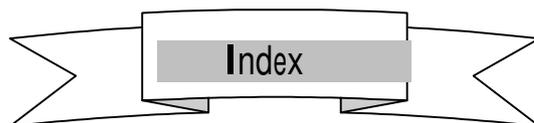


# IP 301 Internet Phone [VoIP]

## **User Manual(V1.0)**



|      |  |    |
|------|--|----|
| 1.   | Introduction.....  | 3  |
| 1.1  | Hardware Overview.....                                       | 3  |
| 1.2  | Software Overview.....                                       | 3  |
| 1.3  | Operating requirements: .....                                | 4  |
| 1.4  | Electric requirements: .....                                 | 4  |
| 1.5  | Size: .....  | 4  |
| 1.6  | Phone diagram.....   | 5  |
| 1.7  | Keypad description.....                                      | 5  |
| 2.   | Installing the IP Phone .....                                | 6  |
| 3.   | Setup the VoIP Phone by Keypad.....                          | 7  |
| 3.1  | Keypad Setting List .....                                    | 7  |
| 4.   | Setup the VoIP Phone by Web Browser .....                    | 11 |
| 4.1  | Login.....   | 11 |
| 4.2  | System Information for the VOIP PHONE .....                  | 11 |
| 4.3  | Phone Book.....  | 11 |
| 4.4  | Phone Setting.....   | 13 |
| 4.5  | Network .....  | 16 |
| 4.6  | SIP Settings.....  | 20 |
| 4.7  | NAT Trans. ....  | 23 |
| 4.8  | Others.....  | 24 |
| 4.9  | System Authority .....                                       | 24 |
| 4.10 | Save Change .....  | 25 |
| 4.11 | Update.....  | 25 |
| 4.12 | Reboot.....  | 26 |
| 5.   | Setup the VoIP Phone by using Console (Hyper Terminal) ..... | 27 |
| 5.1  | Configure the COM port.....                                  | 27 |
| 5.2  | Login into the VoIP Phone.....                               | 27 |
| 5.3  | Using CLI command to configure the VoIP Phone.....           | 29 |
| 6.   | Phone function list.....                                     | 35 |

## Introduction

IP is abbreviation of Internet Protocol. An IP phone is a telephone transporting voice by grouping data package of IP protocol. IP phone can be used on IP Networks from an enterprise-wise LAN to a city-wide WAN.

An IP phone transports voice message over data communication network at extremely low price but of excellent sound quality just like ordinary fixed telephone lines do. With our IP301 Internet phone, you're going to save dramatically on international calls.

With its unique generalized outline and linner line modes. IP301 Internet phone functions much like an ordinary telephone switching between inner and out line, it brings great convenience to the users. When IP301 Internet phone is in generalized inner line mode, it is free of charge to call another IP301 wherever the destination phone is in the world. When IP301 series IP net phone is in generalized out line mode, it can place calls to every ordinary telephones in the world at an extremely low price.

With its stability, sound quality, compatibility and rich functionalities, IP301 leverages your wealth with high quality communications!

## 1.1 Hardware Overview

- Main Chip – CM5000
- Data Memory – 16M
- Program Memory – 64M
- Ethernet Jack – 2 x 10/100M jacks
- AC/DC adapter – Input AC230V, output 9V 1200mA

## 1.2 Software Overview

- Network Protocol
  - SIP V1 (RFC2543), V2 (RFC3261)
  - IP/TCP/UDP/RTP/RTCP
  - ICMP/ARP/RARP/SNTP
  - TFTP Client/DHCP Client/ PPPoE Client, Static IP
  - Telnet/HTTP Server
  - DNS Client
- Codec
  - G.711
  - G.723
  - G.726
  - G.729A
  - G.729B
- IP Assignment
  - Static IP
  - DHCP
  - PPPoE
- Voice Quality

- VAD(Voice Activity Detection)
- CNG(Comfortable Noise Generator)
- LEC(Line Echo Canceller)
- Packet Loss Compensation
- Adaptive Jitter Buffer
  
- Security
  - HTTP 1.1 basic/digest authentication for Web setup
  - MD5 for SIP authentication (RFC2069/ RFC 2617)
  
- Configuration
  - Web Browser
  - Telnet
  - Keypad Configuration
  
- Firmware Upgrade
  - TFTP
  - HTTP

### **1.3 Operating requirements:**

- Operation temperature: 0 to 50 °C
- Storage temperature: -10 to 60 °C
- Humidity: 10 to 90% no dew

### **1.4 Electric requirements:**

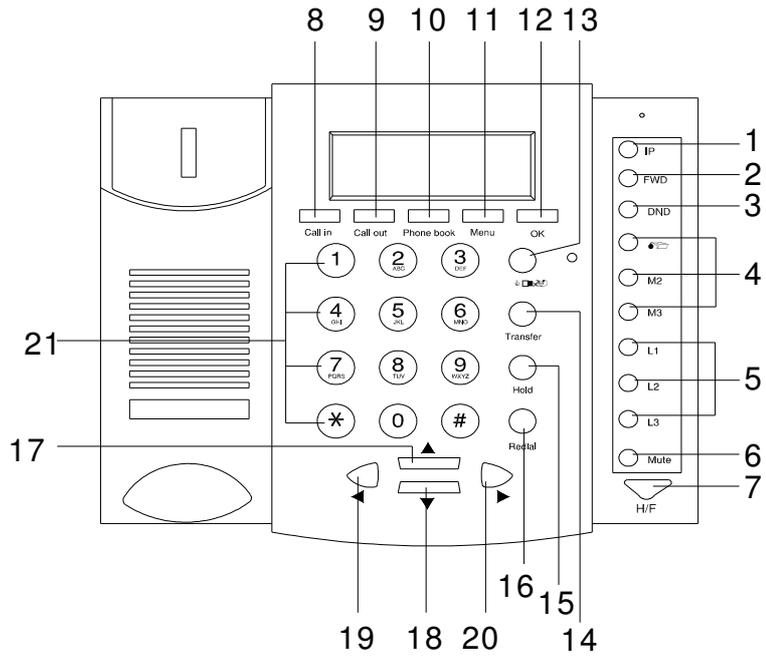
- Voltage: 9V DC
- Power adaptor: AC/DC input 230V, output 9V 1100mA
- Network interface: 2 X RJ-45 Ethernet connectors

### **1.5 Size:**

204 x 187 x 75 mm (L x W x H)

**1.6 Phone diagram**

1. IP key
2. FWD key
3. DND key
4. Men Key
5. L1~L3 key
6. Mute Key
7. Speaker Key
8. Call In Key
9. Call Out key
10. PhoneBook Key
11. Menu Key
12. Confirm Key
13. Conference
14. Transfer
15. Hold key
16. Redial
17. Up Key
18. Down Key
19. Right Key
20. Left key
21. Number key



**1.7 Keypad description**

| Key Name   | Description   |
|------------|---|
| 1          | "1", "-", "3", "!", "?"                                   |
| 2          | "2", "a", "b", "c", "A", "B", "C"                         |
| 3          | "3", "d", "e", "f", "D", "E", "F"                         |
| 4          | "4", "g", "h", "i", "G", "H", "I"                         |
| 5          | "5", "j", "k", "l", "J", "K", "L"                         |
| 6          | "6", "m", "n", "o", "M", "N", "O"                         |
| 7          | "7", "p", "q", "r", "s", "P", "Q", "R", "S"               |
| 8          | "8", "t", "u", "v", "T", "U", "V"                         |
| 9          | "9", "w", "x", "y", "z", "W", "X", "Y", "Z"               |
| 0          | "0", "space"  |
| *          | "*", ".", "@", Switch between PSTN dialing and IP dialing |
| #          | Start dialing process                                     |
| UP/DOWN    | This is Up↑ and Down↓ key                                 |
| MENU       | This is the "Menu" key to set the IP Phone                |
| LEFT/RIGHT | This is Left← and Right→ key                              |
| CONF       | This is three way conference function                     |
| TRANSFER   | This is "Transfer" to the other phone number              |
| HOLD       | This is "HOLD" function                                   |
| Phone book | Go to Phone book directly                                 |
| Mute       | This is "Mute" function                                   |
| OK         | This is "OK", accept setting                              |
| Mute       | This is "Mute" function                                   |
| Reg LED    | Show the status of IP Phone's registration                |
| L1~L3      | Selec different Serve                                     |
| DND        | Dot not Disturb   |
| IP         | Check Local IP  |
| LED        | Register LED  |

## Installing the IP Phone

Before installing your IP phone, it is useful to know what kind of IP addressing your LAN or router is using.

### 2.1 What is the IP addressing of your network?

Most networks, including the Internet, use identification codes IP (Internet Protocol) addresses to identify and locate the devices that share their services.

There are two systems for assigning these addresses:

- DHCP Addressing: With this system, your LAN or router automatically assigns all the required IP parameters to any device connected to it when the device logs on. (Some ISPs who use DHCP require a host name, which must be entered manually.)

---

**Note:** Your IP Phone is shipped from the factory with DHCP on, or enabled. So, if your LAN or router is configured to use DHCP addressing, your IP Phone's LAN parameters will automatically be configured as soon as it is connected to the LAN or router and powered up.

---

- Static Addressing: If your LAN does not use DHCP addressing, each device concurrently connected to it must be assigned its own unique IP address. In this case, your LAN's configuration information must be manually entered into the IP phone. You will need to know the following parameters:
  - IP address
  - Subnet mask
  - Gateway address
  - Primary DNS address
  - Secondary DNS address (required by some systems)

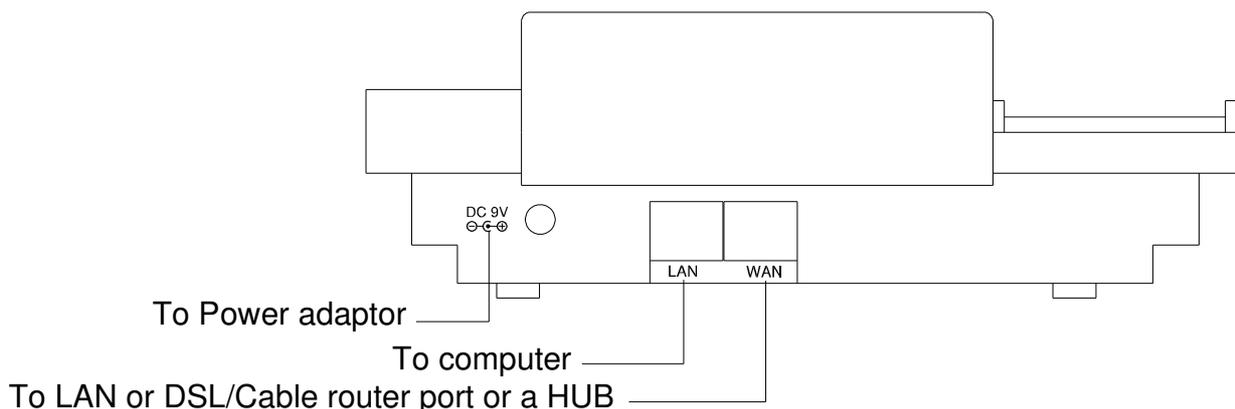
---

**Note:** If your system uses static addressing, this is a good time to make sure you know these parameters, since you will need them to configure your IP Phone. If you don't know them, contact your network administrator to obtain the necessary information.

---

### 2.2 Connection of the IP phone:

1. Connect handset to the base: insert handset cord into handset jack at the left side of the base.
2. Connect IP phone to Internet: Plug a RJ-45 Ethernet cable into the Ethernet jack named 'WAN' of the IP phone. Plug the other end of the cable into an appropriate LAN or DSL/Cable router port or a HUB.
3. Connect IP phone to a computer if necessary: Plug a RJ-45 Ethernet cable into the Ethernet jack named 'LAN' of the IP phone. Plug the other end of the cable to a computer.
4. Power on IP phone: Plug the power adapter into the Power Jack of the IP phone. Then plug the other end of the power adapter into the appropriate wall outlet.



### Setup the VoIP Phone by Keypad

The configuration of the VoIP phone can be set by the keypad on the unit.

## 1.8 Keypad Setting List

### 1. Phone Book

- Search: Search Phone Book.
- Add entry: Add new phone number to phone book
- Speed dial: Add speed dial phone number to speed dial list.
- Erase all: Erase all phone number from Phone Book.

### 2. Call history

- Incoming calls: Show all incoming call.
- Dialed numbers: Show all dialed call.
- Erase record: Delete call history.
  - **All:** Delete all call history.
  - **Incoming:** Delete all incoming call.
  - **Dialed:** Delete all dialed out call.

### 3. Phone setting

- Call forward
  - All Forward.
    - **Activation:** To Enabled/Disabled this function.
    - **Number:** Forward to a Speed Dial Number.
  - Busy Forward.
    - **Activation:** To Enabled/Disabled this function.
    - **Number:** Forward to a Speed Dial Number.
  - No Answer Forward.
    - **Activation:** To Enabled/Disabled this function.
    - **Number:** Forward to a Speed Dial Number.
  - Ring Timeout: **Set the time to start the no answer forward function, ex: 20 means after 20 seconds then forward to the dedicated number.**
- Block Setting
  - **All:** Block all the incoming calls
  - **By Time:** Block all the incoming calls within a period of time

- **Duration:** Set the start time and end time to Block Setting.
  
- Date/Time setting
  - Date & Time: **Set the IP Phone Date and Time.**
  - SNTP setting
    - SNTP : **Enabled / Disable SNTP.**
    - Primary SNTP: **Set Primary SNTP server IP address.**
    - Secondary SNTP: **Set Secondary SNTP server IP address.**
    - Time zone: **Set Time zone.**
    - Adjustment Time: **Set adjustment time period.**
  
- Volume & Gain
  - Handset volume: **Set Handset volume from 0~15 (max.).**
  - Speaker volume: **Set Speaker phone volume from 0~15 (max.).**
  - Handset Gain: **Set Handset gain from 0~15 (max.).**
  - Speaker Gain: **Set Speaker gain from 0~15 (max.).**
  
- Ringer
  - Ringer volume: **Ringer volume setting from 0~15 (max.).**
  - Ringer type: **Ringer tone selection from 1~4.**

## 4. Network

- WAN Setup
  - IP Type
    - Fixed IP client
    - DHCP client
    - PPPoE client
  
  - Fixed IP setting
    - IP Address
    - Subnet mask
    - Default Gateway
    - MAC address
  
  - PPPoE setting
    - User name
    - Password
  
- LAN Setup
  - Bridge
  - NAT
  
- DNS
  - Primary DNS
  - Secondary DNS
  
- VLAN
  - Activation
  - VID: service provider to set your VID Priority(VID 2~4094)
  - CFI: Canonical Format Indicator(0~1)
  - Status: Show WAN and LAN IP addresses

5. SIP Settings

- Service domain
  - First realm
    - Activation
    - User name
    - Display name
    - Register name
    - Register password
    - Proxy server
    - Domain server
    - Outbound proxy
  - Second realm
    - Activation
    - User name
    - Display name
    - Register name
    - Register password
    - Proxy server Proxy
    - Domain server
    - Outbound proxy
  - Third realm
    - Activation
    - User name
    - Display name
    - Register name
    - Register password
    - Proxy server Proxy
    - Domain server
    - Outbound proxy
- Codec
  - Codec type
    - G.711 uLaw
    - G.711 aLaw
    - G.723
    - G.729
    - G.726-16
    - G.726-24
    - G.726-32
    - G.726-40
  - VAD: **To Enabled/Disabled the Voice Active Detection function.**
- RTP setting
  - Outband DTMF
  - Duplicate RTP
    - No duplicate**
    - One duplicate**
    - Two duplicate**
- RPort Setting: RPort Enabled/Disabled
- Hold by RFC : On the talk state selece the hold for disable or enable(RCF:3261)
- Status
  - First Realm:** First SIP server register status

- **Second Realm:** Second SIP server register status
- **Third Realm:** Third SIP server register status

### 6. NAT Transversal

- STUN setting
  - STUN: **To Enabled/Disabled this function.**
  - STUN server

### 7. Administrator

- Auto Config
  - Config mode
    - Disable
    - TFTP
    - FTP
  - TFTP server: **Setting the TFTP server IP address.**
  - FTP server: **Setting the FTP server IP address**
  - FTP Login Name: **Setting the FTP login name**
  - FTP Password: **Setting the FTP password**
- Default setting
- System Authentication: To do the SIP setting from Keypad, need to input the password first. Default is "24234880".
- Version
- Watch dog
- Restart

## Setup the VoIP Phone by Web Browser

The VoIP Phone provides a built-in web server. You can use Web browser to configure the VoIP Phone. First please input the IP address in the Web page. In the end of IP address, please add the port number “:9999”. Ex:http://192.168.1.100:9999

### 1.9 Login

1.9.1 Please input the username and password into the blank field. The default setting is:

1. For Administrator, the username is: admin; and the password is: 24234880. If you use the account login, you can configure all the setting.
2. For normal user, the username is: user; and the password is: user. If you use the account login, but you cannot configure the SIP setting.

1.9.2 Click the “Login” button will move into the VOIP PHONE web based management information page.

1.9.3 If you change the setting in the Web Management interface, please do remember to click the “Submit” button in that page. After you finished the change of the setting, click the “Save” function in the left side, and click the Save Button. When you finished the setting, please click the Reboot function in the left side, and click the Reboot button in that page. After the system restart, all the setting can work properly.



**Login VoIP**

Enter your username and password to login  
VoIP server

Username

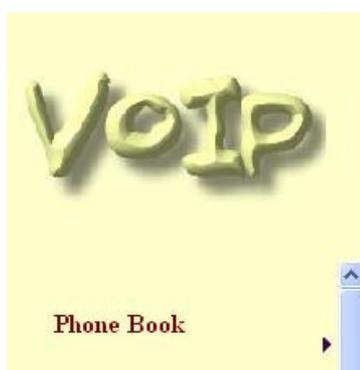
Password

Remember last login

### 1.10 System Information for the VOIP PHONE

1.10.1 When you login the web page, you can see the VOIP PHONE current system information like firmware version, company... etc in this page.

1.10.2 Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.



### System Information

This page illustrate the system related information.

|                   |                           |
|-------------------|---------------------------|
| Model Name:       | VoIP                      |
| Firmware Version: | Mon May 8 13:53:23 2006.  |
| Codec Version:    | Wed Apr 19 18:12:43 2006. |

### 1.11 Phone Book

1.11.1 In Phone Book contains Phone Book and Speed Dial Settings. You can setup the Phone Book and Speed Dial number. The Phone Book can store 140 phone numbers and the

Speed Dial can store 10 phone numbers. If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press “#”.

1.11.2 In the Phone Book function you can add/delete the phone number in the phone book list. You can input maximum 140 entries phone book list.

- ❑ If you need to add a phone number into the phone book, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the “Add Phone” button.
- ❑ If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.
- ❑ If you want to delete all phone numbers, you can click “Delete All” button.

## Phone Book

You could add/delete items in current phone book.

Phone Book Page:

| Phone | Name | URL | Select                   |
|-------|------|-----|--------------------------|
| 90    |      |     | <input type="checkbox"/> |
| 91    |      |     | <input type="checkbox"/> |
| 92    |      |     | <input type="checkbox"/> |
| 93    |      |     | <input type="checkbox"/> |
| 94    |      |     | <input type="checkbox"/> |
| 95    |      |     | <input type="checkbox"/> |
| 96    |      |     | <input type="checkbox"/> |
| 97    |      |     | <input type="checkbox"/> |
| 98    |      |     | <input type="checkbox"/> |
| 99    |      |     | <input type="checkbox"/> |

### Add New Phone

Position:  (0~139)  
Name:   
URL:

1.11.3 In Speed Dial setting function you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list.

- ❑ If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you finished a new phone list, just click the “Add Phone” button.
- ❑ If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.
- ❑ If you want to delete all phone numbers, you can click “Delete All” button.

## Speed Dial Phone List

You could set the speed dial phones in this page.

| Phone | Name | URL                 | Select                   |
|-------|------|---------------------|--------------------------|
| 0     | 0    | 192.168.96.151:5062 | <input type="checkbox"/> |
| 1     | 1    | 192.168.96.153      | <input type="checkbox"/> |
| 2     |      |                     | <input type="checkbox"/> |
| 3     |      |                     | <input type="checkbox"/> |
| 4     |      |                     | <input type="checkbox"/> |
| 5     |      |                     | <input type="checkbox"/> |
| 6     |      |                     | <input type="checkbox"/> |
| 7     |      |                     | <input type="checkbox"/> |
| 8     |      |                     | <input type="checkbox"/> |
| 9     |      |                     | <input type="checkbox"/> |

### Add New Phone

Position:  (0~9)  
 Name:   
 URL:

## 1.12 Phone Setting

1.12.1 In Phone Setting contains Call Forward, SNTP Settings, Volume Settings, Melody Settings, Block Setting, and Call waiting .

1.12.2 Call Forward function: you can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

- All Forward: All incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.
- Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.
- No Answer Forward: : If you can not answer the phone, the incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. Also you have to set the Time Out time for system to start to forward the call to the number you choosed.
- When you finished the setting, please click the Submit button.

## Forward Setting

You could set the forward number of your phone in this page.

All Forward:  Off  On  
 Busy Forward:  Off  On  
 No Answer Forward:  Off  On

|                    | Name                 | URL                  |
|--------------------|----------------------|----------------------|
| All Fwd No.:       | <input type="text"/> | <input type="text"/> |
| Busy Fwd No.:      | <input type="text"/> | <input type="text"/> |
| No Answer Fwd No.: | <input type="text"/> | <input type="text"/> |

No Answer Fwd Time Out:  (2~8 Ring)

- 1.12.3 SNTP Setting function: you can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.

### SNTP Settings

You could set the SNTP servers in this page.

---

SNTP:  On  Off

Primary Server:

Secondary Server:

Time Zone: GMT    (hh:mm)

Sync. Time:    (dd:hh:mm)

- 1.12.4 Volume Setting function: you can setup the Handset Volume, Ringer Volume, and the Handset Gain. When you finished the setting, please click the Submit button.
- Handset Volume is to set the volume you hear from the handset.
  - Speaker Volume is to set the volume you hear from the speaker phone.
  - Ringer Volume is to set the ringer volume.
  - Handset Gain is to set the volume send out from the handset.
  - Speaker Gain is to set the volume send out from the micro phone.

## Volume Setting

You could set the volume of your phone in this page.

---

|                        |                                 |        |
|------------------------|---------------------------------|--------|
| <b>Handset Volume:</b> | <input type="text" value="9"/>  | (0~15) |
| <b>Speaker Volume:</b> | <input type="text" value="10"/> | (0~15) |
| <b>Ringer Volume:</b>  | <input type="text" value="6"/>  | (0~10) |
| <b>Handset Gain:</b>   | <input type="text" value="10"/> | (0~15) |
| <b>Speaker Gain:</b>   | <input type="text" value="9"/>  | (0~15) |

- 1.12.5 Melody Setting function: you can select the melody for the imcoming call. When you finished the setting, please click the Submit button.

### Ringer Settings

You could set your favorite ringer in this page.

Ringer:  On  Off

Ringer Type:

- 1.12.6 Block Setting function: you can setup the Block Setting to keep the phone slience. You can choose Always Block or Block a period.
- Always Block: All incoming call will be blocked until disable this feature.
  - Block Period: Set a time period and the phone will be blocked during the time period. If the "From" time is large than the "To" time, the Block time will from Day 1 to Day 2.
  - When you finished the setting, please click the Submit button.

### Block Setting

You could set the block period of your phone in this page.

Always Block:  On  Off

Block Period:  On  Off

From:   (hh:mm)

To:   (hh:mm)

- 1.12.7 Dial Plan function: This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.

### Dial Plan

You could the set the dial plan in this page.

Replace prefix code:  On  Off

Replace rule:  ->

Dial Plan:

Auto Prefix:  (0000~9999)

Prefix Unset Plan:

Auto Dial Time:  (3~9 sec)

- Symbol explan:

|        |                     |
|--------|---------------------|
| x or X | 0,1,2,3,4,5,6,7,8,9 |
| +      | or                  |

- Replace rule:If replace prefix code is ON and prefix number is matched with rule then 005 will replace prefix.
  - Auto Dial Time : Stop dialing after seconds then send dial number out.
  - Dial Plan: When match with pattern then send dial number out but if fisrt digit is '0' then dial plan will be ignored.
- Example:

|          |   |
|----------|---|
| *xx      | If matched with one of *00,*01....*99 then will send number out |
| #xx      | If matched with one of #00,#01....#99 then will send number out |
| 10x      | If matched with one of 100,101....109 then will send number out |
| 11x      | If matched with one of 110,111....119 then will send number out |
| Xxxxxxxx | If dial with 8 digits then send number out                      |

- ❑ Auto Prefix : Number for add before dial number.
  - ❑ Prefix Unset Plan : When first digit or dial numeb match with pattern then ignore auto prefix.
  - ❑ When you finished the setting, please click the Submit button.
  - ❑ If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.
- 1.12.8 Call Waiting Setting function: If user doesn't want to be inform there is a new incoming call, user can set the function off.

### Call Waiting Setting

You could enable/disable the call waiting setting in this page.

Call Waiting:  On  Off

## 1.13 Network

1.13.1 In Network you can check the Network status, configure the Network Settings and DDNS settings.

1.13.2 Network Status: You can check the current Network setting in this page.

### Network Status

This page shows current status of network interfaces of the system.

| Interface 0   |               |
|---------------|---------------|
| Type:         | DHCP Client   |
| IP:           | 192.168.1.108 |
| Mask:         | 255.255.255.0 |
| Gateway:      | 192.168.1.254 |
| DNS Server 1: | 168.95.192.1  |
| DNS Server 2: | 168.95.1.1    |

| Interface 1   |               |
|---------------|---------------|
| Type:         | DHCP Server   |
| IP:           | 192.168.123.1 |
| Mask:         | 255.255.255.0 |
| Gateway:      | 192.168.123.1 |
| DNS Server 1: | 168.95.192.1  |
| DNS Server 2: | 168.95.1.1    |

- 1.13.3 WAN Settings: You can configure the VoIP Phone Network setting in this page.
- ❑ You can choose Bridge or NAT for the LAN mode. If you set the Bridge On, then the two Fast Ethernet ports will be transparent.
  - ❑ The WAN setting item is to setup the LAN port's network environment. You may refer to your current network environment to configure the VoIP Phone properly.
  - ❑ The PPPoE Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.
  - ❑ When you finished the setting, please click the Submit button.

## WAN Settings

You could configure the WAN settings in this page.

---

|           |                              |                                      |
|-----------|------------------------------|--------------------------------------|
| LAN Mode: | <input type="radio"/> Bridge | <input checked="" type="radio"/> NAT |
|-----------|------------------------------|--------------------------------------|

| WAN Setting  |  |  |                             |
|--------------|--|--|-----------------------------|
| IP Type:     | <input type="radio"/> Fixed IP             | <input checked="" type="radio"/> DHCP Client | <input type="radio"/> PPPoE |
| IP:          | <input type="text" value="192.168.1.108"/> |  |                             |
| Mask:        | <input type="text" value="255.255.255.0"/> |  |                             |
| Gateway:     | <input type="text" value="192.168.1.254"/> |  |                             |
| DNS Server1: | <input type="text" value="168.95.192.1"/>  |  |                             |
| DNS Server2: | <input type="text" value="168.95.1.1"/>    |  |                             |
| MAC:         | <input type="text" value="00094551#01"/>   |  |                             |

| PPPoE Setting |                      |
|---------------|----------------------|
| User Name:    | <input type="text"/> |
| Password:     | <input type="text"/> |

- 1.13.4 LAN Settings: You can configure the VoIP Phone Network setting in this page.

## LAN Settings

You could configure the LAN settings in this page.

---

| LAN Setting |  |
|-------------|--|
| IP:         | <input type="text" value="192.168.123.1"/> |
| Mask:       | <input type="text" value="255.255.255.0"/> |
| MAC:        | <input type="text" value="00094551#02"/>   |

| DHCP Server  |   |                           |
|--------------|---|---------------------------|
| DHCP Server: | <input checked="" type="radio"/> On                                     | <input type="radio"/> Off |
| Start IP:    | <input type="text" value="150"/>  |                           |
| End IP:      | <input type="text" value="200"/>  |                           |
| Lease Time:  | <input type="text" value="1"/> : <input type="text" value="0"/> (dd:hh) |                           |

- 1.13.5 DDNS Setting: You can configure the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP

applications are work with a SIP Proxy Server. When you finished the setting, please click the Submit button

## DDNS Settings

You could set the configuration of DDNS in this page.

---

**DDNS:**                     On    Off

Host Name:               

User Name:               

Password:                

E-mail Address:        

DDNS Server:            

DDNS Server List:       ▼

Type:                       ▼

Wild Card:                 ▼

BACKMX:                  On    Off

Off Line:                  On    Off

- 1.13.6 VLAN Settings: you can set the VLAN setting in this page. There are two parts in this page. First one is to set the packets related to the IP Phone, and the second part is if you use the VLAN setting in the NAT Mode.
- ❑ There are two kinds of destination packets that will come from the WAN port of IP Phone, one kind of packets will go to the IP Phone, the other will go through the LAN port to the PC.
  - ❑ VLAN Packets: if you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be checked with the IP Address and the VID.
  - ❑ VID: You can follow your service provider to set your VID.
  - ❑ User Priority: Defines user priority, giving eight (2<sup>3</sup>) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be defined by your service provider.
  - ❑ CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reasons between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.
  - ❑ When you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the IP address of IP Phone and the same VID will be accepted by the IP Phone. If the incoming packets have the IP Phone IP address but a different VID, then the packets will be discarded by the IP Phone. The other incoming packets with different IP addresses will go through the LAN port to the PC.
  - ❑ NAT VLAN Setting: When you set your device in NAT mode, the IP Phone can help you to filter the wrong incoming packets. You can separate the other devices connected behind the IP Phone into 4 VLAN groups. You can set different VID for these 4 groups. When the incoming packets go through the WAN port of IP Phone, then the IP Phone will check the VID, if the packet is not going to the IP Phone (with the IP address of IP Phone and the correct VID), and the VID is not these four VID you set, then the packets will be discarded by the IP Phone.

## VLAN Settings

You could set the VLAN settings in this page.

---

|                |   |
|----------------|---|
| VLAN Packets:  | <input checked="" type="radio"/> Off <input type="radio"/> On |
| VID:           | <input type="text" value="136"/> (2 ~ 4094)                   |
| User Priority: | <input type="text" value="0"/> (0 ~ 7)                        |
| CFI:           | <input type="text" value="0"/> (0 ~ 1)                        |

| NAT VLAN Setting |   |
|------------------|---|
| VLAN Packets:    | <input checked="" type="radio"/> Off <input type="radio"/> On |
| VID1:            | <input type="text" value="4"/> (2 ~ 4094), 0->Off             |
| VID2:            | <input type="text" value="5"/> (2 ~ 4094), 0->Off             |
| VID3:            | <input type="text" value="6"/> (2 ~ 4094), 0->Off             |
| VID4:            | <input type="text" value="7"/> (2 ~ 4094), 0->Off             |

### 1.14 SIP Settings

1.14.1 In SIP Settings you can setup the Service Domain, Port Settings, Codec Settings, Codec ID Settings, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related informations correctly then you can register to the SIP Proxy Server correctly.

1.14.2 In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the VoIP Phone. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts.

- First you need click Active to enable the Service Domain, then you can input the following items:
  - Display Name: you can input the name you want to display.
  - User Name: you need to input the User Name get from your ISP.
  - Register Name: you need to input the Register Name get from your ISP.
  - Register Password: you need to input the Register Password get from your ISP.
  - Domain Server: you need to input the Domain Server get from your ISP.
  - Proxy Server: you need to input the Proxy Server get from your ISP.
  - Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.
  - Register Period: you need to input the Register Period get from your ISP. This is count in minute.
  - You can see the Register Status in the Status item. If the item shows “Registered”, then your VoIP Phone is registered to the ISP, you can make a phone call directly.
  - If you have more than one SIP account, you can following the steps to register to the other ISP.
  - When you finished the setting, please click the Submit button.

## Service Domain Settings

You could set information of service domains in this page.

---

| Realm 1 (Default)  |   |
|--------------------|---|
| Active:            | <input checked="" type="radio"/> On <input type="radio"/> Off |
| Display Name:      | <input type="text"/>  |
| User Name:         | <input type="text"/>  |
| Register Name:     | <input type="text"/>  |
| Register Password: | <input type="password"/>                                      |
| Domain Server:     | <input type="text"/>  |
| Proxy Server:      | <input type="text"/>  |
| Outbound Proxy:    | <input type="text"/>  |
| Status:            | Not Registered  |

| Realm 2            |   |
|--------------------|---|
| Active:            | <input type="radio"/> On <input checked="" type="radio"/> Off |
| Display Name:      | <input type="text"/>  |
| User Name:         | <input type="text"/>  |
| Register Name:     | <input type="text"/>  |
| Register Password: | <input type="password"/>                                      |
| Domain Server:     | <input type="text"/>  |
| Proxy Server:      | <input type="text"/>  |
| Outbound Proxy:    | <input type="text"/>  |
| Status:            | Not Registered  |

| Realm 3            |   |
|--------------------|---|
| Active:            | <input type="radio"/> On <input checked="" type="radio"/> Off |
| Display Name:      | <input type="text"/>  |
| User Name:         | <input type="text"/>  |
| Register Name:     | <input type="text"/>  |
| Register Password: | <input type="password"/>                                      |
| Domain Server:     | <input type="text"/>  |
| Proxy Server:      | <input type="text"/>  |
| Outbound Proxy:    | <input type="text"/>  |
| Status:            | Not Registered  |

- 1.14.3 Port Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.

### Port Settings

You could set the port number in this page.

SIP Port:  (10~65533)

RTP Port:  (10~65533)

- 1.14.4 Codec Settings: you can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.

### Codec Settings

You could set the codec settings in this page.

**Codec Priority**

Codec Priority 1:

Codec Priority 2:

Codec Priority 3:

Codec Priority 4:

Codec Priority 5:

Codec Priority 6:

Codec Priority 7:

Codec Priority 8:

**RTP Packet Length**

G.711 & G.729:

G.723:

**G.723 5.3K**

G.723 5.3K:  On  Off

**Voice VAD**

Voice VAD:  On  Off

- 1.14.5 Codec ID Settings: you can set the Codec ID to meet the other device’s requirement. When you finished the setting, please click the Submit button.

### Codec ID Setting

You could set the value of Codec ID in this page.

| Codec Type   | ID  | Default Value                           |
|--------------|---|---|
| G726-16 ID:  | <input type="text" value="23"/> (95~255)  | <input checked="" type="checkbox"/> 23  |
| G726-24 ID:  | <input type="text" value="22"/> (95~255)  | <input checked="" type="checkbox"/> 22  |
| G726-32 ID:  | <input type="text" value="2"/> (95~255)   | <input checked="" type="checkbox"/> 2   |
| G726-40 ID:  | <input type="text" value="21"/> (95~255)  | <input checked="" type="checkbox"/> 21  |
| RFC 2833 ID: | <input type="text" value="101"/> (95~255) | <input checked="" type="checkbox"/> 101 |

- 1.14.6 DTMF Setting: you can choose among 2833, Inband DTMF and Send DTMF SIP info in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button. This

### DTMF Setting

You could set the DTMF setting in this page.

2833  
 Inband DTMF  
 Send DTMF SIP Info

- 1.14.7 RPort Function: you can setup the RPort Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button.

### RPort Setting

You could enable/disable the RPort setting in this page.

RPort:  On  Off

- 1.14.8 Other Settings: you can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. When you finished the setting, please click the Submit button. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

### Other Settings

You could set other settings in this page.

Hold by RFC:  On  Off

Voice QoS:  (0~63)

SIP QoS:  (0~63)

SIP Expire Time:  (60~66400 sec)

## 1.15 NAT Trans.

- 1.15.1 In NAT Trans. you can setup STUN function. These functions can help your VoIP Phone working properly behind NAT.
- 1.15.2 STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your VoIP Phone working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

## STUN Setting

You could set the IP of STUN server in this page.

---

STUN:  On  Off

STUN Server:

STUN Port:  (1024~65535)

### 1.16 Others.

- 1.16.1 In Others you can setup Auto config function. The function can configure your VoIP Phone automatically.
- 1.16.2 Auto Config: you can setup the Auto Configuration to Disable, Enable by TFTP or FTP, and auto configuration TFTP or FTP Server IP address in this page. This function can automatically download the configure file to setup your VoIP Phone. When you finished the setting, please click the Submit button.

## Auto Configuration Setting

You could enable/disable the auto configuration setting in this page.

---

Auto Configuration:  Off  By TFTP  By FTP

TFTP Server:

FTP Server:

FTP Username:

FTP Password:

File Path:

- 1.16.3 ICMP Setting: you can setup the ICMP Setting Enable/Disable in this page. When you finished the setting, please click the Submit button.

## ICMP Setting

You could enable/disable the ICMP setting in this page.

---

ICMP Not Echo:  On  Off

### 1.17 System Authority

- 1.17.1 In System Authority you can change your login name and password.

### System Authority

You could change the login username/password in this page.

---

|                     |                          |
|---------------------|--------------------------|
| New username:       | <input type="text"/>     |
| New password:       | <input type="password"/> |
| Confirmed password: | <input type="password"/> |

### 1.18 Save Change

1.18.1 In Save Change you can save the changes you have done. If you want to use new setting in the VoIP Phone, You have to click the Save button. After you click the Save button, the VoIP Phone will automatically restart and the new setting will effect.

### Save Changes

You have to save changes to effect them.

---

Save Changes:

### 1.19 Update

- 1.19.1 In Update you can update the VoIP Phone’s firmware to the new one or do the factory reset to let the VoIP Phone back to default setting.
- 1.19.2 In New Firmware function you can update new firmware via HTTP in this page. You can upgrade the firmware by the following steps:
  - Select the firmware code type, Risc or DSP code.
  - Click the “Browse” button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
  - Select the correct file you want to download to the VoIP Phone then click the Update button.

### Update Firmware

You could update the newest firmware.

---

Method:  HTTP  TFTP

**HTTP**

|                |   |
|----------------|---|
| Code Type:     | <input type="text" value="Risc"/>                             |
| File Location: | <input type="text"/> <input type="button" value="Browse..."/> |

**TFTP**

|              |                                      |
|--------------|--------------------------------------|
| TFTP Server: | <input type="text" value="0.0.0.0"/> |
|--------------|--------------------------------------|

1.19.3 In Default Setting you can restore the VoIP Phone to factory default in this page. You can just click the Restore button, then the VoIP Phone will restore to default and automatically restart again.

## Restore Default Settings

You could click the restore button to restore the factory settings.

---

Restore default settings:

## 1.20 Reboot

- 1.20.1 Reboot function you can restart the VoIP Phone. If you want to restart the VoIP Phone, you can just click the Reboot button, then the VoIP Phone will automatically.

## Reboot System

You could press the reboot button to restart the system.

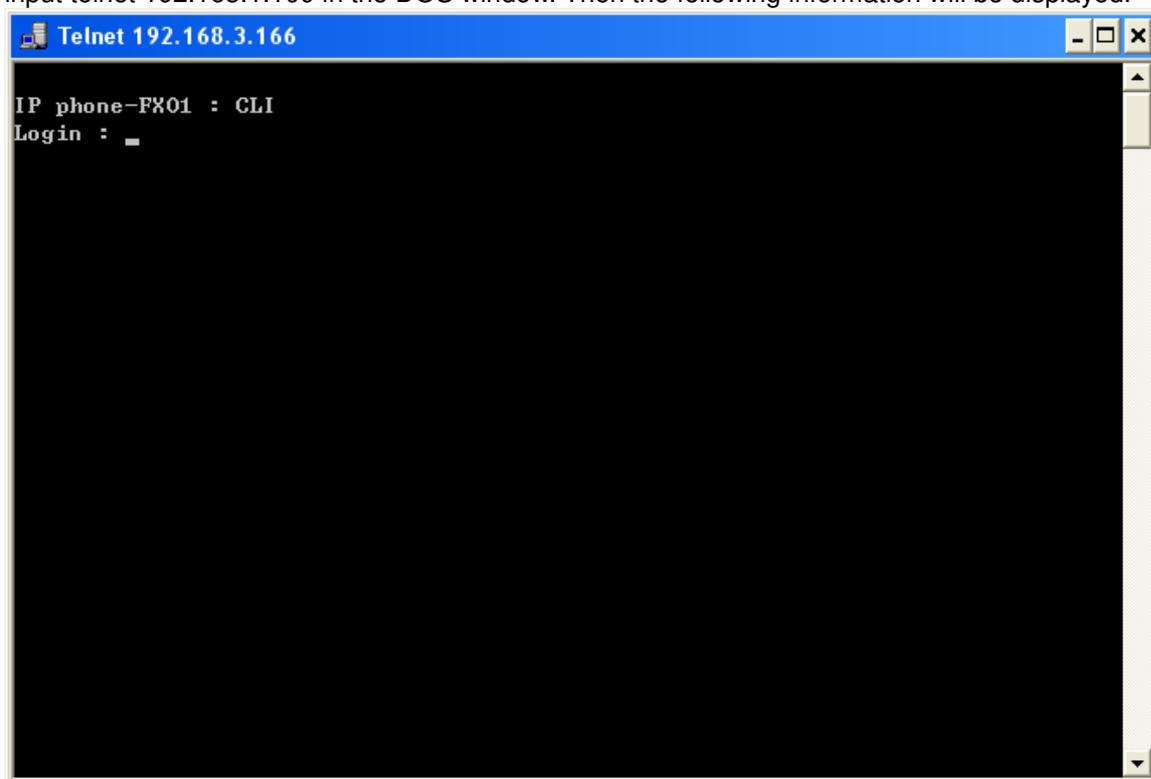
---

Reboot system:

**Setup the VoIP Phone by using Telnet (only support Windoww xp)**

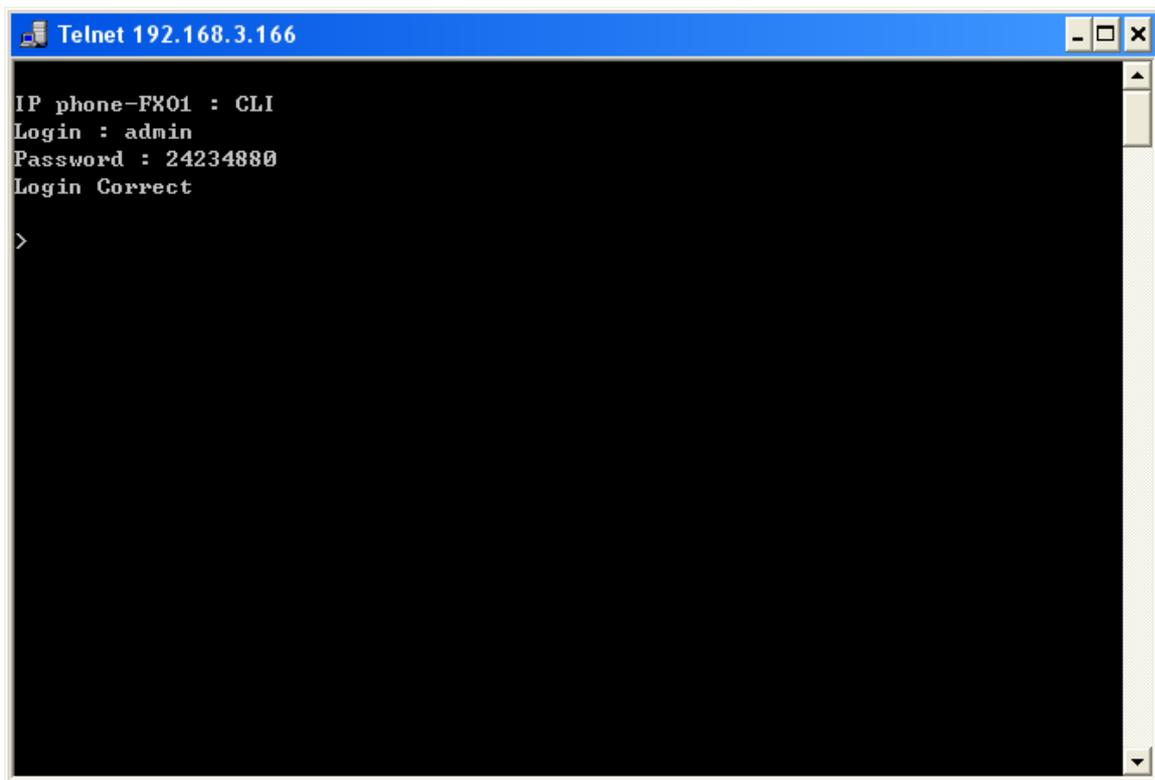
**1.21 Configure**

On the PC choose Start>Run, and then type telnet 192.168.1.100 into Run field in popuping Run dialog. Or input telnet 192.168.1.100 in the DOS window. Then the following information will be displayed.



**Login into the VoIP Phone**

Press "Enter" and the hyper terminal will show the "Login: ". Input "admin" and press the "Enter" button. Then hyper terminal will show the "Password: "Input "2423488" and press the "Enter" button. Now you already login the VoIP Phone.



## 1.22 Using CLI command to configure the VoIP Phone

CLI command list as below:

| ltno | Command   | Description                    |
|------|-----------|--------------------------------|
| 1    | ?         | Show CLI Command               |
| 2    | arp       | ARP Configuration              |
| 3    | ipconfig  | Interface Configuration        |
| 4    | save      | Save to flash                  |
| 5    | reboot    | Reboot                         |
| 6    | exit      | Exit                           |
| 7    | debugmode | Enter Debug Mode               |
| 8    | update    | Update Flash Code/RAM          |
| 9    | auth      | Change User Name and Password  |
| 10   | nat       | NAT Configuration              |
| 11   | dns       | DNS Configuration              |
| 12   | ping      | ping [-IN] [IP-addr host-name] |
| 13   | sip       | SIP Configuration              |
| 14   | ddns      | DDNS Configuration             |
| 15   | sntp      | SNTP Configuration             |
| 16   | vlan      | VLAN Configuration             |
| 17   | time      | Get System Time                |
| 18   | mactab    | Show MAC Learning Table        |
| 19   | dump      | Read/Write Memory              |
| 20   | book      | Edit phone book                |
| 21   | reload    | Reload Factory Setting         |
| 22   | watchdog  | WatchDog Function              |
| 23   | phone     | Phone Setting                  |
| 24   | weblogo   | Change Web's logo              |
| 25   | dsp       | Show dsp type                  |
| 26   | addport   | Add Nat Port Mapping           |
| 27   | cid       | Select slic Cid                |
| 28   | slic      | read or write slic registers   |
| 29   | ver       | Firmware Version               |

- “?” function is to show CLI command list in the screen.

- arp function

| ltno | Command | Description          |
|------|---------|----------------------|
| 1    | ?       | Show 'arp' Option    |
| 2    | -a      | Show ARP Table       |
| 3    | -d      | Delete ARP Table     |
| 4    | -s      | Set Static ARP Table |
| 5    | (null)  | Show ARP Table       |

- ipconfig function

| ltno | Command | Description             |
|------|---------|-------------------------|
| 1    | ?       | Show 'ipconfig' Option  |
| 2    | -if0    | Interface 0             |
| 3    | -if1    | Interface 1             |
| 4    | -if2    | Interface 2             |
| 5    | -h      | Set Host Name           |
| 6    | -a      | Set ARP Cache Expire    |
| 7    | -r      | Restore Current Setting |
| 8    | (null)  | Show IP Setting         |

- ipconfig -ifN function → N is 0, 1, 2

| ltno | Command | Description                 |
|------|---------|-----------------------------|
| 1    | ?       | Show 'ipconfig -ifN' Option |
| 2    | -t      | Set Host Type               |
| 3    | -m      | Set MAC Address             |
| 4    | -i      | Set IP Address              |
| 5    | -nm     | Set Net Mask                |
| 6    | -g      | Set Gateway                 |
| 7    | -dns0   | Set Primary DNS server      |
| 8    | -dns1   | Set Secondary DNS server    |
| 9    | -dr     | Set Default Route           |
| 10   | -nat    | Set NAT                     |
| 11   | on      | Enable Interface            |
| 12   | off     | Disable Interface           |
| 13   | -dhcps  | DHCP Server Setting         |
| 14   | -ddns   | Set DDNS                    |
| 15   | -bridge | Set Bridge                  |
| 16   | -dev0   | Set Device 0 Setting        |
| 17   | -dev1   | Set Device 1 Setting        |
| 18   | -dev2   | Set Device 2 Setting        |
| 19   | (null)  | Show Interface Setting      |

- save function

| ltno | Command | Description         |
|------|---------|---------------------|
| 1    | ?       | Show 'save' Option  |
| 2    | -book   | Save phone book     |
| 3    | -sys    | Save system setting |

- reboot function is to restart the system.
- exit function is to exit the CLI.
- debugmode function is to enter the debugmode.

- update function

| ltno | Command  | Description                        |
|------|----------|------------------------------------|
| 1    | ?        | Show 'update' Option               |
| 2    | -os      | Update OSImage(IP filename)        |
| 3    | -dsp     | Update DSP Image(IP filename)      |
| 4    | -all     | Update All Image(IP filename)      |
| 5    | -server  | Update Server (IP filename length) |
| 6    | -pcm     | PCM(IP filename)                   |
|      | -alaw    | alaw (IP filename)                 |
|      | -ulaw    | ulaw (IP filename)                 |
|      | -g729    | g729 (IP filename)                 |
|      | -g723    | g723 (IP filename)                 |
|      | -g726.16 | g726.16 (IP filename)              |
|      | -g726.24 | g726.24 (IP filename)              |
|      | -g726.32 | g726.32 (IP filename)              |
|      | -g726.40 | g726.40 (IP filename)              |

IP is the TFTP server's IP address, and the filename is the image you want to download into the system.

□ auth function

| ltno | Command | Description                             |
|------|---------|---|
| 1    | ?       | Show 'auth' Option                      |
| 2    | -admin  | Change Administrator user name/password |
| 3    | -sys0   | Change System user0 user name/password  |
| 4    | -sys1   | Change System user1 user name/password  |
| 5    | -sys2   | Change System user2 user name/password  |
| 6    | -sys3   | Change System user3 user name/password  |
| 7    | -sys4   | Change System user4 user name/password  |
| 8    | -norm0  | Change Normal user0 user name/password  |
| 9    | -norm1  | Change Normal user1 user name/password  |
| 10   | -norm2  | Change Normal user2 user name/password  |
| 11   | -norm3  | Change Normal user3 user name/password  |
| 12   | -norm4  | Change Normal user4 user name/password  |
| 13   | -ppp    | Change PPP user name/password           |
| 14   | (null)  | Show auth Setting                       |

In each item includes

| ltno | Command | Description                                |
|------|---------|--|
| 1    | ?       | Show 'auth' Option                         |
| 2    | -user   | Change User Name.'auth -sys3 -user xxx '   |
| 3    | -pass   | Change Password.'auth -sys3 -pass xxx xxx' |
| 4    | (null)  | Show auth's System/PPP Setting             |

If you want to change the password, you need to type the password twice in the CLI.

□ nat function

| ltno | Command | Description           |
|------|---------|-----------------------|
| 1    | ?       | Show 'nat' Option     |
| 2    | -vs     | Set 'nat -vs' Option  |
| 3    | -dmz    | Set 'nat -dmz' Option |
| 4    | (null)  | Show NAT Setting      |

In DMZ item includes

| ltno | Command | Description            |
|------|---------|------------------------|
| 1    | ?       | Show 'nat -dmz' Option |
| 2    | on      | EnableDMZ              |
| 3    | off     | EnableDMZ              |
| 4    | -ip     | Set DMZ IP address     |
| 5    | (null)  | Show DMZ Setting       |

□ dns function

| ltno | Command | Description                   |
|------|---------|-------------------------------|
| 1    | ?       | Show 'dns' Option             |
| 2    | -q      | DNS query. dns -q domain-name |
| 3    | (null)  | Show DNS Table                |

□ ping function

| ltno | Command | Description                     |
|------|---------|---------------------------------|
| 1    | ?       | Show 'ping' Option              |
| 2    | -l      | ping [-l N] [IP-addr host-name] |
| 3    | (null)  | ping [IP-addr host-name]        |

□ sip function

| ltno | Command  | Description                      |
|------|----------|----------------------------------|
| 1    | ?        | Show 'sip' Option                |
| 2    | -proxy0  | sip -proxy0                      |
| 3    | -proxy1  | sip -proxy1                      |
| 4    | -proxy2  | sip -proxy2                      |
| 5    | -upnp    | sip -upnp on/off/show            |
| 6    | -exts    | sip -exts sip upnp external-port |
| 7    | -extr    | sip -extr rtp upnp external-port |
| 8    | -sipp    | sip udp port                     |
| 9    | -rtp     | sip rtp port                     |
| 10   | -stun    | sip -stun on/off                 |
| 11   | -rport   | sip -rport on/off                |
| 12   | -sserver | sip -sserver stun-server         |
| 13   | -out     | sip -out outbound-proxy          |
| 14   | -dump    | sip -dump                        |
| 15   | -log     | sip -log on/off                  |
| 16   | -drtp    | sip -drtp 0/1/2                  |
| 17   | -rtpsc   | sip -rtpsc on/off                |
| 18   | -wanip   | sip -wanip                       |
| 19   | -nattype | sip -nattype                     |
| 20   | -hbyrfc  | sip -hbyrfc                      |
| 21   | -dereg   | sip -dereg                       |
| 22   | -restart | sip -restart                     |
| 23   | -jbt     | sip -jitter buffer Threshold     |
| 24   | (null)   | Show SIP Setting                 |

□ ddns function

| ltno | Command  | Description             |
|------|----------|-------------------------|
| 1    | ?        | Show 'ddns' Option      |
| 2    | -type    | Set DDNS Type           |
| 3    | -host    | Set Host Name           |
| 4    | -wild    | Set Wild Card Mode      |
| 5    | -mx      | Set Mail Exchanger      |
| 6    | -backmx  | Set Mail Exchanger Mode |
| 7    | -offline | Set Offline Mode        |
| 8    | -user    | Set Login User Name     |
| 9    | -pass    | Set Login Password      |
| 10   | (null)   | Show DDNS Setting       |

□ sntp function

| ltno | Command | Description                        |
|------|---------|------------------------------------|
| 1    | ?       | Show 'sntp' Option                 |
| 2    | -on     | Enable SNTP Client                 |
| 3    | -off    | Disable SNTP Client                |
| 4    | -ip1    | Set SNTP Server1 IP                |
| 5    | -ip2    | Set SNTP Server2 IP                |
| 6    | -mode   | Set SNTP Client Mode               |
| 7    | -zone   | Set GMT Time Zone: [+][hour]:[min] |
| 8    | -adjust | Set Adjustment Time: [second]      |
| 9    | (null)  | Show SNTP Setting                  |

- vlan function

| ltno | Command | Description        |
|------|---------|--------------------|
| 1    | ?       | Show 'vlan' Option |
| 2    | -tx     | Tx Vlan setting    |
| 3    | -rx     | Rx Vlan setting    |
| 4    | (null)  | Show Vlan Setting  |

- time function

| ltno | Command | Description                |
|------|---------|----------------------------|
| 1    | ?       | Show 'Time' Option         |
| 2    | -t      | Modify Time: hour:min:sec  |
| 3    | -d      | Modify date: year:mon:date |
| 4    | (null)  | Show Data & Time           |

- mactab function is to show MAC learning table.

- dump function

| ltno | Command | Description         |
|------|---------|---------------------|
| 1    | ?       | Show 'dump' Option  |
| 2    | -r      | dump -r XXXXxxxx    |
| 3    | -w      | dump -w XXXXxxxx XX |

- book function

| ltno | Command | Description        |
|------|---------|--------------------|
| 1    | ?       | Show 'book' Option |
| 2    | -a      | Show answer list   |
| 3    | -c      | Show call list     |
| 4    | -s      | speed dial         |
| 5    | -p      | phone book         |

- reload function is to Reload Factory Setting, please make sure you want to do the factory reset.

- watchdog function

| ltno | Command | Description            |
|------|---------|------------------------|
| 1    | ?       | Show 'WatchDog' Option |
| 2    | on      | Enable WatchDog        |
| 3    | off     | Disable WatchDog       |
| 4    | (null)  | Show WatchDog Setting  |

- phone function

| ltno | Command     | Description                               |
|------|-------------|---|
| 1    | ?           | Show 'phone' Option                       |
| 2    | -autoanswer | phone auto answer                         |
| 3    | -vol        | Volume setting                            |
| 4    | -block      | Block Incoming call                       |
| 5    | -ring       | Set Melody Ringer                         |
| 6    | -forward    | Auto-forward Incall to Phone[0-9] in Book |
| 7    | (null)      | Show Phone Setting                        |

- weblogo function

| ltno | Command | Description           |
|------|---------|-----------------------|
| 1    | ?       | Show 'weblogo' Option |
| 2    | -on     | Vender Logo           |
| 3    | -off    | Crystal media Logo    |
| 4    | (null)  | Show weblogo Setting  |

- dsp function is to show dsp code type.
- addport function is to add Nat Port Mapping

- cid function

| ltno | Command | Description                                |
|------|---------|--|
| 1    | ?       | Show 'cid' Option                          |
| 2    | -off    | Disable Slic Cid signal                    |
| 3    | -1      | Tx FSK after 1st Ring                      |
| 4    | -2      | Tx FSK before 1st Ring                     |
| 5    | -3      | Tx DTMF before 1st Ring                    |
| 6    | -4      | Tx FSK with Line reversal before 1st Ring  |
| 7    | -5      | Tx DTMF with Line reversal before 1st Ring |
| 8    | -time   | FSK cid with time message                  |
| 9    | -single | Single type FSK CID                        |
| 10   | (null)  | Show Cid Option                            |

- slic function

| ltno | Command | Description        |
|------|---------|--------------------|
| 1    | ?       | Show 'slic' Option |
| 2    | -ring   | Issue Ring signal  |
| 3    | -r      | read slic addr     |
| 4    | -w      | write slic addr    |
| 5    | -a      | read all slic reg  |
| 6    | (null)  | Show slic register |

- ver function is to show Firmware Version.

### Phone function list

When your VoIP Phone is configured properly, you can make a phone call to your friend located in the same service provider. If you want to make a phone call, you can dial the phone number then press “#” button to start to dial the phone number or wait for a while then system will dial the number automatically.

The VoIP Phone provides some functions that list as below:

1. **Call Hold:** You can push the ‘Hold’ button to hold the current call for a while, then push ‘Hold’ button again to keep talking.
2. **Call Waiting:** When a new call is coming while you are talking, you can push the ‘Hold’ button to switch to the new call. You can push the ‘Hold’ button to switch between the two calls.
3. **3-Way Conference:** If you want to make a 3-way conference call, you can make a phone call to the first phone number. After the call is established, push the ‘Hold’ button then you can hear the Dial tone, then make a phone call to the second phone number. When the second call is established, press the ‘Conf’ button.
4. **Call Transfer:** Below is the operation method.  
Step 1: A call B  
Step 2: B press "Transfer" then A will be held  
Step 3: Make a new call to C  
Step 4: Before C answers the call, hang up the phone
5. **Redial:** User can push the Redial button to dial the last dialed number.
6. **Speaker Phone:** You can use Speaker phone to make a phone call.
7. **Pre Dial:** User can dial the number first, after finished then raise the handset or push the speaker button; the IP Phone will start to dial.
8. **Phone book:** When Press the phone book key ,the LCD will display to “Seach: [ ]” . Enter you want name, then press the “OK” key .the LCD will display to the name.and press the “SPK” key or pick up the hendset. Auto dail the number.
9. **DND**  
Pess the “ DND” Key , IP-301into the Do Not Disturb” staute. Again press the “ DND” key exit the function.
10. **FWD**  
Pess the “FWD” key mIP-300 into Call Forward function staute. Again press the “FWD” key exit the function.

### USING THE IP PHONE

#### To place a call

Call another IP 300 IP phone under the same Gatekeeper:

1. **Handset:** Pick up the handset and listen for the Internet dial tone. Then dial the phone number you wish to call and press “#”. Once the call connection has been established and the ring tone has sounded, wait for the other party to answer. When the other party answers, you can begin speaking. When the call is over, put back the handset. The dialed number will be saved into the Call Out.
2. **Hand free:** Press”H/F” key and listen for the Internet dial tone. Then input the phone number you wish to call and press “#”. Once the call connection has been established and the ring tone has sounded, wait for the other party to answer. When the other party answers, you can begin speaking. When the call is over, Press”H/F” key again. The dialed number will be saved into the Call out.

### When you're using gatekeeper provided by the manufacturer to call a fixed line or mobile

1. Lift the handset from the cradle or press "**SPK**" and listen for the dial tone

**Note:** Upon lifting the handset, the LCD displays the message **IP Dialing ...**

2. Dial "00" (*or any prefix provided by service provider*) followed by the destination phone number and press the # key. When dialing a number in North America, always dial 00 + 1 + the area code + the number.

As the IP Phone attempts to connect the call, the LCD displays the message **Calling...** as well as the phone number dialed.

If the connection attempt is successful, the **Calling** message will change to **Connected**.

After displaying the **Connected** message, the phone being called will begin to ring. When the call is answered, you may start talking.

## Appendix

### Get a FWD account

1. The website is [www.freeworlddialup.com](http://www.freeworlddialup.com); you can apply an account to use the VoIP communication. You can follow the instruction to input the information. After you finished, you will receive a mail sent by the FWD mail system, you will get the account information in the mail.
2. When you got the account, you can setup the related information into the VoIP Phone.
3. You can setup the related information into the VoIP Phone by web browser. Also you can use Telnet, Console via CLI command to configure the VoIP Phone. You need to input the Proxy Name, Domain Name, Register Name, and password. The Display Name you can input what you want to let others see.

4. After you registered to the SIP Server, you can try to call your friends who also registered in the same SIP Server. You just need to dial your friend's user name (registered name) and press “#” then you can make a phone call to your friend.
5. If you want to make a phone call to the other in the internet, first you need to registered in a Proxy Server (with SIP Server IP, Domain IP, registered name, Password), make sure you already enable Stun function, then you can try