



We Focus • We Deliver



CooFone-D60 VoIP Phone User Manual



Safety Notices

Please read the following safety notices before installing or using this phone. They are crucial for the safe and reliable operation of the device.

- Please use the external power supply that is included in the package. Other power supplies may cause damage to the phone, affect the behavior or induce noise.
- Before using the external power supply in the package, please check power voltage. Inaccurate power voltage may cause fire and damage.
- Please do not damage the power cord. If power cord or plug is impaired, it may cause fire or electric shock.
- The plug-socket combination must be accessible at all times because it serves as the main disconnecting device.
- Do not drop, knock or shake it. Rough handling will break internal circuit boards.
- Do not install the device in places where there is direct sunlight. Also do not put the device on carpets or cushions. It may cause fire or breakdown.
- Avoid exposure the phone to high temperature, below 0°C or high humidity. Avoid wetting the unit with any liquid.
- Do not attempt to open it. Non-expert handling of the device could damage it. Consult your authorized dealer for help, or else it may cause fire, electric shock and breakdown.
- Do not use harsh chemicals, cleaning solvents, or strong detergents to clean it. Wipe it with a soft cloth that has been slightly dampened in a mild soap and water solution.
- When lightning, do not touch power plug or phone line, it may cause an electric shock.
- Do not install this phone in an ill-ventilated place.
- You are in a situation that could cause bodily injury. Before you work on any equipment, be aware of the hazards involved with electrical circuitry and be familiar with standard practices for preventing accidents.

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Chapter 1 Brief Introduction to D60

1.1 Thank You for Purchasing D60

Thank you for purchasing D60 IP Phone. D60 is an advanced and full-feature telephone, which is designed for power users especially in the office. This IP Phone supports many business phone features like call hold, call transfer, intercom, etc.

This guide will help you easily use the various features and services available on your phone. Please read the Quick Installation Guide before setting this device.

1.2 Delivery Content

Please check whether the delivery contains the following parts:

The base unit with display and keypad

The handset

The handset cable

The Ethernet cable

The power supply

Stand

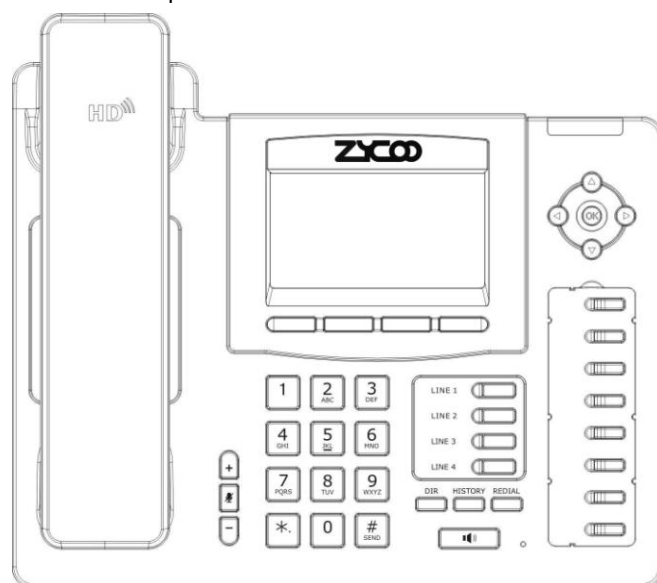
Warranty Card

Quick Installation Guide

Attentions: The D60 may cause damage if you use other power adapter which is not provided by ZYCOO. If the supplied power adapter cannot be used locally due to area or other reasons, please consult your local dealer.

The user manual(you may download from our website)

Here is the appearance of IP Phone description:



1.3 Specification Hardware

Item		D60 VoIP Phone
Adapter (Input / Output)		Input: 100-240V, Output: 5V 1A
Port	WAN	10/100Base- T RJ-45 1 Port
	LAN	10/100Base- T RJ-45 1 Port
Power Consumption		Idle: 2.5W/Active: 2.8W
LCD Size		128x64pixels, 75.49 x 37.73mm
Operation Temperature		0~40℃
Relative Humidity		10~65%
CPU		Broadcom VoIP Chipset
SDRAM		128Mb
Flash		32Mb
Phone Dimension(L x W x H)		175×240×130mm
Net Weight		1.2kg

1.4 Features

➤ Highlights

Support SIP 2.0 (RFC3261) and correlative RFCs

Support IAX2, IAX2 line call

Support 3 SIP lines, users can simultaneously register three SIP server, through any one service

Support HD voice

IEEE 802.3af Power over Ethernet compliant

Supports multiple call queuing

Supports SRTP and BLF

➤ Call Features

Call Transfer(blind/attended)

Call holding/waiting/ Call Park/ Paging and Intercom

Empty calling no. reject service

Voice Codec: G.711A/u, G.723.1, G.729a/b, G.722, G.726-32

Echo cancellation: G.168 Compliance in LEC, additional Acoustic Echo Cancellation(AEC) can reach 96ms max filter length in hands-free mode

9 kinds of ring type and 3 user-defined music ring

500 phonebook records

Support Voice Gain Setting, VAD, CNG

Support full duplex hands-free

Support DTMF type: SIP info, DTMF Relay, RFC2833

➤ **Advanced Features**

8 DSS keys
Support headset RJ9
Support 5 extension console with each consisting of 26 keys
Soft keys programmable, function keys programmable
Customizable multi-language version, the default is English
Flexible call control functions
Support SMS
Support MWI
Support XML phonebook/browser
Support Speed dial
Code synchronization via IP PBX/IMS
Support click to dial via web phone book
Voice codec setting for each SIP line
Support keypad lock, and emergency call during the keypad lock
Headset, speakerphone Ringing Selection

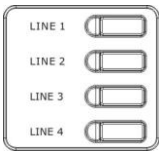







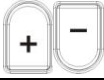

➤ **Network features**

WAN/LAN: support bridge and router model
Support basic NAT and NAPT
Support PPPoE for xDSL
Support VLAN (optional: voice vlan/ data vlan)
NAT Penetrate, Stun Penetrate
Support DMZ
Support VPN (L2TP/OPEN VPN) function
DNS and secondary DNS server
Support DHCP client on WAN
Support DHCP server on LAN
QoS with DiffServ
Network tools in telnet server: including ping, trace route, telnet client

➤ **Maintenance and management**

Upgrade firmware through POST mode
Web ,telnet and keypad management
Management with different account right
Upgrade firmware through HTTP, FTP or TFTP
Telnet remote management/ upload/download setting file
Supports Auto Provisioning to upgrade firmware or configuration file with HTTPS
Support TR-069(optional) and Syslog

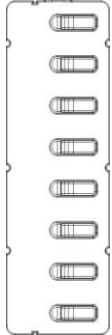
1.5 Key Descriptions

Key	Key name	Function Description
	Line key	Default SIP line key. The user can select SIP line to call, the call can also be defined as a line key, the call for a different way selector.
	Soft key 1/2/3/4	Keys combination, include functions such as History/Directory/DND/Menu/Del/Redial/Send/Quit/Answer/Divert/Reject/Hold/Transfer/Conf/Close and so on.
	Navigation	Navigation key assist users for operating. In desktop, dialer, calling, desktop long pressed state they have special function. You can configure through the web page according to your patterns of use.
DIR 	Phone book	Access to phone book, check the record list and add new records and revise the record. When check the phone book record, press this key again will return to idle mode.
HISTORY 	History	View the Missed call, Incoming Call and Outgoing Call.
REDIAL 	Redial	<ol style="list-style-type: none"> 1. In the hook off/hands-free mode, use the key to dial the last call number; 2. In stand-by mode, it has a function to check the Outgoing Call. 3. You could also find the specify contacts in phone book/call records, and use this number for quick dialing, press this button, you can dial quickly.
	Hands-free	Make the phone into hands-free mode.
	Mute	Press this key in calling mode, you can hear the other side, and the other side cannot hear you. Once you press this key twice, it would leave this mode and become normal.
	Volume +/-	Turn down or turn up the volume by pressing these two keys.
	Indicate light	This light will flash when there is a missed call



Digital
keyboard


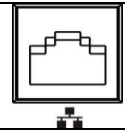
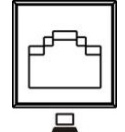
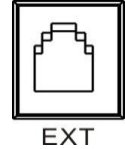
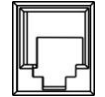
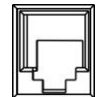
Inputting the phone number or DTMF
















DSS Keys

You can configure them in the web page, the configuration of each key to a different function keys it would more convenient to you.

1.6 Port for Connecting

Port	Name	Description
	Power switch	Input: AC 5V,1A
	WAN	10/100Mbps
	LAN	10/100Mbps
	Expansion Module interface	Port type: RJ-11
	Headset	Port type: RJ-9
	Handset	Port type: RJ-9

1.7 Icon Introduction

Icon	Description
	Call out
	Call in
	Call hold
	Auto answer
	Call mute
	Contact
	DND(Do not Disturb)
	In hand-free mode
	In hook mode
	In headset mode
	SMS
	Missed call
	Call forward

1.8 LED Status Introduction

Table 1 Programmable key for BLF

LED Status	Description
Solid green	The monitored user is idle.
Slow blinking red	The monitored user receives an incoming call.
Solid red	The monitored user is active
Off	The monitored user is failed subscribe
Off	No subscribe

Table 2 Programmable key for Presence

LED Status	Description
Solid green	The object is online.
Slow blinking red	The object is ringing.
Solid red	The object is active.
Off	The object is failed./No subscribe.

Table 3 Programmable key for line

LED Status	Description
Solid green	The account is active.
Fast Blinking green	There is an incoming call to the account.
Slow Blinking green	The call is on hold.
Slow Blinking red	Registration is unsuccessful.
Off	The line is not unapplied or idle.

Table 4 Programmable key LEDs for MWI

LED Status	Description
Blinking green	There are new voice mails.
Off	There is no new voice mail.

Table 5 Power Indication LED

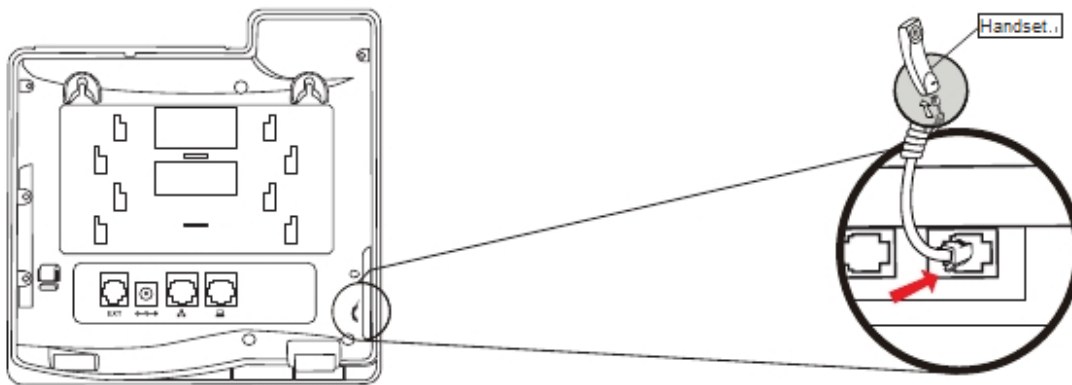
LED Status	Description
Solid red	Power on.
Fast Blinking red	There is an incoming call.
Off	Power off.

Chapter 2 Initial Connection and Login

2.1 Connection

Step 1 Handset Connection

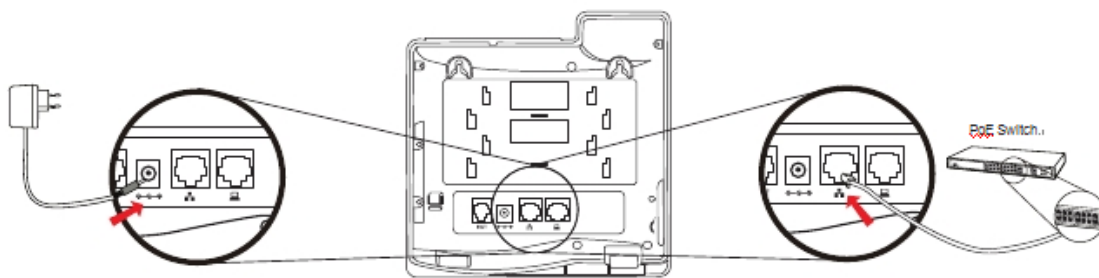
Plug one end of the handset cord into the handset and the other end into the handset jack.



Step 2 Power system Connection

The D60 can be powered either by external AC/DC adapter or by connecting to an IEEE802.3af/at PSE device such as 802.3af injector/hub or 802.3af/at PoE switch.

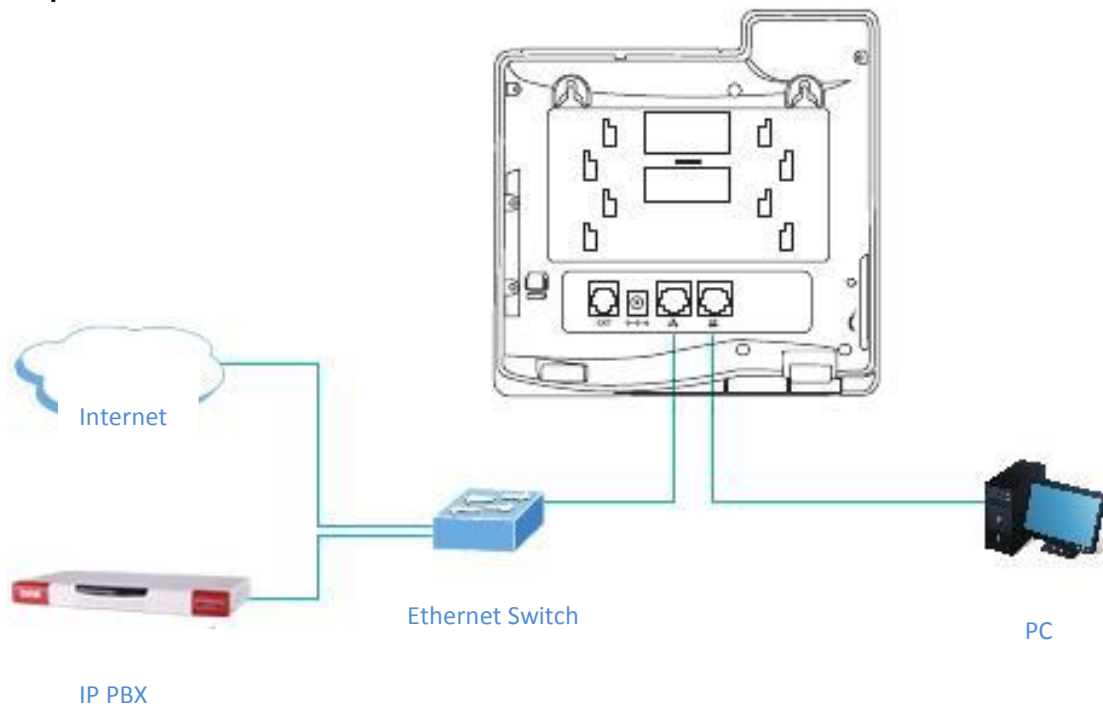
Once the D60 is powered, the LCD screen will display "INICIALIZING". After initializing, the phone will show greeting, current date and time and so forth.



Notice:

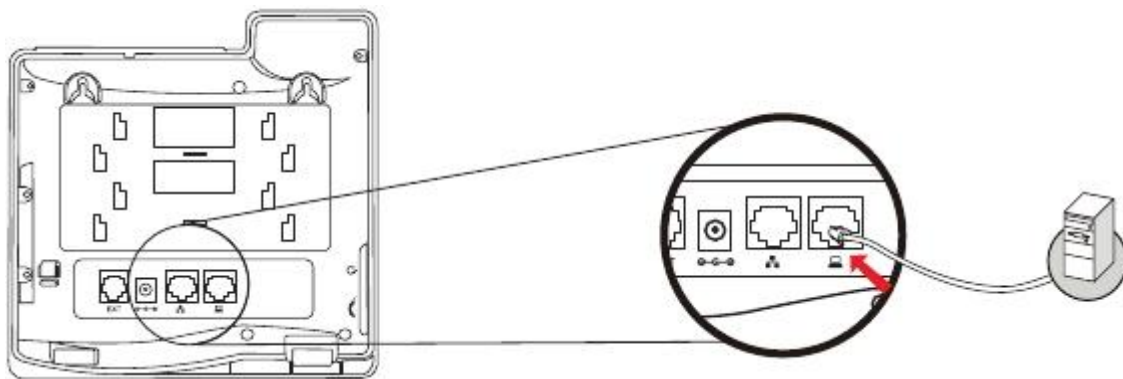
1. Use only the power adapter shipped with the unit to ensure correct functionality.
2. Only WAN support PoE.

Step 3 Network Connection



Step 4 Computer Network Connection

Set your computer's IP address to 192.168.0.x, where x is the number between 2 to 254 (except 1 where is used for the phone by default). If you don't know how to do this, please ask your network administrator. Connect your PC to D60 PC port.



Step 5 Web Login

Use web browser (IE6.0 and above) to connect to 192.168.0.x (type this address in the address bar of web browser).

You are prompted to input the user name and password, default is admin/ admin.

2.2 Network Settings

Please make sure the network is connected already. D60 uses DHCP to get WAN IP, the phone could access to network as long as there is DHCP server. If there is not DHCP server, the phone has to be changed to Static IP or PPPoE.

PPPoE mode (For ADSL connection)

1. Get PPPoE account and password firstly.
2. Press **Menu->Settings->Advanced Settings**, then enter passwords, and choose **Network ->WAN settings->Connection Mode**, enter and choose PPPoE through navigation keys and press the Save key.
3. Press **Back**, then choose **PPPoE Set**, press **Enter**.
4. The screen will show the current information. Press **Del** to delete it, then input your PPPoE user and password and press **Save**.
5. Press **Back** six times to return to the idle screen.
6. Check the status. If the screen shows “**Negotiating...**” it shows that the phone is trying to access to the PPPoE Server; if it shows an IP address, then the phone has already got IP with PPPoE.

Static IP mode (Static ADSL/Cable, or no PPPoE/DHCP network)

1. Get the network’s parameters firstly, such as IP Address, Net mask, Default Gateway and DNS server IP address. If you don’t know this information, please contact the service provider or technician of network.
2. Press **Menu->Settings->Advanced Settings**, then enter password, and choose **Network ->WAN settings->Connection Mode**, enter and choose **Static** through navigation keys and press the Save key.
3. Press **Back**, then choose **Static Set**, press **Enter**.
4. The screen will show the current information, and then press **Del** to delete. Input your IP address, Mask, Gateway, DNS and press **Save**.
5. Press **Back** six times to return to the idle screen.
6. Check the status, the screen shows “**Static**”. If the IP address and gateway is displayed as well, the Static IP mode takes effect.

Settings of DHCP mode




1. Press **Menu->Settings->Advanced Settings**, then enter passwords, and choose **Network ->WAN settings->Connection Mode**, enter and choose **DHCP** through navigation keys and press the Save key.
2. Press **Back** six times to return to the idle screen.
3. Check the status, the screen shows “**DHCP**”, If the IP address and gateway is displayed as well, DHCP mode takes effect.

Chapter 3 Basic Function

3.1 Making a Call

3.1.1 Call Device

You can make a phone call via the following devices:

- 1) Pick up the handset,  icon will be displayed in the idle screen.
- 2) Press the Speaker button,  icon will be displayed in the idle screen.
- 3) Press the headset button if the headset is already connected to the Headset Port. The icon  will be displayed in the idle screen.

And then, you can dial the phone number to make the call. Or, you can also dial the number firstly, and choose the method you prefer to speak.

3.1.2 Call Methods

You can press an available line button to make the call if there is more than one account registered.

- 1) Dial the number you want to call directly.
- 2) Press **History** key, use the navigation buttons to choose the phone number from the lists of Missed Calls, Incoming Calls and Outgoing Calls.
- 3) Press the **REDIAL** button to choose phone number from the list of Dialed Calls.
- 4) Press the programmable key which is set as speed dial button.

Then press the **Send** button or **Dial** key to make the call if necessary.


3.2 Answering a Call

Answering an Incoming Call:

- 1) If you are not on another phone, lift the handset to use, or press the Speaker button/ Answer softkey to answer using the speaker phone, or press the headset button to answer the headset.
- 2) If you are on a call currently, press the **Answer**.


During the conversation, you can alternate between Headset, Handset and Speaker phone by pressing the corresponding buttons or picking up the handset.

3.3 DND

Press DND softkey to active DND Mode. Further incoming calls will be rejected and the LCD shows:  icon.
Press DND twice to cancel DND mode. You can find the incoming call record in the Call History.

3.4 Call Forward

This feature allows you to forward an incoming call to another phone number.

The LCD shows  icon.

The following call forwarding events can be configured:

Off: Call forwarding is deactivated by default.

Always: Incoming calls are immediately forwarded.

Busy: Incoming calls are forwarded when the phone is busy.

No Answer: Incoming calls are forwarded when the phone is not answered after a specific period.

To configure Call Forward via Phone interface:

- 1) Press **Menu ->Features->Enter->Call Forwarding->Enter**.
- 2) There are 4 options: Disabled, Always, Busy, and No Answer.
- 3) If you choose one of them (except Disabled), enter the phone number you want to forward to receiving party.

Press **Save** to save the changes.

3.5 Call Hold


Press the Hold button or Hold softkey to put your active call on hold.

- 1) If there is only one call on hold, press the hold key to retrieve the call.
- 2) If there are more than one call on hold, press the line button, and the Up/Down button to highlight the call, then press the Un-hold button to retrieve the call.

3.6 Call Waiting

- 1) Press **Menu ->Features->Enter->Call Waiting->Enter**.
- 2) Use the navigation keys to activate or deactivate call waiting.
- 3) Then press **Save** to save the changes.

3.7 Mute

Press **Mute** button during the conversation, icon  will be shown on the LCD. Then the caller will not hear you, but you can hear the caller. Press it again to get the phone to normal conversation.

3.8 Call Transfer

1. Blind Transfer

During the call, press the key **Transf**, dial the number that you want to transfer to and finish by pressing "#". The call will be transferred to the third party. After finishing transfer, the call you talk to will be hung up. SIP line is not available for selecting when call is in transferring.

2. Attended Transfer

During the call, press the key **Transf**, input the number that you want to transfer to and press **Send**. After the third party allows to receive the call, then press **Transf** to complete the transfer. (You need enable call waiting and call transfer in advance).

If there are two calls, you can just talk to one, and keep hold to the other one. The one who is keeping hold cannot speak to you or hear from you. In other words, if you want to invite the third party during the call, you can press **Conf** to make calls mode in conference mode. If user wants to stop conference, user can press **Split**. (User must enable call waiting and three way call in advance).

Notice: The server that user uses must support RFC3515 or it might not be used.

3. Alert Transfer

During the call, press **Transf** firstly, press **Send** after inputting the number that you want to transfer. You are waiting for connection, now, press **Transf** and the transfer will be done. (To use this feature, you need enable call waiting and call transfer in advance).

3.9 Three-Way Conference call

- 1) Press the **Conf** softkey during an active call.
- 2) The first call is placed on hold. Then you will hear a dial tone. Dial the number to conference in, then press

Send key.

- 3) When the call is answered, press **Conf** and add the first call to the conference.
- 4) If you want to release the conference, press **Split** key.

Notice: You may need to start the call feature in order to achieve three-way calling mode.

3.10 Multiple-Way Call

If you have 2 lines calls and want to invite the third party during the call, you can press softkey **Conf** or softkey **XFER** "New Call", press OK, enter the number, then press Send and wait for the other party to answer. When there are multiple way calls, you can press an arrow key to select a call.

3.11 Multi-line

D60 supports 6 SIP account numbers at most, that means you can register 6 SIP account numbers, and the 6 accounts can be used at the same time. There are four keys used as SIP line by default to make calls in SIP accounts. It will blink when the account fails to register.

To facilitate the enterprise communication, the phone supports multiple call answering , call hold and multi-line call. The user can answer 10 incoming call phones at most, you can choose any call by pressing the navigation key in talking and the other 9 calls will be on hold.

You can also press the navigation key to change the call and recover the talking, then last call will be hold automatically.

You can also define the six line keys as multi-line keys ,then each line key will be bound to a call and you can choose the talking by pressing the line keys and recover the talking and the light to the line key will be bright all the time in talking, then the light of the call on hold is blinking.

Chapter 4 Advanced Function

4.1 Call Pickup

Call pickup is implemented by simulating pickup function of PBX; that is, when A calls B, B rings but no answer, at this moment, C can hook off and input an appointed prefix plus B's number, pick up A's call and talk to A.

The following chart shows how to configure an appointed prefix in dial peer to have call pick up function.

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
*1*T	0.0.0.0	5060	SIP	rep:pickup	no suffix	3

1 means appointed prefix code. After making the above configuration, C can dial *1* plus B's phone number to pick up A's call. User can set prefix in random, in the case of it does not affect current dialing rules.

4.2 Join Call

When B is calling C, A can join in the existing call by inputting an appointed prefix numbers plus B or C number, if B or C also supports join call.

The following chart shows how to configure an appointed prefix in dial peer to have join call function.

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
*2*T	0.0.0.0	5060	SIP	rep:joincall	no suffix	3

2 means appointed prefix code. After making the above configuration, A can dial *2* plus B or C number to join B and C's call. User can set prefix in random, in the case of no affecting current dialing rules.

4.3 Redial / Un-redial

If B is in busy line when A calls B, A will get notice: busy, please hang up. If A wants to connect B as soon as B is in idle, he can use redial function at the moment and he can dials an appointed prefix number plus B's number to realize redial function.

What is redial function? A cannot build a call with B when B is busy, then A will subscribe B's calling mode at 60 second intervals. Once B is available, A will get reminder of rings to hook off, while a hooks off, A will call B automatically. If A is occupied temporarily at this moment and unwilling to contact B, A can also cancel the redial function by dialing an appointed prefix plus B's number before making the redial function.

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
*3*T	0.0.0.0	5060	SIP	rep:redial	no suffix	3
*4*T	0.0.0.0	5060	SIP	rep:unredial	no suffix	3

3 is appointed prefix code. After making the above configuration, A can dial *3* plus B's phone number to make the redial function.

4 is appointed prefix code. After configuration, A can dial *4* to cancel redial function.

User can set prefix in random, in the case of no affecting current dialing rules.

4.4 Click to Dial

When user A browses on an appointed Web page, user A can click to call user B via a link (this link to user B), then user A's phone will ring, after A hooks off, the phone will dial to B.

Notice: It needs an external software that supports click to dial.

4.5 CallBack

The function allows user to dial out the last phone call user received.

4.6 Auto Answer

When there is an incoming call unanswered, the phone will answer the call automatically.

4.7 Hotline

You can set hotline number for every sip, and then enter the dialer interface and after Warm Line Time, the phone will call out the hotline number automatically.

4.8 Application

4.8.1 SMS

- 1) Press **Menu ->Applications->Enter->SMS->Enter**.
- 2) Use the navigation keys to highlight the options. You can read the message in the Inbox/Outbox.
- 3) After view the new message, you can press **Reply** to reply the message, and use the **2aB** softkey to change the Input Method, when enter the reply message, press **OK**, then use the navigation keys to select the line from which you want to send, then **Send**.
- 4) If you want to write a message, you can press **New** and enter message. Use the **2aB** softkey to change the Input Method. When you input the message you want to send, press **OK**, then use the navigation keys to select the line from which you want to send, then **Send**.
- 5) If you want to delete the message, after view the message, press **Del**, then you have three options to choose from: Yes, All, No.

4.8.2 Memo

You can add some memos to record some important things to remind you.

Press **Menu->Application->Memo->Enter->Add**.

There are some options to configure: Mode, Date, Time, text, Ring. When the configuration is completed, press **Save**.

4.8.3 Voice Mail

- 1) Press **Menu->Application->Voice Mail->Enter**.
- 2) Use the navigation keys to highlight the line for which you want to set, press **Edit**, and use the navigation key to turn on the mode, and input the number. Press **2aB** softkey to choose the proper input method.
- 3) Press **Save** to save the change.
- 4) To view the new voicemail, Press the **Voicemail** key directly. Press **Dial**, then you may be prompted to enter the password, then you can listen to your new and old messages.

4.8.4 Ping

- 1) Press **Menu ->Ping-> Enter**.
- 2) Input the IP you want, then press **start**. You can also press **delete** to modify the IP and change the input method when you input errors.
- 3) After input the IP, LCD will display "**OK**" after a moment, means Ping is successful . Otherwise, Ping failed.

4.8.5 Programmable Key Configuration

The phone has 8 programmable keys which are able to set up to many functions. The following list shows the functions you can set on the programmable keys and provides a description for each function. The default configuration for each key is N/A which means the key hasn't been set for any functions.

Set the type as Memory Key

Press **Menu->Settings->Basic Settings->Enter->Keyboard->DSS Key Settings**, you have two options: Line Key Settings and Function Key Settings. Choose one you want to make the assignment, use the navigation key to choose the type as memory key. In the Dial field, you have some options, such as Normal, Speed Dial, Intercom, BLF, Presence, MWI, Call Park.

Speed dial

You can configure the key as a simplified speed dial key. This key function allows you to easily access your most dialed numbers.

Intercom

You can configure the key for Intercom code and it is useful in an office environment as a quick access to connect to the operator or the secretary.

BLF

BLF is also called “Busy lamp field”, and it is used to prompt the user to pay attention to the state of the object than has been subscribed, and used to cooperate with the server to pick up the phone call. You can configure the key for Busy Lamp Field (BLF) which allows you to monitor the status (idle, ringing, or busy) of other SIP account. User can dial out on a BLF configured key. Please refer to “LED Instruction” for more detail about the LED status in different situations.

Notice: In the Web interface, you can also set the pickup number to activate the pickup function. For example, if you set the BLF number as 212, and the pickup number is 326, when there is an incoming call to 212, press the BLF key, it will call out the 326 automatically to pick up the incoming call on 212.

Presence

Presence is called present, and compared to the BLF, it can also check whether object is online.

Notice: You can not subscribe the BLF and presence station of the same number at the same time.

MWI (Message-Waiting Indicator)

When the key is configured as MWI, you are allowed to access voicemail quickly by pressing this key.

Call Park

1) You need setting a server number, when you have set what represent Call Park. If you have a calling and you busy now, you could press the key and hear a number, then you could choose other phone and input this number. so you can directly recover call.

2) Set the type as Line

You can set these keys as line keys, and press it, it will enter dialer interface.

3) Set the type as Key Event

You can set these keys as Key Event, and the subtype has many options.

Choose one and it will have corresponding function.

- None
- MWI
- DND
- Hold
- Transfer
- Phone Book
- Redial
- Pick up
- Join
- Auto-redial-on
- Auto-redial-off
- Call Forward
- History
- Flash
- Memo

- Headset
- Release: Press this button, you can end all calls
- Lock: You can select whether the keypad lock in the standby interface
- SMS
- Call Back
- Power Light
- Hide DTMF
- Prefix
- Hot Desking: You can clear the current sip information, and register their new sip information.

4) Set the type as DTMF

You can configure the key as DTMF. This key function allows you to easily dial or edit dial number.

5) Set the type as URL

You need to match a XML Phonebook address, pressing the button you can directly access the corresponding remote phonebook.

6) Set the type as BLF List Key

It needs the cooperation with the Broad soft server. The traditional BLF is that every number will need to be subscribed, so if the numbers that are subscribed are so many, it will cause to obstruction. However, BLF List Key will put the numbers that are needed to be subscribed in a group, and the phone use the URL of the group to subscribe and analyze the specific information of each number such as number, name, state and so on according to the notifications from the server. Then set the idle Memory key as BLF List Key, later if the state of an object changes, the corresponding LED will change.

Chapter 5 Other Functions

5.1 Auto Handdown

1. Press **Menu ->Features-> Enter->Auto Handdown-> Enter**.
2. Enable the mode through the navigation key, then set Time, unit is minute, then press **Save**.
3. When the call ends, after the time that you have set, the phone will back to the idle mode.

5.2 Ban Anonymous Call

1. Press **Menu ->Features-> Enter->Ban Anonymous Call-> Enter**.
2. Choose which sip you want to enable Ban Anonymous Call, and then press **Enter**, choose **Enabled** or **Disabled** through navigation key.
3. If you choose Enabled, the others can't call the phone by anonymous. If you choose Disabled, the others can call the phone by anonymous.

5.3 Dial Plan

1. Press **Menu ->Features-> Enter->Dial Plan-> Enter**.
2. The following plans you can set: Press **"#"** to Send, Timeout to Send, Timeout, Fixed Length Number, Press **"#"** to Do BXFER, BXFER On On-hook, AXFER On On-hook. You can enable or disable each dial plan.

5.4 Dial Peer

1. Press **Menu ->Features-> Enter->Dial Peer-> Enter**.
 2. Press **Add** to enter the Edit interface, and then input number and destination. For example: Number:1, Destination:1234, Then press **Save**.
 3. Input 1# number in the dial interface, you can dial out 1234.
- You can refer to 8.3.3.4

5.5 Auto Redial

1. Press **Menu ->Features-> Enter->Auto Redial-> Enter**.
2. Choose Mode **Enabled** or **Disabled** through the navigation key. If you choose Enable, you also need to set Interval and Times, and then press **Save**.
3. After enable auto redial, calling out someone, if he/she is busy, it will pop up a prompt box whether to auto redial, press **OK**, the phone will call out him according the Interval and Times that you set.

5.6 Call Completion

1. Press **Menu ->Features-> Enter->Call Completion-> Enter**.
2. Enable the function through the navigation key, and then **Save**.
3. Call out others, if he/she is busy, it will pop up a prompt Call Completion Waiting number. Press **OK**, when he/she is idle, it will pop up a prompt Call Completion Call number. Press **OK**, the phone will call out the number automatically.

5.7 Ring from Headset

1. Press **Menu ->Features-> Enter->Ring From Headset-> Enter**.
2. Enable this function through the navigation key, the phone connects the headset, when the phone has an incoming call, it will ring from the headset.

5.8 Power Light

1. Press **Menu ->Features-> Enter->Power Light-> Enter**.
2. Enable this function through the navigation key.

5.9 Hide DTMF

1. Press **Menu ->Features-> Enter->Hide DTMF-> Enter**.
2. Through the navigation key to choose: Disabled, All, Delay, Last Show. When you set up a call with others and need to input the DTMF, the DTMF will show as you have set.

5.10 Ban Outgoing

1. Press **Menu ->Features-> Ban Outgoing-> Enter**.
2. Enable this function; you cannot call any number.

5.11 Password Dial

1. Press **Menu ->Features-> Enter->Password Dial-> Enter**.
2. Enable this function, you can also set Prefix and Length. For example, you want to call out 1234567 and you set Password Dial Prefix 123 and Password Length 3, then enter the dial interface and input 1234567, and then the screen will show 123***7.

5.12 Pre Dial

1. Press [Menu ->Features-> Pre Dial-> Enter](#).
2. Enable this function and you will realize Pre-Dial function.

5.13 Action URL & Active URI

1.**Action URL**: The action that the phone carries out. For example, opening DND can produce one URL, and then the phone can send the HTTP to get the URL to PC. The phone can report the action to the PC.

2.**Active URI**: Enter the web page of the phone, PHONE-> FEATURE, input Active URI Limit IP. User can input internet server (e.g. PC'IP), PC can send one URL to the phone.

The phone will produce one action; for example, open DND, so PC can control the phone.

5.14 Push XML

Enter the web page of the phone, PHONE->FEATURE, input Push XML Server(e.g. PC'IP), then PC can push text, SMS, phonebook, advertisement, execute, etc. To phone to update the message or the phone makes an action.

Chapter 6 Basic Settings

6.1 Keyboard

1. Press **Menu ->Settings-> Enter->Basic Settings-> Enter->Keyboard->Enter**.
2. There are four items: DSS Key settings, Programmable Key Settings, Desktop Long Pressed, Soft Key. You can set up respectively on them. Press **Enter** to enter the interface, then use the navigation keys to choose the function for the key according to your requirement.
3. Press **OK** to save.

6.2 Screen Settings

1. Press **Menu ->Settings-> Enter->Basic Settings-> Enter->Screen Settings->Enter**.
2. You can set Contrast, Contrast Calibration and Backlight, press **Enter** and use the navigation keys to set, then press **Save**.

6.3 Ring Settings

1. Press **Menu ->Settings-> Enter->Basic Settings-> Enter->Ring Settings->Enter**.
2. You can set Ring Volume and Ring Type, press **Enter** and use the navigation keys to set, then press **Save**. In the Ring Type, the default system rings have nine and the custom ringtones have three that can be set through the web page.

6.4 Voice Volume

1. Press **Menu ->Settings-> Enter->Basic Setting-> Enter->Voice Volume->Enter**.
2. Use the navigation keys to turn down or turn up the voice volume, then press **Save**.

6.5 Time & Date

1. Press **Menu ->Settings->Enter->Basic Settings-> Enter->Time & Date->Enter**.
2. You have two options to choose: Auto and Manual. Use the navigation keys to choose, then press **Save**.

6.6 Greeting Words

1. Press **Menu ->Settings-> Enter->Basic Settings-> Enter->Greeting Words->Enter**.
2. You can enter the message and press **Save**, it will display on the phone screen when the phone start up.

6.7 Language

1. Press **Menu** ->**Settings**-> **Enter**->**Basic Settings**-> **Enter**->**Language** ->**Enter**.
2. The D60 supports three languages. You can use the navigation keys to choose. The default two languages are English and Chinese.

Chapter 7 Advance Settings

7.1 Accounts

Press **Menu->Enter->Advanced settings**, and input the password to enter the interface, the default password is 123. You can set it through the web page. Then choose **Account** and press **Enter**, you can do some sip settings.

7.2 Network

Press **Menu->Enter->Advanced settings**, and input the password to enter the interface. Then choose **Network** and press **Enter**, you can do network settings, you can refer to 2.2.1 Network settings.

7.3 Security

Press **Menu->Enter->Advanced settings**, and input the password to enter the interface. Then choose **Security**, you can configure Menu Password, Key lock Password, Key lock Status and whether to ban Outgoing.

7.4 Maintenance

Press **Menu->Enter->Advanced settings**, and input the password to enter the interface. Then choose **Maintenance** and press **Enter**, you can configure Auto Provision, Backup, and Upgrade.

7.5 Factory Reset

Press **Menu->Enter->Advanced settings**, and then input the password to enter the interface. Then choose **Factory Reset** and press **Enter**, you can choose **Yes** or **No**.

Chapter 8 Web Configuration

8.1 Introduction of Configuration

8.1.1 Ways to Configure

D60 has three different ways to different users.

- Use phone keypad.
- Use web browser (recommended way).
- Use telnet with CLI command.

8.1.2 Password Configuration

There are two levels to access to phone: root level and general level. User with root level can browse and set all configuration parameters, while user with general level can set all configuration parameters except SIP (1-2) or IAX2's that some parameters cannot be changed, such as server address and port. User will have different access level with different username and password.

- Default user with general level:
 - ◆ Username: guest
 - ◆ Password: guest
- Default user with root level:
 - ◆ Username: admin
 - ◆ Password: admin

The default password of phone screen menu is 123.

8.1.3 Setting via Web Browser

When this phone and PC are connected to network, enter the IP address of the WAN port in this phone as the URL (e.g. <http://xxx.xxx.xxx.xxx/> or <http://xxx.xxx.xxx.xxx:xxxx/>).

If you do not know the IP address, you can look it up on the phone's display by pressing Status button.

The login page is shown as below:



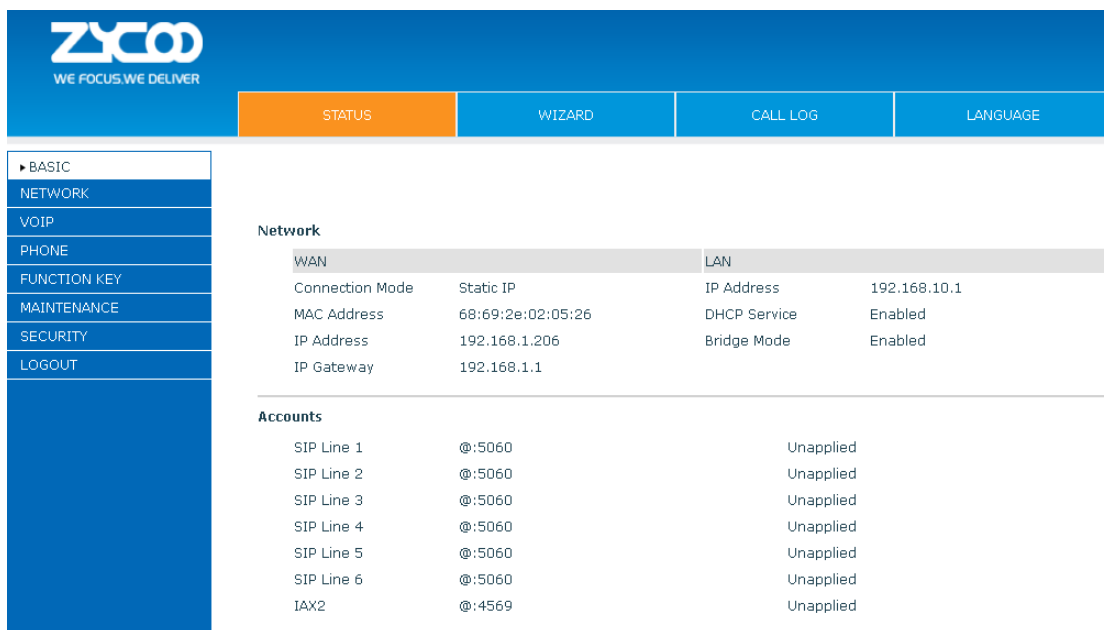
The image shows the login interface for the ZYCOO COOFONE-D60 IP PHONE. It features a blue header with the ZYCOO logo and the text 'WE FOCUS. WE DELIVER.' and 'COOFONE-D60 IP PHONE'. Below the header, there are three input fields: 'User:', 'Password:', and 'Language:'. The 'Language:' field is a dropdown menu currently set to 'English'. A 'Logon' button is located at the bottom right of the form.

After you configure the IP phone, you need click save button in config under Maintenance in the left catalog to save your configuration. Otherwise the phone will lose your modification after power is off and on.

8.2 Configuration via WEB

8.2.1 BASIC

8.2.1.1 STATUS



The image shows the 'STATUS' page of the ZYCOO COOFONE-D60 IP PHONE web interface. The page has a blue header with the ZYCOO logo and the text 'WE FOCUS. WE DELIVER.'. Below the header, there are four tabs: 'STATUS' (selected), 'WIZARD', 'CALL LOG', and 'LANGUAGE'. On the left side, there is a navigation menu with the following items: 'BASIC', 'NETWORK', 'VOIP', 'PHONE', 'FUNCTION KEY', 'MAINTENANCE', 'SECURITY', and 'LOGOUT'. The main content area is divided into two sections: 'Network' and 'Accounts'.

Network			
WAN		LAN	
Connection Mode	Static IP	IP Address	192.168.10.1
MAC Address	68:69:2e:02:05:26	DHCP Service	Enabled
IP Address	192.168.1.206	Bridge Mode	Enabled
IP Gateway	192.168.1.1		

Accounts		
SIP Line 1	@:5060	Unapplied
SIP Line 2	@:5060	Unapplied
SIP Line 3	@:5060	Unapplied
SIP Line 4	@:5060	Unapplied
SIP Line 5	@:5060	Unapplied
SIP Line 6	@:5060	Unapplied
IAX2	@:4569	Unapplied

Status	
Item	Explanation
Network	Shows the configuration information on WAN and LAN port, including the connect mode of WAN port (Static, DHCP, PPPoE), MAC address, the IP address of WAN port and LAN port, ON or OFF of DHCP mode of LAN port and bridge mode
Accounts	Shows the phone numbers provided by the SIP LINE 1-2 servers and IAX2. The last line shows the version number and issued date .

8.2.1.2 WIZARD

Wizard


Please select the proper network mode according to the network condition. D60 provides three different network settings:

- **Static:** If your ISP server provides you the static IP address, please select this mode, and then finish Static mode setting. If you don't know about parameters of Static mode setting, please ask your ISP for them.
- **DHCP:** In this mode, you will get the information from the DHCP server automatically; no need to input this information artificially.
- **PPPoE:** In this mode, you must input your ADSL account and password.

You can also refer to 2.2 Network setting to speedily set your network.

Choose Static IP mode, click **NEXT** can configure the network and SIP(default SIP1) simply. Click **BACK** to return to the last page.

IP Address	Input the IP address distributed to you
Subnet Mask	Input the Netmask distributed to you.
IP Gateway	Input the Gateway address distributed to you.
DNS Domain	Set DNS domain postfix. When the domain which you input cannot be parsed, phone will automatically add this domain to the end of the domain which you input before and parse it again.
Primary DNS	Input your primary DNS server address.
Secondary DNS	Input your standby DNS server address.



STATUS

WIZARD

CALL LOG

LANGUAGE

► BASIC

NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT

Quick SIP Settings

Display Name

Server Address

Server Port


Authentication User

Authentication Password

SIP User

Enable Registration ☐

Display Name	Set the display name.
Server Address	Input your SIP server address.
Server Port	Set your SIP server port.
Authentication User	Input your SIP register account name.
Authentication Password	Input your SIP register password.
SIP User	Input the phone number assigned by your VOIP service provider.
Enable Registration	Start to register or not by selecting it or not.



STATUS

WIZARD

CALL LOG

LANGUAGE

► BASIC

NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT

WAN

Connection Mode Static IP

Static IP Address 192.168.1.206

IP Gateway 192.168.1.1

SIP

Server Address

Account

Phone Number

Registration Disabled

Display detailed information that you manual config.

Choose DHCP MODE, click **Next** to configure SIP (default SIP1) simply. Click **Back** to return to the last page. Like Static IP mode.

Choose PPPoE MODE, click **Next** to configure the PPPoE account/password and SIP (default SIP1) simply. Click **Back** to return to the last page. Like Static IP mode.

PPPoE Settings

Service Name	<input type="text" value="ANY"/>
User	<input type="text" value="user123"/>
Password	<input type="password" value="....."/>
<input type="button" value="Back"/> <input type="button" value="Next"/>	

PPPoE Server Server name, if PPPOE service providers are no special requirements, this name is usually the default value.


User Input your ADSL account

Password Input your ADSL password

Notice: Click **Finish** button after finished your setting, IP Phone will save the setting automatically and reboot. After reboot, you can dial by the SIP account.

8.2.1.3 CALL LOG

You can query all the outgoing through this page.



STATUS

WIZARD

CALL LOG

LANGUAGE

► BASIC

NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT


Call Information

Start Time	Duration	Dialed Calls
------------	----------	--------------

Call log

Item	Explanation
Start Time	Display the start time of the outgoing record.
Duration	Display the conversation time of the outgoing record.
Dialed Calls	Display the account/protocol/line of the outgoing record.

8.2.1.4 LANGUAGE



STATUS

WIZARD

CALL LOG

LANGUAGE

► BASIC

NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT

Language

Language Selection English ▼

Greeting Words

Greeting Words ZYCOO (0-12 character(s))

Apply


LANGUAGE

Item	Explanation
Language	Set the language of phone, English is default.
Greeting Words	The greeting words will display on LCD when phone is idle. It can support 12 characters. the default chars are VOIP PHONE.

Notice: the maximal length of the greeting message is 12 English characters and 5 Chinese characters.

8.2.2 NETWORK

8.2.2.1 WAN



WAN

LAN

QoS&VLAN

SERVICE PORT

DHCP SERVICE

TIME&DATE

BASIC

► NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT

WAN Status

Active IP Address	192.168.1.206
Current Subnet Mask	255.255.255.0
Current IP Gateway	192.168.1.1
MAC Address	68:69:2e:02:05:26
MAC Timestamp	20130918

WAN Settings

Static IP ☒

DHCP ☐

PPPoE ☐

IP Address

192.168.1.206

Subnet Mask

255.255.255.0

IP Gateway

192.168.1.1

DNS Domain

Primary DNS

8.8.8.8

Secondary DNS

8.8.4.4

WAN Status

WE FOCUS.WE DELIVER

WAN
LAN
QoS&VLAN
SERVICE PORT
DHCP SERVICE
TIME&DATE

BASIC

▶ NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT

WAN Status

Active IP Address	192.168.1.206
Current Subnet Mask	255.255.255.0
Current IP Gateway	192.168.1.1
MAC Address	68:69:2e:02:05:26
MAC Timestamp	20130918

Active IP Address	The current IP address of the phone
Current Subnet Mask	The current Net mask address
MAC Address	The current MAC address of the phone
Current IP Gateway	The current Gateway IP address
MAC Timestamp	Shows the time of getting MAC address

SECURITY

LOGOUT

WAN Settings

Static IP ☒
DHCP ☐
PPPoE ☐

IP Address	<input type="text" value="192.168.1.206"/>
Subnet Mask	<input type="text" value="255.255.255.0"/>
IP Gateway	<input type="text" value="192.168.1.1"/>
DNS Domain	<input type="text"/>
Primary DNS	<input type="text" value="8.8.8.8"/>
Secondary DNS	<input type="text" value="8.8.4.4"/>

Please select the proper network mode according to the network condition. D60 provides three different network settings:

- **Static:** If your ISP server provides you the static IP address, please select this mode, and then finish Static Mode setting. If you don't know about parameters of Static Mode setting, please ask your ISP for them.
- **DHCP:** In this mode, you will get the information from the DHCP server automatically; need not to input this information artificially.
- **PPPoE:** In this mode, you must input your ADSL account and password.
You can also refer to 2.2.1 Network setting to speed setting your network.

Obtain DNS server automatically	Select it to use DHCP mode to get DNS address, if you don't select it, you will use static DNS server. The default is selecting it.
---------------------------------	---

WAN Settings

Static IP ☒
DHCP ☐
PPPoE ☐

IP Address	<input type="text" value="192.168.1.206"/>
Subnet Mask	<input type="text" value="255.255.255.0"/>
IP Gateway	<input type="text" value="192.168.1.1"/>
DNS Domain	<input type="text"/>
Primary DNS	<input type="text" value="8.8.8.8"/>
Secondary DNS	<input type="text" value="8.8.4.4"/>

If you user static mode, you need set it.

IP Address	Input the IP address distributed to you.
Subnet Mask	Input the Net mask distributed to you.

IP Gateway	Input the Gateway address distributed to you.
DNS Domain	Set DNS domain postfix. When the domain which you input cannot be parsed, phone will automatically add this domain to the end of the domain which you input before and parse it again.
Primary DNS	Input your primary DNS server address.
Secondary DNS	Input your standby DNS server address.

WAN Settings

Obtain DNS Server Automatically Disabled

Static IP ☐ DHCP ☐ PPPoE ☒

Service Name

User

Password

If you uses PPPOE mode, you need to make the above setting.

Service Name	It will be provided by ISP.
User	Input your ADSL account.
Password	Input your ADSL password.

Notice:

- 1) Click "Apply button " after finishing your setting, IP Phone will save the setting automatically and new setting will take effect.
- 2) If you modify the IP address, the web will not response by the old IP address. You need input new IP address in the address column to login the phone.
- 3) If networks ID which is DHCP server distributed is same as network ID which is used by LAN of system, system will use the DHCP IP to set WAN, and modify LAN's networks ID(for example, system will change LAN IP from 192.168.10.1 to 192.168.11.1) when system uses DHCP client to get IP in startup; If system uses DHCP client to get IP in running status and network ID is also same as LAN's, system will refuse to accept the IP to configure WAN. So WAN's active IP will be 0.0.0.0.

8.2.2.2 LAN

WAN
LAN
QoS&VLAN
SERVICE PORT
DHCP SERVICE
TIME&DATE

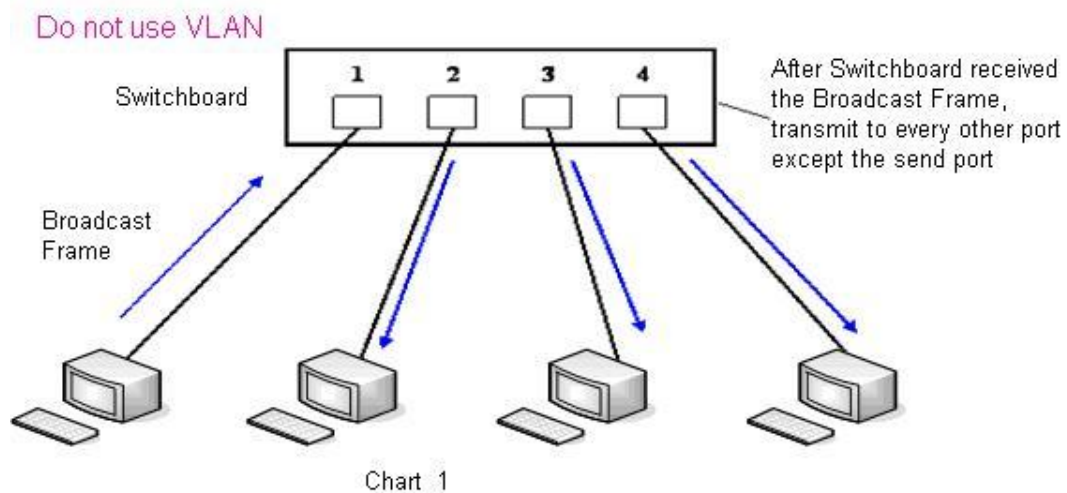
BASIC
▶ NETWORK
VOIP
PHONE
FUNCTION KEY
MAINTENANCE
SECURITY
LOGOUT

LAN Settings ⓘ
IP Address
Subnet Mask
DHCP Service ☒
NAT ☒
Port Mirror ☒ (Only works in the bridge model)
Enable Bridge Mode ☒

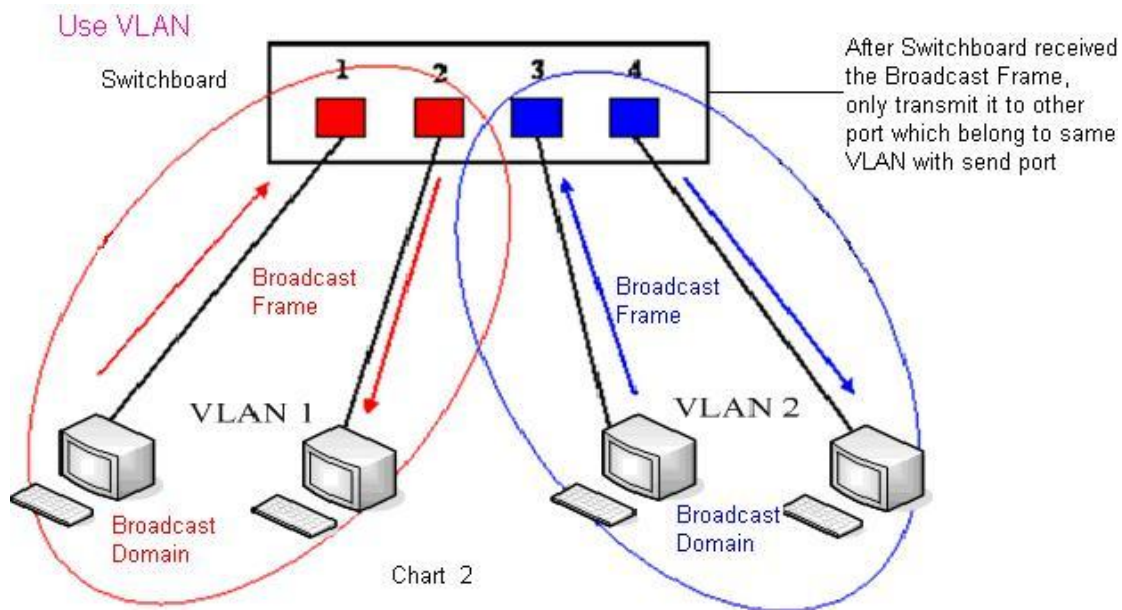
LAN Configuration	
Item	Explanation
LAN IP Address	Specify LAN static IP
Subnet Mask	Specify LAN Net mask
DHCP Service	Select the DHCP server of LAN port or not. After you modify the LAN IP address, phone will amend and adjust the DHCP Lease Table and save the result amended automatically according to the IP address and Net mask. You need reboot the phone and the DHCP server setting will take effect.
NAT	Select NAT or not
Port Mirror	Select Port Mirror or not, it only works in bridge mode, the function of the port mirror is that copy the data stream from the WAN port to the LAN port of the phone.
Enable Bridge Mode	If you select Bridge Mode, the phone will no longer set IP address for LAN physical port; LAN and WAN will join the same network. Click "Apply", the phone will reboot.
Notice: When LAN IP or bridge mode status is changed, the system will reboot! If you choose the bridge mode, the LAN configuration will be disabled.	

8.2.2.3 QoS&VLAN

The VOIP phone support 802.1Q/P protocol and DiffServ configuration. VLAN functionality can use different VLAN IDs by setting voice VLAN and data VLAN. The VLAN application of this phone is very



flexible.



In chart

1, there is a layer 2 that switches without setting VLAN. Any broadcast frame will be transmitted to the other ports except the send port. For example, a broadcast information is sent out from port 1 then transmitted to port 2,3 and 4.

In chart 2, red and blue indicate two different VLANs in the switch, and port 1 and port 2 belong to red VLAN, port 3 and port 4 belong to blue VLAN. If a broadcast frame is sent out from port 1, switch will transmit it to port 2, the other port in the red VLAN and not transmit it to port 3 and port 4 in blue VLAN. By this means, VLAN divide the broadcast domain via restricting the range of broadcast frame transition.

Note: chart 2 use red and blue to identify the different VLAN, but in practice, VLAN uses different VLAN IDs to identify.

WE FOCUS.WE DELIVER

WAN	LAN	QoS&VLAN	SERVICE PORT	DHCP SERVICE	TIME&DATE
-----	-----	----------	--------------	--------------	-----------

BASIC

▶ NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT

Link Layer Discovery Protocol (LLDP) Settings

Enable LLDP ? ☐ Packet Interval(1~3600) second(s)

Enable Learning Function ☐

Quality of Service (QoS) Settings

Enable DSCP ☐ SIP DSCP (0~63)

Audio RTP DSCP (0~63)

WAN Port VLAN Settings

Enable WAN Port VLAN ☐ WAN Port VLAN ID (0~4095)

SIP 802.1P Priority (0~7) Audio 802.1P Priority (0~7)

LAN Port VLAN Settings

LAN Port VLAN Mode Follow WAN ▼ LAN Port VLAN ID (0~4095)

Apply

QoS & VLAN Configuration	
Item	Explanation
Link Layer Discovery Protocol (LLDP) Settings	
Enable LLDP	Enable LLDP by selecting it.
Enable Learning Function	After enabling LLDP Learn, telephone can automatically learn the data of DSCP, 802.1p, VLAN ID from the switch. If the data is different from the data of the LLDP server, telephone will change its own value as the value of the switch (Synchronous with VLAN in switch).
Package Interval(1-3600)	The time interval of sending LLDP Packet.
Quality of Service(QoS) Settings	
Enable DSCP	Enable DSCP by selecting it.
SIP DSCP	Specify the value of the SIP DSCP.
Audio RTP DSCP	Specify the value of the Audio RTP DSCP.
WAN Port VLAN Settings	
Enable WAN Port VLAN	Enable WAN Port VLAN by selecting it.
WAN Port VLAN ID	Specify the value of the WAN Port VLAN ID, the range of the value is 0-4095.
SIP 802.1p Priority	Specify the value of the sip 802.1p priority, the range of the value is 0-7.
Audio 802.1p Priority	Specify the value of the audio 802.1p priority, the range of the value is 0-7.
LAN Port VLAN Settings	
LAN Port VLAN	Follow WAN: Follow the WAN ID. Disable: Disable Port VALN. Enable: Enable Port VLAN and specify the Port VLAN ID different from WAN ID.
LAN Port VLAN ID	Specify the value of the Port VLAN ID different from WAN ID, the range of the value is 0-4095.

8.2.2.4 Service Port

You can set the port of telnet/HTTP/RTP by this page.

WAN
LAN
QoS&VLAN
SERVICE PORT
DHCP SERVICE
TIME&DATE

BASIC
NETWORK
VOIP
PHONE
FUNCTION KEY
MAINTENANCE
SECURITY
LOGOUT

Service Port Settings ⓘ

Web Server Type

HTTP

HTTP Port

80

HTTPS Port

443

Telnet Port

23

RTP Port Range Start

10000

RTP Port Quantity

200


Apply

Service Port	
Item	explanation
Web Server Type	Specify Web Server Type.
HTTP Port	Set web browser port, the default is 80 port, if you want to enhance system safety, you'd better change it into non-80 standard port; Example: The IP address is 192.168.1.70. and the port value is 8090, the accessing address is http://192.168.1.70:8090.
HTTPS Port	Before using the https, you must download https authentication certification into the phone, then set web browser port, the default is 443 port, if you want to enhance system safety, you'd better change it into non-443 standard port. You can access to the web in https after rebooting the phone.
Telnet Port	Set Telnet Port, the default is 23.
RTP Port Range Start	Set the RTP Start Port. It is dynamic allocation.
RTP Port Quantity	Set the maximum quantity of RTP Port, the default is 200.

Notice:

- 1) You need save the configuration and reboot the phone after setting this page.
- 2) Please REBOOT the system if you modify the HTTP or telnet port number (the new number should be greater than 1024).
- 3) If you set 0 for the HTTP port, it will disable HTTP service.

8.2.2.5 DHCP SERVICE



WAN

LAN

QoS&VLAN

SERVICE PORT

DHCP SERVICE

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DHCP Client Table

Leased IP Address	Client MAC Address
-------------------	--------------------

DHCP Lease Table

Name	Start IP	End IP	Leased Time	Subnet Mask	IP Gateway	DNS
lan	192.168.0.2	192.168.0.31	1360	255.255.255.0	192.168.0.1	192.168.0.1

DHCP Lease Table Settings

Leased Table Name

Start IP Address

End IP Address

Leased Time

Subnet Mask

IP Gateway

DNS Server Address

minute(s)

Add

DHCP Lease Table Delete

Leased Table Name

lan

Delete

DNS Relay

Enable DNS Relay

Apply

DHCP SERVICE	
Item	explanation
DHCP Lease Table	IP-MAC mapping table. If the LAN port of the phone connects to a device, this table will show the IP and MAC address of this device.
Shows the DHCP Lease Table, the unit of Lease time is Minute.	
Lease Table Name	Specify the name of the lease table.
Start IP Address	Set the start IP address of the lease table.
End IP Address	Set the end IP address of the lease table, the network device connected to LAN port will get IP address between Start IP and End IP by DHCP.
Leased Time	Set the Lease Time of the lease table.
Subnet Mask	Set the Net mask of the lease table.
IP Gateway	Set the Gateway of the lease table.
DNS	Set the default DNS server IP of the lease table; Click the Add button to submit and add this lease table.

DHCP Lease Table Delete

Leased Table Name

Select name of lease table, click the **Delete** button will delete the selected lease table from DHCP lease table.

DNS Relay

Enable DNS Relay ☒


Enable DNS Relay	Select DNS Relay, the default is enabled. Click the Apply button to become effective.
------------------	---

Notice:

- 1) The size of lease table cannot be larger than the quantity of C network IP address. We recommend you to use the default lease table and not modify it.
- 2) If you modify the DHCP lease table, you need save the configuration and reboot.

8.2.2.6 TIME&DATE

Setting time zone and SNTP (Simple Network Time Protocol) server according to your location, you can also manually adjust date and time in this web page.



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WAN
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SERVICE PORT
DHCP SERVICE
TIME&DATE

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LOGOUT

Simple Network Time Protocol (SNTP) Settings

Enable SNTP ☒

Enable DHCP Time ☐

Primary Server

Secondary Server

Timezone

Resync Period second(s)

12-Hour Clock ☐

Date Format

Apply

Daylight Saving Time Settings

Enable ☐

Offset minutes(s)

Month

Week

Day

Hour

Minute

Apply

Manual Time Settings

Year

Month

Day

Hour

Minute

Apply

TIME&DATE

Item	Explanation
Simple Network Time Protocol (SNTP) Settings	
Enable SNTP	Enable SNTP by selecting it.
Enable DHCP Time	Enable DHCP Time by selecting it, then the phone will automatically synchronize the standard time.
Primary Server	Set SNTP Primary Server IP address.
Secondary Server	Set SNTP Secondary Server IP address.
Time Zone	Select the Time zone according to your location.
Resync Period	Set the time out, the default is 60 seconds.
12 -Hour Clock	Switch the time mechanism between 12 hours and 24 hours. Default is 24 hours mode.
Date format	Specify the date format.
Daylight Saving Time Settings	
Enable	Enable daylight saving time.
Offset(minutes)	Setup the variety length.
Month	Setup start and end month.
Week	Setup start and end week.
Day	Setup start and end day.
Hour	Setup start and end hours.
Minute	Setup start and end minutes.

Manual Time Settings

Manual Time Settings

Year
 Month
 Day
 Hour
 Minute


Apply

Notice: First of all, you need to disable the SNTP service, and above the date hours minutes each of which is required to complete and submit to make manual.

8.2.3 VOIP

8.2.3.1 SIP

Set your SIP server in the following interface.



SIP

IAX2

STUN

DIAL PEER

MCAST

BASIC

NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT

SIP Line SIP 1

Basic Settings >>

Status Unapplied

Server Address

Server Port

Authentication User

Authentication Password

SIP User

Display Name

Enable Registration ☐

Domain Realm

Proxy Server Address

Proxy Server Port

Proxy User

Proxy Password

Backup Proxy Server Address

Backup Proxy Server Port

Server Name

Codecs Settings >>

Advanced SIP Settings >>

Apply

Codecs Settings >>

Disabled Codecs

G.711A
 G.711U
 G.722
 G.723.1
 G.726-32
 G.729AB



Enabled Codecs

↑
↓

Advanced SIP Settings >>

Forward Type	Disabled ▾	Enable Hotline	<input type="checkbox"/>
Forward Number	<input type="text"/>	Hotline Number	<input type="text"/>
No Ans. Fwd Wait Time	60 (0~120)second(s)	Warm Line Wait Time	0 (0~9)second(s)
Transfer Timeout	0 second(s)	BLF Server	<input type="text"/>
SIP Encryption	<input type="checkbox"/>	Enable Auto Answer	<input type="checkbox"/>
SIP Encryption Key	<input type="text"/>	Auto Answer Timeout	60 second(s)
RTP Encryption	<input type="checkbox"/>	Enable Session Timer	<input type="checkbox"/>
RTP Encryption Key	<input type="text"/>	Session Timeout	0 second(s)
Subscribe For MWI	<input type="checkbox"/>	Conference Type	Local ▾
MWI Number	*60	Conference Number	<input type="text"/>
Subscribe Period	3600 second(s)	Registration Expires	3600 second(s)
Enable Service Code	<input type="checkbox"/>	DND Off Code	<input type="text"/>
DND On Code	<input type="text"/>	Always CFwd Off Code	<input type="text"/>
Always CFwd On Code	<input type="text"/>	Busy CFwd Off Code	<input type="text"/>
Busy CFwd On Code	<input type="text"/>	No Ans. CFwd Off Code	<input type="text"/>
No Ans. CFwd On Code	<input type="text"/>	Ban Anonymous Off Code	<input type="text"/>
Ban Anonymous On Code	<input type="text"/>		
Keep Alive Type	SIP Option ▾	Keep Alive Interval	60 second(s)
User Agent	<input type="text"/>	Server Type	COMMON ▾
DTMF Type	AUTO ▾	RFC Protocol Edition	RFC3261 ▾
DTMF SIP INFO Mode	Send 10/11 ▾	Local Port	5060
Ring Type	Default ▾	Anonymous Call Edition	None ▾
Enable Rport	<input type="checkbox"/>	Keep Authentication	<input type="checkbox"/>
Enable PRACK	<input type="checkbox"/>	Ans. With a Single Codec	<input type="checkbox"/>
Enable Long Contact	<input type="checkbox"/>	Auto TCP	<input type="checkbox"/>
Convert URI	<input checked="" type="checkbox"/>	Enable Strict Proxy	<input type="checkbox"/>
Dial Without Registered	<input type="checkbox"/>	Enable GRUU	<input type="checkbox"/>
Ban Anonymous Call	<input type="checkbox"/>	Enable Displayname Quote	<input type="checkbox"/>
Enable DNS SRV	<input type="checkbox"/>	Enable user=phone	<input checked="" type="checkbox"/>
Enable Missed Call Log	<input checked="" type="checkbox"/>	Click To Talk	<input type="checkbox"/>
BLF List Number	<input type="text"/>	Transport Protocol	UDP ▾
Enable BLF List	<input type="checkbox"/>	Use VPN	<input checked="" type="checkbox"/>
Respond 182 when Call waiting	<input type="checkbox"/>	Enable DND	<input type="checkbox"/>

Apply

SIP Global Settings >>

Strict Branch	<input type="checkbox"/>	Enable Group	<input type="checkbox"/>
Registration Failure Retry Time	32 second(s)		

Apply

SIP Configuration

Item	explanation
------	-------------

SIP Line

Choose line to set info about SIP, there are 4 lines to choose. You can switch by **【Load】** button.

Basic Settings

Status	Shows if the phone has been registered the SIP server or not; or so, show Unapplied.
--------	--

Server Address	Input your SIP server address.
----------------	--------------------------------

Server Port	Set your SIP server port.
Authentication User	Input your SIP register account name.
Authentication Password	Input your SIP register password.
SIP User	Input the phone number assigned by your VoIP service provider. Phone will not register if there is no phone number configured.
Display Name	Set the display name.
Proxy Server Address	Set proxy server IP address (Usually, Register SIP Server configuration is the same as Proxy SIP Server. But if your VoIP service provider gives different configurations between Register SIP Server and Proxy SIP Server, you need make different settings).
Proxy Server Port	Set your Proxy SIP server port.
Proxy User	Input your Proxy SIP server account.
Proxy Password	Input your Proxy SIP server password.
Domain Realm	Set the sip domain if needed, otherwise this VoIP phone will use the Register server address as sip domain automatically. (Usually it is same with registered server and proxy server IP address).
Backup Server Address	Input the Backup Server Address, if the primary server is unavailable, then the phone will enable the Backup Server Address.
Backup Server Port	Specify the Backup Server Port.
Enable Registration	Start to register or not by selecting it or not.
Codecs Settings	
Disable Codecs/Enable Codecs	Use the navigation keys to highlight the desired one in the Enable/Disable Codecs list, and press the desired to move to the other list.
Advanced SIP Setting	
Forward Type	Select call forward mode, the default is Disabled.
	Off: Close down calling forward.
	Busy: If the phone is busy, incoming calls will be forwarded to the appointed phone.
	No answer: If there is no answer, incoming calls will be forwarded to the appointed phone after a specific.
	Always: Incoming calls will be forwarded to the appoint phone immediately.
The phone will prompt the incoming while doing forward.	
Forward Number	Specify the number you want to forward.
No Answer Forward Wait Time	Specify the No Answer Forward Delay Time, if the Forward Type is No answer, incoming calls will be forwarded after the no answer forward wait time.
Enable Hot Line	Specify Hot Line by selecting it.
Hot Line Number	Specify Hot Line Number, the phone dial the hot line number automatically at hands-free mode or handset mode after warm line time.
Warm Line Wait Time	Specify the Warm Line Time.
Transfer Timeout	For the phone supports the transfer of certain special features server, set interval time between sending "bye" and hanging up after the phone transfers a call.
BLF Server	Ordinary BLF application is that the phone send subscription package to the registered server, if your server does not support subscription package, please input the BLF server so that it can separate register server and BLF server

SIP Encryption	Enable/Disable SIP Encryption.
SIP Encryption Key	Set the key for sip encryption.
RTP Encryption	Enable/Disable RTP encryption.
RTP Encryption Key	Set the key for RTP encryption.
Enable Auto Answer	Enable Auto Answer by selecting it.
Auto Answer Timeout	Specify Auto Answer Time, the phone auto answers the incoming call after Auto Answer Time.
Enable Session Timer	Set Enable/Disable Session Timer, whether support RFC4028.It will refresh the SIP sessions.
Session Timeout	Set the session timeout.
Subscribe for MWI	Enable the Subscribe for MWI by selecting it, the phone will send subscribe message for MWI to the SIP Server.
MWI Number	Specify the MWI Number; Please contact your system administrator for the connecting code. Different systems have different codes.
Subscribe Period(s)	Overtime of resending subscribe packet. Suggest using the default configuration.
Conference Type	Specify the Conference Type, if you select the local, you needn't input the conference number.
Conference Number	Specify the network conference number, please contact your system administrator for the network conference number.
Registration Expire(s)	Set expire time of SIP server register, default is 60 seconds. If the register time of the server requested is longer or shorter than the expired time set, the phone will change automatically the time into the time recommended by the server, and register again.
Enable Service Code	If you want to realize the following function by the server, please enter the On Code and Off Code option, then when you choose to enable/disable following function on your IP phone, it will send message to the server, and the server will turn on/off the function immediately.
DND On Code	Set the DND On Code, When you press the DND hot key, the phone will send a message to the server, and the server will turn on the DND function. Then any calls to the extension will be rejected by the server automatically. And the incoming call record will not be displayed in the Call History.
DND Off Code	Set the DND Off Code, When you press the DND hot key, the phone will send a message to the server, and the server will turn off the DND function.
Always CFwd On Code	Set the Always CFwd On Code, when you choose to enable the always forward function on your phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will always forward it to the set number automatically. And the IP phone will not show the record in the call history anymore.
Always CFwd Off Code	Set the Always CFwd Off Code, when you choose to disable the always forward function on your phone, it will send message to the server, and the server will turn off the function immediately.
Busy CFwd On Code	Set the Busy CFwd On Code, when you choose to enable the busy forward function v on your phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will forward

	it to the set number automatically based the forward type. And the IP phone will not show the record in the call history anymore.
Busy CFwd Off Code	Set the Busy CFwd Off Code, when you choose to disable the busy forward function on your phone, it will send message to the server, and the server will turn off the function immediately.
No Answer CFwd On Code	Set the No Answer CFwd On Code, when you choose to enable the on answer forward function on your phone, it will send message to the server, and the server will turn on the function immediately. When there are calls to the extension, the server will forward it to the set number automatically based the forward type. And the IP phone will not show the record in the call history anymore.
No Answer CFwd Off Code	Set the No Answer CFwd Off Code, when you choose to disable the busy forward function on your phone, it will send message to the server, and the server will turn off the function immediately.
Anonymous On Code	Set the Anonymous On Code, When you choose to enable the anonymous call function on your IP phone, it will send information to the server, and the server will enable the anonymous call function for your IP phone automatically.
Anonymous Off Code	Set the Anonymous Off Code, When you choose to disable the anonymous call function on your IP phone, it will send information to the server, and the server will disable the anonymous call function for your IP phone automatically.
Keep Alive Type	Specify the keep alive type, if the type is option, the phone will send option sip message to server every NAT Keep Alive Period(s), then the server responses with 200 to keep alive. If the type is UDP, the phone will send UDP message to server to keep alive every NAT Keep Alive Period(s).
Keep Alive Interval	Set examining interval of the server, default is 60 seconds.
User Agent	Set the user agent if have, the default is VoIP Phone 1.0.
DTMF Type	<p>Select DTMF sending mode, there are three modes:</p> <ul style="list-style-type: none"> ● DTMF_RELAY ● DTMF_RFC2833 ● DTMF_SIP_INFO <p>Different VoIP Service providers may provide different modes.</p>
DTMF SIP INFO Mode	There are two options: send 10/11 and send * / #
Local Port	Set sip port of each line.
Ring Type	Set ring type of each line.
Enable Via Rport	Enable/Disable system to support RFC3581. Via Rport is special way to realize SIP NAT.
Enable PRACK	Enable or disable SIP PRACK function, suggest use the default configuration.
Enable Long Contact	Set more parameters in contact field; connection with SEM server.
Convert URI	Convert # to %23 when send the URI.
Dial Without Registered	Set call out by proxy without registration;
Ban Anonymous Call	Set to ban Anonymous Call;
Enable DNS SRV	Support DNS looking up with _sip.udp mode.
Server Type	Select the special type of server which is encrypted, or has some unique requirements or call flows.
RFC Protocol Edition	Select SIP protocol version to adapt for the SIP server which uses the same version

	as you select. For example, if the server is CISCO5300, you need to change to RFC2543; else phone may not cancel call normally. System uses RFC3261 as default.
Transport Protocol	Set transport protocols, TCP or UDP;
Anonymous call Edition	Set Anonymous call out safely; Support RFC3323and RFC3325;
Keep Authentication	Enable/Disable Keep Authentication System will take the last authentication field which is passed the authentication by server to the request packet. It will decrease the server's repeat authorization work, if it is enable.
Answer With A Single Codec	Enable/Disable the function when call is incoming, phone replies SIP message with just one codec which phone supports.
Auto TCP	Set to use automatically TCP protocol to guarantee usability of transport as message is above 1300 byte
Enable Strict Proxy	Support the special SIP server-when phone receives the packets sent from server, phone will use the source IP address, not the address in via field.
Enable GRUU	Set to support GRUU
Enable Display name Quote	Set to make quotation mark to display name as the phone sends out signal, in order to be compatible with server.
Enable user=phone	Enable user=phone by selecting it, it is contained in the invite sip message, in order to be compatible with server.
Enable Missed Call Log	Enable the missed call log by it, the phone will save the missed call log into the call history record and display the missed calls on the idle screen, or won't save the missed call log into the call history record and display the missed calls on the idle screen.
Click to talk	Set click to Talk (need practical software support).
Enable BLF List	Enable BLF List by selecting it, BLF list is a function which can monitor the group status, it is not one to one monitoring, but the information feedback from the server to decide which BLF list will monitor.
Use VPN	Phone use VPN IP to communicate
BLF List Number	Specify the BLF List Number.
SIP Global Settings	
Strict Branch	Enable the Strict Branch, the value of the branch must be in the beginning of z9hG4k in via field of the invite sip message received, or the phone won't response to the invite sip message. Notice: the deployment will become effective in all sip lines.
Enable Group	Enable Group by selecting it, then the phone enable the sip group backup function. Notice: the deployment will become effective in all sip lines.
Registration Failure Retry Time	Specify the registration failure retry time, if the phone register failed, the phone will register again after registration failure retry time. Notice: the deployment will become effective in all sip lines.

8.2.3.2 IAX2

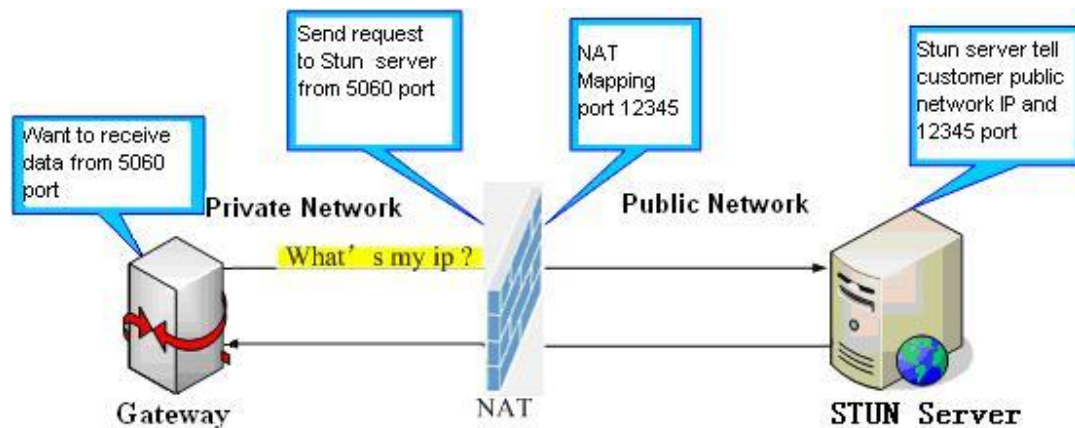
IAX2 Configuration

Item	explanation
Status	Shows if the phone has been registered the IAX2 server or not.
Server Address	Input your IAX2 server address.
Server Port	Set your IAX2 server port, the default is 4569.
Account	Input your IAX2 register account name.
Password	Input your IAX2 register password.
Phone Number	Input your assigned phone number (usually it is same you're your IAX2 account name).
Local Port	Set your local sport, the default is 4569.
Voice Mail Number	Specify the voice mail's number.
Voice Mail Text	Specify the voice mail's name.
Echo Test Number	Set echo test number. If IAX2 server supports echo test, and echo test number is non-numeric, system could set an echo test number to replace the echo test text. So user can dial the numeric number to test echo voice test. This function is provided with server to make endpoint to test whether endpoint could talk through server normally.
Echo Test Text	Specify echo test text's name.
Refresh Time	Set expire time of IAX2 server register, you can set it between 60 and 3600 seconds.
Enable Registration	Start to register the IAX2 server or not by selecting it or not.
Enable G.729AB	Enable or disable code G.729 by selecting it or not.

8.2.3.3 Stun

In this web page, you can configure SIP STUN.

STUN: By STUN server, the phone in private network could know the type of NAT and the NAT mapping IP and port of SIP. The phone might register itself to SIP server with global IP and port to realize the device both calling and being called in private network.



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Simple Traversal of UDP through NATs (STUN) Settings

STUN NAT Traversal

FALSE

Server Address

Server Port

Binding Period

second(s)

SIP Waiting Time

millisecond(s)

Local SIP Port

SIP Line Using STUN

SIP 1

Use STUN

☐

STUN

Item	explanation
Simple Traversal of UDP through NATs (STUN) Settings	
STUN NAT Traversal	Shows STUN NAT Transverse estimation, true means STUN can penetrate NAT, while False means not.
Server Address	Set your SIP STUN Server IP address.
Server Port	Set your SIP STUN Server Port.
Blinding Period(s)	Set STUN blinding period(s). If NAT server finds that a NAT mapping is idle after time out, it will release the mapping and the system need send a STUN packet to keep the mapping effective and alive.
SIP Waiting Time	Specify the sip wait stun time; you can input the time depended on your network condition.
Local SIP Port	Configuration the local SIP Port, the default value is5060(this port immediate effect ,modify, SIP call will use the modified port communication)
Sip Line Using STUN	

SIP Line Using STUN

SIP 1 ▼

Use STUN

☐

Apply

Choose line to set info about SIP, There are 2 lines to choose.

Use STUN

Enable/Disable SIP STUN.

Notice: SIP STUN is used to realize SIP penetration to NAT. If your phone configures STUN Server IP and Port (default is 3478), and enable SIP Stun, you can use the ordinary SIP Server to realize penetration to NAT.

8.2.3.4 DIAL PEER

This functionality offers you more flexible dial rule, you can refer to the following content to know how to use this dial rule. When you want to dial an IP address, the entry of IP addresses is very cumbersome, but by this functionality, you can set number 156 to replace 192.168.1.119 here.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
133	192.168.1.108	5060	SIP	no alias	no suffix	0

When you want to dial a long distance call to Beijing, you need dial an area code 010 before local phone number, but you can also dial number 1 instead of 010 after we make a setting according to this dial rule. For example, you want to dial 01062213123, but you need dial only 162213123 to realize your long distance call after you make this setting.

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
1T	0.0.0.0	5060	SIP	rep:010	no suffix	1

To save the memory and avoid abundant input of user, add the following functions:

Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
1T	0.0.0.0	5060	SIP	rep:010	no suffix	1
18xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
13(5-9)xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0

1. x matches any single digit, for example:

If user makes the above configuration, after user dials 11 digit numbers started with 13, the phone will send out 0 plus the dialed numbers automatically.

2. [] Specifies a range that will match digit. It may be a range, a list of ranges separated by commas, or a list of digits.

If user makes the above configuration, after user dials 11 digit numbers started with from 135 to 139, the phone will send out 0 plus the dialed numbers automatically.

By this phone, you can realize dialing out via different lines without switch in web interface .

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Dial Peer Table

Number	Destination	Port	Mode	Alias	Suffix	Deleted Length
1T	0.0.0.0	5060	SIP	rep:010	no suffix	1
18xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0
13(5-9)xxxxxxxx	0.0.0.0	5060	SIP	add:0	no suffix	0

Add Dial Peer

Phone Number

Destination(Optional)

Port(Optional)

Alias(Optional)

Call Mode

Suffix(Optional)

Deleted Length(Optional)

SIP
▼

Dial Peer Option

▼

DIAL PEER

Examples of Different Alias Application

Set by web	explanation	example
 <p>Phone Number: 9T Destination (optional): 255.255.255.255 Port(optional): Alias(optional): del Call Mode: SIP Suffix(optional): Delete Length (optional): 1</p>	<p>You need set phone number, Destination, Alias and Delete Length. Phone number is XXXT; Destination is 255.255.255.255 (0.0.0.2) and Alias is del.</p> <p>This means any phone No. that starts with your set phone number will be sent via SIP2 line after the first several digits of your dialed phone number are deleted according to delete length.</p>	<p>If you dial “93333”, the SIP2 server will receive “3333”.</p>
 <p>Phone Number: 2 Destination (optional): Port(optional): Alias(optional): all:33334444 Call Mode: SIP Suffix(optional): Delete Length (optional):</p>	<p>This setting will realize speed dial function, after you dialing the numeric key “2”, the number after all will be sent out.</p>	<p>When you dial “2”, the SIP1 server will receive 33334444.</p>
 <p>Phone Number: 8T Destination (optional): Port(optional): Alias(optional): add:0755 Call Mode: SIP Suffix(optional): Delete Length (optional):</p>	<p>The phone will automatically send out alias number adding your dialed number, if your dialed number starts with your set phone number.</p>	<p>When you dial “8309”, the SIP1 server will receive “07558309”.</p>
 <p>Phone Number: 010T Destination (optional): Port(optional): Alias(optional): rep:0086 Call Mode: SIP Suffix(optional): Delete Length (optional): 3</p>	<p>You need set Phone Number, Alias and Delete Length. Phone number is XXXT and Alias is rep: xxx</p> <p>If your dialed phone number starts with your set phone number, the first digits same as your set phone number will be replaced by the alias number specified and New phone number will be send out.</p>	<p>When you dial “0106228”, the SIP1 server will receive “86106228”.</p>
 <p>Phone Number: 147 Destination (optional): Port(optional): Alias(optional): Call Mode: SIP Suffix(optional): 0011 Delete Length (optional):</p>	<p>If your dialed phone number starts with your set phone number. The phone will send out your dialed phone number adding suffix number.</p>	<p>When you dial “147”, the SIP1 server will receive “1470011”.</p>

8.2.3.5 MCAST

MCAST Settings

Priority: 1

Enable Page Priority: ☐

Index/Priority	Name	Host:port
1		
2		
3		
4		
5		
6		
7		
8		
9		
10		

Apply

Use the multicast function to send notice to every member of the multicast is simple and easy. By setting the multicast key on your phone, you can send multicast RTP flow to the pre-configured multicast address. By listening multicast address is configured on the phone, listen and play the multicast address to send the RTP stream.

Send Multicast Settings

On the phone web page, function key-function key, set a function key as shown below.

DSS Key 8 Multicast 239.1.1.1:1366 AUTO G.711A

Value format IP: Port, the IP address of multicast is range from 224.0.0.0 to 239.255.255.255, port is greater than 1024

If multicast codec is G722, the LCD screen will displays "HD", which means the phone is sending high-definition voice stream

Operate steps:

- 1、 When the phone is idle, press multicast key

Multicast RTP stream is sent to pre-configured multicast address (IP: Port).The phone which listens to multicast address in the local network can receive the RTP stream. Multicast function key LED lights yellow.

LCD screen displays the following:



- 2、 Press the hold key to hold the current multicast session
- 3、 Press the end key again or multicast function key, multicast session can be stopped

Notice: RTP stream is one side that is from a sender to a receiver. When the phone initiates a multicast RTP session in a call, the current call is on hold.

Receive Multicast Settings

You can set up the phone monitoring 10 different multicast addresses to receive these multicast RTP stream.

You have two methods to receive RTP stream of multicast that can be set up through the web page:

Enable priorities of normal calls and Enable page Priority:

Enable priorities of normal call by select it, if the incoming RTP stream priority of multicast lower than the priority of current for normal calls, the phone will ignore the RTP stream of multicast. If the incoming RTP stream priority of multicast higher than the priority of current for normal calls, the phone will receive the RTP stream of multicast, and hold the current call.

Disabled priorities of normal call by select disable, the phone will ignore all local networks RTP stream of multicast.

Options as follows:

1-10: the priority defined for normal calls, 1 the highest level, 10 the lowest level

Disabled: Ignore all RTP stream of multicast

Enable Page Priority

Page priority determines the phone how to handle the newly received multicast RTP stream when in a multicast session. Enabled page priority, the phone will automatically ignore the low priority multicast RTP stream and receive the high priority multicast RTP stream and hold the current multicast session; If not enabled, the phone will automatically ignore all incoming multicast RTP stream.

Web page is set as follows:

MCAST Settings		
Priority	1	
Enable Page Priority	<input type="checkbox"/>	
Index/Priority	Name	Host:port
1	ss	239.1.1.1:1366
2	ee	239.1.1.1:1367

Now multicast “ss” has higher priority than multicast “ee”, the highest priority is for normal calls

Notice: When a multicast session begins, multicast sender and receiver will beep

8.2.4 PHONE

8.2.4.1 AUDIO

In this page, you can configure voice codec, input/output volume and so on.

ZYCOO

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Audio Settings

First Codec	G.711A	▼	Second Codec	G.711U	▼
Third Codec	G.722	▼	Fourth Codec	G.729AB	▼
Fifth Codec	None	▼	Sixth Codec	None	▼
Onhook Time	200	millisecond(s)	Tone Standard	China	▼
Handset Volume	5	(1~9)	Default Ring Type	Type 1	▼
Speakerphone Volume	5	(1~9)	Headset Ring Volume	5	(1~9)
Headset Volume	5	(1~9)	Speakerphone Ring Volume	5	(1~9)
G.729AB Payload Length	20ms	▼	G.723.1 Bit Rate	6.3kb/s	▼
G.722 Timestamps	160/20ms	▼	DTMF Payload Type	101	(96~127)
Enable VAD	<input type="checkbox"/>		Enable MWI Tone	<input checked="" type="checkbox"/>	


Apply

AUDIO Configuration

Item	explanation
First Codec	The first preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB, None.
Second Codec	The second preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB, None.
Third Codec	The third preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB, None.
Fourth Codec	The forth preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB, None.
Fifth Codec	The fifth preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB, None.
Sixth codec	The sixth preferential DSP codec: G.711A/u, G.722, G.723.1,726-32 G.729AB, None.
Onhook Time	Specify the least reflection time of Hand down, the default is 200ms.
Default Ring Type	Set up the ring by default.
Handset Output Volume	Specify Output (receiver) Volume grade.
Speakerphone volume	Specify Speakerphone Volume grade.
G729AB Payload Length	Set G729 Payload Length.
Tone Standard	Select Tone Standard.
G722 Timestamps	160/20ms or 320/20ms is available.
G723.1 Bit Rate	5.3 kb/s or 6.3 kb/s is available.
Enable VAD	Select it or not to enable or disable VAD. If enable VAD, G729 Payload length could not be set over 20ms.
DTMF Payload Type	Set DTMF Payload Type.

8.2.4.2 FEATURE

In this web page, you can configure Hotline, Call Transfer, Call Waiting, 3 Ways Call, Black List, White list Limit List and so on.



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Feature Settings

DND (Do Not Disturb) Disabled ▼	Ban Outgoing <input type="checkbox"/>
Enable Call Transfer <input checked="" type="checkbox"/>	Enable Call Waiting <input checked="" type="checkbox"/>
Semi-Attended Transfer <input checked="" type="checkbox"/>	Enable 3-way Conference <input checked="" type="checkbox"/>
Enable Auto Handdown <input checked="" type="checkbox"/>	Accept Any Call <input checked="" type="checkbox"/>
Auto Handdown Time <input type="text" value="3"/> second(s)	Enable Call Completion <input type="checkbox"/>
Enable Auto Redial <input type="checkbox"/>	Enable Pre-Dial <input checked="" type="checkbox"/>
Auto Redial Interval <input type="text" value="10"/> (1~180)second(s)	Enable Silent Mode <input type="checkbox"/>
Auto Redial Times <input type="text" value="10"/> (1~100)	Hide DTMF Disabled ▼
Auto Headset <input type="checkbox"/>	Ring From Headset <input type="checkbox"/>
Enable Intercom <input checked="" type="checkbox"/>	Enable Intercom Mute <input type="checkbox"/>
Enable Intercom Tone <input checked="" type="checkbox"/>	Enable Intercom Barge <input checked="" type="checkbox"/>
P2P IP Prefix <input type="text" value="."/>	DND Return Code 480(Temporarily Not Available) ▼
Turn Off Power Light <input checked="" type="checkbox"/>	Busy Return Code 486(Busy Here) ▼
Emergency Call Number <input type="text" value="110"/>	Reject Return Code 603(Decline) ▼
Enable Password Dial <input type="checkbox"/>	Active URI Limit IP <input type="text"/>
Password Dial Prefix <input type="text"/>	Push XML Server <input type="text"/>
Password Length <input type="text" value="0"/> (0~31)	Enable Call Waiting Tone <input checked="" type="checkbox"/>
Enable Call History <input checked="" type="checkbox"/>	Enable Multi Line <input checked="" type="checkbox"/>
Enable Default Line <input type="checkbox"/>	Enable Auto Switch Line <input checked="" type="checkbox"/>
Allow IP Call <input checked="" type="checkbox"/>	Play Dialing DTMF Tone <input checked="" type="checkbox"/>
Play Talking DTMF Tone <input checked="" type="checkbox"/>	

Block Out Settings

Block Out

▼

Action URL Settings

Setup Completed	<input style="width: 300px;" type="text"/>
Registration Success	<input style="width: 300px;" type="text"/>
Registration Disabled	<input style="width: 300px;" type="text"/>
Registration Failed	<input style="width: 300px;" type="text"/>
Off Hook	<input style="width: 300px;" type="text"/>
On Hook	<input style="width: 300px;" type="text"/>
Incoming Call	<input style="width: 300px;" type="text"/>
Outgoing Call	<input style="width: 300px;" type="text"/>
Call Established	<input style="width: 300px;" type="text"/>
Call Terminated	<input style="width: 300px;" type="text"/>
DND Enabled	<input style="width: 300px;" type="text"/>
DND Disabled	<input style="width: 300px;" type="text"/>
Always Forward Enabled	<input style="width: 300px;" type="text"/>
Always Forward Disabled	<input style="width: 300px;" type="text"/>
Busy Forward Enabled	<input style="width: 300px;" type="text"/>
Busy Forward Disabled	<input style="width: 300px;" type="text"/>
No Ans. Forward Enabled	<input style="width: 300px;" type="text"/>
No Ans. Forward Disabled	<input style="width: 300px;" type="text"/>
Transfer Call	<input style="width: 300px;" type="text"/>
Blind Transfer Call	<input style="width: 300px;" type="text"/>

Attended Transfer Call	<input type="text"/>
Hold	<input type="text"/>
Resume	<input type="text"/>
Mute	<input type="text"/>
Unmute	<input type="text"/>
Missed Call	<input type="text"/>
IP Changed	<input type="text"/>
Idle To Busy	<input type="text"/>
Busy To Idle	<input type="text"/>

Block Out Settings

Block Out			
<input type="text"/>	<input type="button" value="Add"/>	<input type="button" value="▼"/>	<input type="button" value="Delete"/>

FEATURE	
Item	explanation
Do Not Disturb	Select DND, the phone will reject any incoming call, the callers will be reminded by busy, but any outgoing call from the phone will work well.
Ban Outgoing	If you select Ban Outgoing to enable it, and you cannot dial out any number.
Enable Call Transfer	Enable Call Transfer by selecting it.
Semi-Attended Transfer	Enable Semi-Attended Transfer by selecting it.
Enable Auto Redial	Enable Auto Redial, the phone reminds you whether redial when the caller is busy or rejects.
Auto Redial interval	Specify the Auto Redial interval.
Auto Redial Times	Specify the Auto Redial interval.
Enable Call Completion	Enable Call Completion by selecting it.
Enable Pre-Dial	Disable this feature, in standby interface next number, will realize the number rules "send out over the time";Enable the feature ,then the number will not be send out over the time.
Enable Call Waiting	Enable Call Waiting by selecting it. Then the phone reminds whether redial, when the caller is busy or rejects. if it's ok and the phone finds out that the caller is idle by sip message, it will reminds whether redial.
Enable 3-way Conference	Enable 3-way conference by selecting it.
Enable Call Waiting Tone	Disable this function ,you will not hear the tone "beep" when there have multiple incoming calls
Accept Any Call	If select it, the phone will accept the call even if the called number doesn't belong to the phone.
Enable Auto Hand down	The phone will hang up and return to the idle automatically at hands-free mode.
Auto Handdown	Specify Auto Hand down Time, the phone will hang up and return to the idle automatically

Time	after Auto Hand down Time at hands-free mode, and play dial tone Auto Hand down Time at handset mode.
Ring From Headset	Enable Ring From Handset by selecting it, the phone plays ring tone from handset.
Enable Intercom	Enable Intercom Mode by selecting it.
Enable Intercom Mute	Enable mute mode during the intercom call.
Enable Intercom Tone	If the incoming call is intercom call, the phone plays the intercom tone.
Enable Intercom Barge	Enable Intercom Barge by selecting it, the phone auto answers the intercom call during a call. If the current call is intercom call, the phone will reject the second intercom call.
Enable Silent Mode	Enable Silent Mode, the phone light will blink in red to remind that there is a missed call instead of playing ring tone.
Turn Off Power Light	Turn Off the Power Light.
Emergency Call Number	Specify the Emergency Call Number. Despite the keyboard is locked, you can dial the emergency call number.
Enable Password Dial	If we dial the configured passwords prefix, the following N numbers will be hidden and instead of *. N=passwords length. E.g.: If the passwords prefix is 3, and passwords length is 2 numbers; when you input 34567, the phone will display 3**67.
Password Length	Specify the Password length.
DND Return Code	Specify DND Return code.
Busy Return Code	Specify Busy Return Code.
Reject Return Code	Specify Reject Return Code.
Hide DTMF	Specify the hide DTMF mode.
Push XML Server	Specify the Push XML Server, when phone receives request, it will determine whether to display corresponding content on the phone which sent by the specified server or not.
P2P IP Prefix	Set Prefix in peer to peer IP call. For example: what you want to dial is 192.168.1.119, If you define P2P IP Prefix as 192.168.1., you dial only #119 to reach 192.168.1.119. Default is “.”. If there is no “.” Set, it means to disable dialing IP.
Active URI Limit IP	Specify the server IP that remote control phone for corresponding operation.
Action URL Settings	
Action URL Settings	Specify the Action URL that Record the operation of phone; send this corresponding information to server, url: http://InternalServer /FileName.xml? (Internal Server is server IP. Filename is name of xml that contains the action message).
Block Out Settings	
	Set Add/Delete Limit List. Please input the prefix of those phone numbers which you forbid the phone to dial out. For example, if you want to forbid those phones with prefix 001 to be dialed out, you need input 001 in the blank of limit list, and then you cannot dial out any phone number whose prefix is 001.

Block out Wildcard x means matching any single digit. For example, 4xxx expresses any number with prefix 4 which length is 4 will be forbidden to be dialed out; 6 expresses any number with prefix 6 will be forbidden to be dialed out.

Notice: Black List and Limit List can record at most 10 items respectively.

Notice: Black List and Limit List can record at most 10 items respectively.

8.2.4.3 DIAL PLAN

This system supports 7 dial modes:

- 1) End with "#": dial your desired number, and then press #.
- 2) Fixed Length: the phone will intersect the number according to your specified length.
- 3) Time Out: After you stop dialing and waiting time out, system will send the number collected.
- 4) Press # to Do Blind Transfer: input the number you want to transfer to then press "#" you can transfer the current call to the number.
- 5) Blind Transfer on On-Hook : input the number you want to transfer to then hang up handle or press speaker, you can transfer the current call to the number.
- 6) Attend Transfer on On-Hook: hang up handle or press speaker you can realize the blind transfer function
- 7) Press the DSS key Blind: Press DSS key, the current call will turn out blind.
- 8) User defined: you can customize digital map rules to make dialing more flexible. It is realized by defining the prefix of phone number and number length of dialing.

In order to maintain the end-user pbx secondary dial for dialing call mode. When requested to enter a phone number prefix, the system according to the rules in the closing number configuration rules, re-issue the dial tone, the user continues to enter the number, after the end of the closing number, the phone number will be prefixed and analog secondary dial tone is sent to the back of the numbers together server.

For example:

In the list of rules in the configuration of the closing number 9, xxxxxxxx then when the user dials 9, the system to re-play the dial tone, dial the number the user to continue; dial-up is complete, the phone is actually sent containing 9 9 numbers.

DIAL PLAN Configuration	
Item	Explanation
Basic Setting	
Press "#" to Send	Set Enable/Disable the phone ended with “#” dial.
Dial Fixed Length	Specify the Fixed Length of phone ending with.
Press # to Do Blind Transfer	Enable Blind Transfer On Hook, when executing Blind Transfer End with #, press # after inputting the number that you want to transfer, the phone will transfer the current call to the third party.
Blind Transfer on Onhook	Enable Blind Transfer on Onhook, when executing Blind Transfer, hang up after inputting the number that you want to transfer, the phone will transfer the current call to the third party.
Attend Transfer on Onhook	Enable Attend Transfer on Onhook, when executing Attended Transfer, hang up after the third party answers, the phone will transfer the current call to the third party.

Dial Plan Table

Plans:

Below is user-defined digital map rule:

[] Specifies a range that will match digit. Maybe a range, a list of ranges separated by commas, or a list of digits.

* Match any single digit that is dialed.

. Match any arbitrary number of digits including none.

Tn Indicates an additional time out period before digits are sent of n seconds in length. n is mandatory and can have a value of 0 to 9 seconds. Tn must be the last 2 characters of a dial plan. If Tn is not specified it is assumed to be T0 by default on all dial plans.

Dial Plan Table

Plans:

"[1-8]xxx"

"911"

"9xxxxxxx"

"9911x.T4"

"99T4"

[1-8]xxx

Cause

extensions 1000-8999 to be dialed immediately.

Cause 8 digit numbers started with 9 to be dialed immediately.

Cause 911 to be dialed immediately after it is entered.


Cause 99 to be dialed after 4 seconds.

Cause any number started with 9911 to be dialed 4 seconds after dialing ceases.

Notice: End with “#”, Fixed Length, Time out and Digital Map Table can be used simultaneously, System will stop dialing and send number according to your set rules.

8.2.4.4 CONTACT

You can input the name, phone number and select ring type for each name here.



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Phonebook Table

Group: All [Hangup](#)

Index	Name	Office Number	Mobile Number	Other Number	Ring Type	Group	
Page: 	Pre	Next	friend	Add	Add to Blacklist	Delete	Delete All

Add Contact

Name:
Office Number:
Mobile Number:
Other Number:

Ring Type: Default
Line: Auto
Line: Auto
Line: Auto

Group Setting: Unselected

friend
home
work
business
classmate

Add
Modify

Selected

Clear

Import Contact List

Select File: Browse (*.xml,*.vcf,*.csv) Update

Export Contact List

Export XML
Export CSV
Export VCF

Group Option

Group: friend
Name:
Ring Type: Default

Add
Modify
Delete
Delete All

Blacklist Settings

Blacklist Item: Delete Delete All

Type: Number
Value: Add

Line: Auto

Blacklist

Contact

Item

Explanation

Phonebook Table

Name

Shows the name corresponding to the phone number.

Phonebook Table

Group

All

Hangup

Index

Name

Office Number

Mobile Number

Other Number

Ring Type

Group

Page:

Pre

Next

friend

Add

Add to Blacklist

Delete

Delete All

Shows the

detail of current phonebook.

Shows the

detail of current phonebook.

Notice: the maximum capability of the phonebook is 500 items, you can select many or a contact to add to group and add to blacklist, and delete many or a contact, and delete all contacts.

Add Contact List	
Name	Specify the name corresponding to the phone number.
Office Number	Specify the office number.
Mobile Number	Specify the mobile number.
Other Number	Specify the other number.
Ring Type	Specify the ring type for the phone number.
Line	Specify the sip line for the each number.
Group setting	Select the group from the unselected group to selected list for the contact; you can select many groups for the contact.

Notice: the add button for adding a new contact, the modify button for modifying the added contact, the clear all button for clear all input information of the contact.

Group Option	
Group	Select the added groups then modify or delete and so on.
Name	Input the name of the group, then click the add button, you can add a new group.
Ring Type	Specify the ring type for the group as adding a new group.

Blacklist Settings	
Type	Select the blacklist type, you can select number or prefix of number.
Value	Input number or prefix of number.
Line	Select the sip line.

Notice: the add button for adding a new blacklist, the delete button for deleting one item, the delete all button for deleting all items.

If user does not want to answer some phone calls, add these phone numbers to the Black List, and these calls will be rejected x and are wildcard x means matching any single digit. For example, 4xxx expresses any number with prefix 4 which length is 4 will be forbidden to be responded.

DOT (.) means matching any arbitrary number digit. For example, 6. expresses any number with prefix 6 will be forbidden to be responded.

If user wants to allow a number or a series of number incoming, he may add the number(s) to the list as the white list rule. The configuration rule is -number, for example, -123456, or -1234xx.

Blacklist
-4119

Means any incoming number is forbidden except for 4119

Notice: End with DOT (.) when set up the white list.

8.2.4.5 REMOTE CONTACT

You need to match a XML Phonebook address and you can directly access to the corresponding remote phonebook on the phone.

For example: Set the Phonebook Name as ZYCOO, Server URL is
 tftp://192.168.1.3/admin/phonebook/index.xml.

Or Set the Phonebook Name as ldap, Server URL is ldap://192.168.1.3/dc=winline,dc=com.


Remote Phonebook Settings	
Phonebook Name	Custom the phonebook name displayed on the phone.
Server URL	Specify the server URL of the remote phonebook.
SIP Line	Specify the sip line for the remote phonebook.
Authentication	Specify the authentication mode for remote phonebook.
User/password	Input the authentication username and password.

8.2.4.6 WEB DIAL

You can make a call through the WEB DIAL, enter the Dial Number then press Dial, if you want to finish the talk, press Hang-up.

8.2.5 FUNCTION KEY

8.2.5.1 FUNCTION KEY



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SOFTKEY

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Screen Configuration

Contrast (1~9)
 Backlight Time

Enable Backlight ☒

Line Key Settings

Line Key	Type	Value	Line	Subtype	Pickup Number
Line Key 1	<input type="text" value="Line"/>	<input type="text"/>	<input type="text" value="SIP1"/>	<input type="text" value="None"/>	<input type="text"/>
Line Key 2	<input type="text" value="Line"/>	<input type="text"/>	<input type="text" value="SIP2"/>	<input type="text" value="None"/>	<input type="text"/>
Line Key 3	<input type="text" value="Line"/>	<input type="text"/>	<input type="text" value="SIP3"/>	<input type="text" value="None"/>	<input type="text"/>
Line Key 4	<input type="text" value="Line"/>	<input type="text"/>	<input type="text" value="SIP4"/>	<input type="text" value="None"/>	<input type="text"/>

Function Key Settings

Key	Type	Value	Line	Subtype	Pickup Number
DSS Key 1	<input type="text" value="Key Event"/>	<input type="text"/>	<input type="text" value="AUTO"/>	<input type="text" value="Release"/>	<input type="text"/>
DSS Key 2	<input type="text" value="Key Event"/>	<input type="text"/>	<input type="text" value="AUTO"/>	<input type="text" value="MWI"/>	<input type="text"/>
DSS Key 3	<input type="text" value="Key Event"/>	<input type="text"/>	<input type="text" value="AUTO"/>	<input type="text" value="Headset"/>	<input type="text"/>
DSS Key 4	<input type="text" value="None"/>	<input type="text"/>	<input type="text" value="AUTO"/>	<input type="text" value="None"/>	<input type="text"/>
DSS Key 5	<input type="text" value="None"/>	<input type="text"/>	<input type="text" value="AUTO"/>	<input type="text" value="None"/>	<input type="text"/>
DSS Key 6	<input type="text" value="None"/>	<input type="text"/>	<input type="text" value="AUTO"/>	<input type="text" value="None"/>	<input type="text"/>
DSS Key 7	<input type="text" value="None"/>	<input type="text"/>	<input type="text" value="AUTO"/>	<input type="text" value="None"/>	<input type="text"/>
DSS Key 8	<input type="text" value="None"/>	<input type="text"/>	<input type="text" value="AUTO"/>	<input type="text" value="None"/>	<input type="text"/>

Programmable Key Settings

Key	Desktop	Dialer	Calling	Desktop Long Pressed
Up	<input type="text" value="Missed"/>	<input type="text" value="Prev. Line"/>	<input type="text" value="Prev. Call"/>	<input type="text" value="Status"/>
Down	<input type="text" value="Call Forward"/>	<input type="text" value="Next Line"/>	<input type="text" value="Next Call"/>	<input type="text" value="None"/>
Left	<input type="text" value="Memo"/>	<input type="text" value="None"/>	<input type="text" value="Volume Down"/>	<input type="text" value="None"/>
Right	<input type="text" value="Note"/>	<input type="text" value="None"/>	<input type="text" value="Note"/>	<input type="text" value="Speed Dial"/>
OK	<input type="text" value="Menu"/>	<input type="text" value="None"/>	<input type="text" value="None"/>	<input type="text" value="None"/>

Function Key	
Item	Explanation
Contrast	Set contrast of screen.
Enable Backlight	Set enable/disable backlight.
Line Key Settings	
Line: select Auto, SIP1, SIP2 or IAX2 in function key type. After you set it, you pick up handset or hands-free, press this function key, and then you can use the corresponding SIP line.	
Function Key Settings	
key	Show the function key's serial number.
Type	Memory Key: settings can be stored in key storage for each number, the standby or off-hook, select the function keys on the keyboard can call this number. Line, set the dial mode (Auto, SIP1, SIP2, IAX2).Key Event functions, monitor state. DTMF: In the call, send DTMF. URL: You can input remote book url.
Value	Set the type parameter values.
Line	Choose which lines to use this feature.
Subtype	Select the function parameters Key Event and Memory Event.
Pickup Number	Please input the pickup number When SubType is BLF or presence.

Notice:

- Memory keys can be configured through the following:

Speed Dial function, through the configuration of the key corresponding to the number of ways as shown below.

Key	Type	Value	Line	Subtype	Pickup Number
DSS Key 1	Memory Key	4111	SIP1	Speed Dial	

User can press the F1 key to allocate this number by line1 line.

Intercom function, you can press this key in standby to automatically answer the call and make each other.

Key	Type	Value	Line	Subtype	Pickup Number
DSS Key 1	Memory Key	4116	SIP1	Intercom	

User can be configured in accordance with push to talk function the way: 4116 was the other number; Then press the standby button and make it automatically answer the call 4116.

- key can be configured through the following events:

For example:

Key	Type	Value	Line	Subtype	Pickup Number
DSS Key 1	Key Event		SIP1	DND	

8.2.5.2 EXT KEY

EXT KEY has the same usage with the Function key. “In” port connects the phone, “Out” port connects the next one, if there is only, you don’t need for power supply, if there are more than one, you need supply 5V power for the first one, and use RJ-45 direct connector.



FUNCTION KEY

EXT KEY

SOFTKEY

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Expansion Module Selection

Expansion Module 1 ▼


Load

Not Connected

Key	Type	Value	Line	Subtype	Pickup Number
F 1	None ▼		AUTO ▼	None ▼	
F 2	None ▼		AUTO ▼	None ▼	
F 3	None ▼		AUTO ▼	None ▼	
F 4	None ▼		AUTO ▼	None ▼	
F 5	None ▼		AUTO ▼	None ▼	
F 6	None ▼		AUTO ▼	None ▼	
F 7	None ▼		AUTO ▼	None ▼	
F 8	None ▼		AUTO ▼	None ▼	
F 9	None ▼		AUTO ▼	None ▼	
F 10	None ▼		AUTO ▼	None ▼	
F 11	None ▼		AUTO ▼	None ▼	
F 12	None ▼		AUTO ▼	None ▼	
F 13	None ▼		AUTO ▼	None ▼	
F 14	None ▼		AUTO ▼	None ▼	
F 15	None ▼		AUTO ▼	None ▼	
F 16	None ▼		AUTO ▼	None ▼	
F 17	None ▼		AUTO ▼	None ▼	
F 18	None ▼		AUTO ▼	None ▼	
F 19	None ▼		AUTO ▼	None ▼	
F 20	None ▼		AUTO ▼	None ▼	
F 21	None ▼		AUTO ▼	None ▼	
F 22	None ▼		AUTO ▼	None ▼	
F 23	None ▼		AUTO ▼	None ▼	
F 24	None ▼		AUTO ▼	None ▼	
F 25	None ▼		AUTO ▼	None ▼	
F 26	None ▼		AUTO ▼	None ▼	

Apply

8.2.5.3 SOFTKEY



FUNCTION KEY

EXT KEY

SOFTKEY

BASIC
 NETWORK
 VOIP
 PHONE
 ►FUNCTION KEY
 MAINTENANCE
 SECURITY
 LOGOUT

Softkey Settings

Softkey Mode: More ▼

Screen: Call Dialer ▼

Unselected Softkeys

- None
- Call Back(CBack)
- Clear
- History
- In
- Join
- Missed
- MWI
- Next Line(Next)
- Out
- Pause
- Phonebook(Dir)
- Pickup
- Prev. Line(Prev.)
- Redial

→
←

Selected Softkeys

- Delete
- None
- Dial
- Exit

↑
↓

Apply

SOFTKEY

You can configure different functions in different screens for every softkey.

8.2.6 Maintenance

8.2.6.1 Auto Provision Settings

ZYCOO endpoint supports PnP and DHCP and Phone Flash to obtain the parameters. The PnP and DHCP and Phone Flash are all deployed, endpoint will go by the following process to try to obtain the server address and other parameters, when it boots up:

DHCP option → PnP server → Phone Flash

Auto Provision	
Item	Explanation
Auto Update Setting	
Current Config Version	Show the current config file's version. If the version of the configuration downloaded is higher than the version of the running configurations, the auto provision would upgrade, or stop here. If the endpoints confirm the configuration by Digest method, the endpoints wouldn't upgrade configuration unless the configuration in the server is different with the running configuration.
Common Config Version	Show the common config file's version. If the configuration downloaded and the running configurations are the same, the auto provision would stop here. If the endpoints confirm the configuration by Digest method, the endpoints wouldn't upgrade configuration unless the configuration in the server is different with the running configuration.
CPE Serial Number	Show CPE Serial Number.
User	Specify FTP/HTTP/HTTPS server Username. System will use anonymous if username keep blank.
Password	Specify FTP/HTTP/HTTPS server Password.
Config Encrypt Key	Input the Encrypt Key, if the configuration file is encrypted.
Common Config Encrypt Key	Input the Common Encrypt Key, if the Common Configuration file is encrypted.
Save Autoprovision Information	Save the username and password authentication message of http/https/ftp and input ID message in the phone until the url in the server changes.
DHCP Option Setting	
DHCP Option Setting	Specify DHCP Option. DHCP option supports DHCP custom option and DHCP option 66 and DHCP option 43 to obtain the parameters. You could choose one method among them; the default is DHCP option disable.
Custom DHCP Option	A valid Custom DHCP Option is from 128 to 254. The Custom DHCP Option must be in accordance with the one defined in the DHCP server.
Plug and Play	
Enable PnP	Enable PnP by selecting it, than the phone will send SIP SUBSCRIBE messages to a multicast address when it boots up. Any SIP server understanding that message will reply with a SIP NOTIFY message containing the Auto Provisioning Server URL where the phones can request their configuration.
PnP Server	Specify the PnP Server.
PnP Port	Specify the PnP Server.

PnP Transport	Specify the PnP Transfer protocol.
PnP Interval	Specify the Interval time, unit is hour.
Phone Flash	
Server Address	Set FTP/TFTP/HTTP server IP address for auto update. The address can be IP address or Domain name with subdirectory.
Config File Name	Set configuration file's name which need to update. System will use MAC as config file name if config file name keep blank. For example, 000102030405.
Protocol Type	Specify the Protocol type FTP, TFTP or HTTP.
Update Interval	Specify update interval time, unit is hour.
Update Mode	Different update modes: 1. Disable: means no update. 2. Update after reboot: means update after reboot. 3. Update at time interval: means periodic update.

8.2.6.2 SYSLOG

Syslog is a protocol which is used to record the log messages with client/server mechanism. Syslog server receives the messages from clients, and classifies them based on priority and type. Then these messages will be written into log by some rules which administrator can configure. This is a better way for log management.

8 levels in debug information:

Level 0---emergency: This is highest default debug info level. Your system cannot work.

Level 1---alert: Your system has deadly problem.

Level 2---critical: Your system has serious problem.

Level 3---error: The error will affect your system working.

Level 4---warning: There are some potential dangers. But your system can work.

Level 5---notice: Your system works well in special condition, but you need to check its working environment and parameter.

Level 6---info: the daily debugging info.

Level 7---debug: the lowest debug info Professional debugging info from R&D person.

At present, the lowest level of debug information is info; debug level only can be displayed on telnet.

Syslog Configuration	
Item	Explanation
Syslog Setting	
Server Address	Set Syslog server IP address.
Server Port	Set Syslog server port.
MGR Log Level	Set the level of MGR log.
SIP Log Level	Set the level of SIP log.
IAX2 Log Level	Set the level of IAX2 log.
Enable Syslog	Select it or not to enable or disable syslog.
Web Capture	
Start	Click the start button when you need capture the WAN packet stream of the phone, then open or save the file as the interface.
Stop	Click the end button to stop capturing the packet stream.

8.2.6.3 CONFIG

Config Setting	
Item	Explanation
Save Configuration	You can save all changes of configurations. Click the Save button, all changes of configuration will be saved, and be effective immediately.
Backup Configuration	Right clicks on "Right click here..." and select "Save Target As config File(.txt)" then you will save the config file in .txt format, or select "Save Target As config File(.xml)" then you will save the config file in .xml format.
Clear Configuration	User can restore factory default and reboot the phone. If you login as Admin, the phone will reset all configuration and restore factory default; If you login as Guest, the phone will reset all configuration except for VoIP accounts (SIP1-2 and IAX2) and version number.

8.2.6.4 UPDATE

You can update your configuration with your config file in this web page.

Update

Item	Explanation
Web Update	
Web Update	Click the browse button, find out the config file saved before or provided by manufacturer, download it to the phone directly, press “Update” to save. You can also update downloaded update file, logo picture, ring, mmiset file by web.
TFTP/FTP Update	
Server Address	Set the FTP/TFTP server address for download/upload. The address can be IP address or Domain name with subdirectory.
User	Set the FTP server Username for download/upload.
Password	Set the FTP server password for download/upload.
File name	Set the name of update file or config file. The default name is the MAC of the phone, such as 000102030405.
Notice: You can modify the exported config file. And you can also download config file which includes several modules that need to be imported. For example, you can download a config file just keep with SIP module. After reboot, other modules of system still use previous setting and are not lost.	
Type	Action type that system want to execute: <ol style="list-style-type: none"> 1. Application update: download system update file. 2. Config file export: Upload the config file to FTP/TFTP server, name and save it. 3. Config fie import: Download the config file to phone from FTP/TFTP server. The configuration will be effective after the phone is reset. 4. Phone book export (.vcf): Upload the phonebook file to FTP/TFTP server, name and save it. 5. PhoneBook import (.vcf): Download the phonebook file to phone from FTP/TFTP server.
Protocol	Select FTP/TFTP server.
Update Logo File	
Select File	Specify the URL of the logo file.
Delete Logo File	

Select File Select the logo that you want to delete.

Logo File

Logo File Show the logo file.

8.2.6.5 ACCESS

You can add or delete user account, and change the authority of each user account in this web page.

Access Configuration

Item	Explanation
Keyboard Password	Set the password for entering the setting menu of the phone by the phone's key board. The password is digit.

User Settings

User	User Level
admin	Root
guest	General

This table shows the current user existed.

User	Set account user name.
User Level	Set user level, Root user has the right to modify configuration, General can only read.
Password	Set the password.
Confirm	Confirm the password.

Select the account and click the **Modify** to modify the selected account, and click the **Delete** to delete the selected account.

General user only can add the user whose level is General.

8.2.6.6 REBOOT

If you modified some configurations which need the phone's reboot to be effective, you need click the Reboot, then the phone will reboot immediately.

Notice: Before reboot, you need confirm that you have saved all configurations.

8.2.7 SECURITY

8.2.7.1 WEB FILTER

WEB Filter

User could make some device own IP, which is pre-specified, access to the MMI of the phone to config and manage the phone.

Item	Explanation
------	-------------

Web Filter Table Settings:	
----------------------------	--

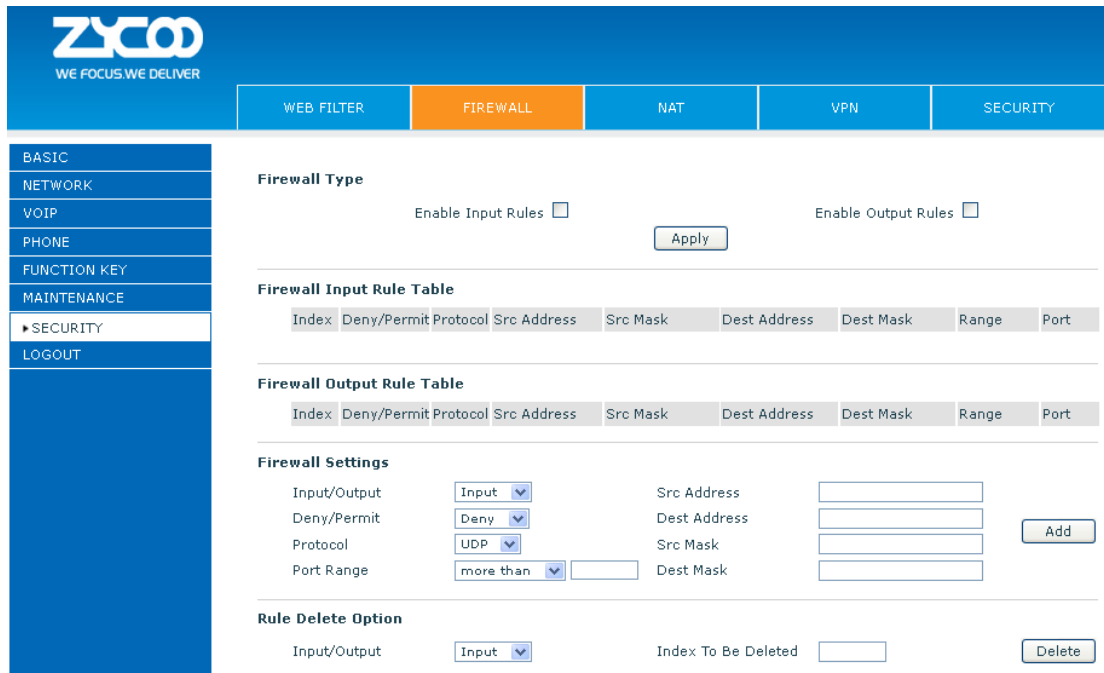
Add or delete the IP address segments that access to the phone.	
---	--

Set initial IP address in the Start IP column, Set end IP address in the End IP column, and click Add to add this IP segment. You can also click Delete to delete the selected IP segment.	
--	--

Web Filter setting	Select it or not to enable or disable Web Filter. Click Apply to make it effective.
--------------------	--

Notice: Do not set your visiting IP outside the Web filter range, otherwise, you cannot logon through the web.

8.2.7.2 FIREWALL



Firewall Configuration

In this web interface, you can set up firewall to prevent unauthorized Internet users from accessing private networks connected to the Internet (input rule), or prevent unauthorized private network devices from accessing the Internet (output rule).

Firewall supports two types of rules: input access rule and output access rule. Each type supports at most 10 items.

Through this web page, you could set up and enable/disable firewall with input/output rules. System could prevent unauthorized access, or access other networks set in rules for security. Firewall, is also called access list, is a simple implementation of a Cisco-like access list (firewall). It supports two access lists: one for filtering input packets, and the other for filtering output packets. Each kind of list could be added 10 items.

We will give you an instance for your reference.

Item	Explanation
Enable Input Rules	Select it to Enable Input Rules.
Enable Output Rules	Select it to Enable Output Rules.
Input / Output	Specify current adding rule by selecting input rule or output rule.
Deny/Permit	Specify current adding rule by selecting Deny rule or Permit rule.
Protocol	Filter protocol type. You can select TCP, UDP, ICMP, or IP.
Port Range	Set the filter Port range.
Src Address	Set source address. It can be single IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.0.
Des Address	Set the destination address. It can be IP address, network address, complete address 0.0.0.0, or network address similar to *.*.*.*.
Src Mask	Set the source address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.

Dest Mask Set the destination address' mask. For example, 255.255.255.255 means just point to one host; 255.255.255.0 means point to a network which network ID is C type.

Click the **Add** button if you want to add a new output rule.

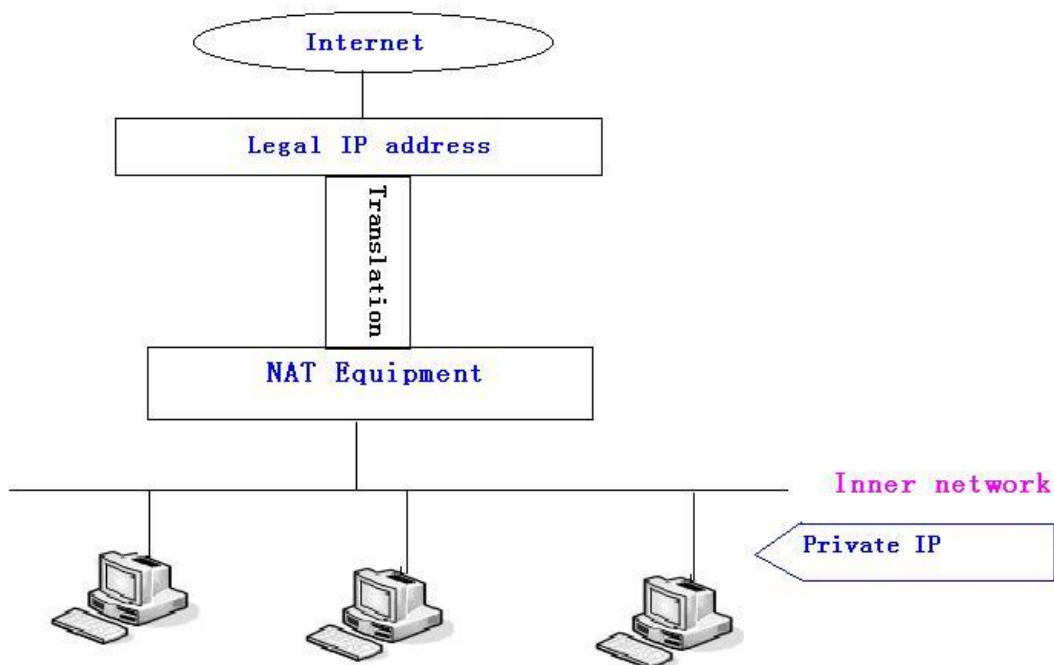
Then enable out access, and click the Apply button.

So when devices execute to ping 192.168.1.118, system will deny the request to send icmp request to 192.168.1.118 for the out access rule. But if devices ping other devices which network ID is 192.168.1.0, it will be normal.

Click the **Delete** button to delete the selected rule.

8.2.7.3 NAT

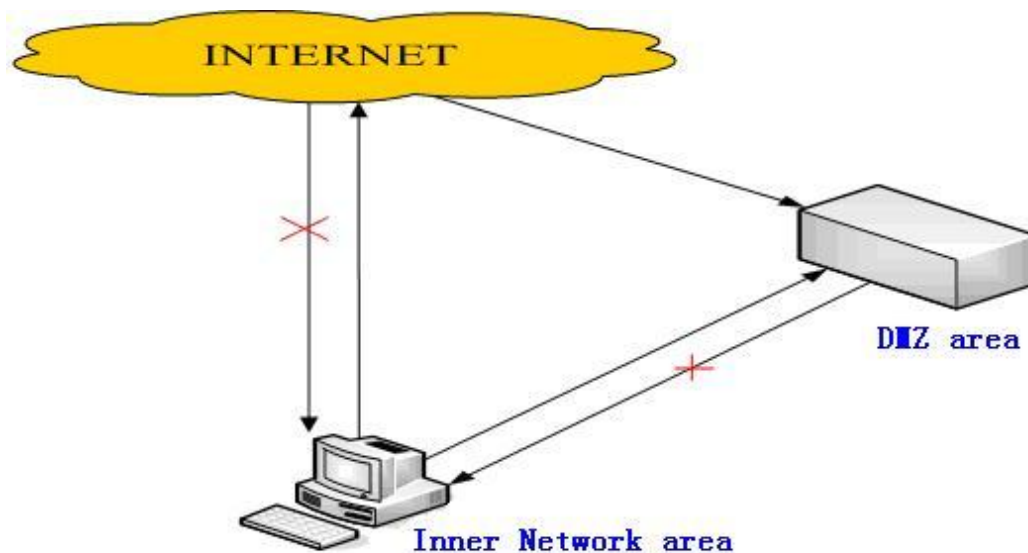
NAT is abbreviated from Net Address Translation; it's a protocol responsible for IP address translation. In other word, it is responsible for transforming IP and port of private network to public, also is the IP address mapping which we usually say.



DMZ config:

In order to make some intranet equipment support better service for extranet, and make internal network security more effectively, these equipment open to extranet need be separated from the other equipment not open to extranet by the corresponding isolation method according to different demands. We can provide the different security level protection in terms of the different resources by building a DMZ region which can provide the network level protection for the equipment environment, reduce the risk which is caused by providing service to distrust customer, and is the best position to put public information

The following chart describes the network access control of DMZ.



ZYCOO
 WE FOCUS WE DELIVER

WEB FILTER
FIREWALL
NAT
VPN
SECURITY

BASIC
 NETWORK
 VOIP
 PHONE
 FUNCTION KEY
 MAINTENANCE
 ► SECURITY
 LOGOUT

Application Layer Gateway (ALG) Settings

IPSec ALG ☒
FTP ALG ☒
PPTP ALG ☒

Network Address Translation (NAT) Table

Inside IP Address	Inside TCP Port	Outside TCP Port
Inside IP Address	Inside UDP Port	Outside UDP Port

NAT Table Option

Transfer Type TCP

Inside IP Address

Outside Port

Inside Port

NAT Configuration

Item	Explanation
IPSec ALG	It is an encryption technology. Select it to enable IPSec ALG, the default is enabled.
FTP ALG	FTP is a service of connection layer which can transform intranet IP into extranet IP when intranet IP is sending out packet. Select it to enable FTP ALG, the default is enabled.
PPTP ALG	Select it enable PPTP ALG, the default is enabled.
Shows the NAT TCP mapping table	
Shows the NAT UDP mapping table	
Transfer Type	Select the NAT mapping protocol style, TCP or UDP
Inside IP	Set the IP address of device which is connected to LAN interface to do NAT mapping.
Inside Port	Set the LAN port of the NAT mapping
Outside Port	Set the WAN port of the NAT mapping

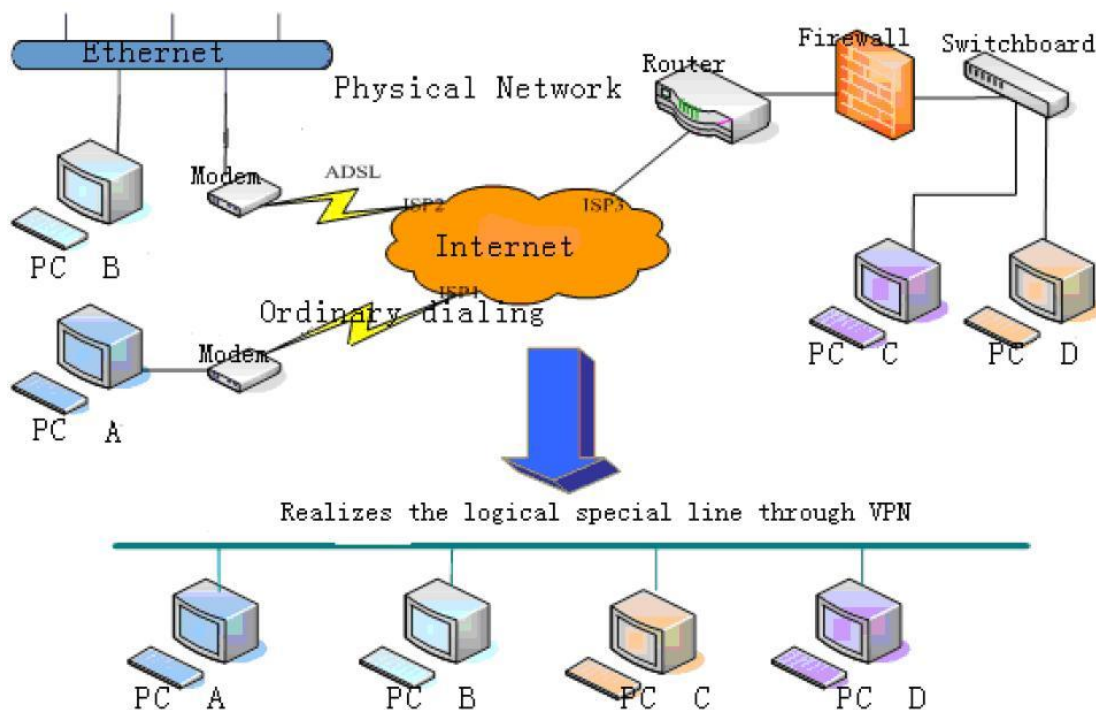
Notice: After finish setting, click the Add button to add new mapping table; click the Delete button to delete the selected mapping table.

Shows the outside WAN port IP address and the inside LAN port IP address.

Notice: 10M/100M adaptive means the network card, and other equipment physical consultations speed, testing speed under bridge mode near to 100M, in order to ensure the quality of voice and communications real-time performance, we made some sacrifices of NAT under the transmission performance. Transmit with full capability only when system is idle, so cannot guarantee that the transmission speed reach to 100M.

8.2.7.4 VPN

This web page provides us a safe connect mode by which we can make remote access to enterprise inner network from public network. That is to say, you can set it to connect public networks in different areas into inner network via a special tunnel.



WE FOCUS.WE DELIVER

WEB FILTER	FIREWALL	NAT	VPN	SECURITY
------------	----------	-----	-----	----------

BASIC

NETWORK

VOIP

PHONE

FUNCTION KEY

MAINTENANCE

SECURITY

LOGOUT

Virtual Private Network (VPN) Status

IP Address 0.0.0.0

VPN Mode

Enable VPN ☐

Layer 2 Tunneling Protocol (L2TP)

VPN Server Address

VPN Password

VPN User

VPN Configuration

Item	Explanation
VPN IP	Shows the current VPN IP address.
Select L2TP. You can choose only one for current state. After you select it, you'd better save configuration and reboot your phone.	
Enable VPN	Select it or not to enable or disable VPN.
VPN Server Address	Set VPN L2TP Server IP address.
VPN User	Set User Name access to VPN L2TP Server.
VPN Password	Set Password access to VPN L2TP Server.

8.2.7.5 SECURITY

Security













Item	Explanation
Update Security File	
Select Security File	Select the security file you want to update, then click Update button to update.
Delete Security File	
Select Security File	Select the security file you want to delete, then click Delete button to update.
SIP TLS File	Show SIP TLS authentication certification file.
HTTPS File	Show HTTPS authentication certification file.
Open VPN Files	Show Open VPN File authentication certification file.

8.2.8 LOGOUT

Click **Logout**, and you will exit web page. If you want to enter it next time, you need input user name and password again.

Chapter 9 Appendix

Digit-character map table

Keypad	Character	Keypad	Character
	1 @		7 P Q R S p q r s
	2 A B C a b c		8 T U V t u v
	3 D E F d e f		9 W X Y Z w x y z
	4 G H I g h i		*./
	5 J K L j k l		0
	6 M N O m n o		#/SEND