



IP101 VoIP Phone

User Manual

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Telephone function:

- 1、 Protocol support: SIP (RFC2543, RFC3261) ;
- 2、 Display and store 64 groups of incoming calls;
- 3、 Display and store 64 groups of outgoing calls;
- 4、 Phone Book with 140 groups memory
- 5、 Hand-free dialing functions, count time of calling automatically;
- 6、 Review, delete, redial Incoming calls/outgoing calls;
- 7、 Do Not Disturb function;
- 8、 Roaming function;
- 9、 Check local number;
- 10、 Pre-dial function;
- 11、 Display date and time;
- 12、 Music on Hold;
- 13、 PPPoE;
- 14、 Dynamic and static IP address;
- 15、 Support voice codec: G.711A/U, G.723.1, G.729A/B, G.726; VAD/CNG
- 16、 User authentication;
- 17、 TCP/IP(ARP/RARP 、 IP/ICMP、 UDP/TCP/IP、 RTP/RTCP);
- 18、 TFTP and Console function;
- 19、 IEEE 802.1P/802.1Q/ToS u 10 Base T/100Base TX;
- 20、 DNS;
- 21、 Call Transfer function;
- 22、 Three-way conversation;
- 23、 Call waiting function;
- 24、 Call forward function;
- 25、 DDNS function
- 26、 Remote software updates/upgrades;
- 27、 Hand-free;

Main technology parameter:

1. AC/DC adaptor-----Input: AC 240/100V;Output: 9V DC, 1000mA
2. Support DHCP client and static IP address
3. Support PPPoE (for ADSL, Cable Modem)
4. Upgradeable software
5. Support voice codec: G.711A/U, G.723.1, G.729A/B, G.726
6. Voice activity detection, Comfortable noise generator, Adaptive jitter buffer
7. G.165 16ms echo cancellation
8. Signaling tone of ITU-T standard, DTMF
9. Support dialing standard base on E.164 code

1 Introduction

This user manual is for IP101 IP phone. Before using the IP Phone, some configurations are required to make the IP Phone work properly. The manual will illustrate how to configure the ip phone via keypad, web page and console.

1.1 Hardware Overview

Two RJ-45 Networking interfaces support 10/100Mbps Fast Ethernet. User can connect one of them to ADSL or Switch, the other one to PC computer. By default, WAN is a DHCP client.

LAN's ip is 192.168.123.1 and can provide DHCP service.

Passed FCC, CE authentication

Wide-range power adaptor (From 100V to 240V)

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 Programming Tone
Codec	Phone Function
<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 key Phonebook Flash Hand-free function
Voice Quality	IP Assignment
<ul style="list-style-type: none"> VAD: Voice activity detection CNG: Comfortable noise generator LEC: Line echo canceller Packet Loss Compensation Adaptive Jitter Buffer 	<ul style="list-style-type: none"> IP (Static IP) DHCP PPPoE
Call Function	Security
<ul style="list-style-type: none"> Call Hold Call Waiting Call Forward Caller ID 3-way conference 	<ul style="list-style-type: none"> HTTP 1.1 basic/digest authentication for Web setup MD5 for SIP authentication (RFC2069/ RFC 2617)
DTMF	QoS
<ul style="list-style-type: none"> In-Band DTMF RFC 2833 DTMF SIP Info 	<ul style="list-style-type: none"> ToS field
SIP Server	NAT Traversal
<ul style="list-style-type: none"> Registrar Server (three SIP account) Outbound Proxy 	<ul style="list-style-type: none"> STUN
	Configuration
	<ul style="list-style-type: none"> Web Browser Console/Telnet Keypad
	Firmware Upgrade
	<ul style="list-style-type: none"> TFTP Console HTTP FTP

2 Setup the IP Phone system by using keypad

2.1 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	
PHONEBOOK	Phonebook	
CID	Incoming call list	
Out	Out going call list	
ENTER	“OK”, accept setting	
MENU	“Menu” key to set the IP Phone	
UP/DOWN	Up↑ and Down↓ key	
LEFT/RIGHT	Left← and Right→ key	
Volume +/-	Volume control	
FLASH	Flash key	
DEL	“Delete”, delete word or phone number	
REDIAL	“REDIAL” the same number again	
H-F	Hand Free	
M1~M4	4 speed dial numbers	
Line1~Line2	the Line1 to Line2	
HOLD	“HOLD” function	
CONF	Three way conference	
FWD	Forward key	
DND	Do Not Disturb	

2.1.1 Outgoing calls

Press “OUT” button, LCD will display the outgoing. Press “UP” or “DOWN” to review all outgoing call numbers and conversation time.

2.1.2 Caller ID function

Press “CID” key, LCD will show incoming call numbers and conversation time, using “UP” or “DOWN” to review all incoming calls number.

2.1.3 Talking volume control

During conversation, press “VOL+” or “VOL-” to adjust volume.

2.1.4 Phone-Book function (Total 140 groups of numbers)

- 1) Press “PHONBOOK” key to enter into display mode, LCD shows the total groups you have stored. If nothing stored, LCD will display “P-Book is empty”.
- 2) In display mode, input the name of your stored number, press “ENTER”, LCD will display the number. Then press “H-F” or pick up the handset, the present number will be dialed out automatically.
- 3) In display mode, press “ENTER” button to enter into name check status, using “UP” or “DOWN” to review.
- 4) In display mode, press “ENTER” button to enter into name check status, press “ENTER” key again to enter into modify mode. Use keypad to modify name. Press “ENTER” one more time to modify call numbers. After finishing, press “ENTER” to confirm.
- 5) In display mode, press “ENTER” key to enter into name check status, using “UP” or “DOWN” to review. At this time, press “DEL” to delete the present call number.

2.1.5 DND function

Press “DND” button, LCD will display “#DND#”. The ip phone will refuse all of incoming calls. Press “DND” again to quit DND.

2.1.6 Hold with music

- 1) When you are receiving a call on LINE1, there is an incoming call from LINE2. Now press “HOLD” button to hold on the line1, and press “Line2” to start conversation with LINE2.
- 2) Press “Line1” to quit holding status.

2.1.7 Call forward

After setting forward number, press FWD key to enter into forward mode. LCD display “# Forward #”, press FWD key again to quit. The forward number must be a speed dial number.

2.1.8 Three-way conference

During conversation on Line1, press “HOLD”, and then press LINE2 to dial another number. After connecting though with Line2, press “CONF” to start three-way conference.

2.1.9 Speed dial function

After setting speed dial number, the M1~M4 button stands for 0-3 speed dial number. Press “0~9” and “#” button to dial out the corresponding speed dial number.

2.1.10 Transfer between LINE1 and LINE2

- 1) When you are receiving a call, there is another incoming call. The LCD will show you which line the incoming call comes from. Press HOLD and LINE1 or LINE2 (the line that the latter incoming call comes from), then you can answer the latter call. Press LINE2 or LINE1, which the former call is on, you will return to the former call.
- 2) When you are receiving a call, there is another incoming call. The LCD will show you which line the incoming call comes from. Press HOLD and LINE1 or LINE2 (the line that the latter incoming call comes from), then you can answer the latter call. Press CONF to start three-way conference.

2.2 Keypad Function and Setting List

A. Press “MENU” button to enter into setting list. Press “UP” or “DOWN” to choose items, then press “ENTER” to enter into submenu of the item. Move cursor through “LEFT/RIGHT” and modify content by keypad. After finishing setting, press ENTER to confirm and return to the

parent menu. Press MENU to return to parent menu without confirm.

B、 Menu list:

- 1、 Phone Book
- 2、 Call history
- 3、 Phone setting
- 4、 Network
- 5、 SIP Settings
- 6、 NAT transversal
- 7、 Administrate

C、 All configuration will be effective after reboot.

2.2.1 Phone Book

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

2.2.2 Call history

- 1) Incoming calls: Show all incoming call.
- 2) Dialed numbers: Show all dialed call.
- 3) Erase record: Delete call history.
 - All: Delete all call history.
 - Incoming: Delete all incoming call.
 - Dialed: Delete all dialed call.

2.2.3 Phone setting

2.2.3.1 Call forward

2.2.3.1.1 All Forward.

1. Activation
2. Number:

2.2.3.1.2 Busy Forward.

1. Activation
2. Number:

2.2.3.1.3 No Answer Forward.

1. Activation
2. Number

2.2.3.1.4 Ring Timeout: set call transfer time. For example, if you input 3 here, incoming call will be forwarded after 3 rings.

2.2.3.2 Do Not Disturb Setting

- 1) Always
- 2) By period
- 3) Period time

2.2.3.3 Alarm setting

- 1) Activation
- 2) Alarm time

- 2.2.3.4 Date/Time setting
 - 2.2.3.4.1 Date & Time: Set the IP Phone Date and Time.
 - 2.2.3.4.2 SNTP
 - 1) SNTP: Enabled/Disable SNTP.
 - 2) Primary SNTP
 - 3) Secondary SNTP
 - 4) Time zone
 - 5) Adjustment Time
- 2.2.3.5 Volume&Gain
 - 1) Handset volume: Set Handset volume from 0 to 15.
 - 2) Speaker volume: Set Speaker phone volume from 0 to 15.
 - 3) Handset Gain Set handset gain from 0 to 15.
 - 4) Speaker Gain: Set speaker gain from 0 to 15.
- 2.2.3.6 Ringer
 - 1) Ringer volume: from 0 to 10.
 - 2) Ringer type: Ringer type selection from 1 to 4.
- 2.2.3.7 Auto Dial (3sec—9sec)
 - After auto dial time, the number will be sent out without dialing #. Default is 5 seconds.
- 2.2.4 Network
 - 2.2.4.1 WAN Setup
 - 2.2.4.1.1 IP Type
 - 1) Fixed IP
 - 2) DHCP client
 - 3) PPPoE client
 - 2.2.4.1.2 Fixed IP setting
 - 1) Host IP
 - 2) Network mask:
 - 3) Gateway IP
 - 4) MAC address
 - 2.2.4.1.3 PPPoE
 - 1) User name
 - 2) Password
 - 2.2.4.2 LAN Setup
 - 1) Bridge
 - 2) NAT
 - 2.2.4.3 DNS
 - 1) Primary DNS
 - 2) Secondary DNS
 - 2.2.4.4 VLAN
 - 1) Activation: enable/disable VLAN
 - 2) VID: 2~4094
 - 3) Priority: 0~7

4) CFI: 0 or 1

2.2.4.5 Status: show the ip and mac addresses of the ip phone.

2.2.5 SIP Settings

2.2.5.1 Service domain

2.2.5.1.1 First realm:

- 1) Activation
- 2) User name
- 3) Display name
- 4) Register name
- 5) Register password
- 6) Proxy server:
- 7) Domain server
- 8) Outbound proxy

2.2.5.1.2 Second realm:

- 1) Activation
- 2) User name
- 3) Display name
- 4) Register name
- 5) Register password
- 6) Proxy server
- 7) Domain server
- 8) Outbound proxy

2.2.5.1.3 Third realm:

- 1) Activation:
- 2) User name
- 3) Display name
- 4) Register name
- 5) Register password
- 6) Proxy server
- 7) Domain server
- 8) Outbound proxy

2.2.5.2 Codec

2.2.5.2.1 Codec type:

- 1) G.711 uLaw
- 2) G.711 aLaw
- 3) G.723
- 4) G.729
- 5) G.726-16
- 6) G.726-24
- 7) G.726-32
- 8) G.726-40

2.2.5.2.2 VAD: Voice Activity Detection

- 2.2.5.3 RTP setting
 - 2.2.5.3.1 Outband DTMF
 - 2.2.5.3.2 Duplicate RTP
 - 1) No duplicate
 - 2) One duplicate
 - 3) Two duplicate
- 2.2.5.4 RPort Setting: Enable/Disable RPort
- 2.2.5.5 Hold by RFC
- 2.2.5.6 Status:
 - 1) First Realm
 - 2) Second Realm
 - 3) Third Realm
- 2.2.6 NAT Transversal
 - 2.2.6.1 STUN setting
 - 1) STUN: enable/disable STUN
 - 2) STUN server
- 2.2.7 Administrate
 - 2.2.7.1 Auto Config:
 - 1) Config mode
 - 2) TFTP server
 - 3) FTP server
 - 4) FTP Login Name
 - 5) FTP Password
 - 2.2.7.2 Upgrade system
 - 1) Upgrade Now
 - 2) Status
 - 3) Reset time
 - 2.2.7.3 Default setting:
 - 1) Load default:
 - 2) Abort
 - 2.2.7.4 System Authentication: if you want to configure SIP settings by keypad, you have to input password here.
 - 2.2.7.5 Version
 - 2.2.7.6 Watch dog: enable/disable Watch dog.
 - 2.2.7.7 Restart

3 Setup the IP Phone system by using Web Browser

You can use Web browser to configure the IP Phone. First input the IP address in the Web browser. In the end of IP address, please add the port number “:9999”. Ex: <http://192.168.123.1:9999>. By default, NAT is on, LAN’s IP address is 192.168.123.1, WAN is DHCP client. In order to configure the IP Phone, your PC must be in the same subnet as the IP Phone. Go to NETWORK menu to check the WAN’s IP.

3.1 Login

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: user; and the password is: test. If you use the account login, you cannot configure the SIP setting.

Click the “Login” button to go to configuration web page.

Note: If you change the setting in a web page, please do remember to click the “Submit” button in that page. After finishing setting, click the “Save change” function in the left side, and then click the Save Button. The IP Phone will reboot automatically, and the new configuration will be effective after reboot.

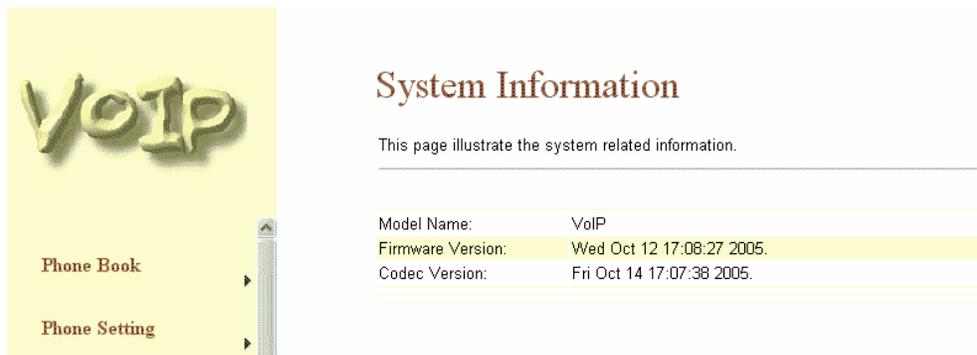


The image shows a login form titled "Login VoIP". It has a red header bar with the text "Login VoIP". Below the header, there is a yellow background area with the text "Enter your username and password to login VoIP server". There are two input fields: "Username" and "Password". Below the input fields, there are two buttons: "Login" and "Clear". At the bottom, there is a checkbox labeled "Remember last login".

3.2 System Information

After login the web page, you can see the system information like firmware version, model name and so on in this page.

In addition, you can see the function list in the left side. You can use mouse to click the function you want to set up.



The image shows a screenshot of the "System Information" page. On the left side, there is a vertical menu with the text "Phone Book" and "Phone Setting". The main content area has the title "System Information" and a subtitle "This page illustrate the system related information." Below this, there is a table with the following information:

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

3.3 Phone Book

In Phone Book function, you can setup the Phone Book and Speed Dial setting. The phone book can store up to 140 groups of number. The speed dial can have 10 entries. Press any one of 0~9, the corresponding number will dial out.

- 3.3.1 In phone book, you can add/delete phone number.
- 3.3.1.1 If you want to add phone number in Phone Book, you need input position, name, URL (ie. Phone number). After input, please click “Add Phone” button.

Phone Book

You could add/delete items in current phone book.

Phone Book Page:

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

Add New Phone

Position: (0~139)

Name:

URL:

- 3.3.1.2 To delete a group of number, first select the number, and then click “Delete Selected” button to delete selected number.
- 3.3.1.3 To delete all number, click “Delete All” button, a dialogue window will show up. Click OK button to delete all numbers.



- 3.3.2 In Speed Dial setting function, you can add/delete Speed Dial number. You can input maximum 10 entries into speed dial list.
- 3.3.2.1 To add a phone number into the Speed Dial list, you need input position, name, and URL. After finishing input, click “Add Phone” button.

Speed Dial Phone List

You could set the speed dial phones in this page.

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

Add New Phone

Position: (0-9)

Name:

URL:

- 3.3.2.2 To delete a group of number, first select the number, and then click “Delete Selected” button to delete selected number.
- 3.3.2.3 To delete all number, click “Delete All” button, a dialogue window will show up. Click OK button to delete all numbers.



3.4 Phone Setting

Phone Setting contains Call Forward, SNTP Settings, Volume Settings, Ringer Setting, DND Setting, Dial Plan, Call Waiting.

- 3.4.1 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 between All Forward, Busy Forward, and No Answer Forward.
- 3.4.1.1 All Forward: All incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.
- 3.4.1.2 Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed.
- 3.4.1.3 No Answer Forward: If you can not answer the phone, the incoming call will forward to the number you choosed. Also you have to set the Time Out time for system to start forwarding the call to the number you choosed.
- After finishing the setting, please click the Submit button.

Forward Setting

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

All Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> On
Busy Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> On
No Answer Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> On

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

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

- 3.4.2 SNTP settings: you can set primary and secondary SNTP server to get date/time information. You also need set Time Zone and Sync Time. After finishing 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)

- 3.4.3 Volume Setting: you can setup Handset Volume, Speaker Volume, Ringer Volume Handset Gain and Speaker Gain. When you finished the setting, please click the Submit button.

- 1) Handset Volume: (0—15)
- 2) Speaker Volume: (0—15)
- 3) Ringer Volume: (0—10)
- 4) Handset Gain: (0—15)
- 5) Speaker Gain: (0—15)

Volume Setting

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

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

- 3.4.4 Ringer setting: you can select a ringer for incoming call. (total 4 types). When you finished the setting, please click the Submit button.

Ringer Settings

You could set your favorite ringer in this page.

Ringer: On Off

Ringer Type:

- 3.4.5 DND Setting function: you can setup the DND Setting to keep the phone silence. You can choose DND Always or DND period.
- 3.4.5.1 DND Always: All incoming call will be blocked until disable this feature.
- 3.4.5.2 DND Period: Set a time period and the phone will be blocked during the time period. If the “From” time is larger than the “To” time, the “To” time will be regarded as next day’s time.

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

DND Setting

You could set the do not disturb period of your phone in this page.

DND Always:	<input type="radio"/> On <input checked="" type="radio"/> Off
DND 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)

3.4.6 Dial Plan: This function provides basic dial number replacement rule.

Dial Plan

You could the set the dial plan in this page.

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 2:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 3:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 4:	<input type="text"/> + <input type="text"/>
Auto Dial Time:	<input type="text" value="5"/> (3~9 sec)

Drop prefix: If No, when the number you dialed begins with the number in right blank, the number in left blank will be added automatically before your dialed number. If Yes, when the number you dialed begins with the number in right blank, the number will be replaced with the number in left blank.

Auto Dial Time: the number you dialed will be sent out automatically after the time you set here.

Dial Plan

You could the set the dial plan in this page.

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	<input type="text" value="002"/> + <input type="text" value="8613+8662"/>
Drop prefix :	<input checked="" type="radio"/> Yes <input type="radio"/> No
Replace rule 2:	<input type="text" value="006"/> + <input type="text" value="002+003+004+005+007+009"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 3:	<input type="text" value="009"/> + <input type="text" value="12"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 4:	<input type="text" value="007"/> + <input type="text" value="5xxx+35xx+21xx"/>
Auto Dial Time:	<input type="text" value="5"/> (3~9 sec)

Example 1: Drop prefix: No; Replace rule 1: 002, 8613+8662

When your dialed number begins with 8613 or 8662, 002 will be added to the beginning of your dialed number.

Example 2: Drop prefix: Yes, Replace rule 2: 006, 002+003+004+005+007+009

When your dialed number begins with 002 or 003, 004, 005, 007, 009, 002 or 003, 004, 005, 007, 009 will be replaced with 006.

Example 3: Drop prefix: No; Replace rule 4: 007, 5xxx+35xx+21xx

When your dialed number begins with 5 or 35, 21, and is 4 digits, 007 will be added to

the beginning of your dialed number.

- 3.4.7 Call Waiting Setting function: You can Enable/Disable the Call Waiting function, When you are talking with someone, there is a new incoming call, you will hear the call waiting tone.

Call Waiting Setting

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

Call Waiting: On Off

- 3.4.8 Soft-key Setting: you can set Pick up key and Voice mail key here.

Soft-key Setting

You could configure the soft-key setting in this page.

Pick up key:

Voice mail key:

- 3.4.9 Hot line setting

Hot line Setting

You could set the hot line in this page.

Use Hot Line : Enable Disable

Hot line number:

If enabled, when you lift handset, the hot line number will be dialed out automatically.

3.4.10 Alarm setting

Alarm Settings

You could set the alarm time in this page.

Alarm: ON OFF

Alarm Time: : (hh:mm)

Current time: 2006-10-05 17:47

If On, the phone will alarm at alarm time.

3.5 Network

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

3.5.1 Network Status: in this page you can check the current network setting.

Network Status

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

Interface 0	
Type:	DHCP Server
IP:	192.168.123.1
Mask:	255.255.255.0
Gateway:	192.168.2.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

Interface 1	
Type:	DHCP Client
IP:	192.168.0.121
Mask:	255.255.255.0
Gateway:	192.168.0.1
DNS Server 1:	202.96.134.133
DNS Server 2:	202.96.128.68

3.5.2 WAN Settings

3.5.2.1 Bridge settings: Enable/disable bridge mode. If you set the Bridge On, then the two Fast Ethernet ports will be transparent

3.5.2.2 WAN Settings: Set network parameters for WAN. WAN can obtain IP address through 3 methods: fixed IP, DHCP client, PPPoE. You may refer to your current network environment to configure the IP Phone properly.

3.5.2.3 The PPPoE Setting: Set PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.

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

WAN Settings

You could configure the WAN settings in this page.

LAN Mode: Bridge NAT

WAN Setting

IP Type: Fixed IP DHCP Client PPPoE

IP:

Mask:

Gateway:

DNS Server1:

DNS Server2:

MAC:

PPPoE Setting

User Name:

Password:

- 3.5.3 LAN Settings: Set network parameters for LAN. You may refer to your current network environment to configure the IP Phone properly. When you finished the setting, please click the Submit button.

LAN Settings

You could configure the LAN settings in this page.

LAN Setting

IP:

Mask:

MAC:

DHCP Server

DHCP Server: On Off

Start IP:

End IP:

Lease Time: : (dd:hh)

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

BACKMX: On Off

Off Line: On Off

- 3.5.5 VLAN Settings

VLAN Settings

You could set the VLAN settings in this page.

VLAN Packets: On Off

VID: (2 ~ 4094)

User Priority: (0 ~ 7)

CFI: (0 ~ 1)

NAT VLAN Setting

VLAN Packets: On Off

VID1: (2 ~ 4094), 0->Off

VID2: (2 ~ 4094), 0->Off

VID3: (2 ~ 4094), 0->Off

VID4: (2 ~ 4094), 0->Off

- 3.5.6 DMZ setting

DMZ Setting

You could configure your demilitarized zone setting in this page.

DMZ: On Off

DMZ Host IP:

3.5.7 Virtual Server

Virtual Server Settings

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].

Virtual Server Page: page 1

Num	Enable	Protocol	In Port	Ex Port	Server IP	Select
0	<input type="checkbox"/>					<input type="checkbox"/>
1	<input type="checkbox"/>					<input type="checkbox"/>
2	<input type="checkbox"/>					<input type="checkbox"/>
3	<input type="checkbox"/>					<input type="checkbox"/>
4	<input type="checkbox"/>					<input type="checkbox"/>
5	<input type="checkbox"/>					<input type="checkbox"/>
6	<input type="checkbox"/>					<input type="checkbox"/>
7	<input type="checkbox"/>					<input type="checkbox"/>

Add Virtual Server

Num: (0~23)
 Server IP:
 Protocol:
 Internal Port: External Port:

3.6 SIP Settings

In SIP Settings you can setup the Service Domain, Port Settings, Codec Settings, Codec ID Setting, DTMF Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you have to setup the related information correctly so that you can register to the SIP Proxy Server successfully.

3.6.1 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 accounts in the IP Phone. You can dial the VoIP phone to your friends via first enabled SIP account and receive the phone from these three SIP accounts.

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

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

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

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

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

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

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

Outbound Proxy: you need to input the Outbound Proxy got from your ISP. If your ISP does not provide the information, you can skip this item.

Subscribe for MWI: When set to “On” a Subscribe for Message Waiting Indication will

be sent periodically.

Status: You can see the Register Status in the Status item. If the item shows “Registered”, then your IP Phone register to the ISP, you can make a phone call now.

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

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

Service Domain Settings

You could set information of service domains in this page.

Realm 1 (Default)	
Active:	<input 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"/>
Subscribe for MWI:	<input type="radio"/> On <input checked="" type="radio"/> Off
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"/>
Subscribe for MWI:	<input type="radio"/> On <input checked="" type="radio"/> Off
Status:	Not Registered

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

Port Settings

You could set the port number in this page.

SIP Port:	<input type="text" value="5060"/>	(1024~65535)
RTP Port:	<input type="text" value="60000"/>	(1024~65535)

- 3.6.3 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:	G.711 u-law
Codec Priority 2:	G.711 a-law
Codec Priority 3:	G.729
Codec Priority 4:	G.723
Codec Priority 5:	G.726 - 16
Codec Priority 6:	G.726 - 24
Codec Priority 7:	G.726 - 32
Codec Priority 8:	G.726 - 40

RTP Packet Length	
G.711 & G.729:	20 ms
G.723:	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

- 3.6.4 Codec ID Setting: Sometimes 2 VoIP devices with different Codec ID will cause some problems. If there are some problems when you are talking with others, you may ask the other one what Codec ID he is using, and change your Codec ID to the same as his. 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

- 3.6.5 DTMF Setting: Choose between 2833, Inband DTMF and Send DTMF SIP Info. Please refer to your ISP for correct setting. When you finished the setting, please click the Submit button.

DTMF Setting

You could set the DTMF setting in this page.

2833
 Inband DTMF
 Send DTMF SIP Info

- 3.6.6 RPort Setting: Enable/Disable RPort. To change this setting, please following your ISP's 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

- 3.6.7 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's information. The QoS setting is to set the voice packets' priority. If you set the value higher, the voice packets will get higher priority. But the QoS function still need to cooperate with the others Internet devices. When you finished the setting, please click the Submit button.

Other Settings

You could set other settings in this page.

Hold by RFC: On Off

Voice QoS: (0~63)

SIP QoS: (0~63)

SIP Expire Time: (60~86400 sec)

3.7 NAT Trans

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

STUN Setting

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

STUN: On Off

STUN Server:

STUN Port: (1024~65535)

3.8 Other

This page is for “Auto Config” and “ICMP Setting” configuration.

- 3.8.1 Auto Config: Please set these according to your need. When you finished the setting, please click the Submit button.

Auto Configuration Setting

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

Auto Configuration: Off TFTP FTP HTTP

TFTP Server:

HTTP Server:

HTTP Path:

FTP Server:

FTP Username:

FTP Password:

File Path:

- 3.8.2 MAC Clone Setting: Enable/disable MAC Clone.

MAC Clone Setting

You could enable/disable the MAC clone setting in this page.

MAC Clone: On Off

3.8.3 Tones Settings

Tones Settings

You could configure your tones settings in this page.

	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
Cadence On:	<input type="checkbox"/>	<input checked="" type="checkbox"/>				
Hi-Tone Freq.:	440	480	620	620	480	440
Lo-Tone Freq.:	350	440	480	480	440	350
Hi-Tone Gain:	4522	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	50	30	200	30
Off Time 1:	0	400	50	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

3.8.4 Advanced Settings

Advanced Setting

You could change advanced setting in this page.

ICMP Not Echo: Yes No

Send Anonymous CID: Yes No

Priority Reversal: Yes No

Send Flash event:

SIP Encrypt:

3.9 System Auth.

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:

New password:

Confirmed password:

3.10 Save Change

In Save Change you can save the changes you have made. If you want to make new setting into effect, you have to click the Save button. After you click the Save button, the IP Phone will automatically restart and the new setting will take effect.

Save Changes

You have to save changes to effect them.

Save Changes:

3.11 Update

In Update you can update the IP Phone's firmware to the latest or restore the factory setting.

3.11.1 In New Firmware function you can update new firmware via Local PC/TFTP in this page.

You can upgrade the firmware by the following steps:

- 1) Select the firmware code type, Risc or DSP code.
- 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) Select the correct file you want to download to the IP Phone then click the Update button.

Update Firmware

You could update the newest firmware.

Method: Local PC TFTP

Local PC

Code Type:

File Location:

TFTP

TFTP Server:

3.11.2 Auto Update

Auto Update Settings

You could set auto update settings in this page.

Update via: Off TFTP FTP HTTP

TFTP Server:

HTTP Server: Exp. 60.35.187.30

HTTP File Path: Exp. /download/

FTP Server:

FTP Username:

FTP Password: Exp. 60.35.17.1

FTP File Path: Exp. /file/load

Check new firmware: Power ON Scheduling

Scheduling (Date): (1~30 days)

Scheduling (Time):

Automatic Update: Notify only Automatic

Firmware File Prefix:

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

Restore Default Settings

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

Restore default settings:

3.12 Reboot

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

Reboot System

You could press the reboot button to restart the system.

Reboot system: