# VoIP Phone User Manual

V.03

2005/10/19

# 1 Introduction

This user's manual is for VoIP Phone. This user's manual will explain the keypad instruction, web configuration and command line configuration for the VoIP Phone. Before using the VoIP Phone, some setup processes are required to make the VoIP Phone work properly. Please refer to the Setup Menu for further information.

## 1.1 Hardware Overview

The VoIP Phone has the following interfaces for Networking, telephone interface, LED indication, and power connector.

1.1.1 Two RJ-45 Networking interface, these two interfaces support 10/100Mps Fast Ethernet. you can connect one RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the other one to your computer.

1.1.2 LED Indication: There are some LED indicators in the VoIP Phone to show the functions, like speaker phone, .Rggister, ….

## 1.2 Software Overview

| **Network Protocol** | **Tone** |
|---|---|
| • SIP v1 (RFC2543), v2(RFC3261)<br>• IP/TCP/UDP/RTP/RTCP<br>• IP/ICMP/ARP/RARP/SNTP<br>• TFTP Client/DHCP Client/ PPPoE Client<br>• Telnet/HTTP Server<br>• DNS Client | • Ring Tone<br>• Ring Back Tone<br>• Dial Tone<br>• Busy Tone<br>• User Programming Tone |
| | **Phone Function** |
| **Codec** | • Volume Adjustment<br>• Speed dial, Phone book<br>• Flash<br>• Speaker Phone |
| • G.711: 64k bit/s (PCM)<br>• G.723.1: 6.3k / 5.3k bit/s<br>• G.726: 16k / 24k / 32k / 40k bit/s (ADPCM)<br>• G.729A: 8k bit/s (CS-ACELP)<br>• G.729B: adds VAD & CNG to G.729 | **IP Assignment** |
| | • Static IP<br>• DHCP<br>· PPPoE |
| **Voice Quality** | **Security** |
| • VAD: Voice activity detection<br>• CNG: Comfortable noise generator<br>• LEC: Line echo canceller<br>• Packet Loss Compensation<br>· Adaptive Jitter Buffer | • HTTP 1.1 basic/digest authentication for Web setup<br>· MD5 for SIP authentication (RFC2069/ RFC 2617) |
| | **QoS** |

| Call Function | · ToS field |
|---|---|
| • Call Hold<br>• Call Waiting<br>• Call Forward<br>• Caller ID<br>· 3-way conference | **NAT Traversal**<br>· STUN<br>**Configuration** |
| **DTMF Function**<br>• In-Band DTMF<br>• Out-of Band DTMF<br>• SIP Info | • Web Browser<br>• Console/Telnet<br>• Keypad |
| **SIP Server** | **Firmware Upgrade** |
| • Registrar Server (three SIP account)<br>• Outbound Proxy | • TFTP<br>• Console<br>• HTTP |

## 2 Keypad interface for IP Phone demo system

1.3 Keypad description

| Key Name | Description | Note |
|---|---|---|
| 1 | "1", "-", ",", "!", "?" | |
| 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" | |
| * | "*", "•", ":", "@" | |
| # | Start dialing process | 開始撥號鍵 |
| XFER | This is "Transfer" to the other phone number | 轉接鍵 |
| REDIAL | This is "REDIAL" the same number again | 上通重播鍵 |
| HOLD | This is "HOLD" function | 通話保留鍵 |
| Mute | This is "Mute" function | 靜音鍵 |
| DND | This is "Reject" function | 拒接所有來電 |
| OK | This is "OK", accept setting | 設定確認鍵 |
| DEL | This is "Delete", Delete word or phone number | 刪除鍵 |
| UP/DOWN | This is Up↑ and Down↓ key | 上下鍵 |
| LEFT/RIGHT | This is Left← and Right→ key | 左右鍵 |
| MENU | This is the "Menu" key to set the IP Phone | 菜單 |
| SPK | This the Speaker Phone | 免提鍵 |
| Line1~Line4 | This is the Line1 to Line4, will be 4 line | 4 線對外撥號 |
| M1~M4 | This is the M1 to M4, this is 4 speed dial number. | 紀錄 4 組快速鍵 |
| Conf | This is three way conference function | 三方通話 |
| Call In | This is Incoming call list | 來電紀錄 |
| Call Out | This is out going call list | 去電紀錄 |
| Volume -/+ | This is volume setting | 音量鍵 |

1.4 Keypad Function and setting List

2.1.1 Phone Book

2.1.1.1 Search:Search Phone Book. 搜尋電話簿清單

2.1.1.2 Add entry:Add new phone number to phone book. 加入新的電話號碼

2.1.1.3 Speed dial:Add speed dial phone number to speed dial list. 加入新的速撥號碼

2.1.1.4 Erase all:Erase all phone number from Phone Book. 刪除整個電話簿

2.1.2 Call history

2.1.2.1 Incoming calls: Show all incoming call. 顯示所有來電

2.1.2.2 Dialed numbers: Show all dialed call. 顯示所有已撥號

2.1.2.3 Erase record: Delete call history. 刪除通話紀錄

1. All: Delete all call history. 刪除所有通話紀錄
2. Incoming: Delete all incoming call. 刪除所有來電紀錄
3. Dialed: Delete all dialed out call. 刪除所有撥號紀錄

2.1.3 Phone setting

2.1.3.1 Call forward

2.1.3.1.1 All Forward. 所有來電轉接

1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.1.3.1.2 Busy Forward. 忙線時來電轉接

1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.1.3.1.3 No Answer Forward. 無人接聽時來電轉接

1. Activation: To Enabled/Disabled this function.
2. Number: Forward to a Speed Dial Number.

2.1.3.1.4 Timeout: Set the time to start the no answer forward function, ex: 20 means after 20 seconds then forward to the dedicated number.

2.1.3.2 Block Setting

1. All: 拒接所有來電
2. By Time: 一段時間拒接所有來電
3. Duration: Set the start time and end time to Block Setting.

2.1.3.3 Date/Time setting: Date and Time Setting. 日期時間設定功能

2.1.3.3.1 Date & Time: Set the IP Phone Date and Time. 修改日期時間

2.1.3.3.2 SNTP setting

2.1.3.3.2.1 SNTP : Enabled / Disable SNTP. 啓動/ 關閉 網路時間伺服器

2.1.3.3.2.2 Primary SNTP: Set Primary SNTP server IP address. 第一網路時間伺服器

2.1.3.3.2.3 Secondary SNTP: Set Secondary SNTP server IP address. 第二網路時間伺服器

2.1.3.3.2.4 Time zone: Set Time zone. 時區設定

2.1.3.3.2.5 Adjustment Time: Set adjustment time period. 自動對時設定

2.1.3.4 Volume

2.1.3.4.1 Handset volume: Set Handset volume from 0~15 (max.). 話筒音量調整

2.1.3.4.2 Speaker volume: Set Spearer phone volume from 0~15 (max.). 免持聽筒音量調整

2.1.3.5 Ringer

2.1.3.5.1 Ringer volume: Ringer volume setting from 0~15 (max.). 鈴聲音量調整

2.1.3.5.2 Ringer type: Ringer tone selection from 1~4. 鈴聲旋律選擇

2.1.3.6 Auto Dial: Set Auto Dial time from 3~9 seconds.

2.1.3.7 Auto Answer: Set Auto Answr for user can re-dial a call from IP call to PSTN call or from PSTN call to IP call. This function will active on IP Phone with FXO interface.

2.1.3.8 Answer Counterl: Set Auto Answer will active after the numbers of ring. This function will active on IP Phone with FXO interface.

2.1.4 Network

2.1.4.1 General

2.1.4.1.1 IP Type

1. Fixed IP client: 以手動方式設定網路地址
2. DHCP client: 以 DHCP 方式取得網路地址
3. PPPoE client: 以 PPPoE 方式取得網路地

2.1.4.1.2 Fixed IP setting

2.1.4.1.2.1 Host IP: 此話機之網路地址設定

2.1.4.1.2.2 Network mask: 網路遮罩設定

2.1.4.1.2.3 Gateway IP: 網關IP地址設定

2.1.4.1.2.4 MAC address: MAC地址設定

2.1.4.1.3 PPPoE setting

2.1.4.1.3.1 User name: PPPoE使用者名稱設定

2.1.4.1.3.2 Password: PPPoE使用者密碼設定

2.1.4.1.4 DNS Server

2.1.4.1.4.1 Primary DNS: 第一DNS伺服器地址設定

2.1.4.1.4.2 Secondary DNS: 第二DNS伺服器地址設定

2.1.4.2 Status: Show IP address and MAC address, 網路設定狀況, 顯示IP地址及MAC 地址

2.1.5 SIP Settings

2.1.5.1 Service domain

2.1.5.1.1 First realm

2.1.5.1.1.1 Activation: 第一SIP 伺服器啓動/停止

2.1.5.1.1.2 User name: SIP 使用者名稱設定

2.1.5.1.1.3 Display name: SIP 顯示名稱設定

2.1.5.1.1.4 Register name: SIP登錄名稱設定

2.1.5.1.1.5 Register password: SIP登錄密碼設定

2.1.5.1.1.6 Proxy server: SIP Proxy伺服器地址設定

2.1.5.1.1.7 Domain server: Domain伺服器地址設定

2.1.5.1.1.8 Outbound proxy: Outbound Proxy 伺服器地址設定

2.1.5.1.1.9 Register period: The time period to send register information to the SIP Proxy (Minute) 向SIP Proxy Server註冊週期.

2.1.5.1.2 Second realm

2.1.5.1.2.1 Activation: 第二SIP 伺服器啓動/停止

2.1.5.1.2.2 User name: SIP 使用者名稱設定

2.1.5.1.2.3 Display name: SIP 顯示名稱設定

2.1.5.1.2.4 Register name: SIP登錄名稱設定

2.1.5.1.2.5 Register password: SIP登錄密碼設定

2.1.5.1.2.6 Proxy server Proxy: 伺服器地址設定

2.1.5.1.2.7 Domain server: Domain伺服器地址設定

2.1.5.1.2.8 Outbound proxy: Outbound Proxy 伺服器地址設定

2.1.5.1.2.9 Register period: The time period to send register information to the SIP Proxy (Minute) 向SIP Proxy Server註冊週期.

2.1.5.1.3 Third realm

2.1.5.1.3.1 Activation: 第三SIP 伺服器啓動/停

2.1.5.1.3.2 User name: SIP 使用者名稱設定

2.1.5.1.3.3 Display name SIP: 顯示名稱設定

2.1.5.1.3.4 Register name: SIP登錄名稱設定

2.1.5.1.3.5 Register password: SIP登錄密碼設定

2.1.5.1.3.6 Proxy server Proxy: 伺服器地址設定

2.1.5.1.3.7 Domain server: Domain伺服器地址設定

2.1.5.1.3.8 Outbound proxy: Outbound Proxy 伺服器地址設定

2.1.5.1.3.9 Register period: The time period to send register information to the SIP Proxy (Minute) 向SIP Proxy Server註冊週期.

2.1.5.2 Codec

2.1.5.2.1 Codec type

1. G.711 uLaw: 選擇優先用 G.711 uLaw 語音壓縮方式
2. G.711 aLaw: 選擇優先用 G.711 aLaw 語音壓縮方式
3. G.723: 選擇優先用 G.723.1 語音壓縮方式
4. G.729: 選擇優先用 G.729A 語音壓縮方式
5. G.726-16: 選擇優先用 G.726 16Kbps 語音壓縮方式
6. G.726-24: 選擇優先用 G.726 24Kbps 語音壓縮方式
7. G.726-32: 選擇優先用 G.726 32Kbps 語音壓縮方式
8. G.726-40: 選擇優先用 G.726 40Kbps 語音壓縮方式

2.1.5.2.2 VAD: Voice Active Detection 啓動/停止設定

2.1.5.3 RTP setting

2.1.5.3.1 Outband DTMF: Outband DTMF 啓動/停止設定

2.1.5.3.2 Duplicate RTP

1. No duplicate: 語音封包重送 0 次
2. One duplicate: 語音封包重送 1 次
3. Two duplicate: 語音封包重送 2 次

2.1.5.4 RPort Setting: RPort Enabled/Disabled 啓動/停止 RPORT 設定

2.1.5.5 Hold by RFC: 通話保留啓動/停止設定 (依照RFC3261標準)

2.1.5.6 Status 顯示對SIP Proxy 的註冊狀態

1. First Realm: 第一 SIP 伺服器註冊狀態
2. Second Realm: 第二 SIP 伺服器註冊狀態
3. Third Realm: 第三 SIP 伺服器註冊狀態

2.1.6 NAT Transversal

2.1.6.1 STUN setting

2.1.6.1.1 STUN: STUN啓動/停止設定

2.1.6.1.2 STUN server: STUN伺服器地址設定

2.1.7 Administrate

2.1.7.1 Auto Config

2.1.7.1.1 TFTP config: Enable/Disable auto config function.

2.1.7.1.2 TFTP server: Setting the TFTP server IP address.

2.1.7.2 Default setting: 還原成出廠設定值.

2.1.7.3 System Authentication: To do the SIP setting from Keypad, need to input the password first. Default is “test”.

2.1.7.4 Restart: 重新開機

# 3 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

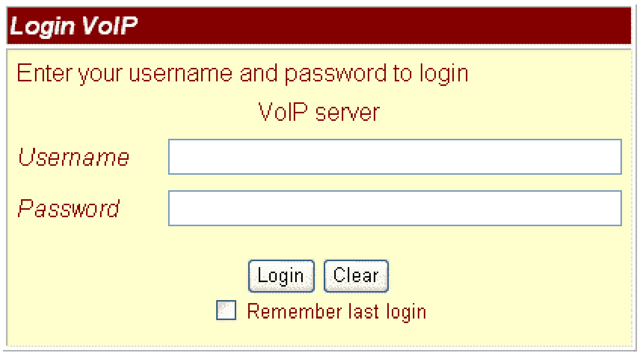
1.5 Login.

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

1. For Administrator, the username is: root; and the password is: test. If you use the account login, you can configure all the setting.
2. For normal user, the username is: system or user; and the password is: test. If you use the account login, but you can not configure the SIP setting.

3.1.2 Click the "Login" button will move into the VOIP PHONE web based management information page.

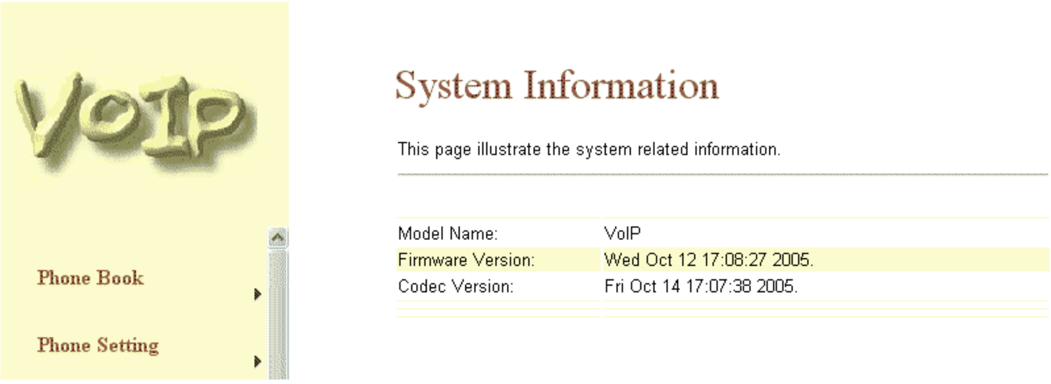
3.1.3 If you change the setting in the Web Management interface, please do remember to click the "Submit" button in that page. After you finished the change of the setting, click the "Save" function in the left side, and click the Save Button. When you finished the setting, please click the Reboot function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



1.6 System Information for the VOIP PHONE.

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

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




1.7 Phone Book

3.1.6 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 "#".

3.1.7 In the Phone Book function you can add/delete the phone number in the phone book list. You can input maximum 100 entries phone book list.

3.1.7.1 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.

3.1.7.2 If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

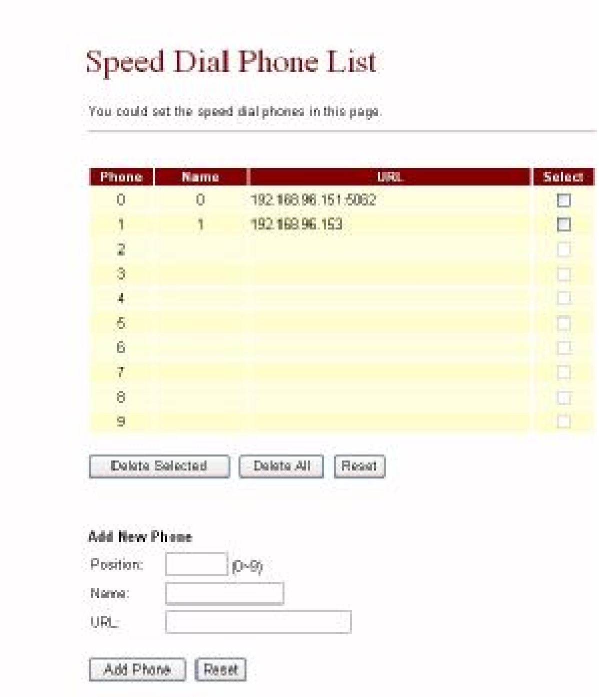
3.1.7.3 If you want to delete all phone numbers, you can click “Delete All” button.


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

3.1.8.1 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.

3.1.8.2 If you want to delete a phone number, you can select the phone number you want to delete then click “Delete Selected” button.

3.1.8.3 If you want to delete all phone numbers, you can click “Delete All” button.




1.8 Phone Setting

3.1.9 In Phone Setting contains Call Forward, SNTP Settings, Volume Settings, Melody Settings, Block Setting, and Auto Dial Setting functions.

3.1.10 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.

3.1.10.1 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.

3.1.10.2 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.

3.1.10.3 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.

3.1.10.4 When you finished the setting, please click the Submit button.


3.1.11 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.


3.1.12 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.

3.1.12.1 Handset Volume is to set the volume you hear from the handset.

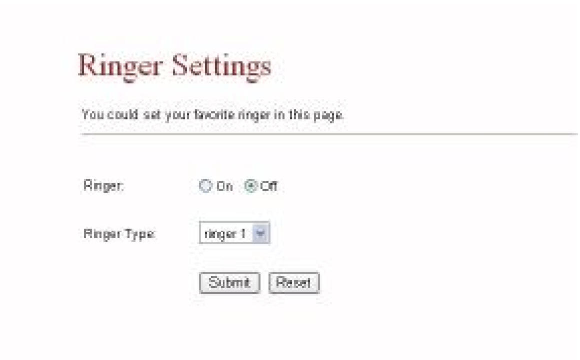
3.1.12.2 Speaker Volume is to set the volume you hear from the speaker phone.

3.1.12.3 Ringer Volume is to set the ringer volume.

3.1.12.4 Handset Gain is to set the volume send out from from the handset.

3.1.12.5 Speaker Gain is to set the volume send out from from the micro phone.


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



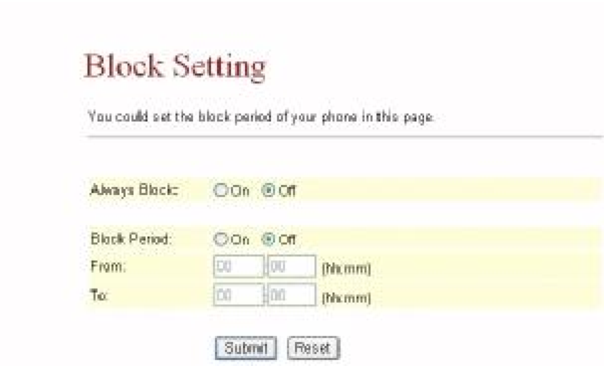

3.1.14

3.1.15 Block Setting function: you can setup the Block Setting to keep the phone slience. You can choose Always Block or Block a period.

3.1.15.1 Always Block: All incoming call will be blocked until disable this feature.

3.1.15.2 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.

3.1.15.3 When you finished the setting, please click the Submit button.



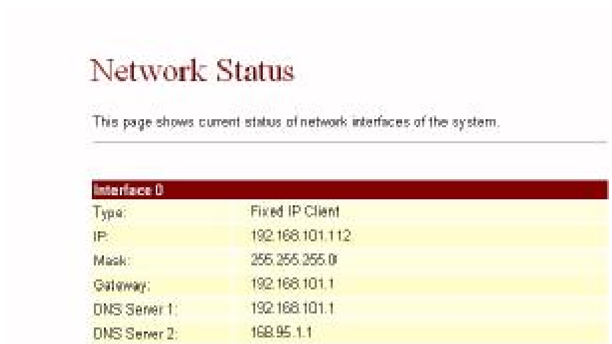
3.1.16 Auto Dial Setting function: This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.


3.1.17 Call Waiting Setting function: If user doesn't want to be inform there is a new incoming call, user can set the function off.


1.9 Network

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

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



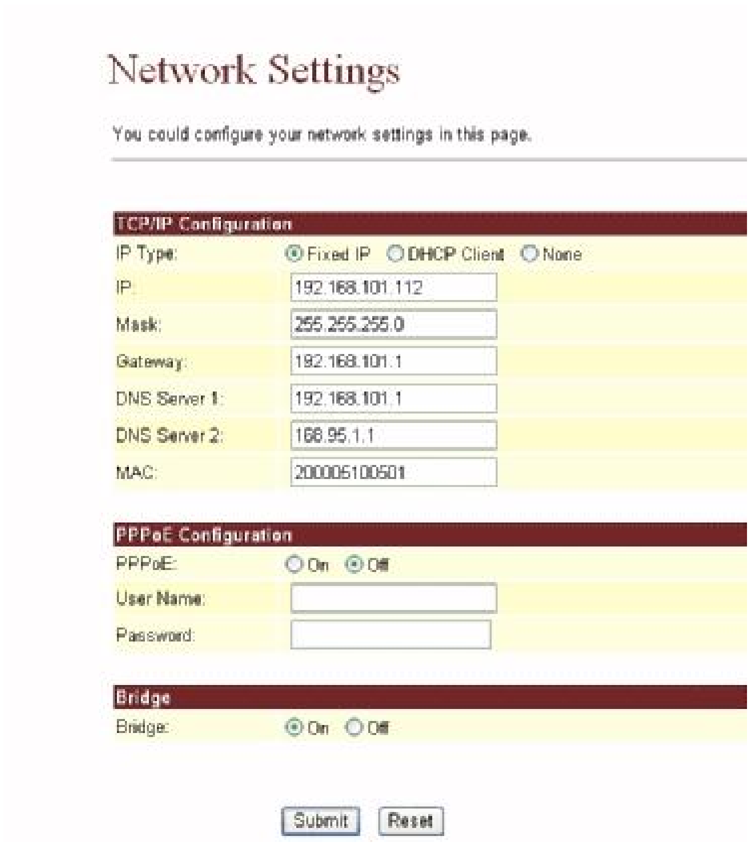
3.1.20 Network Settings: You can configure the VoIP Phone Network setting in this page.

3.1.20.1 The TCP/IP Configuration item is to setup the LAN port's network environment. You may refer to your current network environment to configure the VoIP Phone properly.

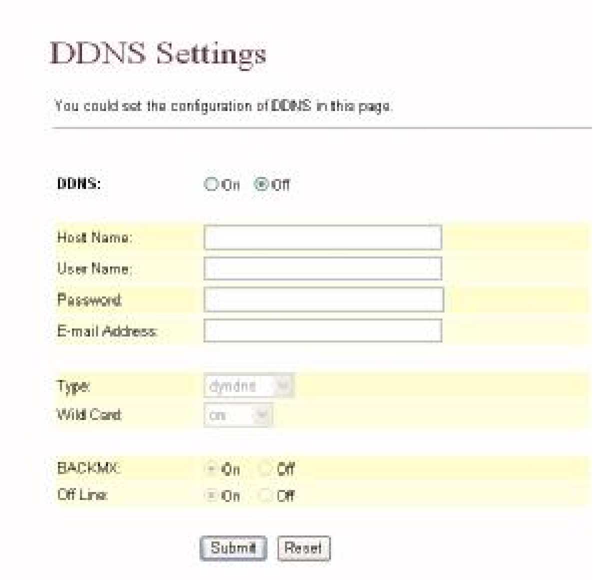
3.1.20.2 The PPPoE Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.

3.1.20.3 The Bridge Item is to setup the VoIP Phone Bridge mode Enable/Disable. If you set the Bridge On, then the two Fast Ethernet ports will be transparent.

3.1.20.4 When you finished the setting, please click the Submit button.




3.1.21 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.




## 1.10 SIP Settings

3.1.22 In SIP Settings you can setup the Service Domain, Port Settngs, 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.

3.1.23 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.

3.1.23.1 First you need click Active to enable the Service Domain, then you can input the following items:

3.1.23.1.1 Display Name: you can input the name you want to display.

3.1.23.1.2 User Name: you need to input the User Name get from your ISP.

3.1.23.1.3 Register Name: you need to input the Register Name get from your ISP.

3.1.23.1.4 Register Password: you need to input the Register Password get from your ISP.

3.1.23.1.5 Domain Server: you need to input the Domain Server get from your ISP.

3.1.23.1.6 Proxy Server: you need to input the Proxy Server get from your ISP.

3.1.23.1.7 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.

3.1.23.1.8 Register Period: you need to input the Register Period get from your ISP. This is count in minute.

3.1.23.1.9 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 direcly.

3.1.23.1.10 If you have more than one SIP account, you can following the steps to register to the other ISP.

3.1.23.1.11 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:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Register Period:
0
(0~99) [0: 30 sec ,1~99 min]
Status:
Not Registered
Realm 2
Active:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Register Period:
15
(0~99) [0: 30 sec ,1~99 min]
Status:
Not Registered
Realm 3
Active:
On
Off
Display Name:
User Name:
Register Name:
Register Password:
Domain Server:
Proxy Server:
Outbound Proxy:
Register Period:
15
(0~99) [0: 30 sec ,1~99 min]
Status:
Not Registered
Submit
Reset

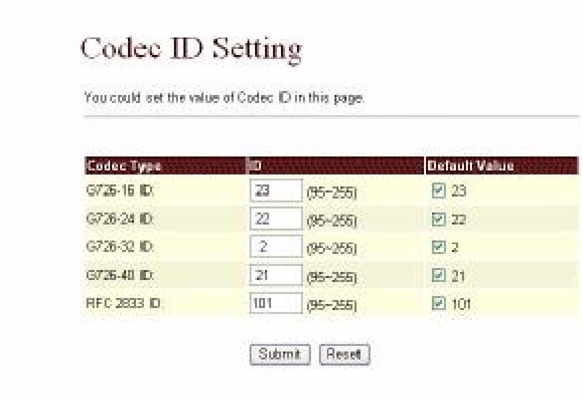
3.1.24 Port Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.


3.1.25 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.



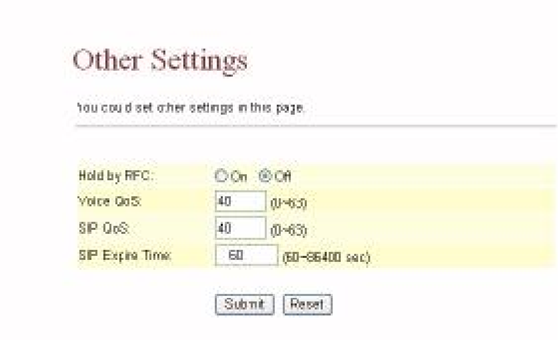
3.1.26 Codec ID Settings: you can set the Codec ID to meet the other device's requirement. When you finished the setting, please click the Submit button.




3.1.27 RTP Setting: you can setup the Out-Band DTMF and Send DTMF SIP Info Enable/Disable in this page. To change this setting, please following your ISP information. When you finished the setting, please click the Submit button. This


3.1.28 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.

3.1.29 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.




## 1.11 NAT Trans.

3.1.30 In NAT Trans. you can setup STUN function. These functions can help your VoIP Phone working properly behind NAT.

3.1.31 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.


## 1.12 Others.

3.1.32 In Others you can setup Auto config function. The function can configure your VoIP Phone automatically.

3.1.33 Auto Config: you can setup the Auto Configuration Enable/Disable and auto configuration TFTP 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.

1.13 System Auth.

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


1.14 Save Change

3.1.35 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.

## 1.15 Update

3.1.36 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.

3.1.37 In New Firmware function you can update new firmware via HTTP in this page. You can ugrade the firmware by the following steps:

3.1.37.1 Select the firmware code type, Risc or DSP code.

3.1.37.2 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.

3.1.37.3 Select the correct file you want to download to the VoIP Phone then click the Update button.


3.1.38 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.


## 1.16 Reboot

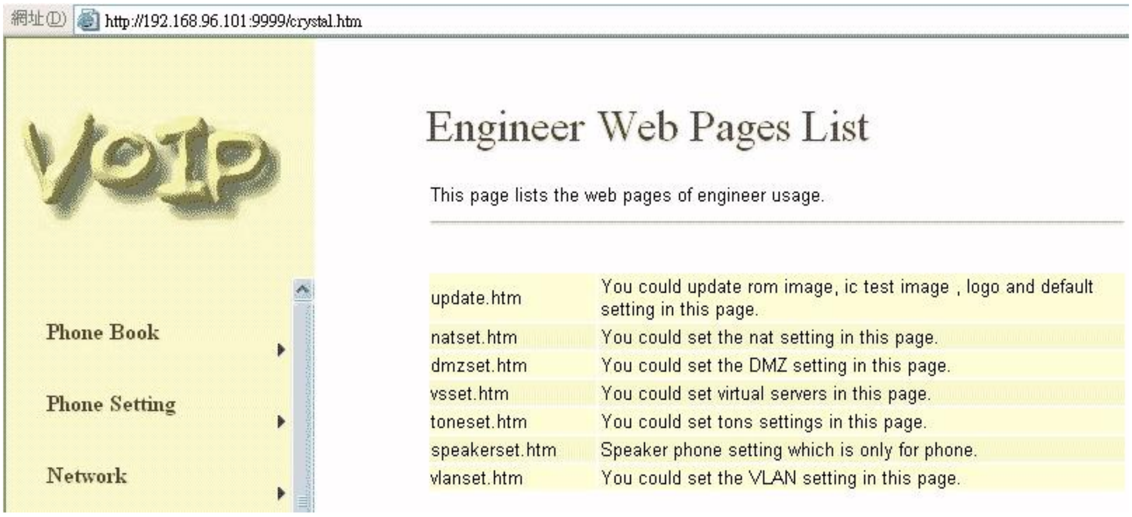
3.1.39 Reboot function you can restart the VoIP Phone. If you want to restart the VoIP Phone, you can just click the Reboot button, then the VoIP Phone will automatically.


Reboot System
You could press the reboot button to restart the system.
Reboot system:
Reboot

# 4 Engineering webpage

## 4.1 Engineer usage webpage list

4.1.1 You have to login the system first then change the webpage to crystal.htm manually. In this webpage you will see the list about engineer webpage. You can change the webpage to what you want.

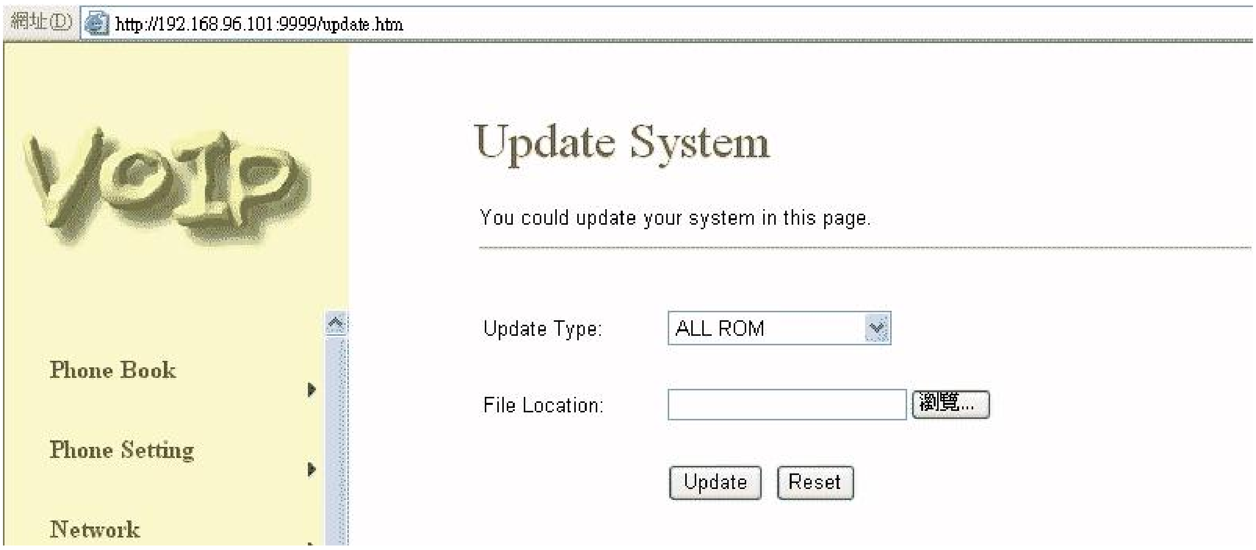



## 4.2 Update.htm

4.2.1 In this page you can update the system's ROM code, IC Test , Default setting and Logo.

4.2.2 Update ROM code. You can update the ROM code from this function. Please be noted that if you update the wrong file or during the update process the power is off, the system will be crashed.

4.2.3 Update Logo. The Logo specification is 220x170 jepg file.



4.3 Natset.htm

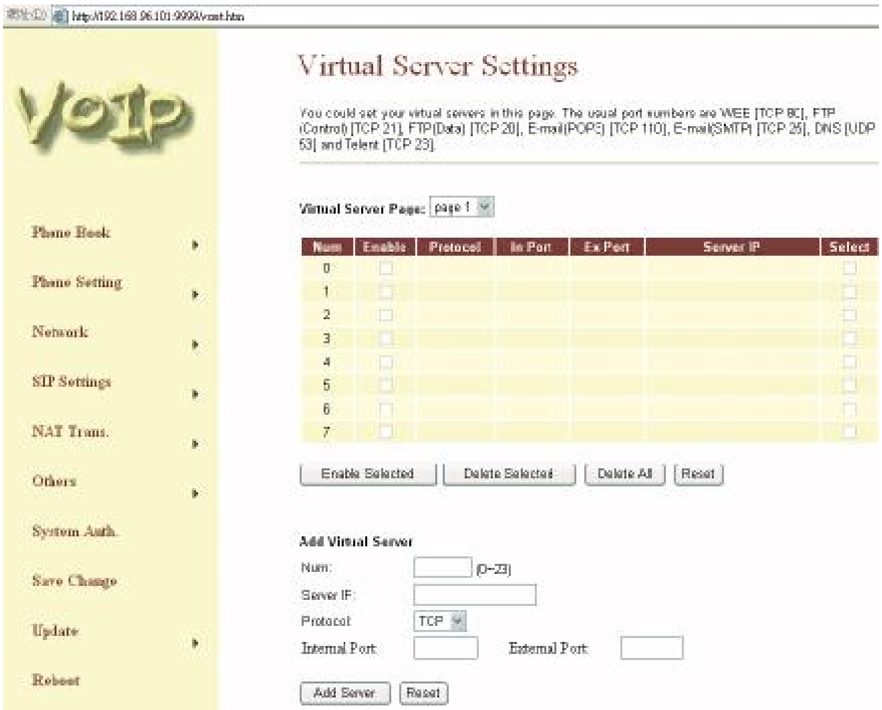
4.3.1 In this page you can setup the nat function. The WAN setting is for you to set how the get the IP address for the device. The LAN setting is for the other devices to get the IP address from the device. You can choose to use DHCP server or not.

## 4.4 Dmzset.htm

4.4.1 In this page you can setup the DMZ function. You need to enable/disable this function and set the IP address for DMZ.

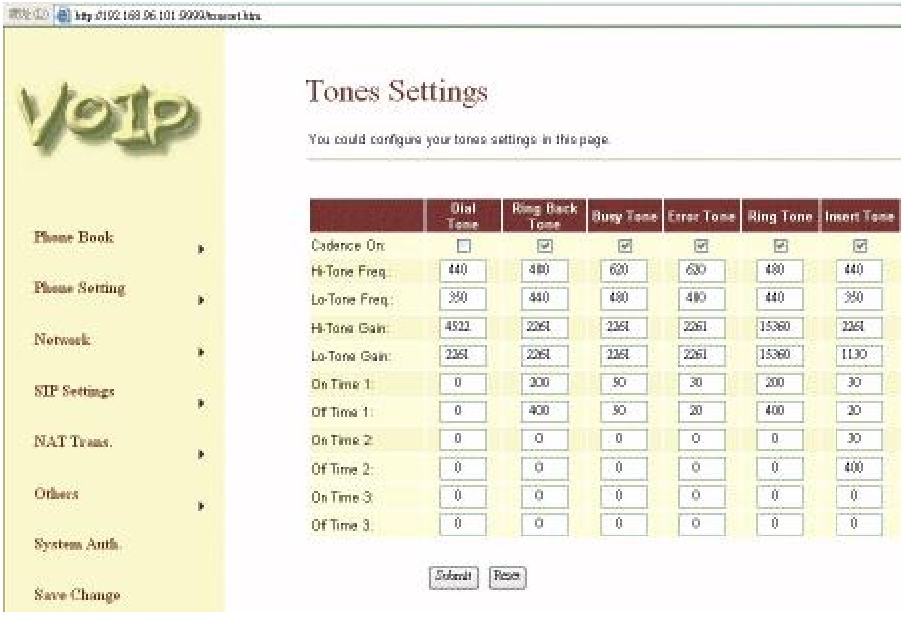

## 4.5 Vsset.htm

4.5.1 In this page you can setup the Virture Server function.



4.6 Toneset.htm

4.6.1 In this page you can setup the Tone frequency and cadence to meet the requirement.

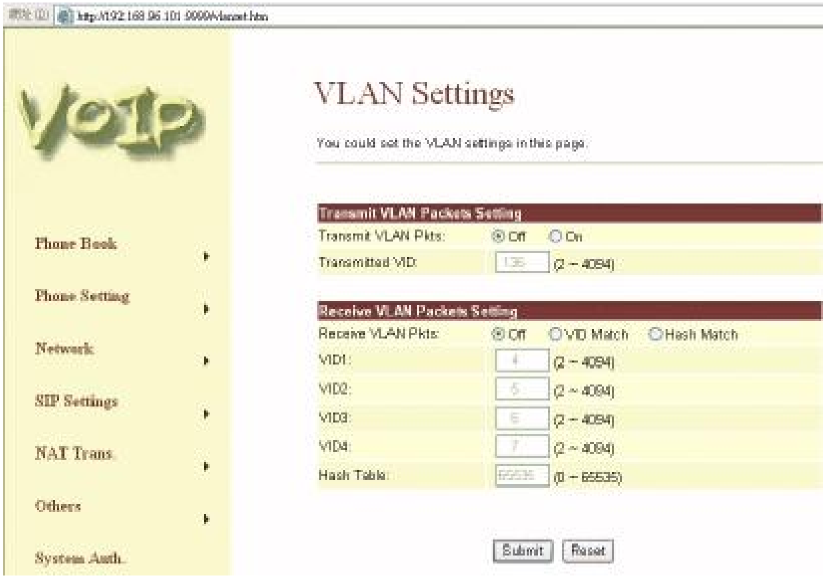



4.7 Speakerset.htm

4.7.1 In this page you can setup the Speaker function.

4.8 Vlanset.htm

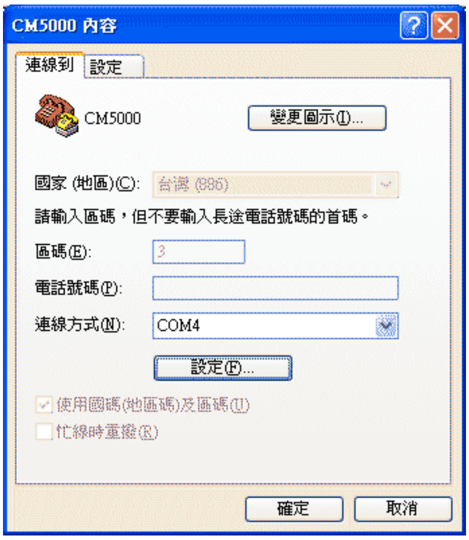
4.8.1 In this page you can setup the VLAN function.



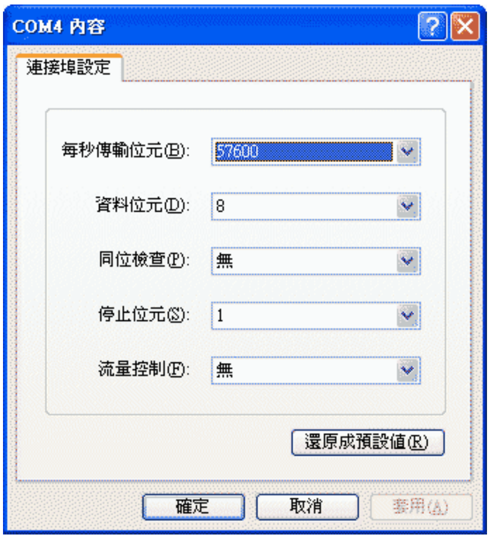
# 5 Setup the VoIP Phone by using Console (Hyper Terminal)

## 1.17 Configure the COM port

First Open the hyper terminal window, select the connection by the COM port, then click the “Setting” button.

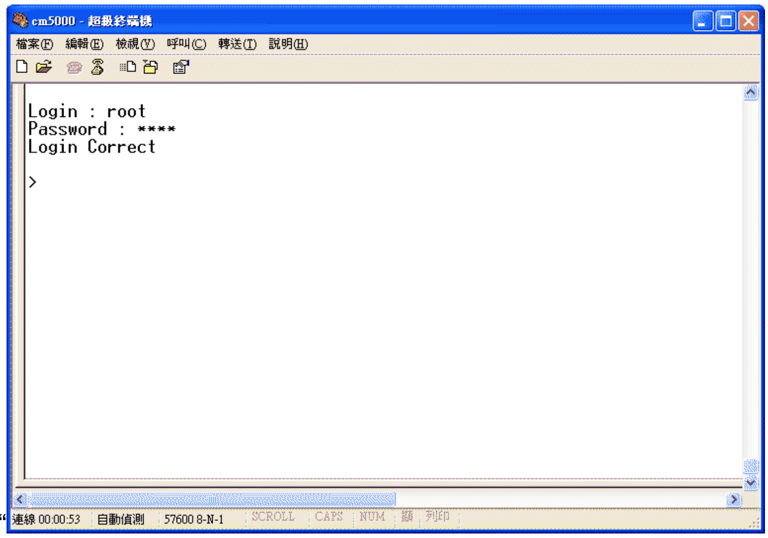



Set the COM port’s setting as following setting. Then click OK.



## 1.18 Login into the VoIP Phone

After finished the setting, click the “Connect” button (looks like a telephone icon). Then the hyper terminal is ready to connect to the VoIP Phone. Press “Enter” and the hyper terminal will show the “Login: “. Input “root” and press the “Enter” button. Then hyper terminal will show the “Password: “. Input “test” and press the “Enter” button. Now you already login the VoIP Phone. Please follow the CLI command list to configure the VoIP Phone with proper instruction and value.



## 1.19 Using CLI command to configure the VoIP Phone

### 5.1.1 CLI command list as below:

| Itno | 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 |

#### 5.1.1.1 "?" function is to show CLI command list in the screen.

#### 5.1.1.2 arp function

| Itno | 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 |

#### 5.1.1.3 ipconfig function

| Itno | 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 |

5.1.1.3.1 ipconfig –ifN function à N is 0, 1, 2

| Itno | 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 |

5.1.1.4 save function

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

5.1.1.5 reboot function is to restart the system.

5.1.1.6 exit function is to exit the CLI.

5.1.1.7 debugmode function is to enter the debugmode.

5.1.1.8 update function

| Itno | 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.

5.1.1.9 auth function

| Itno | 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

| Itno | 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.

5.1.1.10 nat function

| Itno | 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

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

5.1.1.11 dns function

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

5.1.1.12 ping function

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

5.1.1.13 sip function

| Itno | 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 | -rtpp | 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 | -rtpnc | sip -rtpnc 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 |

5.1.1.14 ddns function

| Itno | 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 |

5.1.1.15 sntp function

| Itno | 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 |

5.1.1.16 vlan function

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

5.1.1.17 time function

| Itno | 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 |

5.1.1.18 mactab function is to show MAC learning table.

5.1.1.19 dump function

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

5.1.1.20 book function

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

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

5.1.1.22 watchdog function

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

5.1.1.23 phone function

| Itno | 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 |

5.1.1.24 weblogo function

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

5.1.1.25 dsp function is to show dsp code type.

5.1.1.26 addport function is to add Nat Port Mapping

5.1.1.27 cid function

| Itno | 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 |

5.1.1.28 slic function

| Itno | 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 |

5.1.1.29 ver function is to show Firmware Version.

## 6 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 key to hold the current call for a while, then push Hold key again to keep talking.

2. Call Waiting: When a new call is coming while you are talking, you can push the Flash button to switch to the new call. You can push the Flash 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 Flash 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 Flash button again.

4. Call Transfer: Current we can support 3 kind of transfer application. Below is the operation method.

   A. Normal Transfer: A call B then transfer to C
      Step 1: A call B
      Step 2: B press "Flash" then A will be held
      Step 3: Make a new call to C
      Step 4: After C answer the call the press "Flash" to complete transfer

   B. 2 way Transfer: A call B then transfer to C but C is busy or C reject transfer then B want the call back to A
   Step 1: A call B
   Step 2: B press "Flash" then A will be held
   Step 3: Make a new call to C
   Step 4: If C is busy or reject then B hang up the phone
   Step 5: Press "Line 1" to restore call with A

C. Blind Transfer: A call B then transfer to C and do not care about C's situation
Step 1: A call B
Step 2: B press "Blind transfer" then A will be held
Step 3: Make a new call to C
Step 4: B hang up the phone

5. Redial: User can push the Redial button to dial the last dialed number.

6. Flash: User can push the Flash button to make the IP Phone to dial mode.

7. Speaker Phone: You can use Speaker phone to make a phone call.

8. 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.

## 7 Get a FWD account

1. The website is 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