone.

Version	
Firmware Version	2.0.0.4
Hardware Version	1.0.0.0

Network	
WAN Port Type	Static IP
WAN IP Address	192.168.0.24
Subnet Mask	255.255.240.0
MAC Address	00-15-65-c0-39-00
Link Status	Connected
LAN IP Address	0.0.0.0
Device Type	Bridge
DHCP Server Status(PC)	Enabled

**NOTE**

**Version:**  
This option shows you the version of firmware.

**Network:**  
This option shows you the information about WAN port and LAN port.

# Individual Account Settings

## 1. Set Basic Information

**Yealink**  
UNIVERSAL SYSTEM

Status | **Account** | Network | Phone | Contacts | Upgrade | Security

Account 1 | Account 2 | Account 3 | Account 4 | Account 5 | Account 6

**Basic >>**

Register Status: REGISTERED

Line Active:  On  Off

Display Name:

User Name:

Register Name:

Password:

SIP Server: 192.168.1.199 Port: 5060

Enable Outbound Proxy Server: Disabled

Outbound Proxy Server:  Port: 5060

NAT Traversal: STUN

STUN Server: stun.xten.com Port: 3478

Voice Mail: \*97

Proxy Require:

**NOTE**

**Display Name:** SIP service subscribers name which will be used for Caller ID display.

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

**Register Name:** SIP service subscribers Authenticate ID used for authentication.

**Codecs >>**

**Advanced >>**

Confirm Cancel

The following table describes the labels in this screen.

Field Name	Description
<i>Register Status</i>	Show the register status of the account.
<i>Line Active</i>	Enable or disable this line.
<i>Password</i>	Account password.
<i>SIP Sever</i>	SIP server's IP address or Domain name provided by VoIP service provider.
<i>SIP port</i>	SIP port, the default is 5060.
<i>Enable Outbound Proxy Server</i>	Defines whether to active the outbound server. If your server provider does not inform you about that, leave it as disabled.

<i>Outbound Proxy Server</i>	Outbound server's IP address or Domain name provided by VoIP service provider.
<i>Outbound Port</i>	<i>Outbound Port</i> , the default is 5060.
<i>NAT Traversal</i>	Defines whether to active the STUN server. If your server provider doesn't inform you about the STUN server, leave it disable.
<i>STUN Server</i>	SIP Extension notifies the SIP server that the unit is behind the NAT/Firewall.
<i>STUN Port</i>	This parameter defines the STUN server port.
<i>Voice Mail</i>	Voice Mail number provided by VoIP service provider.
<i>Proxy Require</i>	You shouldn't config it unless you are register to the Nortel server.

## 2. Set Codecs

The screenshot displays the Yealink SIP Manager web interface for configuring codecs. The 'Codecs' section is active, showing a list of disabled codecs (G722) and enabled codecs (PCMU, PCMA, G729). A note on the right side of the page provides additional information:

- Display Name:** SIP service subscribers name which will be used for Caller ID display.
- User Name:** User account information, provided by VoIP service provider.
- Register Name:** SIP service subscribers Authentic ID used for authentication.

### 3. Set Advanced Information

**Yealink**  
SIP Manager

Status | **Account** | Network | Phone | Contacts | Upgrade | Security

Account 1 | Account 2 | Account 3 | Account 4 | Account 5 | Account 6

**Basic >>**

**Codecs >>**

**Advanced >>**

UDP Keep-alive Message	Enabled	
UDP Keep-alive Interval	30	(seconds)
Login Expire	3600	(seconds)
Local SIP Port	5060	
Local RTP Port	11780	
RPort	Disabled	
SIP Session Timer T1	0.5	(seconds)
SIP Session Timer T2	4	(seconds)
SIP Session Timer T4	5	(seconds)
SubscribePeriod	3600	(seconds)
DTMF Type	RFC2833	

Confirm Cancel

**NOTE**

**Display Name:**  
SIP service subscribers name which will be used for Caller ID display.

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

**Register Name:**  
SIP service subscribers Authenticate ID used for authentication.

The following table describes the labels in this screen.

<b>Field Name</b>	<b>Description</b>
<i>UDP Keep-alive Message</i>	Defines whether to active the phone UDP Keep-alive mechanism. The default is Enabled.
<i>UDP Keep-alive Interval</i>	This parameter specifies how often the phone will send a packet to the SIP server. Default is 30 seconds.
<i>Login Expire</i>	This parameter specifies the time frequency that phone refreshes its registration. The default interval is 3600 seconds.
<i>Local SIP Port</i>	Local SIP port. The default min value is 5060.
<i>Local RTP Port</i>	Defines the local RTP port that the phone will listen and transmit. The default value is 11780.
<i>RPort</i>	The parameter allows you configuring the proxy to send responses back to a particular address and port. The default is disabled.
<i>SIP Session Timer</i>	This document defines an extension to the Session Initiation Protocol (SIP). This extension allows for a periodic refresh of SIP sessions through a re-INVITE or UPDATE request. The refresh allows both user agents and proxies to determine if the SIP session is still active.
<i>Subscribe Period</i>	This parameter could set the period of the subscription. The default value is 3600.
<i>DTMF Type</i>	Select the DTMF type.

# Network Settings

## 1. Set WAN Port Information

The screenshot shows the 'Network' tab in the Yealink web interface. The 'Internet Port (WAN)' is selected. There are three radio button options for IP configuration: 'Obtain an IP Address Automatically', 'Use the Following IP Address', and 'Behind xDSL Modem (PPPoE)'. The 'Use the Following IP Address' option is selected. Below it are input fields for IP Address, Subnet Mask, Default Gateway, Primary DNS, and Secondary DNS. The 'Behind xDSL Modem (PPPoE)' option has input fields for User and Password. There are 'Confirm' and 'Cancel' buttons at the bottom. A 'NOTE' section on the right explains each option.

The following table describes the labels in this screen.

Field Name	Description
<i>Obtain an IP address automatically</i>	If this mode is enabled, the phone will obtain its IP address from the DHCP server.
<i>Use the following IP address</i>	If this mode is enabled, IP address, Subnet Mask, Default Router IP address, Primary DNS, Secondary DNS fields will need to be configured.
<i>Behind xDSL Modem (PPPoE)</i>	To use the PPPoE function, the PPPoE account settings need to be set. Please input the Username and the Password correctly.



## 2. Set PC Port LAN Information

The screenshot shows the Yealink web interface for configuring the PC Port (LAN). The 'Network' tab is selected, and the 'PC Port (LAN)' section is active. There are two radio buttons: 'As an Bridge' (selected) and 'As an Router'. Below 'As an Router', there are input fields for IP Address (10.0.0.1), Subnet Mask (255.255.255.0), Enable DHCP Server (Enabled), Starting IP Address (10.0.0.10), and Ending IP Address (10.0.0.100). There are 'Confirm' and 'Cancel' buttons. A 'NOTE' section on the right explains the Bridge and Router modes.

The following table describes the labels in this screen.

Field Name	Description
<i>As an Bridge</i>	If you select the Bridge mode, then the two Fast Ethernet ports will be transparent.
<i>As an Router</i>	If you select the Router mode, the SIP phone will work as a router
<i>IP address</i>	User could configure the LAN port IP address.
<i>DHCP Server</i>	If you set the DHCP server on, the device connected to the LAN port will get the IP address automatically between the start IP address and the end IP address. But if you select the bridge mode, the DHCP server can not work.

# Phone Settings

## 1. Set the Preference Settings

**Yealink**  
VoIP UC

Status | Account | Network | **Phone** | Contacts | Upgrade | Security

Preference | Features | DSS Key | SMS

Language: English

Ring Type: Ring4.wav

Time Zone: +8\_China, Philippines, Malaysia

Primary NTP Server: cn.pool.ntp.org

Secondary NTP Server: cn.pool.ntp.org

Update Interval: 1000 (seconds)

Confirm Cancel

**NOTE**

**Time Zone:**  
Choose the time zone you live in.

**NTP Server:**  
Specify the server that is used to synchronize the time of unit.

**Update Interval:**  
Specify the time frequency that unit refresh the time automatically.

## 2. Set the Features information

**Yealink**  
VoIP UC

Status | Account | Network | **Phone** | Contacts | Upgrade | Security

Preference | **Features** | DSS Key | SMS

**Forward**

Cancel Forward

Always forward to: [ ]

Busy forward to: [ ]

No answer forward to: [ ]

After ring time: 10 (seconds)

Busy/No answer forward: [ ]

After ring time: [ ] (seconds)

Call Waiting: Enabled

Auto Answer: Disabled

Confirm Cancel

**NOTE**

**Voice Mail Number:**  
Put in your voice mail number provide by VoIP service provider.

The following table describes the labels in this screen.

<b>Field Name</b>	<b>Description</b>
Call Waiting	If you disable this function, the second incoming call will be declined when you are on the call.
Auto Answer	If you set this option as Enabled, when someone calls in, the phone will be put through automatically.

### 3. Set DSS information

The screenshot shows the Yealink web interface with the 'Phone' tab selected. Under the 'DSS Key' sub-tab, there is a list of ten DSS keys (DSS Key1 to DSS Key10). Each key has a 'Key Mode' dropdown menu, all of which are currently set to 'N/A'. At the bottom of the list are 'Confirm' and 'Cancel' buttons. On the right side, there is a 'NOTE' section with the following text:

**NOTE**

**Key Mode:**  
There are four kinds of key mode for you to choose to config these ten multi-functional button.

**BLF:**  
The buttons can be configured for Asterisk Busy Line Field function with specified account. This feature must be supported by the sip server.

**Call Pickup:**  
This feature lets you use one phone to answer another phone that ringing. This feature must be supported by sip server.

### 4. Edit SMS

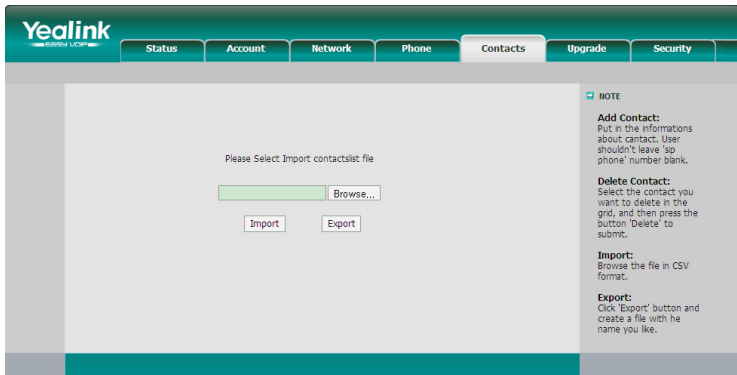
The screenshot shows the Yealink web interface with the 'Phone' tab selected. Under the 'SMS' sub-tab, there are three input fields: 'Account' (a dropdown menu), 'Number' (a text input field), and 'Message' (a large text area). At the bottom are 'Send' and 'Cancel' buttons. On the right side, there is a 'NOTE' section with the following text:

**NOTE**

**SMS Target Number:**  
Put in the phone number that you are going to send message to.

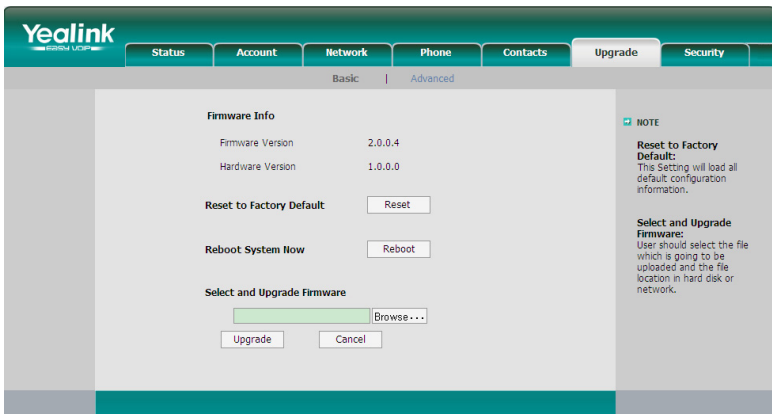
## Contacts

- Import Contact  
Click the Browse button and select the contact you want to import, then click the **Import** button.
- Export Contact  
Click the Export button and name a file you want to restore.



## Update

### 1. Set Basic Update Information



**Note:** Do not power off when you are updating the firmware.

## 2. Set Advanced Update

The screenshot shows the 'Advanced' update settings page in the Yealink web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Phone', 'Contacts', 'Upgrade', and 'Security'. The 'Upgrade' tab is active, and the 'Advanced' sub-tab is selected. The main content area contains the following fields and buttons:

- Update Via:
- Check new config:
- Scheduling (Date):  (1~30days)
- Click here to autoprovision Now:
- Export system log:
- Buttons:

A note on the right side of the page reads: **NOTE**  
**Auto Provision when Power On:**  
When you set yes, it will auto update the settings when power on.

## Security

Advanced user could change the login user type and the password in this page.

The screenshot shows the 'Security' user settings page in the Yealink web interface. The top navigation bar includes 'Status', 'Account', 'Network', 'Phone', 'Contacts', 'Upgrade', and 'Security'. The 'Security' tab is active. The main content area contains the following fields and buttons:

- User Type:  user  admin
- Old Password:
- New Password:
- Confirm Password:
- Buttons:

A note on the right side of the page reads: **NOTE**  
**Password Type:**  
Select your pinview. If you log in as a user, you couldn't change the administrator password.

If you want to configure through keypad, please check the **Menu** structure at page 4 and functions of these keys listed at page 5.

# Troubleshooting

## **I can not register to the server?**

1. Check the IP address. If you set your WAN port in DHCP mode, please make sure that your DHCP server is on.
2. Check your gateway.
3. Check your DNS server.
4. Make sure your account information is the same as you have got from your ISP.
5. Check whether the SIP server is on.
6. Check the SIP register port, the default value is 5060.

## **I can't get the IP address?**

1. Make sure you have plugged the Ethernet cable into the WAN port.
2. Make sure that the DHCP server is on, and there are available IP addresses in the server.
3. Try to set your WAN port to static IP client mode.

## **During a call, I can not hear any voice?**

1. Make sure your handset is tightly connected with the phone.
2. Check whether you have muted the conversation or not.
3. Consult the outbound server details with your ISP.

## **Have DTMF problem?**

1. Check which kind of DTMF you are using, and whether it is compatible with the server.
2. Consult the payload value with your ISP.

### **How to change the time?**

Select the time zone on the webpage.

**Note:** You can't change the time manually because that our phone will automatically get the time from the SNTP server.

### **How to answer the incoming calls during a call?**

If a call comes in when you are in a conversation, press the **Answer** key to answer the incoming call.

### **How to refuse incoming calls during a call?**

You can turn off the function of call waiting, and then our phone will refuse all the incoming calls when you are in a conversation.

### **How to send SMS?**

You could edit the SMS in the MENU-> Messages->Text Messages.

**Note:** Make sure that the SIP server you have registered supports SMS function.

### **How to update the firmware?**

1. Enter the webpage of your phone, go to Upgrade, then you can find the option "Select and Upgrade Firmware" at the bottom of the page.
2. Select the file to update, then click the Upgrade button.

**Note:** Make sure the firmware you choose is provided by VOPTech, or the device will probably crash after the update.

### **How to auto provision?**

Consult the auto provision server address with your ISP.

#### **Notice:**

This document is subjected to change without notice. The latest electronic version of this user manual is available to download from the following location: <http://www.voptech.com>.