



VoIP Phone User Manual

V.03

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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
<ul style="list-style-type: none"> SIP v1 (RFC2543), v2(RFC3261) IP/TCP/UDP/RTP/RTCP IP/ICMP/ARP/RARP/SNTP TFTP Client/DHCP Client/ PPPoE Client Telnet/HTTP Server DNS Client 	<ul style="list-style-type: none"> Ring Tone Ring Back Tone Dial Tone Busy Tone User Programming Tone
	Phone Function
Codec <ul style="list-style-type: none"> G.711: 64k bit/s (PCM) G.723.1: 6.3k / 5.3k bit/s G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) G.729A: 8k bit/s (CS-ACELP) G.729B: adds VAD & CNG to G.729 	<ul style="list-style-type: none"> Volume Adjustment Speed dial, Phone book Flash Speaker Phone
	IP Assignment
	<ul style="list-style-type: none"> Static IP DHCP PPPoE
Voice Quality <ul style="list-style-type: none"> VAD: Voice activity detection CNG: Comfortable noise generator LEC: Line echo canceller Packet Loss Compensation Adaptive Jitter Buffer 	Security
	<ul style="list-style-type: none"> HTTP 1.1 basic/digest authentication for Web setup MD5 for SIP authentication (RFC2069/ RFC 2617)
	QoS



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Call Function	· ToS field
<ul style="list-style-type: none"> • Call Hold • Call Waiting • Call Forward • Caller ID · 3-way conference 	NAT Traversal
	· STUN
DTMF Function	Configuration
<ul style="list-style-type: none"> • In-Band DTMF • Out-of Band DTMF • SIP Info 	<ul style="list-style-type: none"> • Web Browser • Console/Telnet • Keypad
	Firmware Upgrade
SIP Server	
<ul style="list-style-type: none"> • Registrar Server (three SIP account) • Outbound Proxy 	<ul style="list-style-type: none"> • TFTP • Console • 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. 啟動/ 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 Speaker 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 伺服器啟動/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 伺服器啟動/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 伺服器啟動/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 啟動/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.

Login VoIP

Enter your username and password to login
VoIP server

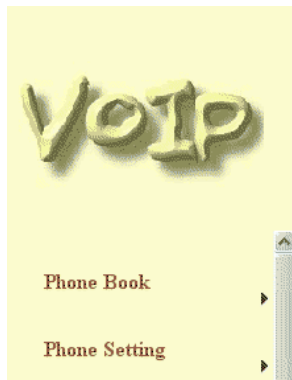
Username

Password

Remember last login

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.



System Information

This page illustrate the system related information.

Model Name:	VoIP
Firmware Version:	Wed Oct 12 17:08:27 2005.
Codec Version:	Fri Oct 14 17:07:38 2005.

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 "#".



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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.

Phone Book

You could add/delete items in current phone book.

Phone Book Page: page 10

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"/>

Delete Selected Delete All Reset

Add New Phone

Position: (0-139)
Name:
URL:

Add Phone Reset

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.

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"/>

Delete Selected Delete All Reset

Add New Phone

Position: (0-9)
Name:
URL:

Add Phone Reset

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.

Forward Setting

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

All Forward: Busy Forward: No Answer Forward:

	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"/>
Time Out:	20 (10-90 sec)	

- 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.

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)

- 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.

Volume Setting

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

Handset Volume:	<input type="text" value="10"/>	(0~12)
Ringer Volume:	<input type="text" value="10"/>	(0~10)
Handset Gain:	<input type="text" value="9"/>	(0~15)

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

Ringer Settings

You could set your favorite ringer in this page.

Ringer:	<input type="radio"/> On <input checked="" type="radio"/> Off
Ringer Type:	<input type="text" value="ringer 1"/>

3.1.14

3.1.15 Block Setting function: you can setup the Block Setting to keep the phone silence. 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.

Block Setting

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

Always Block:	<input type="radio"/> On <input checked="" type="radio"/> Off
Block Period:	<input type="radio"/> On <input checked="" type="radio"/> Off
From:	<input type="text" value="00"/> <input type="text" value="00"/> (hh:mm)
To:	<input type="text" value="00"/> <input type="text" value="00"/> (hh:mm)



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.

Auto Dial Setting

You could the time slice to auto dial in this page.

Auto Dial Time: (3-9 sec)

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.

Call Waiting Setting

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

Call Waiting: On 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.

Network Status

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

Interface 0	
Type:	Fixed IP Client
IP:	192.168.101.112
Mask:	255.255.255.0
Gateway:	192.168.101.1
DNS Server 1:	192.168.101.1
DNS Server 2:	168.95.1.1



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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.

Network Settings

You could configure your network settings in this page.

TCP/IP Configuration	
IP Type:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input type="radio"/> None
IP:	<input type="text" value="192.168.101.112"/>
Mask:	<input type="text" value="255.255.255.0"/>
Gateway:	<input type="text" value="192.168.101.1"/>
DNS Server 1:	<input type="text" value="192.168.101.1"/>
DNS Server 2:	<input type="text" value="166.95.1.1"/>
MAC:	<input type="text" value="200005100501"/>

PPPoE Configuration	
PPPoE:	<input type="radio"/> On <input checked="" type="radio"/> Off
User Name:	<input type="text"/>
Password:	<input type="text"/>

Bridge	
Bridge:	<input checked="" type="radio"/> On <input type="radio"/> Off

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.



DDNS Settings

You could set the configuration of DDNS in this page.

DDNS: On Off

Host Name:

User Name:

Password:

E-mail Address:

Type:

Wild Card:

BACKMK: On Off

Off Line: On Off

1.10 SIP Settings

3.1.22 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.



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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 directly.
- 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:	<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="text"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text"/>
Outbound Proxy:	<input type="text"/>
Register Period:	<input type="text" value="0"/> (0~99) [0: 30 sec, 1~99 min]
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="text"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text"/>
Outbound Proxy:	<input type="text"/>
Register Period:	<input type="text" value="15"/> (0~99) [0: 30 sec, 1~99 min]
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="text"/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text"/>
Outbound Proxy:	<input type="text"/>
Register Period:	<input type="text" value="15"/> (0~99) [0: 30 sec, 1~99 min]
Status:	Not Registered



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3.1.24 Port Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTP port 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: (1024-65535)
RTP Port: (1024-65535)

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.

Codec Settings

You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	<input type="text" value="G.711 u-law"/>
Codec Priority 2:	<input type="text" value="G.711 a-law"/>
Codec Priority 3:	<input type="text" value="G.729"/>
Codec Priority 4:	<input type="text" value="G.723"/>
Codec Priority 5:	<input type="text" value="G.726 - 16"/>
Codec Priority 6:	<input type="text" value="G.726 - 24"/>
Codec Priority 7:	<input type="text" value="G.726 - 32"/>
Codec Priority 8:	<input type="text" value="G.726 - 40"/>

RTP Packet Length	
G.711 & G.729:	<input type="text" value="20 ms"/>
G.723:	<input type="text" value="30 ms"/>

G.723 5.3K	
G.723 5.3K:	<input type="radio"/> On <input checked="" type="radio"/> Off

Voice VAD	
Voice VAD:	<input type="radio"/> On <input checked="" type="radio"/> Off

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.

Codec ID Setting

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

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

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

RTP Setting

You could set the RTP setting in this page.

Outband DTMF: On Off

Send DTMF SIP Info: On Off

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.

RPort Setting

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

RPort: On Off



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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.

Other Settings

You could set other settings in this page.

Hold by RFC:	<input type="radio"/> On <input checked="" type="radio"/> Off
Voice QoS:	<input type="text" value="40"/> (0-63)
SIP QoS:	<input type="text" value="40"/> (0-63)
SIP Expire Time:	<input type="text" value="60"/> (60-65400 sec)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

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.

STUN Setting

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

STUN:	<input type="radio"/> On <input checked="" type="radio"/> Off
STUN Server:	<input type="text" value="88.7.238.210"/>
STUN Port:	<input type="text" value="3478"/> (1024-65535)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

1.12 Others.

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



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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.

Auto Configuration Setting

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

Auto Configuration: On Off

TFTP Server:



1.13 System Auth.

3.1.34 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"/>

1.14 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.

Save Changes

You have to save changes to effect them.

Save Changes:

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 upgrade 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.

Restore Default Settings

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

Restore default settings:

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

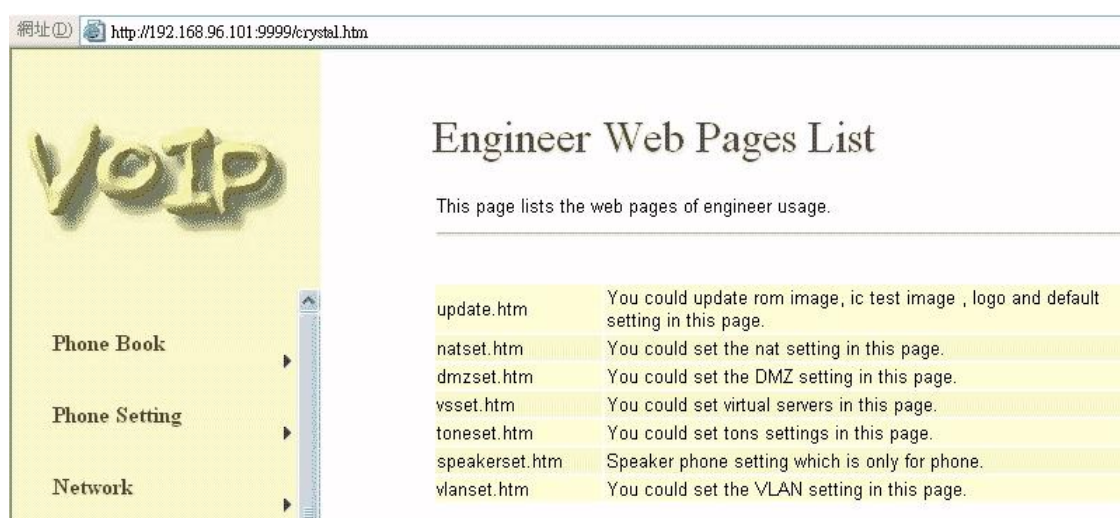
You could press the reboot button to restart the system.

Reboot system:

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.

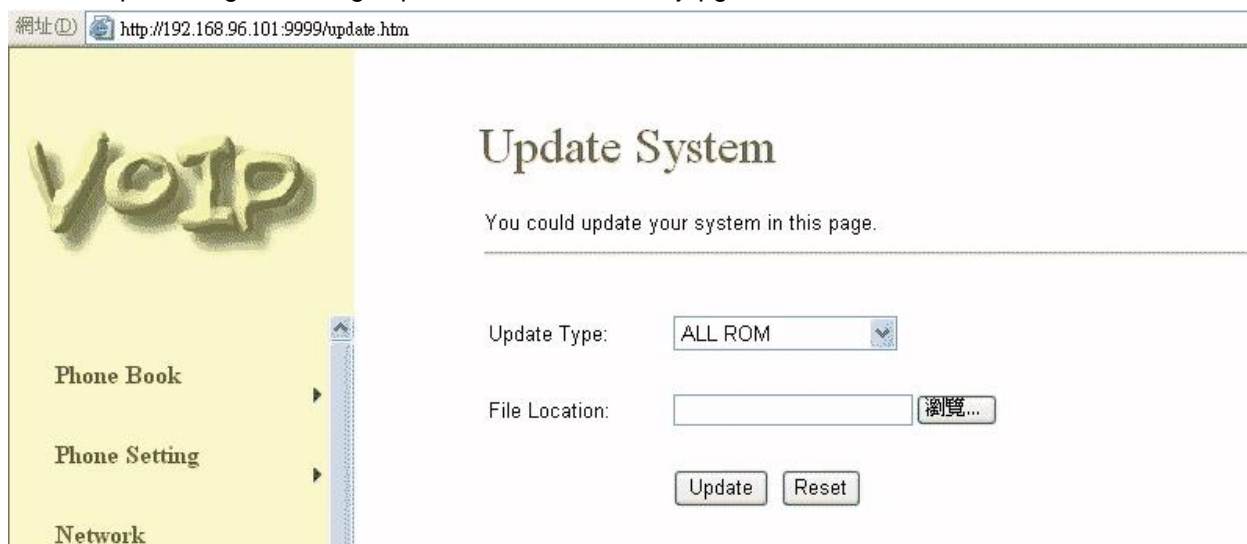


4.2 Update.htm

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 jpeg file.

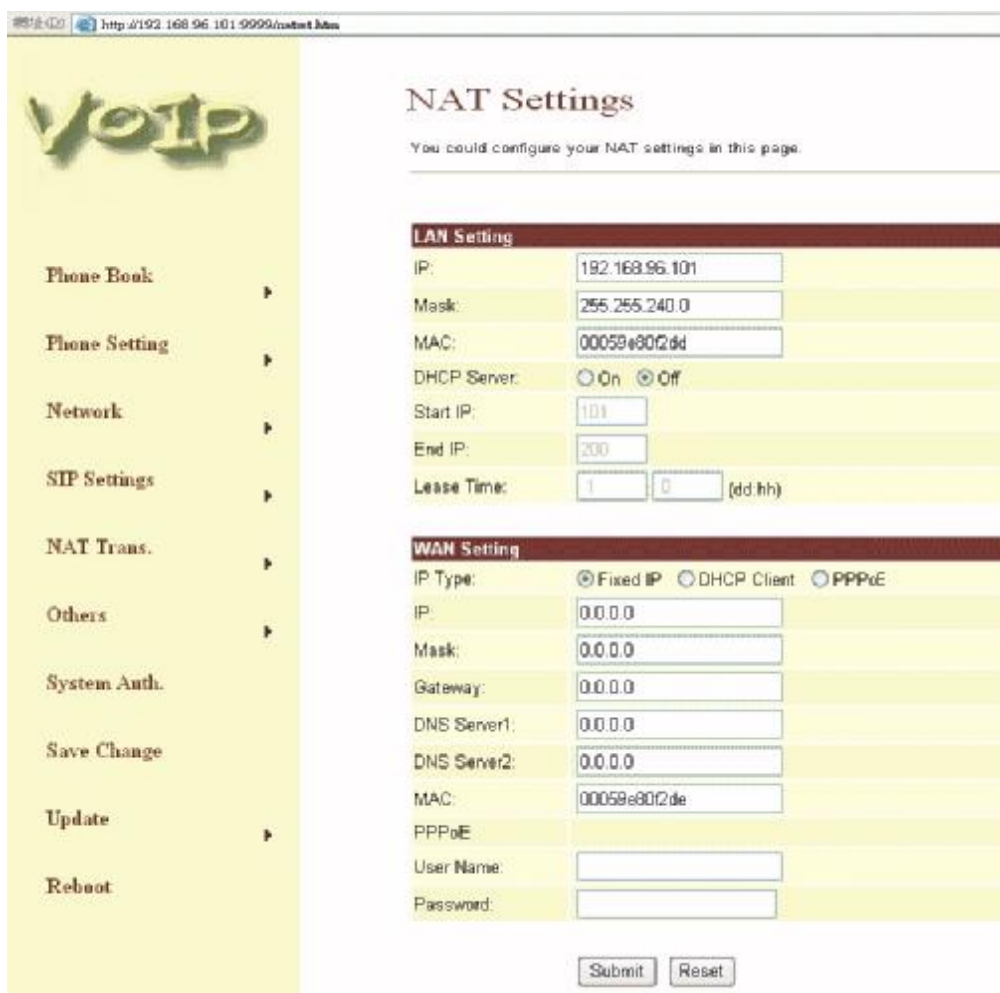




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4.3 Natset.htm

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.



The screenshot shows a web browser window with the URL <http://192.168.96.101:9999/natset.htm>. The page title is "NAT Settings" and it includes a sub-header "You could configure your NAT settings in this page." The interface is divided into two main sections: "LAN Setting" and "WAN Setting".

LAN Setting

IP:	192.168.96.101
Mask:	255.255.240.0
MAC:	00059e802dd
DHCP Server:	<input type="radio"/> On <input checked="" type="radio"/> Off
Start IP:	101
End IP:	200
Lease Time:	1 0 (dd:hh)

WAN Setting

IP Type:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input type="radio"/> PPPoE
IP:	0.0.0.0
Mask:	0.0.0.0
Gateway:	0.0.0.0
DNS Server1:	0.0.0.0
DNS Server2:	0.0.0.0
MAC:	00059e802de
PPPoE	
User Name:	
Password:	

At the bottom of the form, there are two buttons: "Submit" and "Reset".

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.



The screenshot shows the 'DMZ Setting' page in a web browser. The address bar shows 'http://192.168.06.101:9999/dmzset.htm'. On the left is a yellow sidebar with 'VoIP' and menu items: 'Phone Book', 'Phone Setting', and 'Network'. The main content area has the title 'DMZ Setting' and a sub-header 'You could configure your demilitarized zone setting in this page.' Below this, there are two radio buttons for 'DMZ: On' and 'Off', with 'Off' selected. A text input field for 'DMZ Host IP:' contains '0.0.0.0'. At the bottom are 'Submit' and 'Reset' buttons.

4.5 Vsset.htm

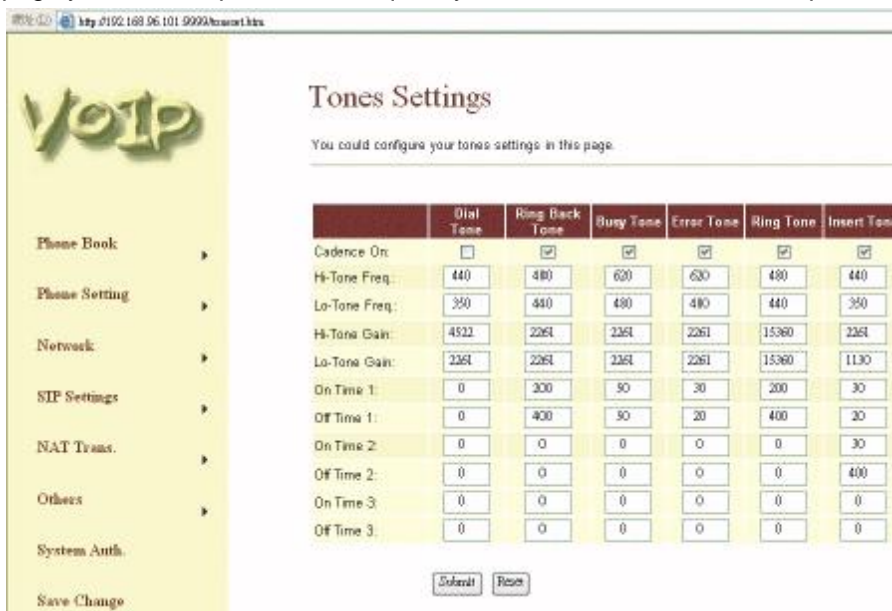
4.5.1 In this page you can setup the Virture Server function.



The screenshot shows the 'Virtual Server Settings' page in a web browser. The address bar shows 'http://192.168.06.101:9999/vsset.htm'. The left sidebar is yellow with 'VoIP' and menu items: 'Phone Book', 'Phone Setting', 'Network', 'SIP Settings', 'NAT Trans.', 'Others', 'System Auth.', 'Save Change', 'Update', and 'Reboot'. The main content area has the title 'Virtual Server Settings' and a sub-header 'You could set your virtual servers in this page. The usual port numbers are: WEB [TCP 80], FTP (Control) [TCP 21], FTP (Data) [TCP 20], E-mail (POP3) [TCP 110], E-mail (SMTP) [TCP 25], DNS (UDP 53) and Telnet [TCP 23]'. Below this is a 'Virtual Server Page:' dropdown set to 'page 1'. A table lists 8 virtual servers (Num 0-7) with columns for 'Enable', 'Protocol', 'In Port', 'Ex Port', 'Server IP', and 'Select'. Below the table are buttons for 'Enable Selected', 'Delete Selected', 'Delete All', and 'Reset'. At the bottom is the 'Add Virtual Server' form with fields for 'Num:' (0-23), 'Server IP:', 'Protocol' (dropdown set to 'TCP'), 'Internal Port:', and 'External Port:', with 'Add Server' and 'Reset' buttons.

4.6 Toneset.htm

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



The screenshot shows the 'Tones Settings' page in a web browser. The page title is 'Tones Settings' and it includes a sub-header: 'You could configure your tones settings in this page.' Below this is a table with columns for 'Dial Tone', 'Ring Back Tone', 'Busy Tone', 'Error Tone', 'Ring Tone', and 'Insert Tone'. Each column has a corresponding 'Cadence On' checkbox and numerical input fields for 'Hi-Tone Freq.', 'Lo-Tone Freq.', 'Hi-Tone Gain', and 'Lo-Tone Gain'. There are also input fields for 'On Time' and 'Off Time' for each tone type. At the bottom of the table are 'Submit' and 'Reset' buttons.

	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
Cadence On:	<input type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>
Hi-Tone Freq.:	440	480	600	600	480	440
Lo-Tone Freq.:	300	440	480	480	440	300
Hi-Tone Gain:	4522	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	90	30	200	30
Off Time 1:	0	400	90	20	400	20
On Time 2:	0	0	0	0	0	30
Off Time 2:	0	0	0	0	0	400
On Time 3:	0	0	0	0	0	0
Off Time 3:	0	0	0	0	0	0

4.7 Speakerset.htm

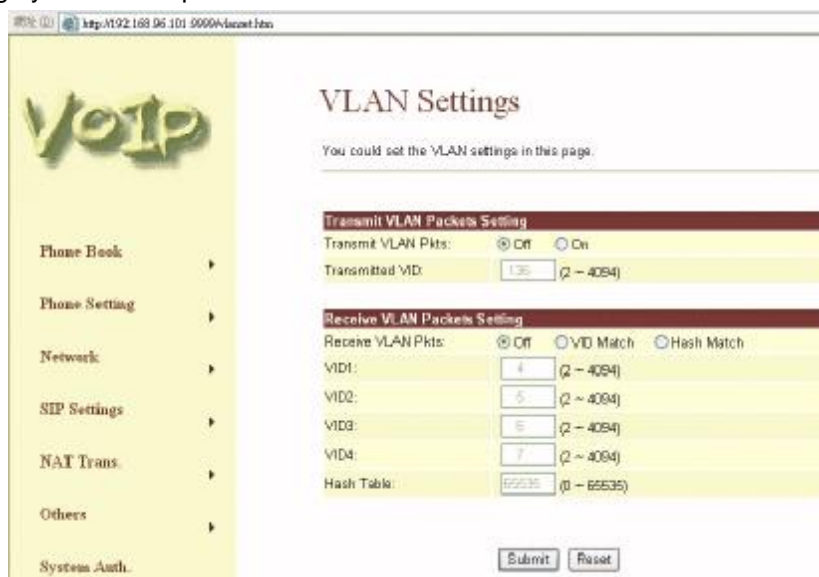
4.7.1 In this page you can setup the Speaker function.



The screenshot shows the 'Speaker Phone Setting' page in a web browser. The page title is 'Speaker Phone Setting' and it includes a sub-header: 'You could set the speaker phone in this page.' Below this are radio buttons for 'Half Duplex' (selected) and 'Full Duplex'. There are three input fields: 'Cut-off Threshold' (0010), 'Cut-off Time Constant' (4000), and 'Cut-off Hold Time' (0014). At the bottom are 'Submit' and 'Reset' buttons.

4.8 Vlanset.htm

4.8.1 In this page you can setup the VLAN function.



The screenshot shows a web browser window with the address <http://192.168.96.101:8080/Vlanset.htm>. The page title is "VLAN Settings". Below the title, it says "You could set the 'VLAN settings in this page." The interface is divided into two main sections: "Transmit VLAN Packets Setting" and "Receive VLAN Packets Setting".

Transmit VLAN Packets Setting

Transmit VLAN Pkts:	<input checked="" type="radio"/> Off	<input type="radio"/> On
Transmitted VID:	<input type="text" value="136"/>	(2 ~ 4094)

Receive VLAN Packets Setting

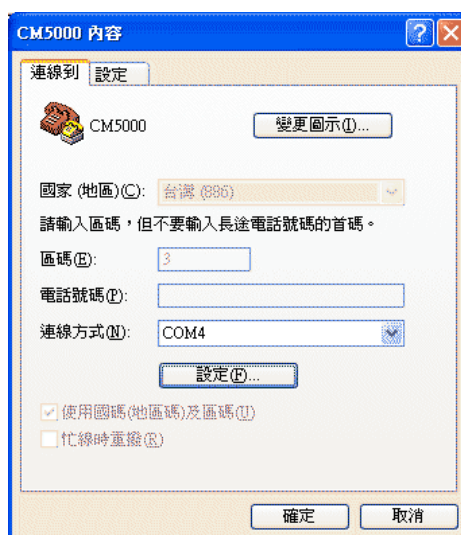
Receive VLAN Pkts:	<input checked="" type="radio"/> Off	<input type="radio"/> VID Match	<input type="radio"/> Hash Match
VID1:	<input type="text" value="4"/>	(2 ~ 4094)	
VID2:	<input type="text" value="5"/>	(2 ~ 4094)	
VID3:	<input type="text" value="6"/>	(2 ~ 4094)	
VID4:	<input type="text" value="7"/>	(2 ~ 4094)	
Hash Table:	<input type="text" value="FFFF"/>	(0 ~ 65535)	

At the bottom of the form, there are two buttons: "Submit" and "Reset".

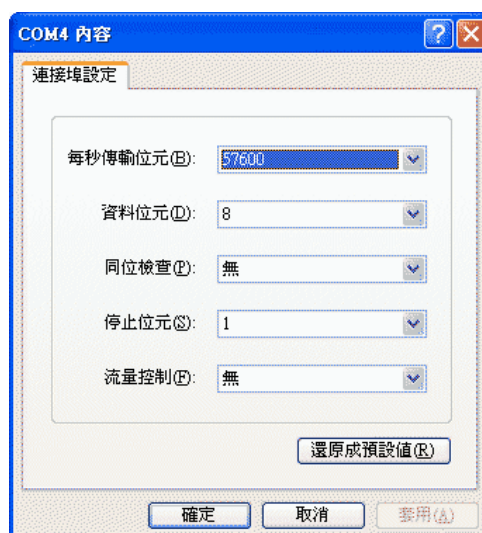
5 Setup the VoIP Phone by using Console (Hyper Terminal)

1.17 Configure the COM port

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

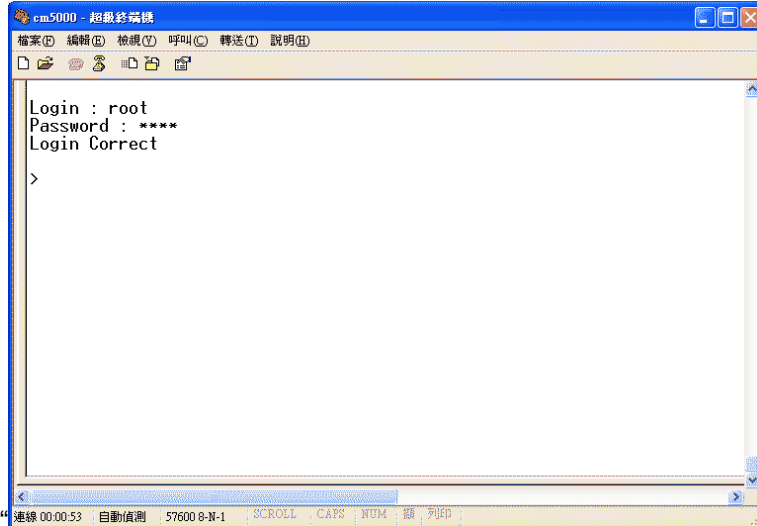


Set the COM port's setting as following setting. Then click OK.



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.



```
cm5000 - 超級終端機
檔案(F) 編輯(E) 檢視(V) 呼叫(C) 轉送(T) 說明(H)
Login : root
Password : ****
Login Correct
>
```



1.19 Using CLI command to configure the VoIP Phone

5.1.1 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

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

5.1.1.2 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

5.1.1.3 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

5.1.1.3.1 ipconfig -ifN function à 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

5.1.1.4 save function

ltno	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

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	-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	-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

5.1.1.14 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

5.1.1.15 sntp function



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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



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



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7	(null)	Show Phone Setting
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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.



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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