Micronet®
Faster and Easier Networks

User Guide

VoIP Telephone with PoE

Model No.: SP5102

World Wide Web: www.micronet.com.tw ; www.micronet.info

TABLE OF CONTENTS

1. INTRODUCTION 5
1.1 Packet Content 5
1.2 Key Features 5
1.3 Telephone Appearance 7
1.4 Key Definition 9
1.5 Keys with “123” number mode 11
1.6 Keys with “ABC” character mode 12
1.7 Keys with “abc” character mode 13
1.8 Menu Keys 14

2. START UP 15
2.1 Installation 15
2.2 LCD Display 16
2.2.1 LCD Display in initializing mode 16
2.2.2 LCD Display in IDLE state 16
2.2.3 LCD Display in OFF-HOOK state 16
2.2.4 LCD Display in DISCONNECTED state 17
2.2.5 LCD Display for RINGING state 17
2.2.6 LCD Display for DIALING state 17
2.2.7 LCD Display for Incoming Call state 17
2.2.8 LCD Display in CONNECTED state 17
2.2.9 LCD Display in Call Waiting state 18
2.2.10 LCD Display for firmware upgrading mod 18
2.2.11 LCD Display for saving and rebooting mode 19
2.2.12 LCD Display for edit mode 19
2.3 Editing Display 20
2.4 LED Display 21

3. LCD MENU OPERATION 23
3.1 Call Records 23

3.1.1 Received 24
3.1.2 Dialed Calls 25
3.1.3 Missed 25
3.1.4 Delete All 26
3.2 Phone Book 27
3.2.1 New Entry 27
3.2.2 View Entry 28
3.2.3 Search Entry 29
3.2.4 Memory Check 29
3.3 Networking Setting 30
3.3.1 IP Mode 30
3.3.2 IP Address 30
3.3.3 Net Mask 30
3.3.4 Default GW 31
3.3.5 DNS Setting 31
3.3.6 PPPoE 31
3.3.7 SNTP 32
3.4 SIP Setting 34
3.4.1 Proxy Setting 34
3.4.2 User Setting 35
3.5 Phone Setting 36
3.5.1 Alarm Setting 36
3.5.2 Ring Setting 38
3.6 Mail Box 39
3.6.1 Information 39
3.6.2 MailBox No. 39
3.6.3 MailBox Key 39
3.6.4 Voice Mail Dial 39
3.7 Function Keys 40
3.7.1 New Entry 40
3.7.2 From Phone Book 40
3.7.3 View Entry 40
3.8 Reboot 41
4. BASIC FUNCTION 42

4.1 Power on and Initialization 42
4.2 Making Calls 42
4.2.1 OFF-HOOK Dialing 42
4.2.2 Redial (OFF-HOOK Dialing) 42
4.2.3 Pre-Dial 42
4.2.4 Redial (ON-HOOK Dialing) 43
4.2.5 Dial during connected 43
4.2.6 Memory Dial 43
4.3 Answer Calls 44
4.3.1 Answer the call in the ringing state 44
4.3.2 Answer the call in the connect state 44
4.4 Hold and Retrieve Calls 45
4.5 Transfer 46
4.5.1 Consultant Transfer 46
4.5.2 Blind Transfer 46
4.6 Conference 47

5. WEB ADMINISTRATION 48
5.1 Factory Default 48
5.2 Configuring the SIP Phone through Web Pages 49
5.3 Installation Wizard 52
5.4 Advanced Configuration 55
5.4.1 Network Configuration 55
5.4.2 SIP Configuration 60
5.4.3 System Configuration 61
5.4.4 Number Configuration 63
5.4.5 Media Configuration 66
5.4.6 Device Management 70
5.5 System Status 74
5.5.1 Network Status 74
5.5.2 Version Information: 75
5.6 Reboot 76

6. TELNET COMMAND LINES 77
6.1 [help] command 78

6.2 [quit] command 79
6.3 [debug] command 80
6.4 [reboot] command 81
6.5 [pbook] command 82
6.6 [commit] command 83
6.7 [ping] command 84
6.8 [time] command 85
6.9 [ifaddr] command 86
6.10 [pppoe] command 88
6.11 [flash] command 89
6.12 [sysconf] command 90
6.13 [sip] command 92
6.14 [security] command 94
6.15 [line] command 95
6.15.1 Busy Forward 95
6.15.2 No Response 95
6.15.3 Unconditional 95
6.16 [voice] command 96
6.17 [phone] command 97
6.18 [tos] command 98
6.19 [prefix] command 100
6.20 [rom] command 101
6.21 [auth] command 102
6.22 [passwd] command 103

1. Introduction

Micronet SP5102 VoIP desktop telephone integrates DSP/codec system-on-chip solutions to provide the industry's highest voice quality. For ease-of-use functionality, SP5102 provides well compatibility with many well-known IP-PBX, rich telephony features, and many user-friendly function buttons of conference, call pickup, transfer, redial, Hold …etc. Moreover, Power-over-Ethernet technology (IEEE 802.3af) is incorporated in SP5102. Not only for toll-saving, but it also facilitates enterprises to benefit from reduced expense of electrical wiring, effective network control, and reliable power-supply.

1.1 Packet Content

Before you start installing the device, verify the following items are in the package:

- SP5102 IP Telephone
- Quick Installation Guide
- User's Manual CD
- Power Adaptor (optional)

1.2 Key Features

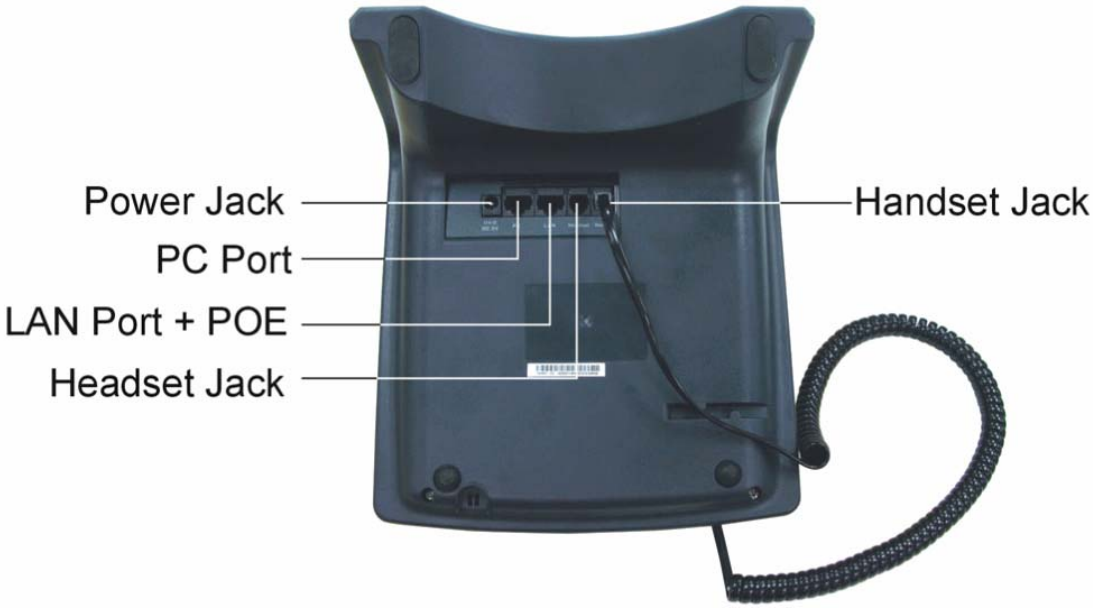
- Compliant with IETF SIP standards
- Compliant with IEEE 802.3af standard
- Support G.711A/μ-law, G.723 and G.729 codecs
- Well compatibility with well-known IP-PBX (Micronet SP5211, Asterisk, Siemens, Alcatel, Lucent and NEC)
- Rich call features
  - DND (Do Not Disturb)
  - Call waiting
  - Call Conference (3-way)
  - Call hold / retrieve
  - Call forward (Busy, No-Answer and Unconditional)
  - Call transfer (attended / unattended)
  - Call pickup
- User-friendly function buttons
- Call details list and message waiting indicator
- Support outbound proxy and STUN for NAT traversal

- Support QoS to ensure voice quality
- Cost-effective, suitable for pure VoIP environment

1.3 Telephone Appearance

Front View

Bottom shell



LCD
Line 1
Line 2
F1
F2
F3
Forward
DND
Missed
VMS
Blind Tran.
Mute
Headset
K1
K2
K3
K4
K5
K6
K7
K8
K9
K10
K11
K12
M1
M2
M3
M4
2
1
C
OK
LED1
LED2
Conf
Pick
Trans
Redial
Hold
K13
K14
K15
K16
K17
4
5
3

1.4 Key Definition

NO	KEYPAD	FUNCTION
1	OK	1. Press this key to confirm a menu selection, a setting, or a phone entry. 2. When in menu mode, pressing this key will select the menu selection highlighted. 3. When entering phone setting or menu selection entry, pressing this key will confirm the entry.
2	C	1. Cancel for the menu setting or number typing. 2. Reject the incoming calls
3		Adjust the volume
4		Adjust the volume
5		Speaking without picking up handset
M1		UP arrow key
M2		Left arrow and return key for the menu setting
M3		1. Right arrow 2. Enter key for the menu setting
M4		1. Down arrow key 2. Enter Phone Book directory
K1	Line1	Line1 switch key
K2	Line2	Line2 switch key
K3	F1	Memory key 1
K4	F2	Memory key 2
K5	F3	Memory key 3
K6	Forward	Direct forward

NO	KEYPAD	FUNCTION
K7	**DND**	DND
K8	**Missed**	Missed calls
K9	**VMS**	Voice Mail
K10	**Blind Tran**	Blind transfer
K11	**Mute**	Mute
K12	**Headset**	Headset mode switch
K13	**Conf**	Conference
K14	**Pick**	Call Pickup
K15	**Trans**	Transfer
K16	**Redial**	Redial
K17	**Hold**	Call hold / Call retrieve

This phone could support three methods for the text input, such like this: “123”, “ABC”, “abc”. “123” is for the number inputs, “ABC” is for the uppercase character inputs; “abc” is for the lowercase character inputs.

User could press the # button to exchange these three methods.

1.5 Keys with “123” number mode

Users could only enter the digits in the “123” mode as following:

Key	Numbers in “123” input method
1 S33	**1**
ABC 2 S30	**2**
DEF 3 S28	**3**
GHI 4 S32	**4**
JKL 5 S23	**5**
MNO 6 S18	**6**
PQRS 7 S29	**7**
TUV 8 S10	**8**
WXYZ 9 S11	**9**
* S04	*
0 S17	**0**
# S12	**Input method switch key**

1.6 Keys with “ABC” character mode

Users could enter the characters with the uppercase in this mode. The key [1] could only input the special symbols and press more times to pick up the symbols you want. If users press two times on the key [DEF 3], the input character will be E.

Key	Characters with uppercase in “ABC” input method
[1] S33	. , : ; / - _ @ & ? ¿ ! ¡ ‘ “
[ABC 2] S30	A B C
[DEF 3] S28	D E F
[GHI 4] S32	G H I
[JKL 5] S23	J K L
[MNO 6] S18	M N O
[PQRS 7] S29	P Q R S
[TUV 8] S10	T U V
[WXYZ 9] S11	W X Y Z
[*] S04	* + / ( ) < = > % ¥ £ $
[0] S17	Space
[#] S12	Input method switch key

1.7 Keys with “abc” character mode

Users could enter the characters with the uppercase in this mode. The key [1] could only input the special symbols and press more times to pick up the symbols you want. If users press two times on the key [DEF 3], the input character will be e.

Key	Characters with uppercase in “ABC” input method
[1] S33	. , : ; / - _ @ & ? ¿ ! ¡ ‘ “
[ABC 2] S30	a b c
[DEF 3] S28	d e f
[GHI 4] S32	g h i
[JKL 5] S23	j k l
[MNO 6] S18	m n o
[PQRS 7] S29	p q r s
[TUV 8] S10	t u v
[WXYZ 9] S11	w x y z
[*] S04	* + / ( ) < = > % ¥ £ $
[0] S17	Space
[#] S12	Input method switch key

1.8 Menu Keys

There are four keys to help users enter, exit or pick up the configuration tables for changing. Please check out the following info:

Key	Functions descriptions
	Press the right arrow key in the IDLE mode to enter the main menu.
	Press the left arrow key will help users to return the original table or exit the main menu
	Press the up arrow key to scroll up configuration items.
	1. Press the down arrow key to scroll down configuration items. 2. Enter Phone Book directory.
	Enter the selected configuration table.
	Return to the original table or exit the main menu key.

2. Start Up

2.1 Installation

Using **Power over Ethernet (PoE)**, which uses standard twisted-pair cable to convey electricity to IP phones, wireless access points and other devices, is ideal. PoE isn't a new idea. Utilities have been putting power and voice over the same lines for decades, but until recently it wasn't widely used for data networks. PoE technology is highly reliable and an ideal way for businesses to save on their communications infrastructures by transmitting power and data over the same wire.

You have two options for network and power source connections. Please use one of the following methods:

A. To connect the AC power (optional):

a. Connect the supplied network cable between the LAN jack on the phone and the LAN port on the wall or hub/switch device port.

b. Connect the DC plug on the power adapter to the DC 9V1A jack on the phone.

c. Connect the country-specific AC cord to the power adapter.

B. Power over Ethernet (PoE)

Using a regular CAT5 cable, the phone can be powered from a PoE (IEEE 802.3af) compliant switch or hub.

Plug the supplied network cable between the LAN jack on the phone and an available power and data port on the in-line power switch.

2.2 LCD Display

2.2.1 LCD Display in initializing mode

1970/01/01 00:00 Initializing…	Status during the system initializing

2.2.2 LCD Display in IDLE state

2006/05/10 10:10 SipPhone SP5101	Displaying in the IDLE mode while the network and registration were ok. (User can change display name from SIP configuration)
2006/05/10 10:10 Register Fail	Registrations fail during the IDLE mode

2.2.3 LCD Display in OFF-HOOK state

2006/05/10 10:10 Network Fail	Network fail during the IDLE mode (WAN port detection or get IP failed with DHCP and PPPoE)[1]
Line 1 Dial…	LCD displaying while the phone was in the OFFHOOK state.
Line 2 Dial…	LCD displaying while the phone was in the OFFHOOK state.

2.2.4 LCD Display in DISCONNECTED state

2006/05/10 10:10 **SipPhone** **SP5102**	After the remote side sends the BYE message, LPSP5102 will automatically return IDLE state.

2.2.5 LCD Display for RINGING state

Incoming call… 208	Incoming call with the calling name or number

2.2.6 LCD Display for DIALING state

Line 1 Dialing… 9999	Making a outgoing call with the dialed number
Line 2 Dialing… 9999	Making a outgoing call with the dialed number

2.2.7 LCD Display for Incoming Call state

Jaosn 65605	When it received incoming call, it will show incoming call display name and Line number

2.2.8 LCD Display in CONNECTED state

Line 1 Talking… 00:00:10	During the talking state and running the timer
Line 2 Talking… 00:00:10	During the talking state and running the timer

Mute… 00:00:10	The timer will still go on while the mute function was enabled during the conversation.
Transfer…	Transfer action during the call.
Hold…	Hold the established call.
Parking…	Parking the call.
In Conference….. 00:10:10	Enable the conference function and start the timer from the beginning.

2.2.9 LCD Display in Call Waiting state

Line 1 Talking……. 65605	Under conversation and LPSP5102 received incoming call. LCD of LPSP5102 will display incoming call number. User can press Line button to pick up call waiting or presses C button to reject call waiting.

2.2.10 LCD Display for firmware upgrading mod

Download…	Firmware file downloading mode
Download… Writing…	Writing the firmware after the firmware downloading
Completed… Please reboot	Request for the rebooting after the firmware upgrading

2.2.11 LCD Display for saving and rebooting mode

Saving…	Saving the configurations while users had be changed
Rebooting…	Users reboot this unit from the remote side
Rebooting?	Request for the keypad pressing to reboot
Please Wait…	Users reboot this unit by the keypad pressing

2.2.12 LCD Display for edit mode

The characters and digits were entered from the left to the right side. The cursor will be blinking every 500ms just like the example as above:

John → John_

208 → 208_

If the users enter the incorrect info, the LCD will show as following:

Invalid Input…

2.3 Editing Display

a Cursor
Under the characters or digits, the cursor will be displayed and the characters or digits will be blinking. It was shown as "_" with the LCD showing. Users could press the right or left arrow keys to make the cursor move to the right or left side. The blinking time for the on and off will be 500ms.

b Insert the characters or numbers
All the characters and numbers were entered from the left to the right side. While users enter a new character, the cursor will move to the right side of the entered character or numbers and wait for the next one.

c Delete the characters or numbers
User could use the cancel key for erasing the entered characters or digits. It will only erase the characters or digits, which are in the left side of the cursor.

d Rapid delete the characters or numbers
All the info will be dropped while users press the keypad for 3 seconds and all the entered info will be deleted and exit the edit mode.

e Exit the edit mode
Users could press the keypad for 3 seconds to exit the edit mode during the editing.
Or move the cursor to the left side of the LCD and without any characters and numbers; press the cancel key to exit the edit mode.

2.4 LED Display

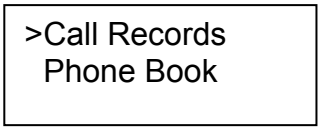
While users power on this phone set, all the LED will be lighted up before the system initializing procedure finished. The function LED will be light while users press the function keys; and will be blink while users press the function keys for twice. The following definitions are for the LED:

LED	Key	Function description
LED1	●	1 Flashing for the incoming call. Flash time : 200ms on; 200ms off 2 Lighting for the OFFHOOK.
LED2	●	
K1 LED	● K1	1. Flashing for the line holding Flash time : 500ms on; 500ms off 500ms on; 1000ms off 2. Lighting for the line in use.
K2 LED	● K2	
K3 LED	● K3	Light up while user press this for the speed dial
K4 LED	● K4	Light up while user press this for the speed dial
K5 LED	● K5	Light up while user press this for the speed dial
K6 LED	● K6	1. Light up while user enabled forward function (unconditional/ No Answer/ Busy). 2. Light off when user diable forward function.
K7 LED	● K7	1. Light up while users enable the DND function 2. Off while users disable the DND function
K8 LED	● K8	1. Light up while user has missed call. 2. Light Off while users checked out the miss called records
K9 LED	● K9	1. Light up while users have the voice mail 2. Off while users checked out the voice mail records
K10 LED	● K10	Press during communication to do blind transfer. 1. A communicate with B→Blind Tran. + C's number→hear nothing (C ring) → Blind transfer 2. A communicate with B→Blind Tran. + C's number→C Busy → hear special tone then retrieve B immediately 3. A communicate with B→Blind Tran. + C's number (not finished) →Blind Tran. Retrieve call
K11 LED	● K11	1. Light up while users enable the mute 2. Off while users cancel the mute
K12 LED	● K12	1. Light up when user is using headset mode. 2. HEADSET button works the same like

		hook/speaker button. When having incoming call, press HEADSET button can pick up call and press again can hang up call. 3. When in communication, press HEADSET button can switch voice path to headset.

3. LCD Menu Operation

During the menu operation, there is no cursor for the configuration tables selecting. The LCD will show just like this:



Users could press the up or down arrow keys to move on different item. This sign will guide you to choose the configuration tables. In the main menu, it will exit the menu and back to the IDLE mode if there is no action for 30 seconds. The LCD menu includes eight tables in the menu tree: Call Records, Phone Book, Networking setting, SIP setting, Phone setting, Mail Box, Function Keys, and reboot.

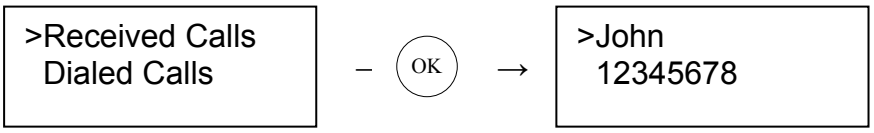
3.1 Call Records

- Received Calls
  - Call Record List
    - Dial Out
    - Detail
    - Add to Book
    - Delete
- Dialed Calls
  - Call Record List
    - Dial Out
    - Detail
    - Add to Book
    - Delete
- Missed Calls
  - Call Record List
    - Dial Out
    - Detail
    - Add to Book
  - Delete
- Delete All
  - Delete all?

This phone set could support 10 entries for the received / dialed / missed calls. The sorting will be based on the latest period. If there could be matched with the phone book, LCD will show the name; or show the number only.

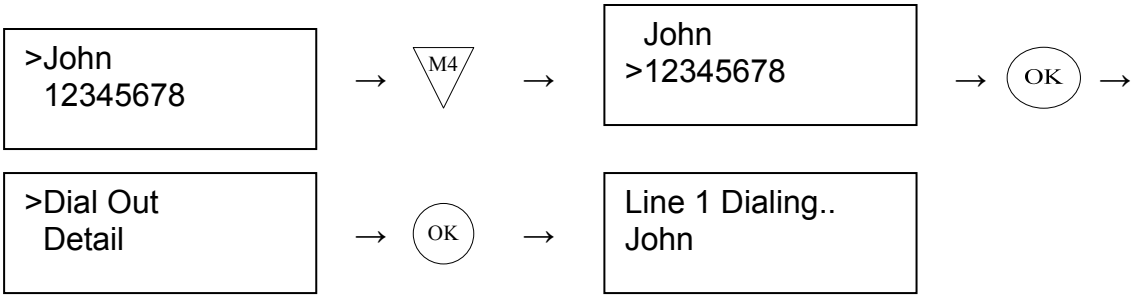
3.1.1 Received

Showing the received calls; pressing the M4 and M1 button to check out the other received calls.



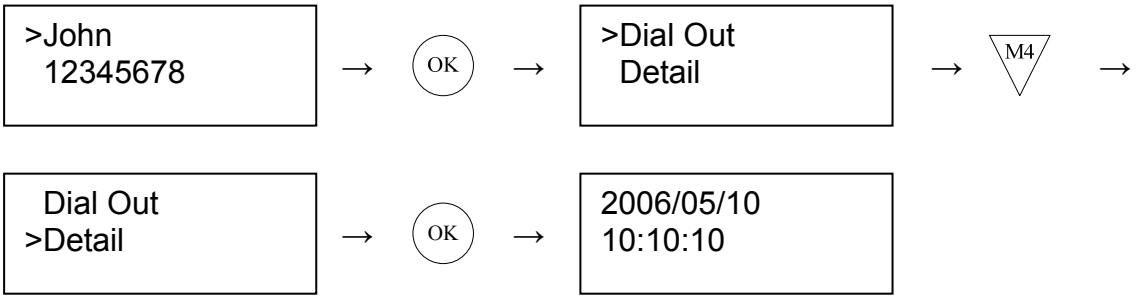
The LCD will show as following while there is no any record for the received calls:

- **Dial out**



- **Detail**

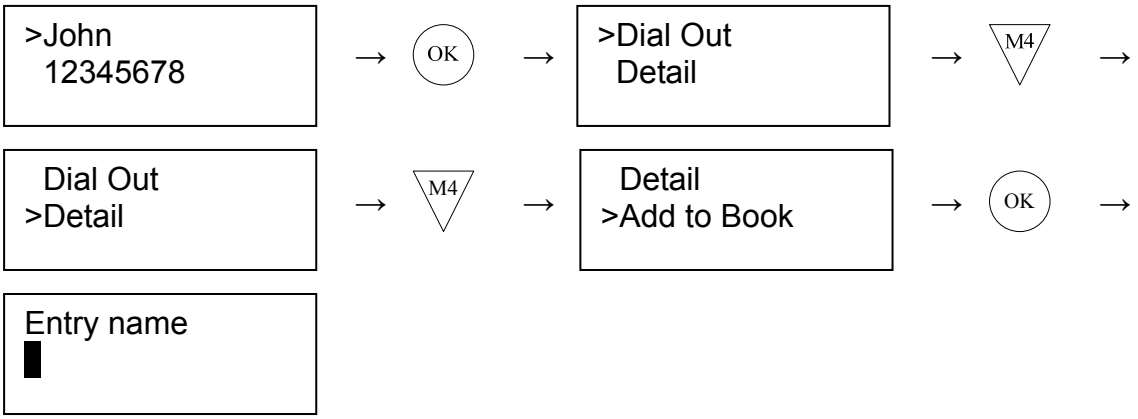
The Detail will show the date and time for the calls as following:



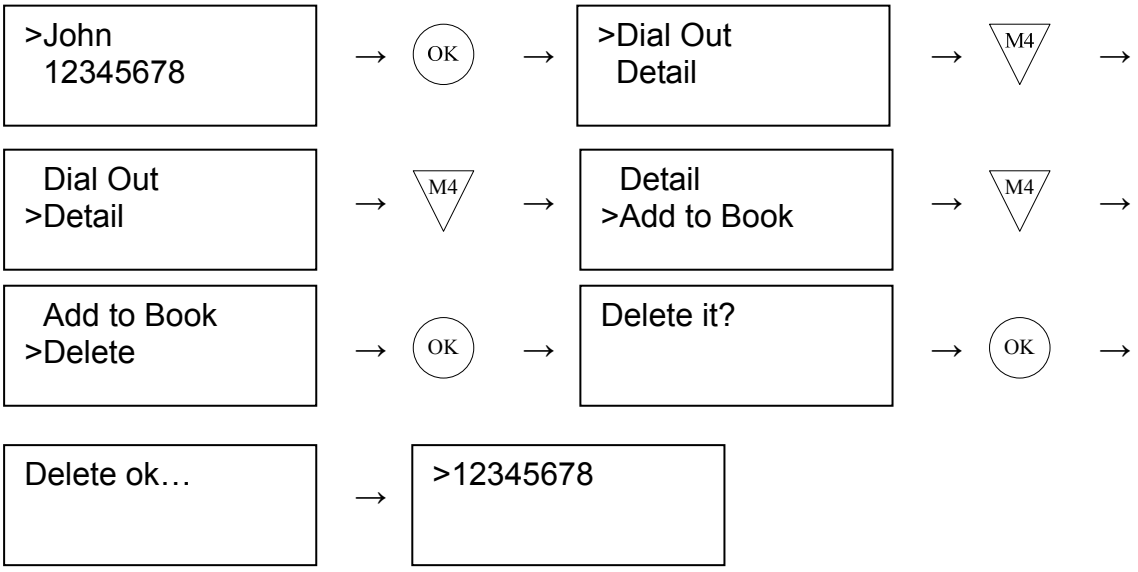
- **Add to Phone**

This allow user add the unknown number into the phone book. After the new

entry added, show all the entries in the phone book to verify by users. If users press the OK button, the flow will be just like the actions in the phone book



- **Delete**



3.1.2 Dialed Calls

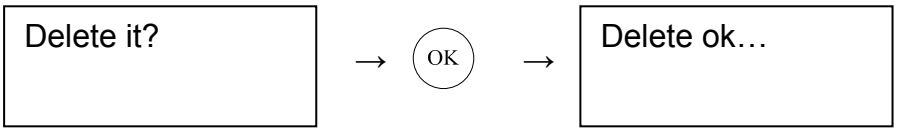
All the LCD displaying or actions are as same as the flow of Received. Please check out the chapter 6.1 for the detail as above.

3.1.3 Missed

All the LCD displaying or actions are as same as the flow of Received. Please check out the chapter 6.1 for the detail as above.

3.1.4 Delete All

This will delete all the records of the received, dialed and missed calls.



3.2 Phone Book

This phone set could support 90 entries of the phone book. Users could add, modify, delete, and dial out all the entries in the phone book. If the name and number had been added in the phone book, the LCD has to show the name if it is the incoming call.
In the view mode, the name has to be sorted by the characters.

- New Entry
  - Input Name
  - Input Number
- View Entry
  - Phone Book List
    - Dial Out
    - Modify Entry
      - Entry Name
      - Entry Number
    - Delete Entry
      - Delete it?
    - Detail
- Search Entry
  - Input Name
  - Memory Check

3.2.1 New Entry

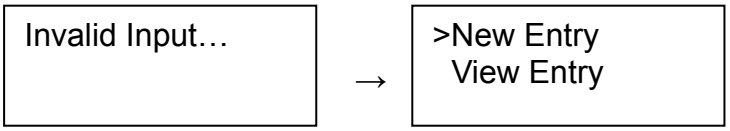
A Input Name

It could support the number and character typing in this mode.
In the phone book entries, this isn't permitted for the same name. If users enter the different number with the same name, the LCD will show as following:

Input Name Joh
Overwrite Entry?

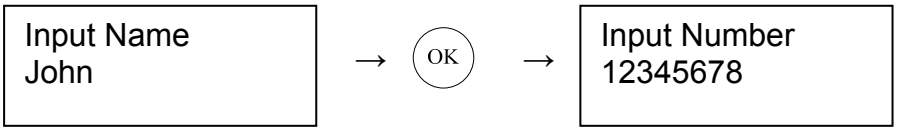
This is allowed the same number with the different name. For the name of the incoming call displaying, the phone set will show up the latest record configured by users.
This isn't permitted for the empty info about the name in this mode. If users press the OK button without any info for the name, the LCD will show as

following and return to the New Entry table after a few seconds:



B Input Number

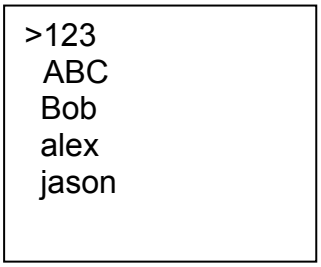
After users enter the valid name for the New Entry, the LCD will show as following and it only supports the number typing in this mode:



After saving the name and number for the new entry, LCD will show up the exist records:

3.2.2 View Entry

In this mode, the phone has to sort the name by the characters. The priority of sorting just like this: Number > Uppercase character > Lowercase character.



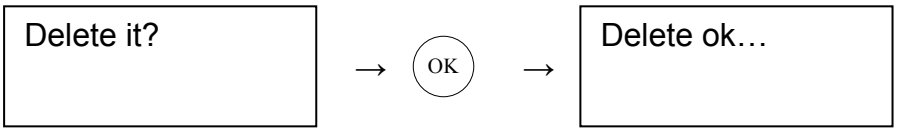
Users could press the OK button to select the action in the view mode.
If there is no any entry for the phone book, LCD will show the massage as following:

Press OK on specific phone book entry, phone will be off-hook and dial out automatically.

Modify Entry

Here user can modify name and number of existed phone book data.

Delete Entry

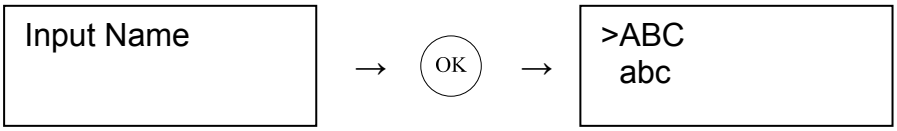


Detail

Press to see detail name and number of this phone book entry.

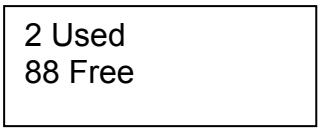
3.2.3 Search Entry

This mode could support the numbers or characters typing. If the entered name couldn't be matched with the existed records, on LCD will show the first entry of this phone book.



3.2.4 Memory Check

Phone set could support about 90 entries for the phone book; it will show the records for the unused and used entries.



3.3 Networking Setting

- IP Mode
  - Fixed

DHCP
PPPoE

- IP Address
- Net Mask
- Default GW
- DNS Setting
  - Primary DNS
  - Second DNS
- PPPoE
  - ID
  - Password
  - Reconnect
  - SNTP
- SNTP
  - Server IP
  - Time Zone
  - Mode

3.3.1 IP Mode

Press the

or

button to select the IP mode.

3.3.2 IP Address

Enter IP address for Fixed IP mode. Under this mode can only input digits.

IP Address: 10.1.1.3

3.3.3 Net Mask

All the operation is just like the IP Address configuring. It will be shown as following from the LCD:

IP Mask: 255.0.0.0

3.3.4 Default GW

All the operation is just like the IP Address configuring. It will be shown as following from the LCD:

```
Default GW:
10.1.1.254
```

If users input the invalid value for the IP address, Net Mask or Default Gateway, LCD will show as following and return to the original setting of this configuration table:

```
Invalid Input…
```

3.3.5 DNS Setting

Primary DNS

It supports the number typing in this table only.

```
Primary DNS:
168.95.192.1
```

Second DNS

It supports the number typing in this table only.

```
Second DNS:
168.95.1.1
```

3.3.6 PPPoE

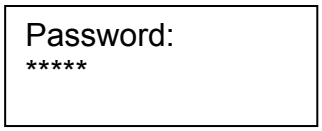
ID

It could support the number or character typing in this mode.

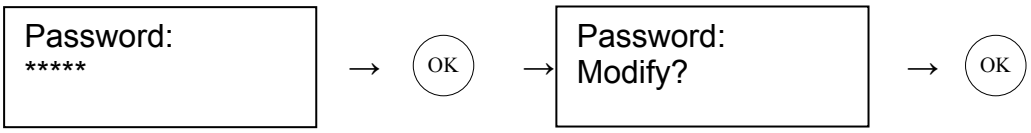
```
User Name:
pppoe
```

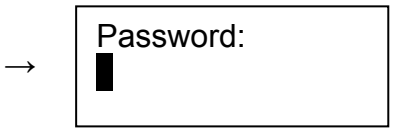
Password

For the protecting policy, LCD will use the asterisk to replace the info showing.



For the password modify displaying, the LCD will clean all the asterisks and showing the cursor as following:





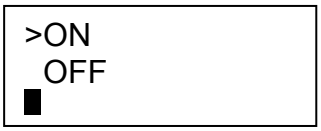
Reconnet

Please press the button

or

to enable or disable this function.

3.3.7 SNTP

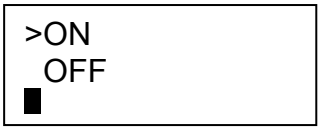


Mode

Please press the button

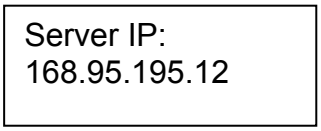
or

to enable or disable this function.



Server IP

To enter SNTP server IP address. It supports the number typing and Domain typing.



Time Zone

Pressing the

for decreasing and

for increasing the zone value.

The

button will save the changed.

Zone: GMT +8:00

3.4 SIP Setting

All the menu operations are as same as the Networking setting.

- Proxy Setting
  - Proxy IP
  - Proxy Port
  - OutPx IP
  - OutPx Port
- User Setting
  - ID
  - Password
  - Phone Num
  - Local Port

3.4.1 Proxy Setting

3.4.1.1. Proxy IP

It could support the number and character typing for the IP or domain.

Proxy IP: 10.1.1.2	Proxy IP: proxy.com

3.4.1.2. Proxy Port

Configuring the Proxy port in this table; it could only support the number typing. The max value is 65535.

Proxy Port: 5060

3.4.1.3. OutPx IP

This is the setting for the Outbound Proxy. It could support the number and character typing for the IP or domain.

OutPx IP: 10.1.1.2	OutPx IP: outpx.com

3.4.1.4. OutPx Port

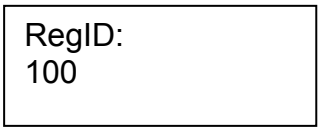
This could only support the number typing only. The max value is 65535

OutPx Port: 5060

3.4.2 User Setting

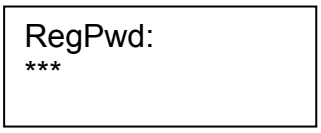
3.4.2.1. ID

It could support the number or character typing for the ID.

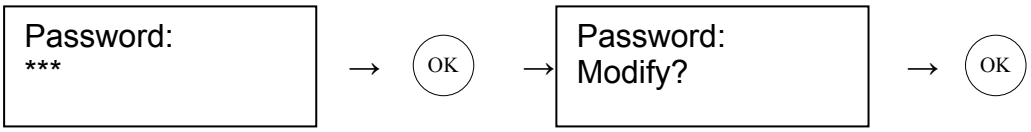


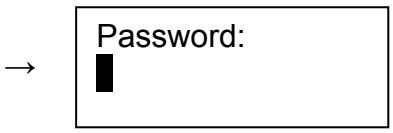
3.4.2.2. Password

It could support the number or character typing for the ID. LCD will show the asterisks to replace the password.



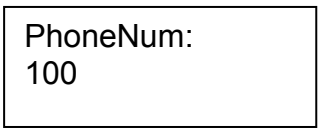
For the modify displaying, the LCD will empty the all the asterisks and showing the cursor as following:





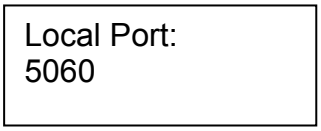
3.4.2.3. Phone Num

It could support the number typing only.



3.4.2.4. Local Port

It could support the number typing only and the max value is 65535.



3.5 Phone Setting

- Alarm Setting
  - Add
  - View All
  - Del All
- Ring Setting
  - Ringer Volume
    - Volume 1
    - Volume 2
    - Volume 3
    - Volume 4
  - Ringer Melody
    - Melody 1
    - Melody 2
    - Melody 3

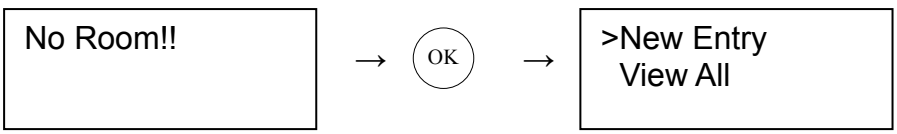
3.5.1 Alarm Setting

There are some definitions for the Alarm clock, please check out the detail as following:
If the alarm clock had been set and the time is up:

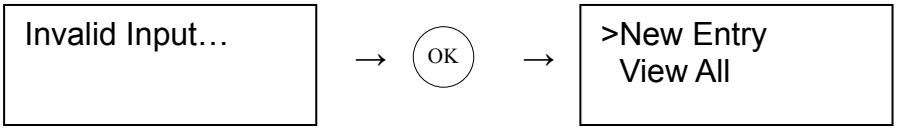
- The phone will be ring in the IDLE and MENU state only.
- If the phone isn't in the IDLE or MENU state, the phone will ring while the state back to the IDLE or MENU.
- If the state don't return to the IDLE or MENU for 30 minutes, the clock will cancel and don't ring the phone.
- Ringing the phone every 20 seconds.
- For the ringing tone, depends on the ring melody setting.
- Stop ringing until users pick up the phone set or hand free.

3.5.1.1. Add

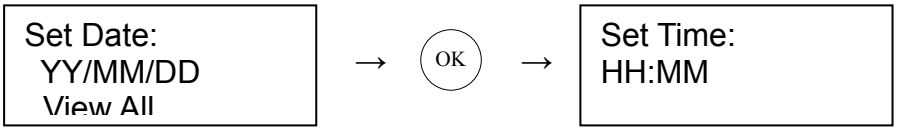
This phone set could support 3 entries for the Alarm.
If all the three entries had been taken, LCD will show the message as following and back to the original page:



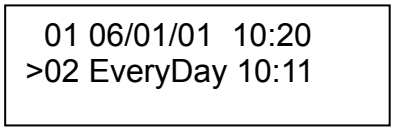
If users enter the incorrect month or date, please show the message as following and back to the original page:



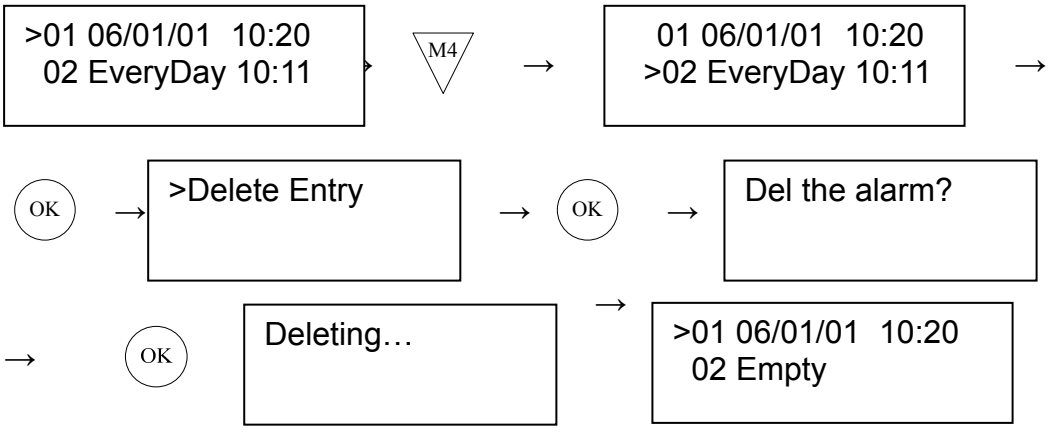
Switch to the Time setting while users press the OK button in the Date setting:



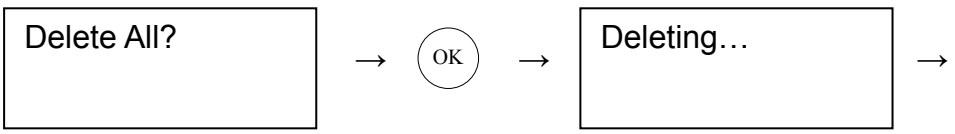
The entire format for the date and time are as following:
Year: 2000 ~ 2099; Month: 01 ~ 12; Date: 01~31; Hour: 00 ~ 23;
Minute: 00~59
The schedule will be everyday while user didn't enter the date for the alarm clock.
The format of the time setting is as following:



3.5.1.2. View All



3.5.1.3. Del All



3.5.2 Ring Setting

3.5.2.1. Ringer Volume

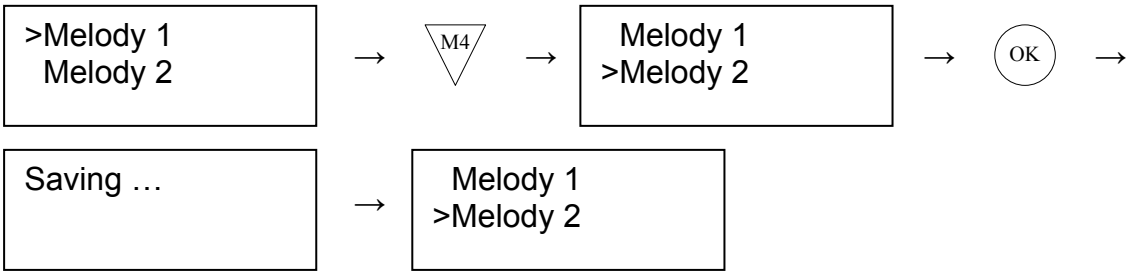
There are four levels for the volume adjustment and the default is level 2.
The symbol “>” will point out the current level for this phone set.

Pressing the M4 and M1 button to select the volume level for the ring.

While users stop the selection for 1 second, phone will play the ring to verify the ring volume.

3.5.2.2. Ringer Melody

This phone set could support four types of ring melody.



3.6 Mail Box

- Information
- MailBox No.
- MailBox Key
- Voice Mail Dial

3.6.1 Information

If VNS LED lights up, you can view voice mail information from this item. Below:

Information: 2 new, 1 old

3.6.2 MailBox No.

3.6.3 MailBox Key

3.6.4 Voice Mail Dial

3.7 Function Keys

- New Entry
  - Input Number
  - Press the key…
- From PhoneBook
  - Phone Book List
  - Press the key…
- View Entry
  - Function Key list
    - F1:
    - F2:
    - F3:

Users could configure the Function Keys for the speed dial or special IP-PBX function. For the function keys configuration, users could input the new number or select the entries from the Phone Book.

3.7.1 New Entry

Set function key number and key button. User needs to input number first and then defines which key to match the number.

3.7.2 From Phone Book

User can define one phone book data to match memory function key, so that user can press function key to do speed dial.

3.7.3 View Entry

The view mode could only for the entries showing. If users want to modify the entry, only to add the New Entry or change Phone Book configurations.

3.8 Reboot

4. Basic Function

4.1 Power on and Initialization

During the initialization procedure:

1 All the LED will light on till the initialization procedure was finished.

2 All the LED will light on if there is error during the procedure.

3 On LCD will show “Initializing”.

4.2 Making Calls

There are some ways to make outgoing calls:

1 OFF HOOK Dialing

2 Redial (OFF-HOOK Dialing)

3 Pre-Dial (ON-HOOK Dialing)

4 Redial (ON-HOOK Dialing)

5 Dial during connected

6 Memory Dial

4.2.1 OFF-HOOK Dialing

The maximum digit is 16. If users enter more than 16 digits, the number will move on the LCD displaying. User can press Speaker button, pick up handset, or press line button to dial out. The

button could delete the digits.

4.2.2 Redial (OFF-HOOK Dialing)

User can press redial button during off-hook mode, phone will dial out the latest record of the dialed list.

4.2.3 Pre-Dial

User can press the numbers while the phone is in the IDLE state then pressing the

button will change to OFF-HOOK state and dial out this number. The

button could delete the digits as same as the OFF-HOOK dialing

mode.
During the Pre-dial function:

1 This is permitted to enter and dial out the digits “*” and “#”.

2 If the dialed number could be matched with the list in the phone book, LCD will display the name of this record.

4.2.4 Redial (ON-HOOK Dialing)

This will show all the records of the dialed list. After the records selected, pressing the SPK or OK button, the number will be dialed out.

4.2.5 Dial during connected

Pressing the K1 or K2 buttons could switch the Line 1 and Line 2. If the Line 1 was connected, it will be put on hold.

If the line has been put on hold, it can not dial out another call.

4.2.6 Memory Dial

The memory dial function could support the dialing from the IDLE of OFF-HOOK state. User can press memory function key or make phone off-hook first to dial out memory key.

4.3 Answer Calls

The LCD will show up the name if the incoming calls could be matched with the records from the phone book; or show the phone number if couldn't find out the records.

4.3.1 Answer the call in the ringing state

User can press Speaker button, Line button, or pick up handset to answer the incoming call.

4.3.2 Answer the call in the connect state

During the call was established, press the button K1 and K2 to hold the current line and answer another incoming call.

4.4 Hold and Retrieve Calls

There are two ways to hold the current calls and retrieve them back:

1 Press another idle line button K1 or K2 to hold the current line and switch to another IDLE line.

2 Press the Hold function keys to hold the current calls.

During the hold status, the LED of Line will flash.

4.5 Transfer

SP5102 could support the call transfer with two types, one is the Consultant, and another is the Blind transfer.
The transfer could be initialized only for the state as above.

i Dial another call during the connect state

ii Answer another call during the connect state

4.5.1 Consultant Transfer

Consultant Transfer scenario:

- A communicate with B→Tranf or Line hear dial tone+ C's number→C pick up → A hangs up → B communicate with C
- A communicate with B→Tranf or Line hear dial tone+ C's number → C busy/ring →Tranf or original Line →Retrieve call with B
- A communicate with B→Tranf or Line hear dial tone+ C's number → C reject →Tranf →Retrieve call with B
- A communicate with B→Tranf or Line hear dial tone+ C's number→C reject → press Line to hang up C and hear dial tone again → press original Line to retrieve call
- A communicate with B→Tranf or Line hear dial tone+ C's number→C ring→ A hangs up → Blind transfer

4.5.2 Blind Transfer

Blind Transfer Scenario:

- A communicate with B→Blind Tran. + C's number→hear nothing (C ring) → Blind transfer
- A communicate with B→Blind Tran. + C's number→C Busy → hear special tone then retrieve B immediately
- A communicate with B→Blind Tran. + C's number (not finished) →Blind Tran. Retrieve call

4.6 Conference

User needs to specify conference function to be local conference or server-based conference.
Conference Scenario:

A. Local Conference:

(1) A communicate with B→Conf or Line hear dial tone+ C's number→C pick up → press Conf again to build conference

(2) A communicate with B→Conf or Line hear dial tone+ C's number→C refuse to join conference → press Line to hang up C and hear dial tone again → press original Line to retrieve call

B. Server-based Conference:

(1) A communicate with B→Conf implement sever-based conference scenario

5. Web Administration

5.1 Factory Default

DESCRIPTION	DEFAULT VALUE
IP	10.1.1.3
Net Mask:	255.255.0.0
Default Gateway:	192.168.1.254
SIP Proxy:	10.1.1.2
SIP account	1001
SIP Port:	5060
Proxy Port:	5060
RTP Port:	16384
Digit Timeout:	5s
Call waiting:	Off
Call transfer	On

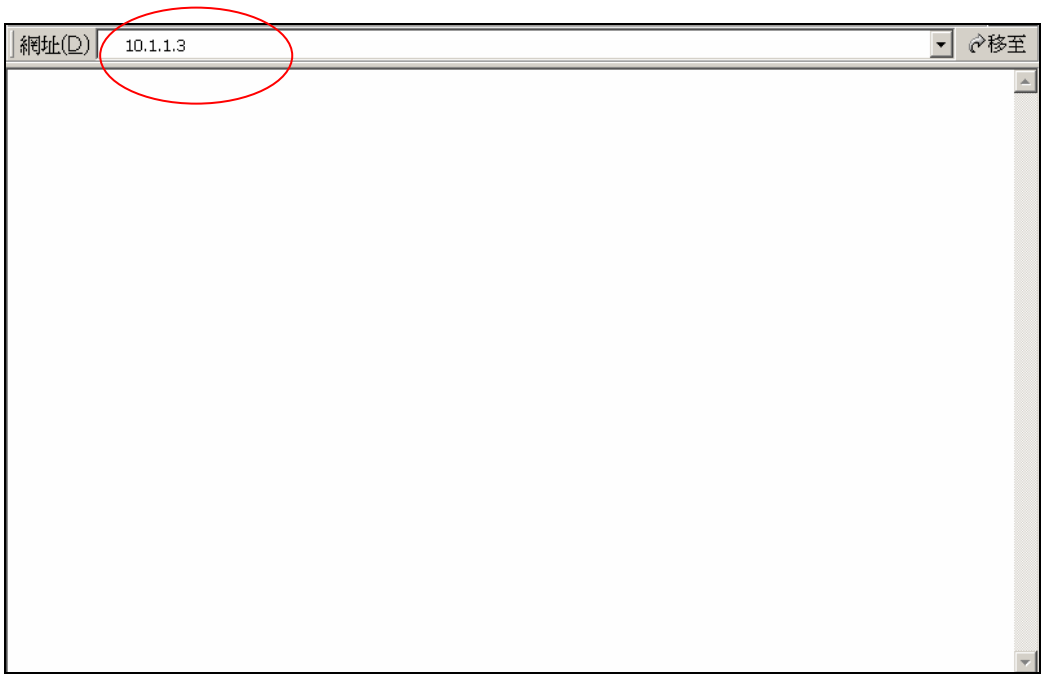
5.2 Configuring the SIP Phone through Web Pages

The HTTPD web management interface provides user an easier way to configure rather than command line method through TELNET.

The configuration function and steps are similar with the way through command line. Please refer to the chapter 4-Configuring the SIP Phone through Telnet command lines for more detail information. Below is a guide for user to configure via web interface.

Step 1. Browse the IP Address predefined via Keypad

Please enter IP address (user have to set via LCD menu first) of SIP Phone in web browser . If user failed to set IP address via LCD menu, the **default IP address of SIP Phone is 10.1.1.3**, user can try to connect to SIP Phone via this default IP via web interface.



Step 2. Input the login name and password

- Login name: **root** / user

> ***Note:***
> Login with “user” only has authority as below:
> 1. Modify network configuration
> 2. Modify Phone Book
> 3. Change login password of “user”
> 4. Reboot

- Password (The same with TELNET): Null (just press confirm, no need to key in password in default value)

Note: **User can set password later via web interface.**

Step 3. Enter the web interface main screen

After enter login name and password, user can see web interface main screen as below.

Step 4. Start to configure

After enter web management interface, user can see four main items.

1. **Installation Wizard:** User can follow steps in wizard to make first-time initial configuration.
2. **Advanced Configuration:** This menu includes other advanced configuration items. Please press triangle figure to list all items below Advanced Configuration.
3. **System Status:** User can check SIP Phone current status here.
4. **Reboot:** After make any change, user has to reboot SIP Phone to apply change.

Button Definition:

1. **OK**: After change or input any parameter, press this button will save data into SIP Phone.

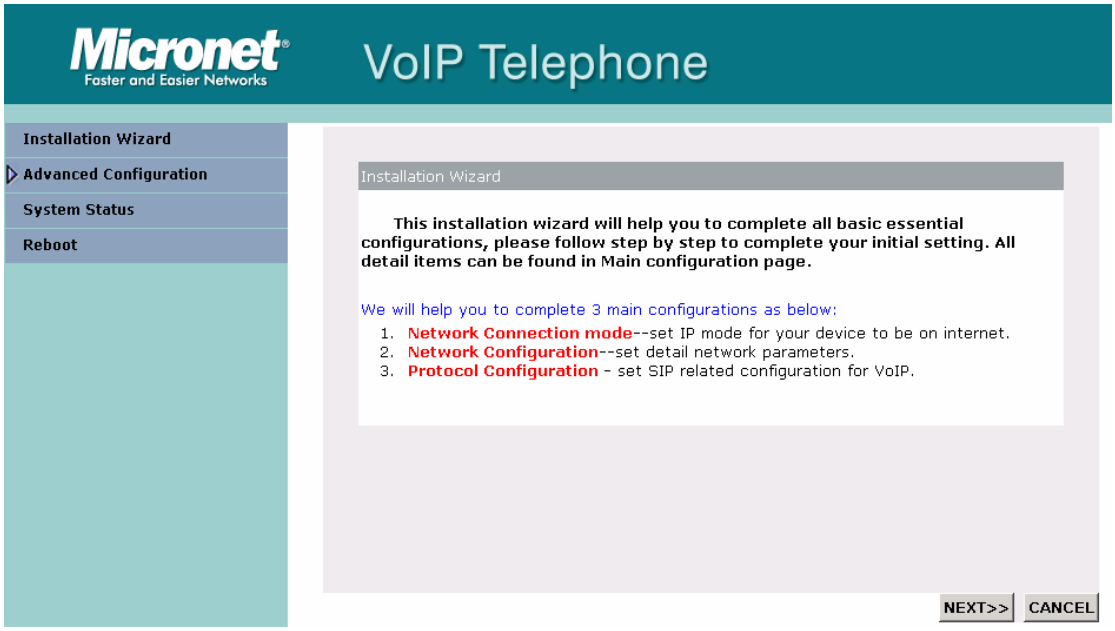
2. **CANCEL**: Press this button will clean data input by user and restore to original data.

   (A) **ADD**: Add a new data.

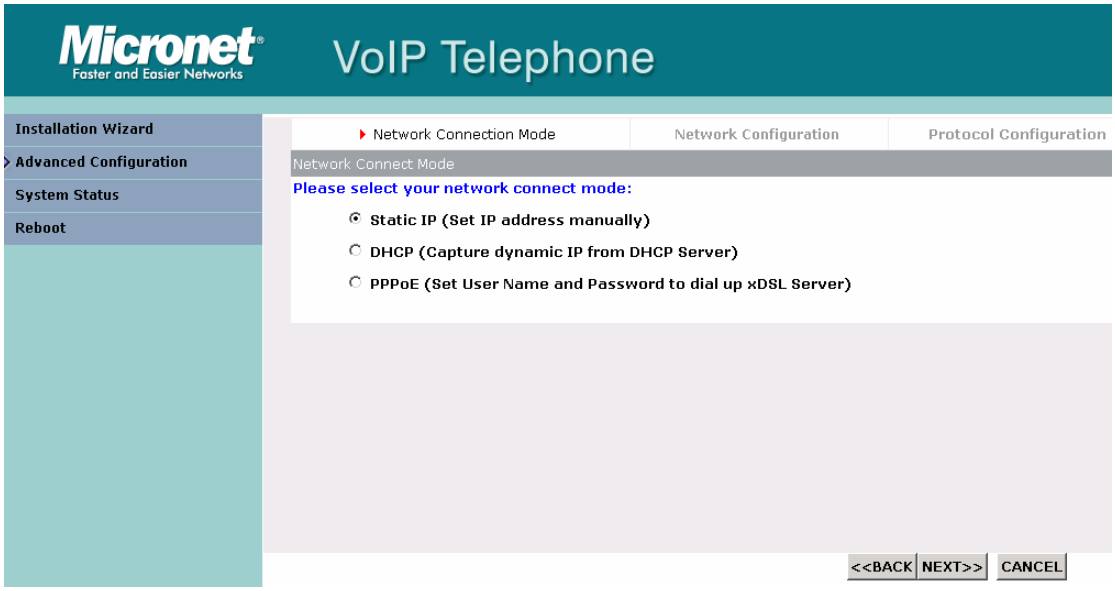
   (B) **DELETE**: Delete a specific data according to index number.

5.3 Installation Wizard

Installation Wizard includes three steps:

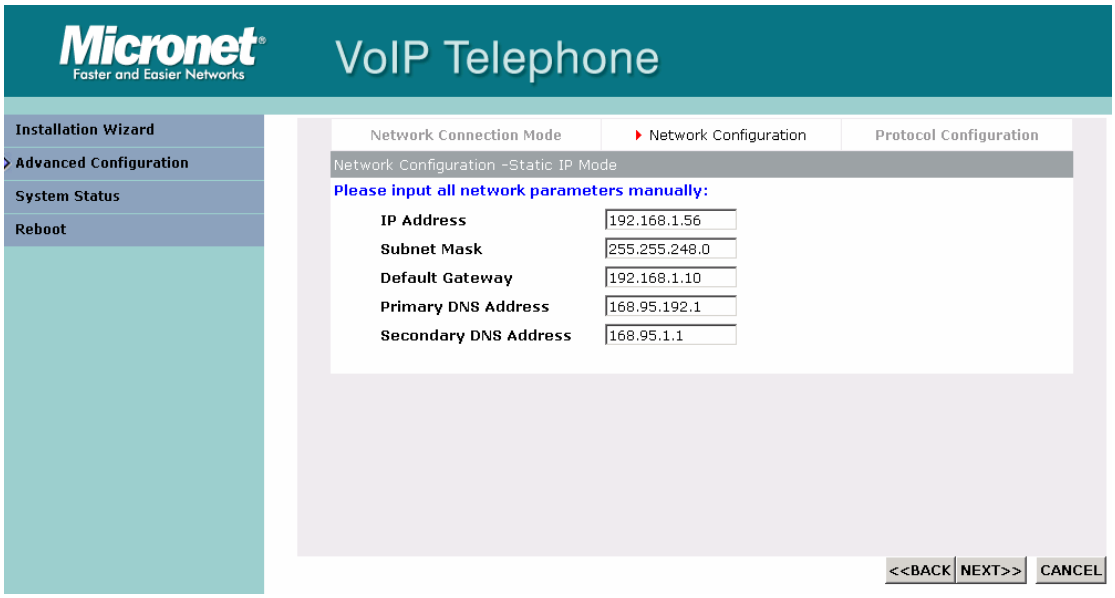


(A) Network Connection Mode: User has to select SIP Phone network mode as Static IP, DHCP or PPPoE.

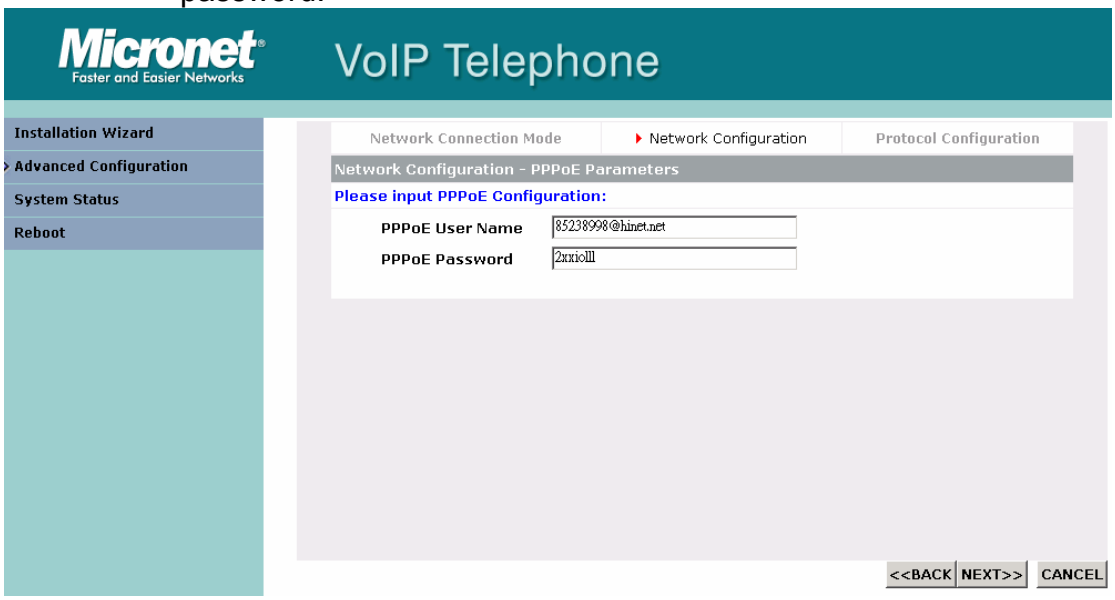


(B) Network Configuration: After selecting network connection mode, user has to input related network parameters.

(1) Static IP: User has to input IP, subnet mask, default gateway, and DNS server address.

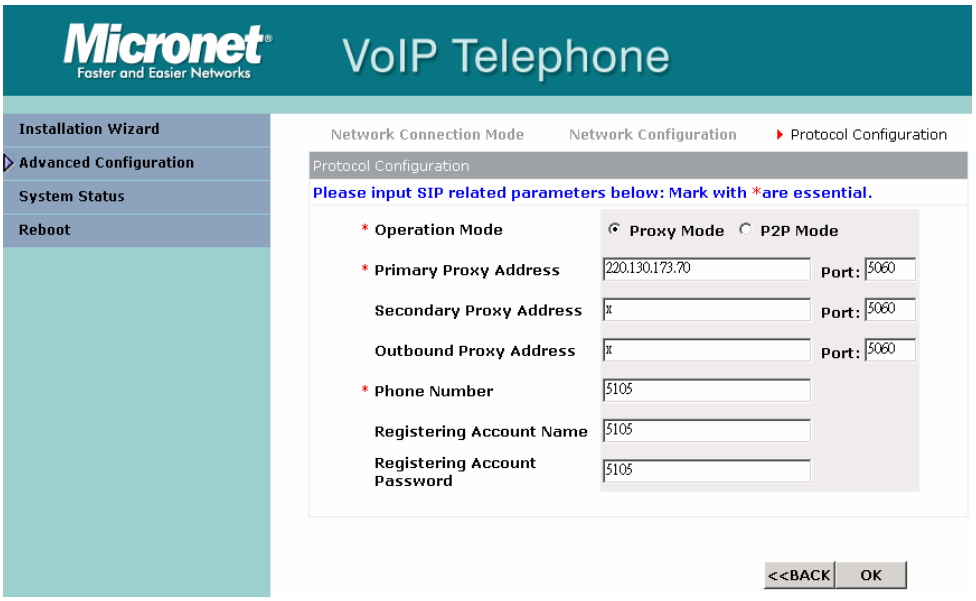


(2) PPPoE: User has to input PPPoE connection user name and password.



(C) Protocol Configuration: After setting network, user has to set SIP related parameters.

- Primary Proxy Address and port: If user select Proxy mode in item A, please input Primary Proxy address and signaling port of Proxy.
- Secondary Proxy Address and port: User can also input secondary Proxy server and port for backup.
- Outbound Proxy Address and port: User can input outbound Proxy and port if necessary.
- Phone Number: Registering Phone number of SIP Phone.
- Registering Account Name and password: If Proxy server need registration authentication please input user name and password here.



5.4 Advanced Configuration

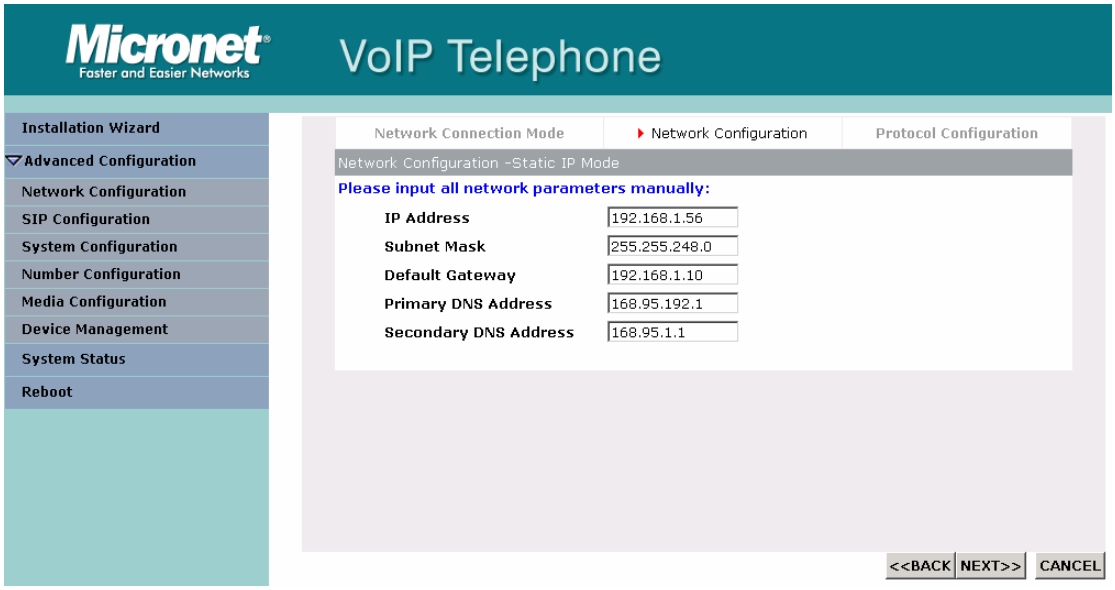
5.4.1 Network Configuration

(1) Network Configuration

Network Connection Mode: User has select network configuration mode first, then configuration page will display related items.

(a) **Static IP:** If user selects Static IP mode, following items will be displayed.

- IP Address: Set IP Address of SIP Phone
- Subnet Mask: Set the Subnet Mask of SIP Phone
- Default Gateway: Set Default routing gateway of SIP Phone
- Primary DNS Address: Set Primary Domain Name Server IP address.
- Secondary DNS Address: Set Secondary Domain Name Server IP address.
- HTTP Port for WEB Management: Set port number for user to configure SIP Phone via WEB management. Default value is 80.



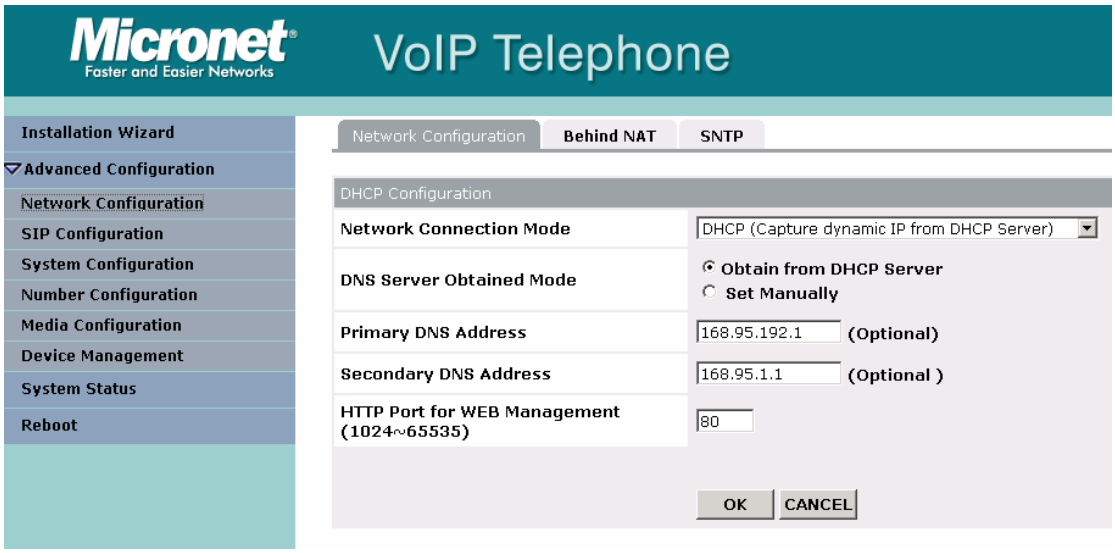
(b) **DHCP**: If user selects DHCP mode, following items will be displayed.

- DNS Server Obtained Mode: When SIP Phone is in DHCP or PPPoE mode, user can determine DNS address is obtained from server or set manually.
- Primary DNS Server: If user determines to set DNS address manually, please set Primary Domain Name Server IP address here.
- Secondary DNS Server: If user determines to set DNS address manually, please set Secondary Domain Name Server IP

address here.

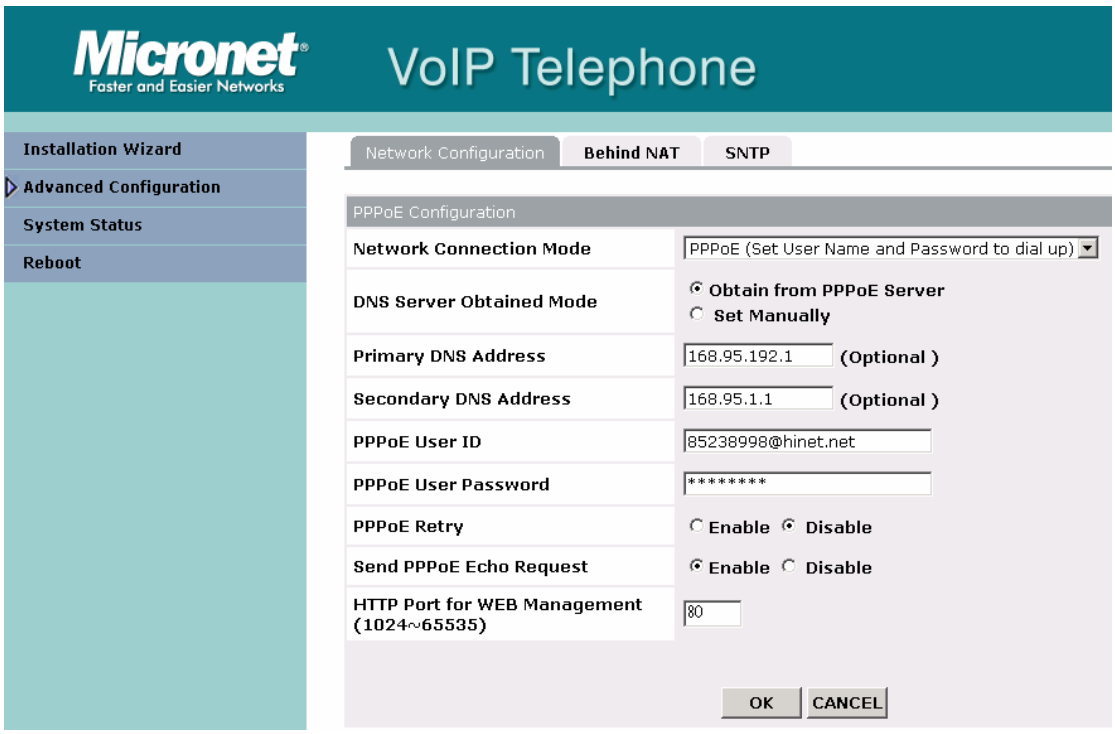
- HTTP Port for WEB Management: Set port number for user to configure SIP Phone via WEB management. Default value is 80.

-



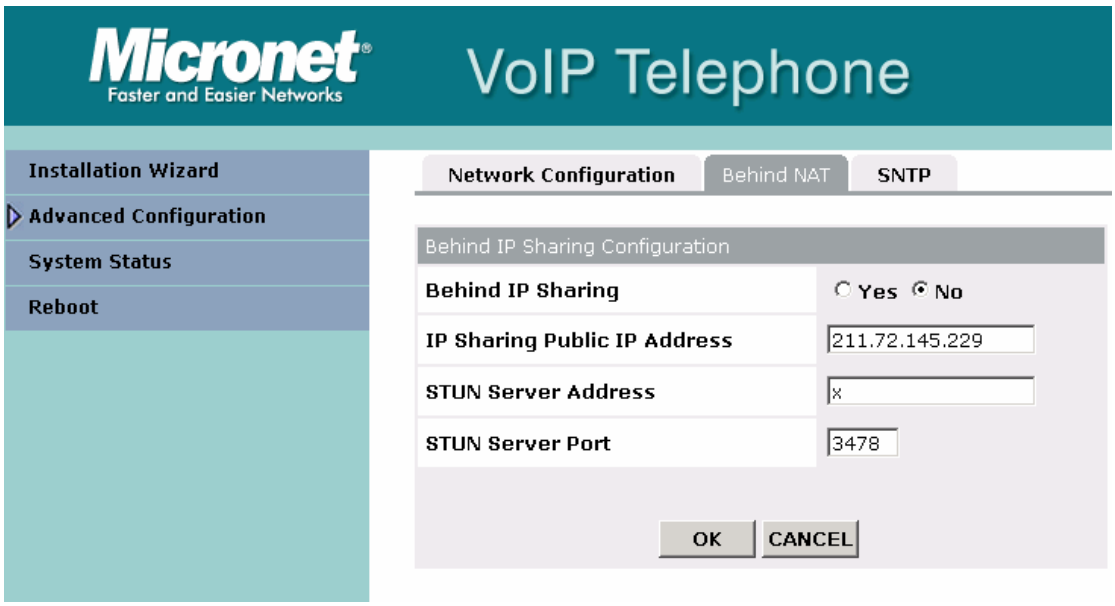
(c) **PPPoE:** If user selects PPPoE mode, following items will be displayed.

- DNS Server Obtained Mode: When SIP Phone is in DHCP or PPPoE mode, user can determine DNS address is obtained from server or set manually.
- Primary DNS Server: If user determines to set DNS address manually, please set Primary Domain Name Server IP address here.
- Secondary DNS Server: If user determines to set DNS address manually, please set Secondary Domain Name Server IP address here.
- PPPoE User ID: Set PPPoE authentication User Name.
- PPPoE User Password: Set PPPoE authentication password.
- PPPoE Retry: Enable/Disable auto-retry function after PPPoE disconnection. If user enables this function, after PPPoE being disconnected, SIP Phone will automatically reboot to re-connect, and after reboot, if SIP Phone still can't get contact with server, SIP Phone will keep trying to connect. After re-connected, SIP Phone will also restart system. On the other hand, if user disables this function, SIP Phone won't reboot and keep trying to connect.
- Send PPPoE Echo Request: Enable or Disable PPPoE Echo function. If user enabled this feature, SIP Phone will send out echo request to check PPPoE connection status. Please notice that if user disables this function, SIP Phone cannot detect if PPPoE connection is still alive or not.
- HTTP Port for WEB Management: Set port number for user to configure SIP Phone via WEB management. Default value is 80.



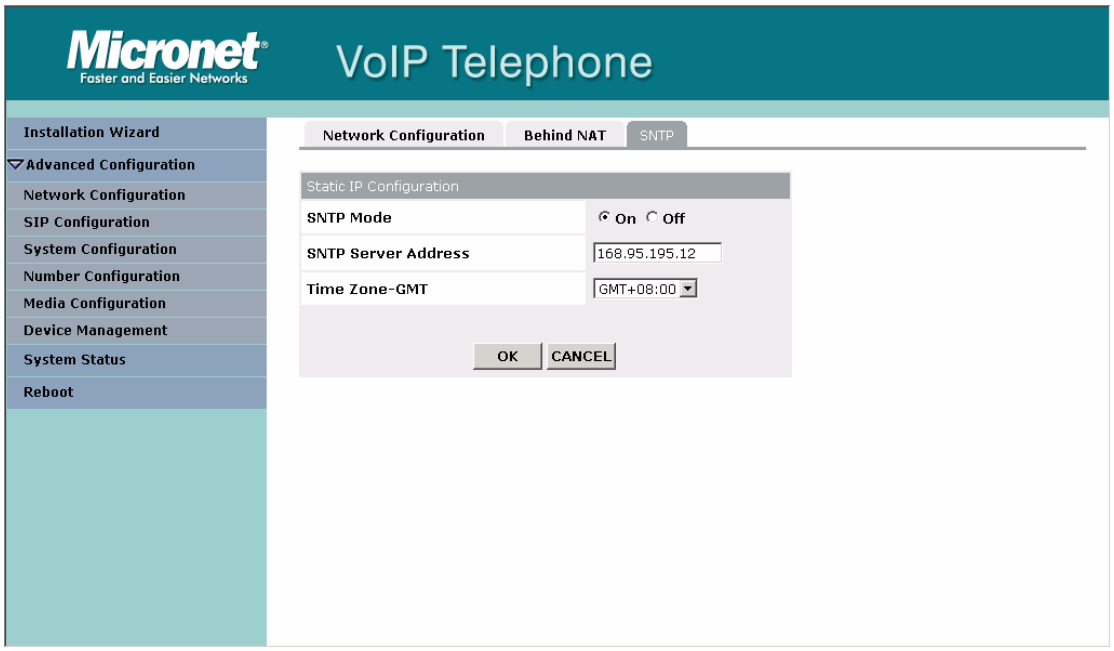
(2) Behind NAT

- Behind IP Sharing: Select if enable SIP Phone behind IP Sharing router function.
- IP Sharing Public IP Address: Set Public IP Address of IP Sharing router for SIP Phone to work behind IP sharing.
- STUN Server address: If user wants to use STUN function, user must enable Behind NAT Device function then inputting STUN Server address.
- STUN Server port: If the STUN server port doesn't any restriction, you don't input any port data.
-



(3) SNTP

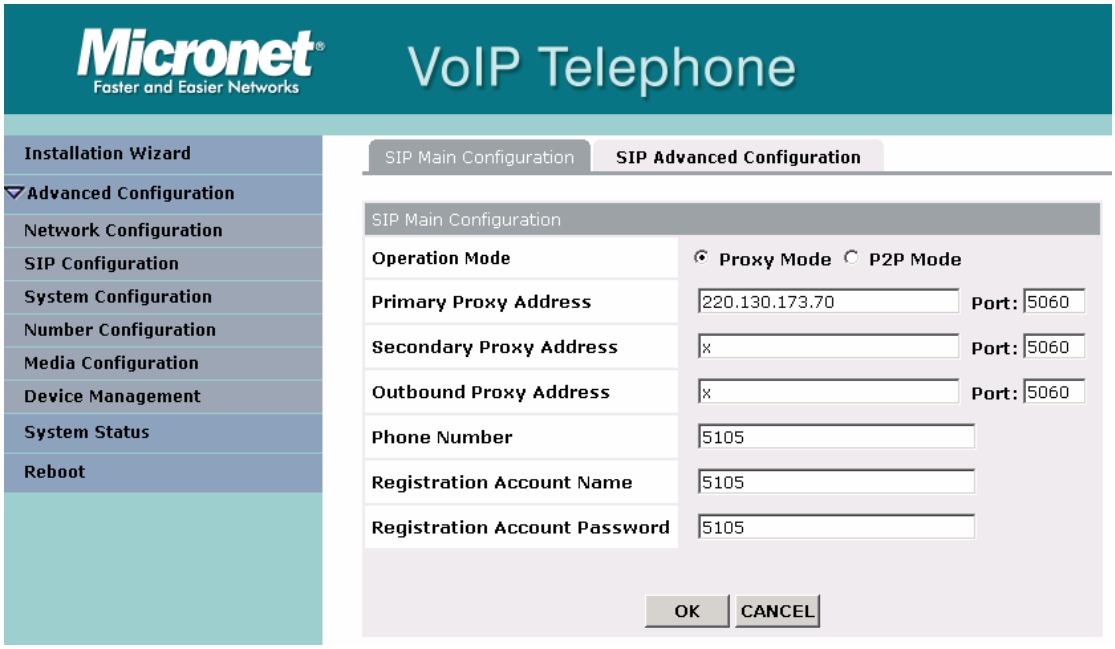
- SNTP Mode: Enable / Disable the Simple Network Time Protocol function
- SNTP Server Address: Set SNTP Server Address.
  When SNTP server is available, enable SIP Phone SNTP function to point to SNTP server IP address so that SIP Phone can get correct current time.
- Time Zone-GMT: Set time zone for SNTP Server time.
  User can set different time zone according to the location of SIP Phone. For example, in Taiwan the time zone should be set as 8,which means GMT+8.



5.4.2 SIP Configuration

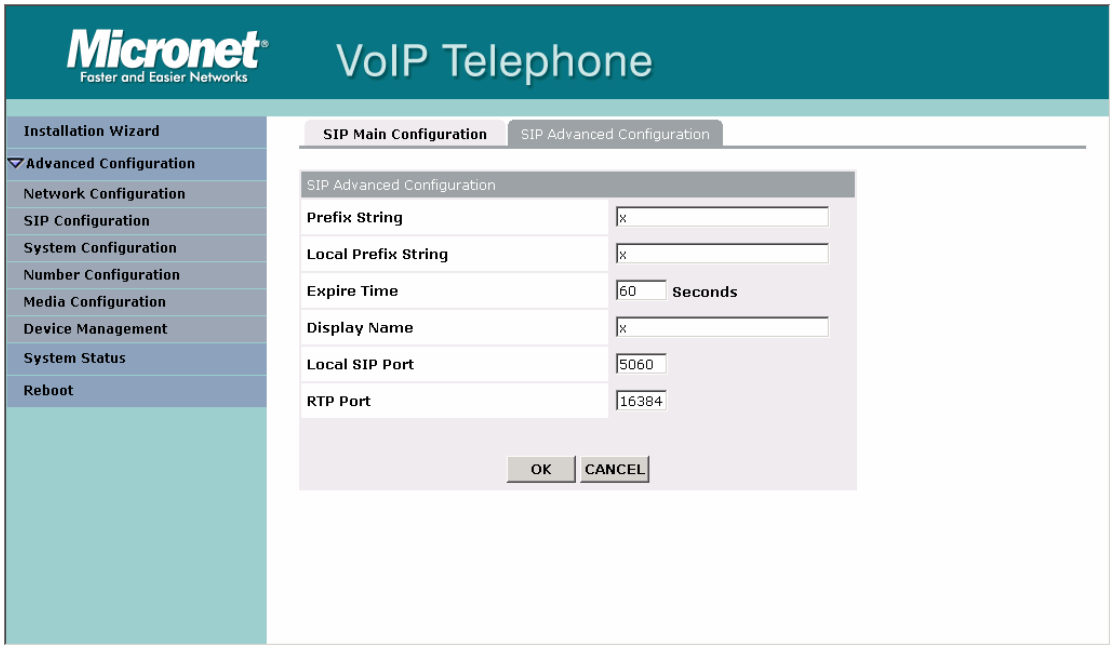
(1) SIP Main Configuration

- Primary Proxy Address and port: If user select Proxy mode in item A, please input Primary Proxy address and signaling port of Proxy.
- Secondary Proxy Address and port: User can also input secondary Proxy server and port for backup.
- Outbound Proxy Address and port: User can input outbound Proxy and port if necessary.
- Phone Number: Registering Phone number of SIP Phone.
- Registering Account Name and password: If Proxy server need registration authentication please input user name and password here. Registering Account Name and password: If Proxy server need registration authentication please input user name and password here.



(2) SIP Advanced Configuration

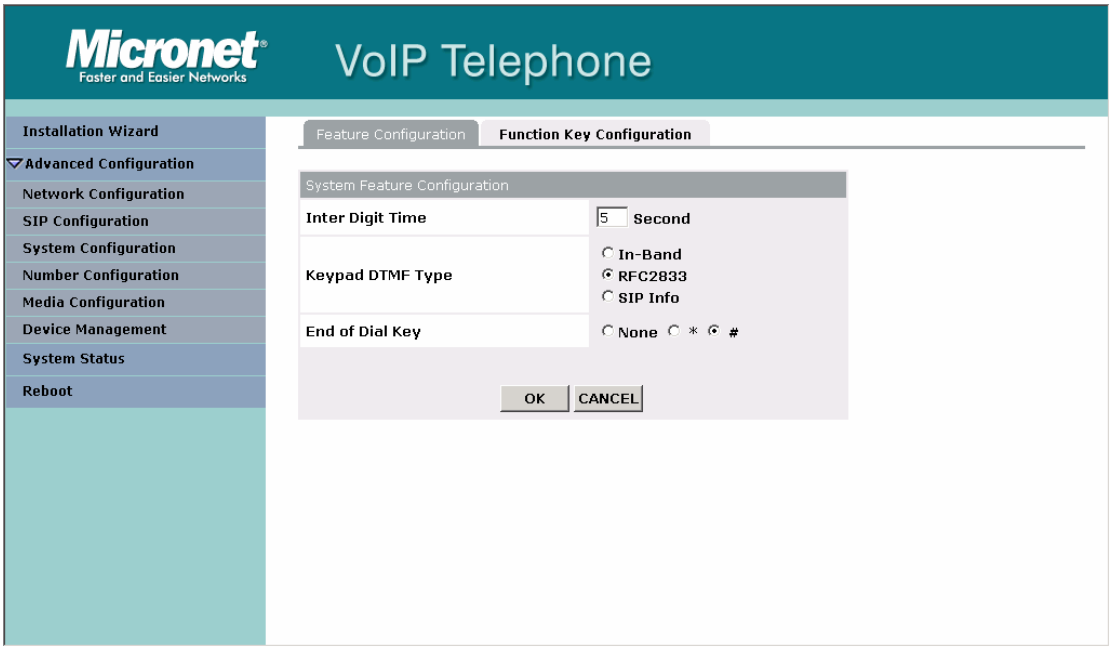
- Prefix String: set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, SIP Phone will sent out messages as Account Name @"IP address of Proxy", if user set prefix as abc, SIP Phone will set out as abc123@"IP address of Proxy". This function is for special proxy server.
- Expire Time: Set expire time of registration. SIP Phone will keep re-registering to proxy server before expire timed out
- Display Name: Set SIP Phone display name for caller ID information.
- Local SIP Port: Set SIP UDP port.
- RTP Port: Set RTP port for sending voice data.



5.4.3 System Configuration

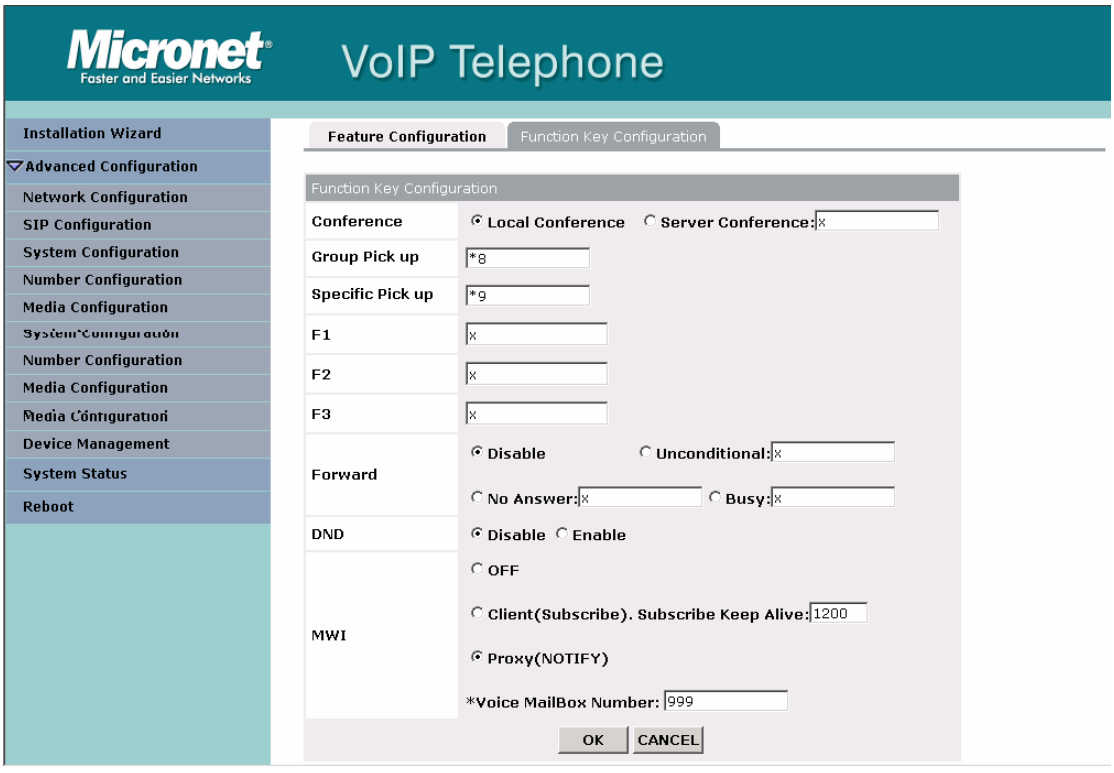
(1) Feature Configuration:

- Inter Digit Time: Set the DTMF inter digit time (second). To set the duration (in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, SIP Phone will dial out all number pressed.
- Keypad DTMF Type: set DTMF type. User can select DTMF type SIP Phone transmits.
- End of Dial Key: select end of dialing key, e.g. set end of dial key as * button, after finished pressing dialing number then press * will dial out.



(2) Functional Key Configuration:

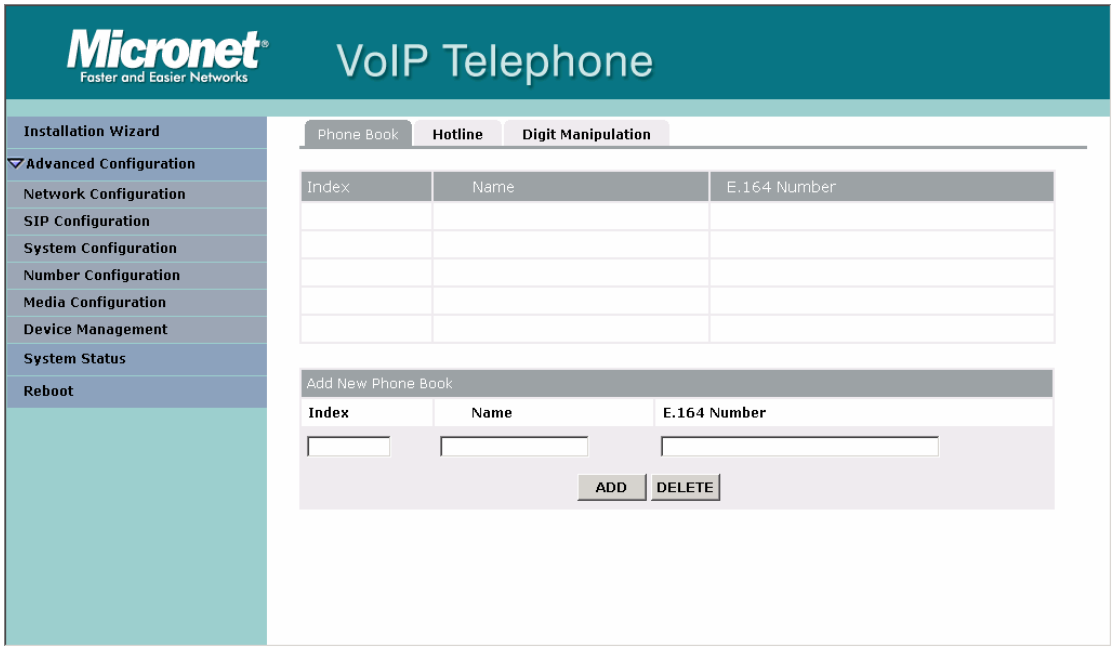
- Conference : Set the Conference mode. Default: Local Conference. But now it is not support Server Conference mode yet!
- Group Pick up: Set Group Pick up code, you might contact with your IP-PBX system administrator.
- Specific Pick up: Set Specific Pick up code, you might contact with your IP-PBX system administrator.
- F1: User-defined function key.
- F2: User-defined function key.
- F3: User-defined function key.
- Forward: “Local forward”. Set Forward type and number.
- DND: ”Local Do Not Disturb”
- MWI: Set support which type Voice Mail. Default: Proxy(NOTIFY).



5.4.4 Number Configuration

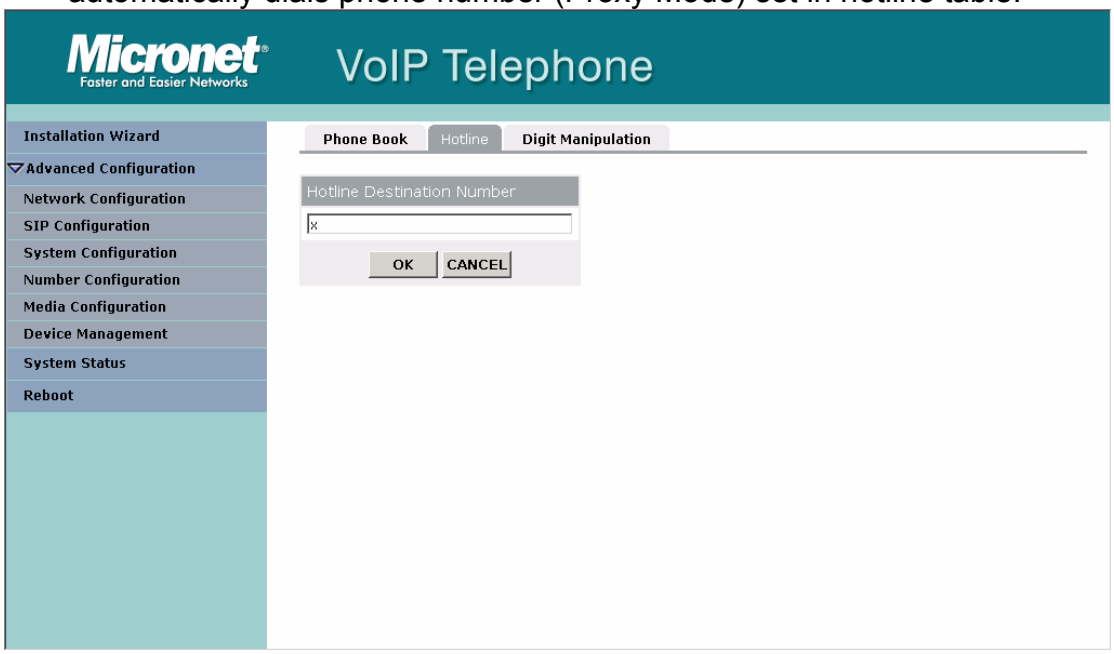
(1) Phone Book

- Add Data: User can specify 50 sets of phone book via web interface. Please input index, Name, E.164 number, IP Address, port of the destination device, drop prefix, and insert number.
  1. name: Specify name for one pbook data
  2. e.164: set phone number of callee.
- Delete Data: User can delete any configured phone book data by index.



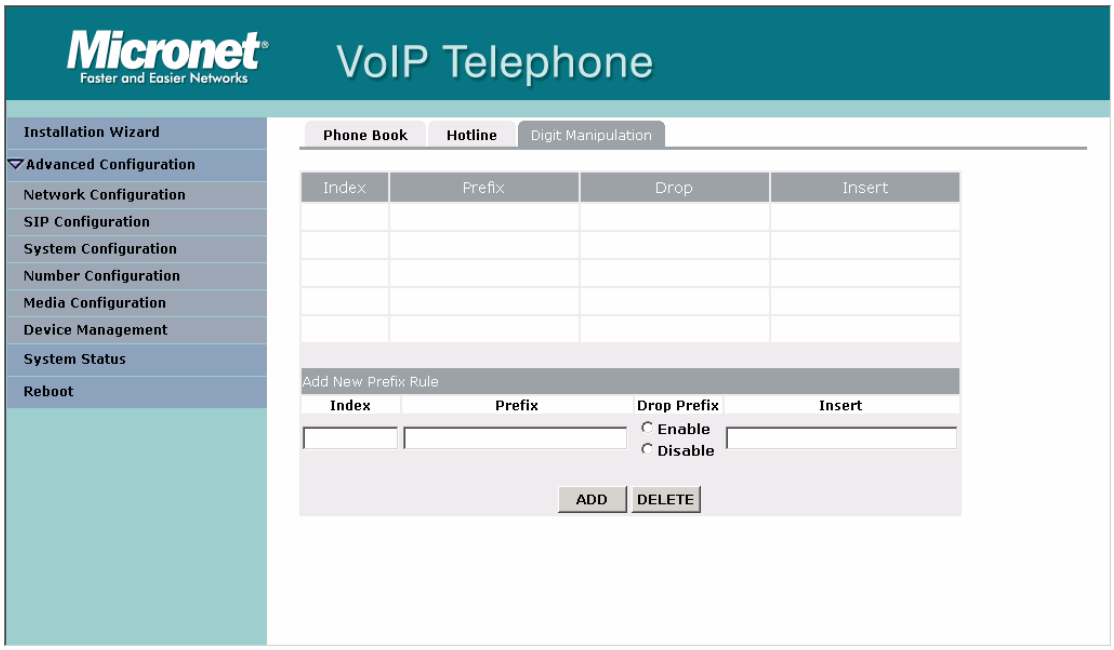
(2) Hotline

If user set SIP Phone as hotline mode, once SIP Phone is off-hook, it will automatically dials phone number (Proxy Mode) set in hotline table.



(3) Digit Manipulation

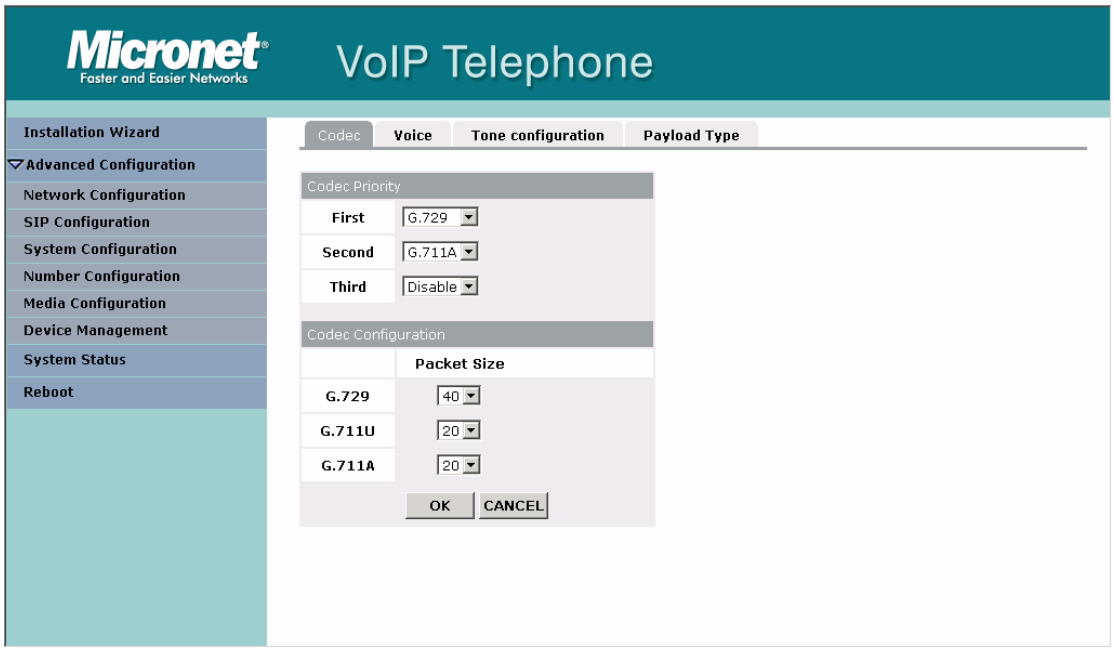
- add: Add a rule to drop or insert prefix digits of incoming call.
- prefix: Set which prefix number to implement digit manipulation rule.
- drop: Enable or disable drop function. If this function is enabled, Phone will drop prefix number on incoming call.
- insert: Set which digit to insert.
- delete: Delete a digit manipulation rule by index.



5.4.5 Media Configuration

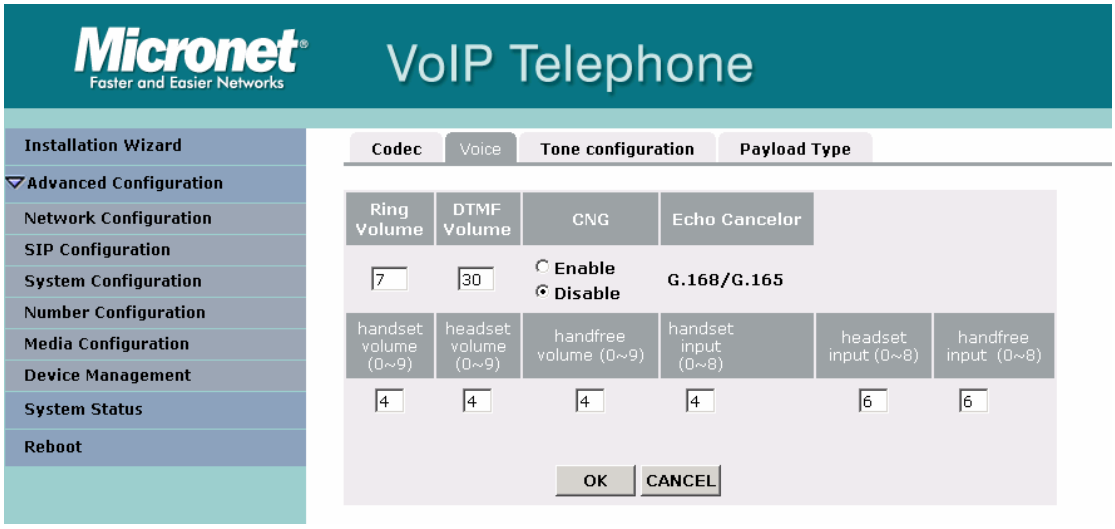
(1) Codec

- Codec Priority: set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need. For example, user can only set first priority as G.723.1, and set the others as x, that means only G.723.1 is available.
- Packet Size: User can set different packet size for each codec.



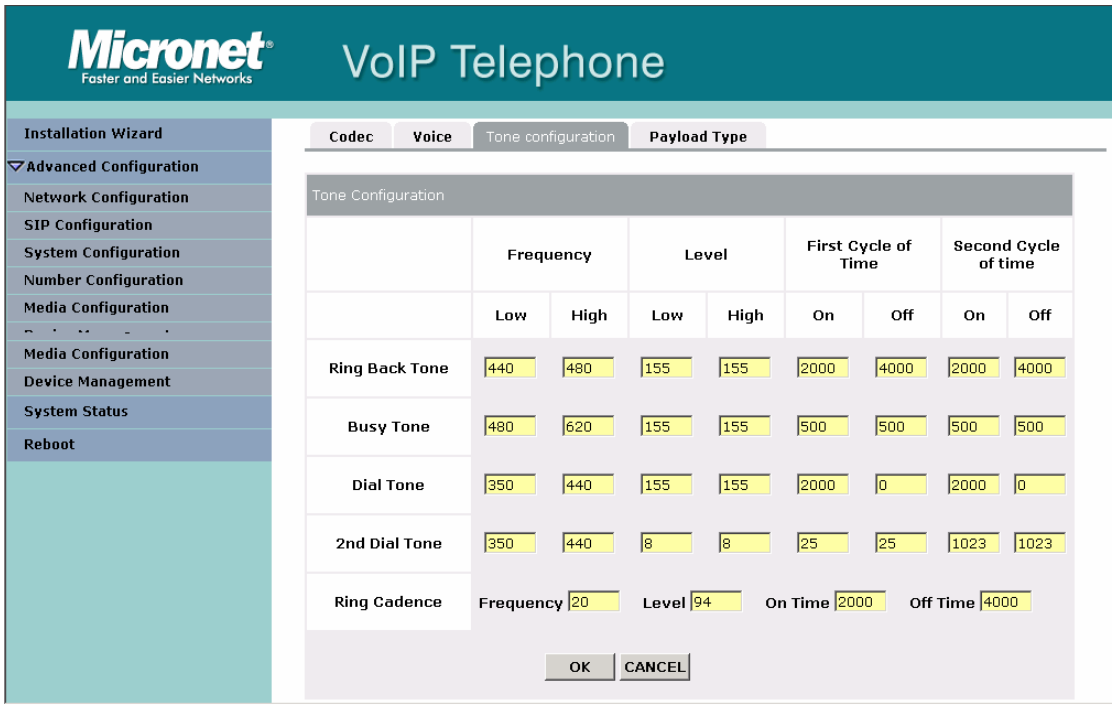
(2) Voice

- Volume: Adjust the volume of Ringer, Receive (Local side hearing), transmit (remote side hearing), DTMF.
- Echo Cancel: Enable / Disable (it is suggested to always Enable this function).



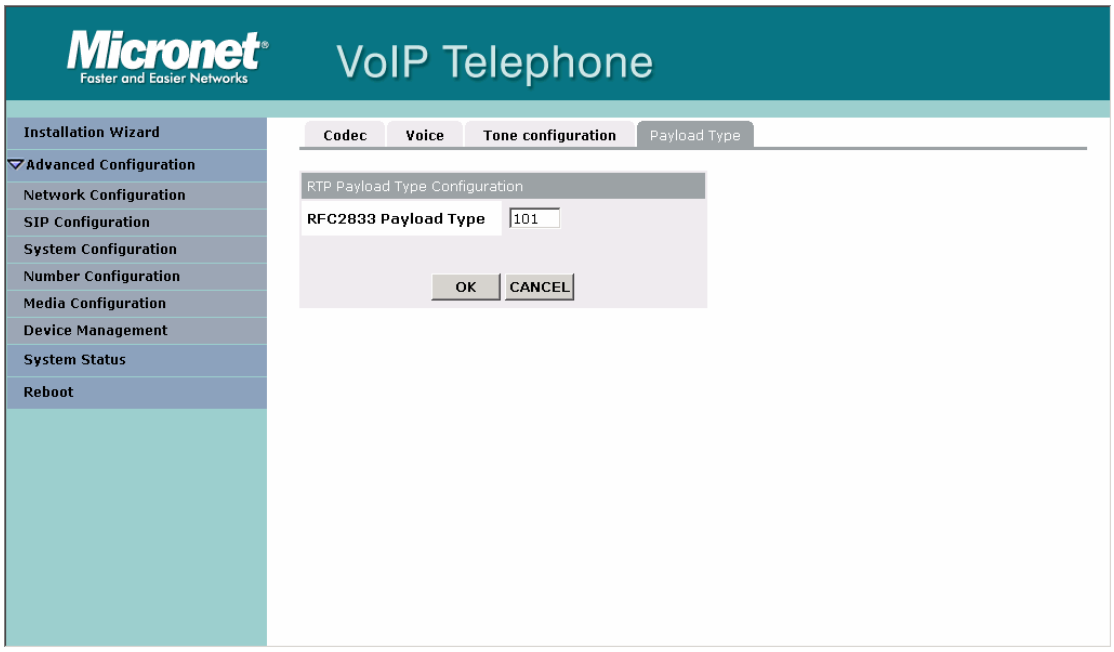
(3) Tone Configuration

- Ring Back Tone: Set ring back tone parameters.
- Busy Tone: Set busy tone parameters.
- Dial Tone: Set dial tone parameters.
- 2nd Dial Tone: Set 2nd dial tone parameters.
- Ring Tone: Set ring tone parameters.



(4) Payload Type

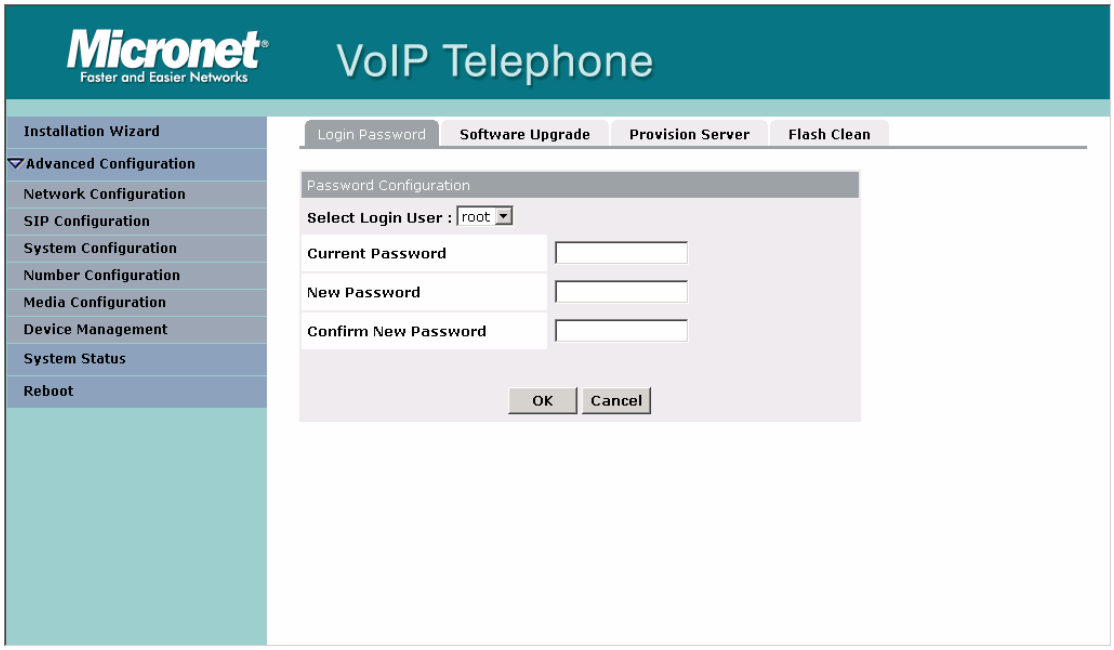
RFC2833 Payload Type: Change RFC2833 Payload type. This is for special request from the other site, if RFC2833 payload types of 2 sites are different, it may cause some problem of connection.



5.4.6 Device Management

(1) Login Password

Change password configuration: First select login name as root or user, then enter current password, new password and confirm new password again to set new password.



(2) Software Upgrade

Download Mode: Select download method as TFTP or FTP

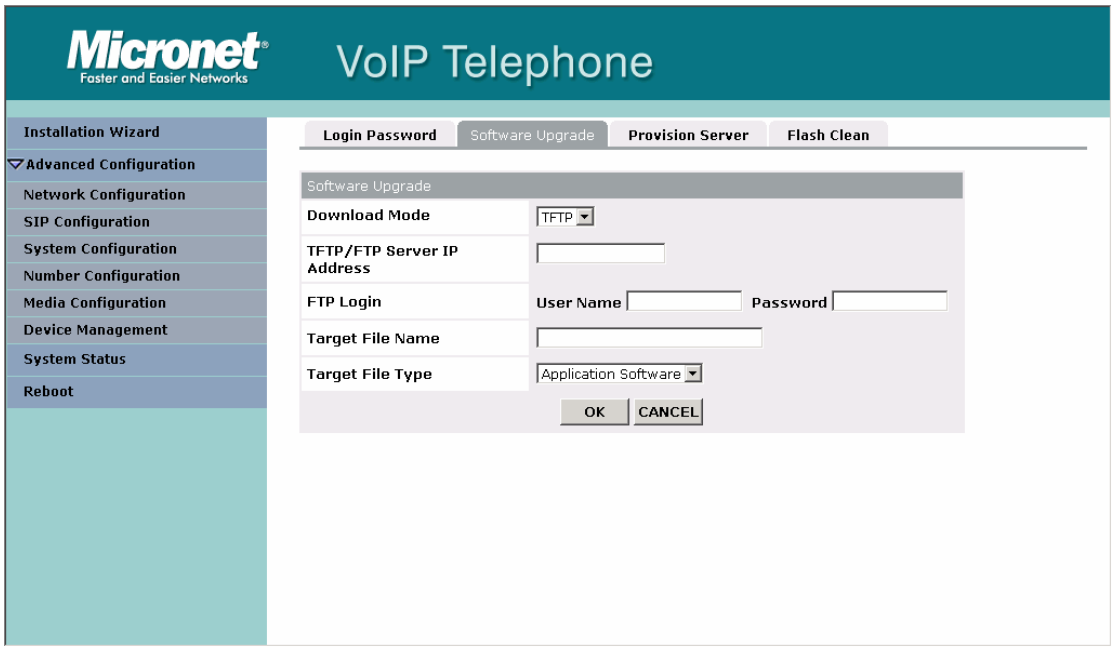
TFTP/FTP Server IP Address: Set TFTP server IP address

FTP Login: Set FTP login name and password

Target File name: Set file name prepared to upgrade

Target File Type: Select which sector of SIP Phone to upgrade

> Note:
> After upgrade Application, please remember to execute Flash Clean, which will clean all configurations become factory values except Network settings.



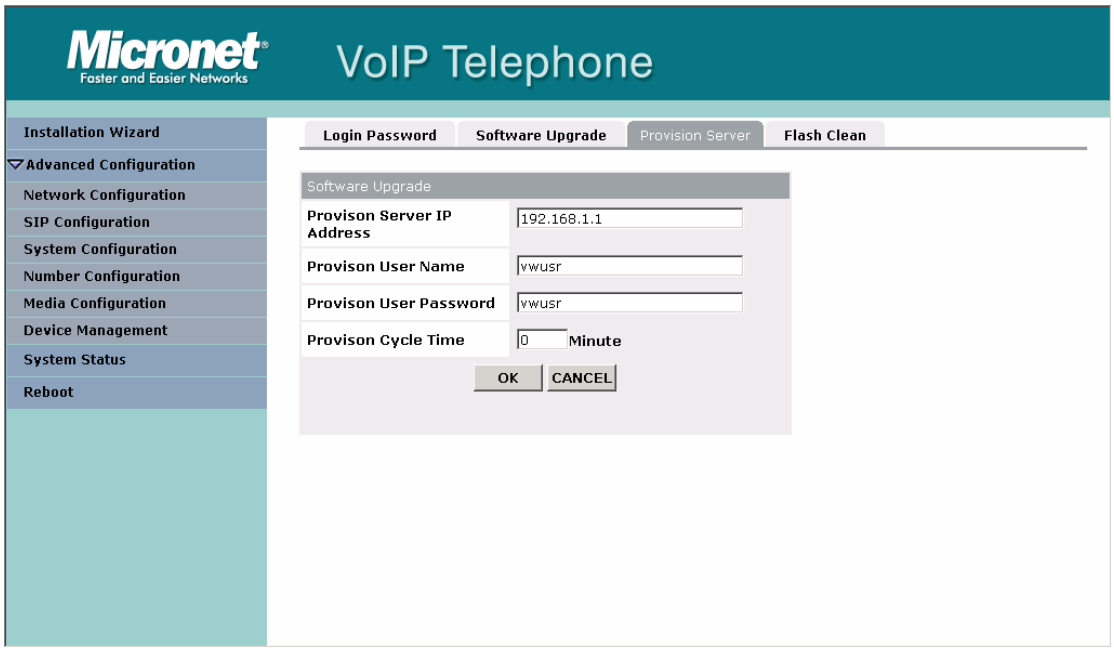
(3) Provision Server

Provision Server Address: set Provision server IP address.

Provision Server Login User Name: set Provision Server login user ID.

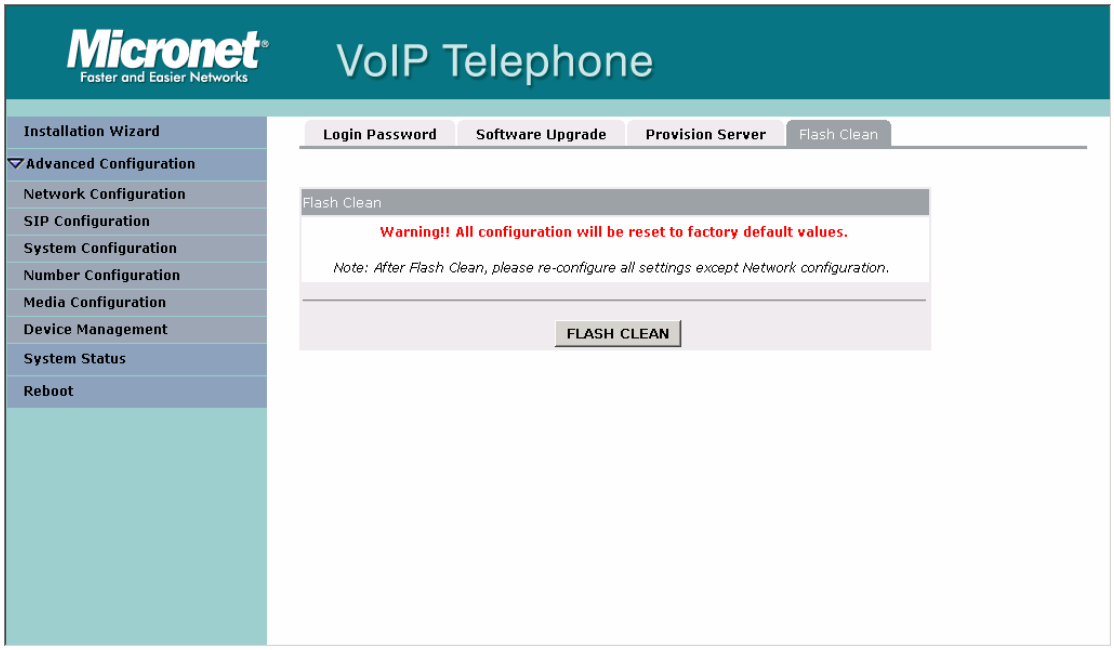
Provision Server Login User Password: set Provision Server login user password.

Provision Server Cycle Time: set Provision Server update cycle time.



(4) Flash Clean

Press CLEAN will clean all configurations except Network Configuration of SIP Phone and reset to factory default value.

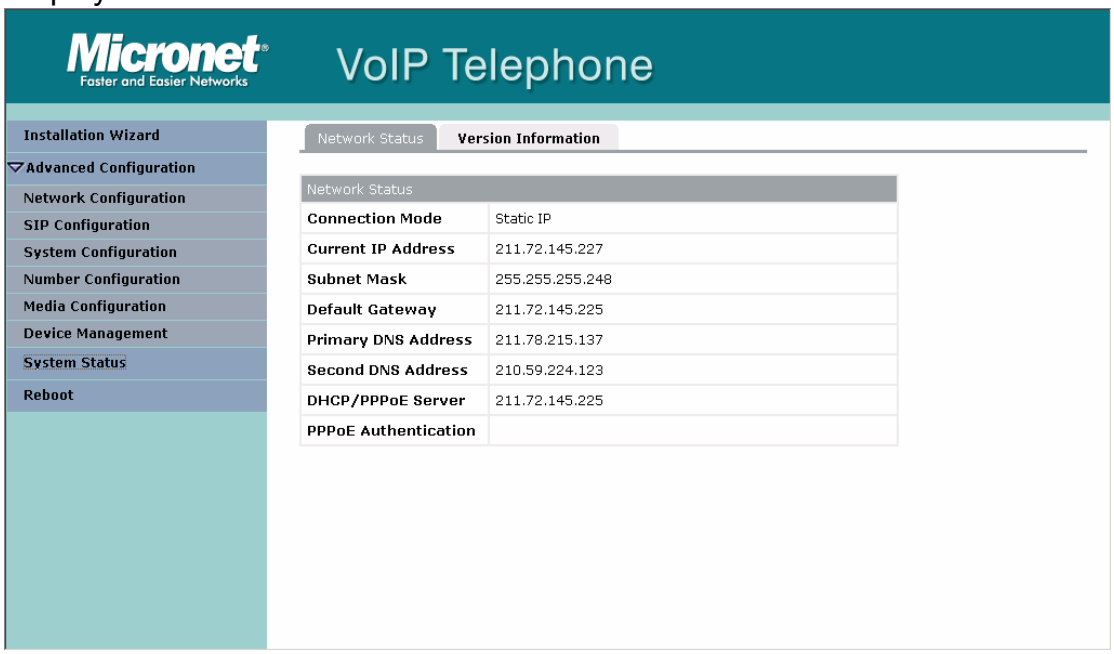


Note: User must re-configure all commands all over again (except Network Configuration).

5.5 System Status

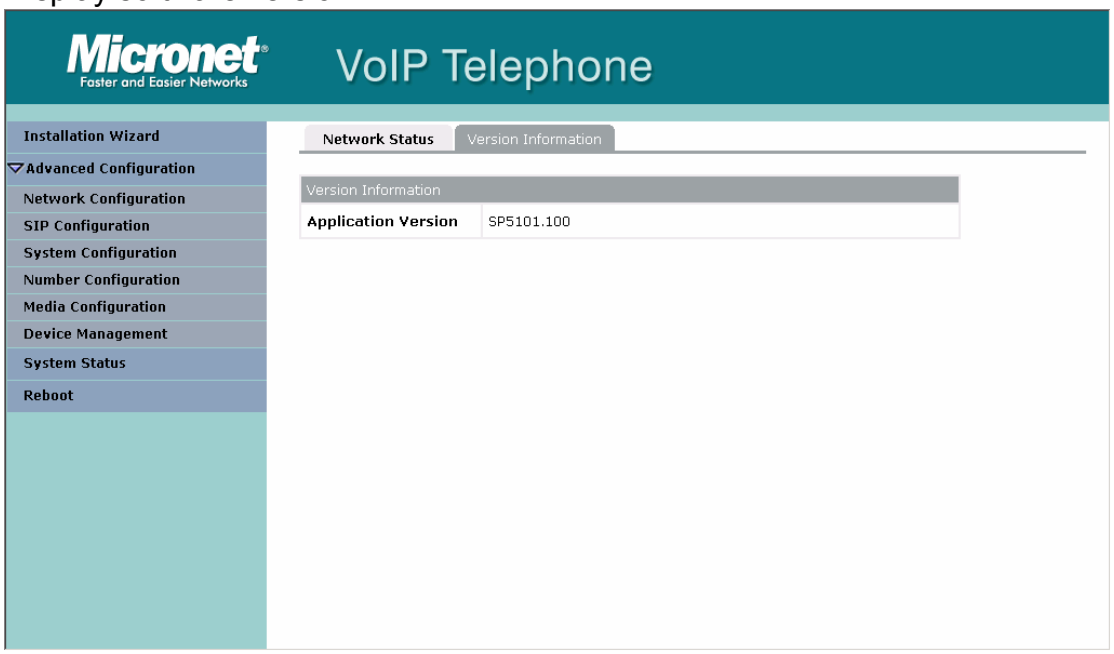
5.5.1 Network Status

Display all current network status of SIP Phone.



5.5.2 Version Information:

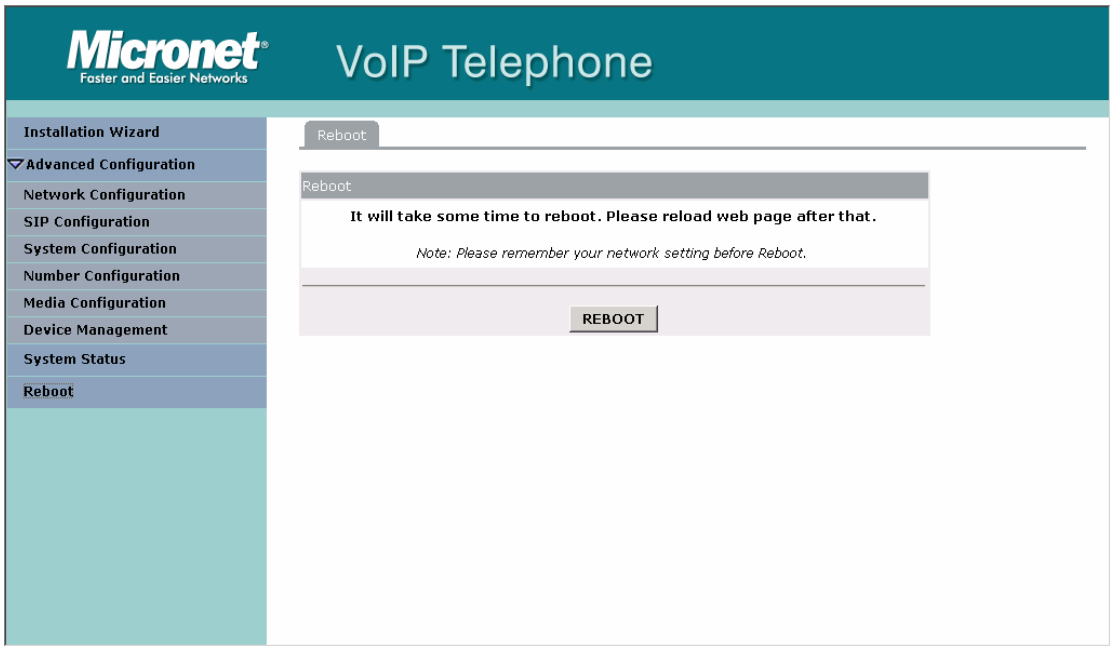
Display software version.



5.6 Reboot

Press reboot will reset SIP Phone.

Note: To execute reboot via web browser, SIP Phone will automatically save all data before reboot. To execute reboot via TELNET command, please remember to do Commit Data before Reboot System.



6. Telnet command lines

After setting the IP Address of SIP Phone and reboot, (please refer to LCD Menu: 5-3.4.5), user can enter into Telnet command lines.

Note:

1. After user enter SIP Phone configuration via telnet, please use login: ”root”, password: null, press enter to enter command lines. If user forgets password, please contact with the distributor, we will generate a specific password according to your MAC address of SIP Phone. MAC address is on a label at the bottom of your case in format with “0001a8xxxxxx”.
2. User must input lower-case command, but contents of configurations such as SIP alias or user name etc, user can set as capital case.
3. If user wants to disable or clean any input data, please set value as “x”.
4. After any change of configuration, please remember to do commit command to save changes and then reboot command to reboot system.

Telnet Commands:

	Command	Description
1	help	help/man/? [command]
2	quit	quit/exit/close
3	debug	show debug message
4	reboot	reboot local machine
5	pbook	Phonebook information and configuration
6	commit	commit flash rom data
7	ping	test that a remote host is reachable
8	time	show current time
9	ifaddr	internet address manipulation
10	pppoe	PPPoE stack manipulation
11	flash	clean configuration from flash rom
12	sysconf	System information manipulation
13	sip	SIP information manipulation
14	security	Security information manipulation
15	line	Line information manipulation
16	voice	Voice information manipulation
17	phone	Setup of call progress tones and ringing
18	tos	IP Packet ToS (Type of Service)values
19	prefix	Prefix drop/insert information manipulation
20	rom	ROM file update
21	auth	Set configuration items for “administrator” user.
22	passwd	Password setting information and configuration

1. After setting the IP Address of SIP Phone SP5102 and reboot, (please refer to LCD Menu), user can enter Telnet command lines.

Note:

1. After user enter SIP Phone configuration via telnet, please use login: **"root"**, password: null, press enter to enter command lines.
2. If user forget login password, please contact with your distributor, we will generate one new password according to LAN Phone's MAC address. Please login with **"mac"** and this new password.
3. User must input lower-case command, but contents of configurations such as SIP alias or user name etc, user can set as capital case.
4. After any change of configuration, please remember to do commit command to save changes and then reboot command to reboot system.

6.1 [help] command

Type **help** or **man** or **?** to display all the command lists. The following figure is shown all commands of SIP Phone SP5102.

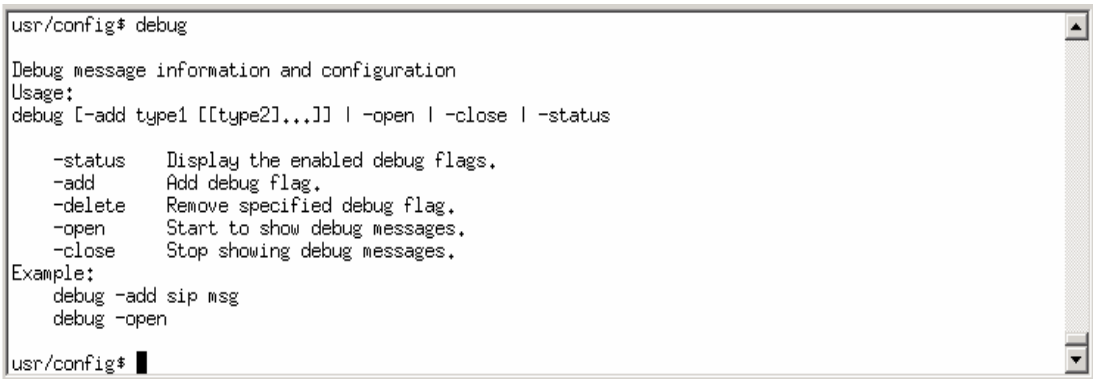
	Command	Description
1	help	help/man/? [command]
2	quit	quit/exit/close
3	debug	show debug message
4	reboot	reboot local machine
5	pbook	Phonebook information and configuration
6	commit	commit flash rom data
7	ping	test that a remote host is reachable
8	time	show current time
9	ifaddr	internet address manipulation
10	pppoe	PPPoE stack manipulation
11	flash	clean configuration from flash rom
12	sysconf	System information manipulation
13	sip	SIP information manipulation
14	security	Security information manipulation
15	line	Line information manipulation
16	voice	Voice information manipulation
17	phone	Setup of call progress tones and ringing
18	tos	IP Packet ToS (Type of Service)values
19	prefix	Prefix drop/insert information manipulation
20	rom	ROM file update
21	auth	Set configuration items for "administrator" user.
22	passwd	Password setting information and configuration

6.2 [quit] command

Type **quit/exit/close** will logout SIP Phone SP5102 and Telnet Program.

6.3 [debug] command

This command is for engineers to debug system of SIP Phone SP5102. User can add debug flag via command **debug –add "debug flags"**, and then start debug function via command **debug –open**. When SIP Phone SP5102 is working on screen will display related debug messages. Most frequently used debug flag are "sip", "fsm", "msg"…etc.



6.4 [reboot] command

After typing **commit** command, type **reboot** to restart the SIP Phone SP5102.

6.5 [pbook] command

SIP Phone SP5102 can support 90 phone book data.

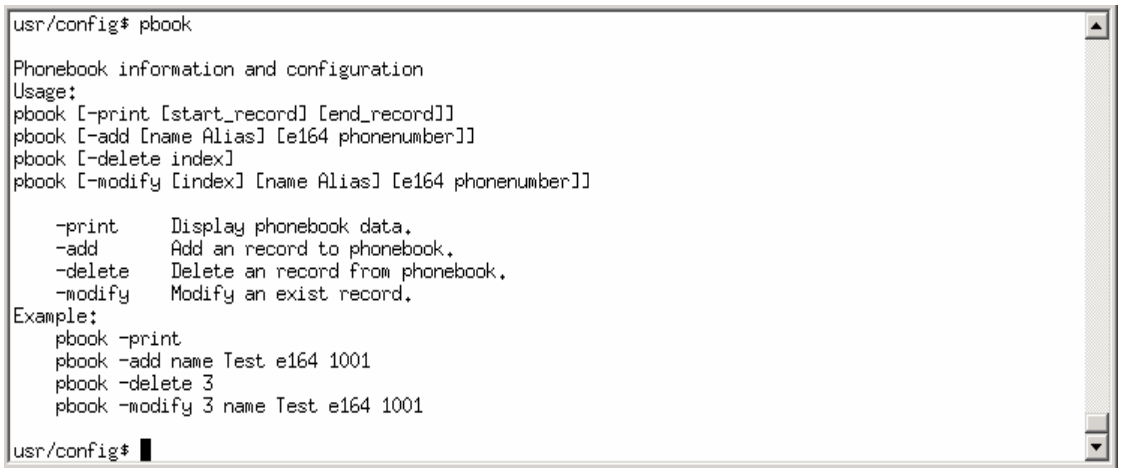
1. **-print:** display phone book data. User can print all data in phone book by command (**pbook –print**).

2. **-add**: add a new record in phone book table by giving name and e.164 number of callee endpoint.**(pbook –add name "X" e164 "X")**

3. **-delete**: delete a record of certain listed index in phone book table. (**pbook –delete "index number"**)

4. **-modify**: modify record of a certain index in phone book .

(pbook –modify "index" name "X" e164 "X")

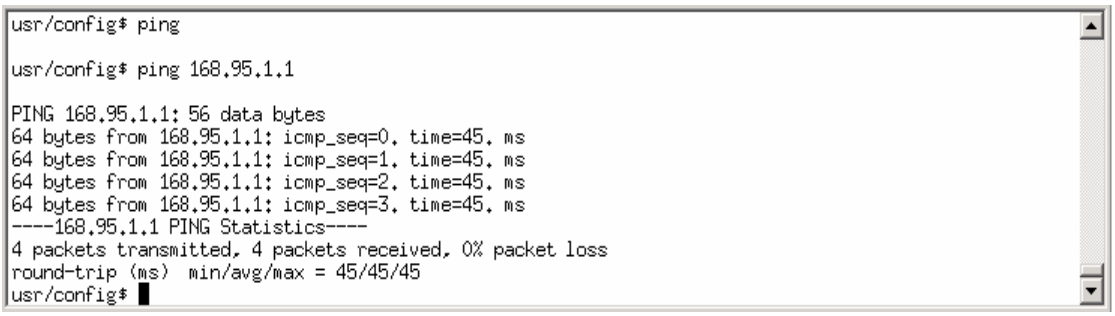


6.6 [commit] command

Save any changes after configuring the SIP Phone SP5102.

6.7 [ping] command

Command **ping** can test which the IP address is reachable or not.
Usage: ping “IP address”
The message will display packets transmitting condition or no answer from the IP address.



6.8 [time] command

When SIP Phone SP5102 enable SNTP function and be able to connect with SNTP server, type **time** command will show the current time retrieved from SNTP server.

6.9 [ifaddr] command

Configure and display the SIP Phone SP5102 IP information.

1. **–print:** print out all current configurations of ifaddr command.
2. **-ip, -mask, -gate:** Set SIP Phone SP5102 IP Address, subnet mask and default gateway respectively.
3. **-ipmode:** Set SIP Phone network mode to be Fixed IP, DHCP or PPPoE.

   When User set IP mode to be fixed IP, please set IP, subnet Mask, default gateway as mentioned in item 2.

   If User set IP mode to be DHCP, SIP Phone will search for DHCP server to capture IP address after reboot.

   If user set IP mode to be PPPoE, please remember to set related parameters under **[pppoe] command**.
4. **-sntp:** When SNTP server is available, enable SIP Phone SP5102 SNTP function and assign SNTP server IP address so that SIP Phone can capture current time from SNTP server. **(ifaddr –sntp 1 "xxx.xxx.xxx.xxx**")
5. **–autodns:** specify the way to obtain DNS server address. When phone is under DHCP or PPPoE mode, user can let phone to capture DNS server address from server automatically or specify the address manually.
6. **-dns:** User can set primary and secondary Domain Name Server IP address. Once SIP Phone can connect with DNS server, user can specify URL address instead of IP address for Proxy Server and phone book IP address...etc. **(ifaddr –dns 1 "primary DNS server address**" **–dns 2 "secondary DNS server address**")
7. **–timezone:** User can set different time zone according to the location SIP Phone is. For example, in Taiwan the time zone should be set as 8,which means GMT+8. (GMT+8: **ifaddr –timezone 8**)
8. **–timeformat**: Set time display format as 12 or 24 hours. (**ifaddr –timeformat 0/1**, 0 as 24 hours, 1 as 12 houts)
9. **-ipsharing:** If SIP Phone SP5102 is behind a IP-sharing , user must enable IP sharing function and specify public IP address.(**ifaddr –ipsharing 0/1 "public IP address of IP sharing" ,** 0 for disable and 1 for enable)
10. **–server:** Set Provision Server address.
11. **–id:** Input ID of Provision server.
12. **–pwd:** Input Password of Provision server.
13. **–emstime:** Set provision cycle time.
14. **–stun:** Input STUN server address. (ifaddr –stun 1 61.220.2.2) But you must take notice when you use this function, you must enable

"ipsharing" function.

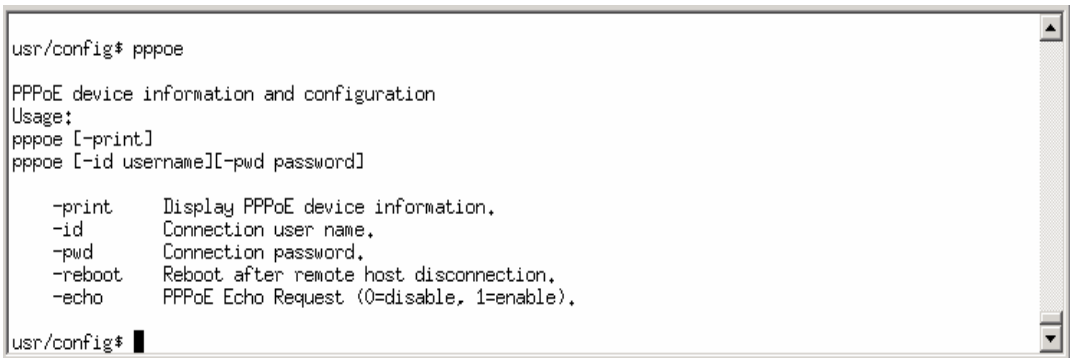
15. **–stunport:** Input port of STUN server.

> ***Note:***
> Some Proxy servers support endpoint behind NAT function, in this case SIP Phone doesn't have to enable IP sharing function, please contact with your Proxy Server vendor for detail information.

```
usr/config$ ifaddr

LAN information and configuration
Usage:
ifaddr [-print]|[-dhcp used]|[-sntp mode [server]]
ifaddr [-ip ipaddress] [-mask subnetmask] [-gate defaultgateway]
ifaddr [-dns index [dns server address]]
ifaddr [-autodns used]

    -print     Display LAN information and configuration.
    -ip        Specify WAN ip address.
    -mask      Set Internet subnet mask.
    -gate      Specify default gateway ip address
    -ipmode    Set get IP mode(0:Fixed IP/1:DHCP/2:PPPoE)
    -sntp      Set SNTP server mode and specify IP address.
    -autodns   Specify the way to obtain DNS Server (0:Manual/1:Auto).
    -dns       specify IP address of DNS Server.
    -timezone  Set local timezone.
    -timeformat Set time format(1:12/0:24)
    -ipsharing Specify usage of an IP sharing device and specify IP address.
    -server    Specify Provison Server IP address
    -id        Specify Provison Server ID
    -pwd       Specify Provison Server password
    -emstime   Specify Provison cycle time
    -stun      Specify STUN Server address(x:disable STUN).
    -stunport  Specify STUN Server port.
Note:
    Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
    SNTP mode (0=no update, 1=specify server IP, 2=broadcast mode).
Example:
    ifaddr -ip 210.59.163.202 -mask 255.255.255.0 -gate 210.59.163.254
    ifaddr -ipmode 1
    ifaddr -sntp 1 210.59.163.254
    ifaddr -ipsharing 1 210.59.163.254
    ifaddr -dns 1 168.95.192.1
    ifaddr -autodns 1

usr/config$
```

6.10 [pppoe] command

1. **-print:** display all current configurations and information.
2. **–id:** to set PPPoE authentication user name.
3. **–pwd:** to set PPPoE authentication password.
4. **–reboot:** Select enable or disable this function. If user enables this function, after PPPoE disconnected, SIP Phone will automatically reboot to re-connect, and after reboot, if SIP Phone still can't connect with server, SIP Phone will keep trying to connect. On the other hand, if user disables this function, SIP Phone won't reboot and keep trying to connect. (**pppoe –reboot 0/1**)
5. **–echo**: to set PPPoE send echo request function or not. Under some ISP sending echo request will cause abnormal behavior for LAN Phone, however, if user disable echo function, when ISP disconnect, LAN Phone will not try to reconnect. Suggest for most ISPs this function need to be enabled. Please refer to **pppoe –reboot** function.

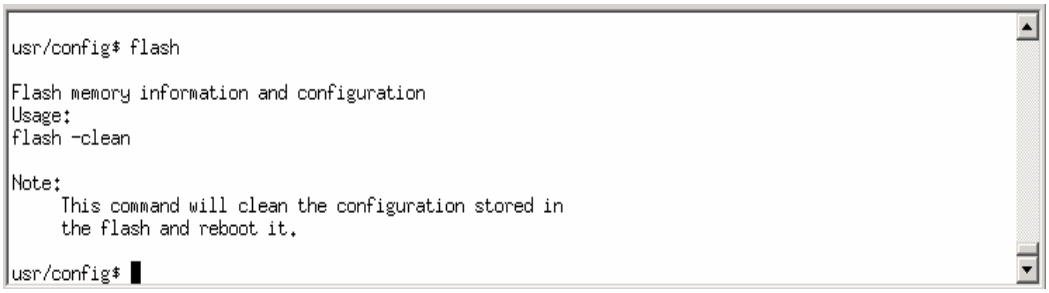


6.11 [flash] command

This command will clean the configuration stored in the flash rom to default value and reboot the SIP Phone SP5102.

Note:

1. After user upgrade new software version, suggested to execute this command to make sure new software work well on SIP Phone SP5102.
2. To execute the command **flash –clean,** all configuration of SIP Phone SP5102 stored in flash will be cleaned. It is authorized for the user whose login name is “root” only.



6.12 [sysconf] command

1. **-print:** display all current configurations.
2. **-idtime:** set the duration(in second) of two pressed digits in dial mode as timed out. If after the duration user hasn't pressed next number, SIP Phone will dial out all number pressed.
3. **-keypad:** set DTMF type .User can select DTMF type SIP Phone receive and transmit.(**sysconf –keypad 0/1 ,** 0 for in band ,1 for RFC2833.)
4. **-2833type:** change RFC2833 Payload type.
5. **-eod**: select end of dialing key, e.g. set end of dial key as "＊" button , after finished pressing dialing number then press "＊" will dial out. (**sysconf –eod 0/1/2** , 0 for no end of dial key , 1 for "＊" button, 2 for "#" button)
6. **–privacy**: this function can only work in Japan and also user's service platform supports Japan standard telecom CLIR specification. When this function is set as Japan mode, other users can hide their caller ID by press special code before dial out phone number.
7. **–phone:** set in SIP message to add user=phone parameter or not. If user enables this function, in SIP message will add this parameter.
8. **–waiting**: User can enable or disable Call Waiting function.
9. **–transfer:** User can enable or disable Transfer function.
10. **–pick:** Set Call Group Pick up Code. After setting this code, user can do Group Call Pick Up with this special code.
11. **–spick:** Set Call Specific Pick up Code. After setting this code, user can do Single Call Pick Up with this special code
12. **–park:** Set Call Park Code. Set special access code for Call Park function.
13. **–mwi:** Enable/Disable MWI function and specify MWI method to be client or server-based. (0: OFF, 1: Client(Subscribe),2: Proxy(NOTIFY))
14. **–mwiexpires:** If user set MWI method as client-based. Here can define the interval time of phone to send subscribe message to check voice mail.
15. **–mwinumber**: Set voice mail box number.
16. **–atanswer:** enable/disable auto answer function of Phone. If this function is enabled, phone will answer incoming automatically.

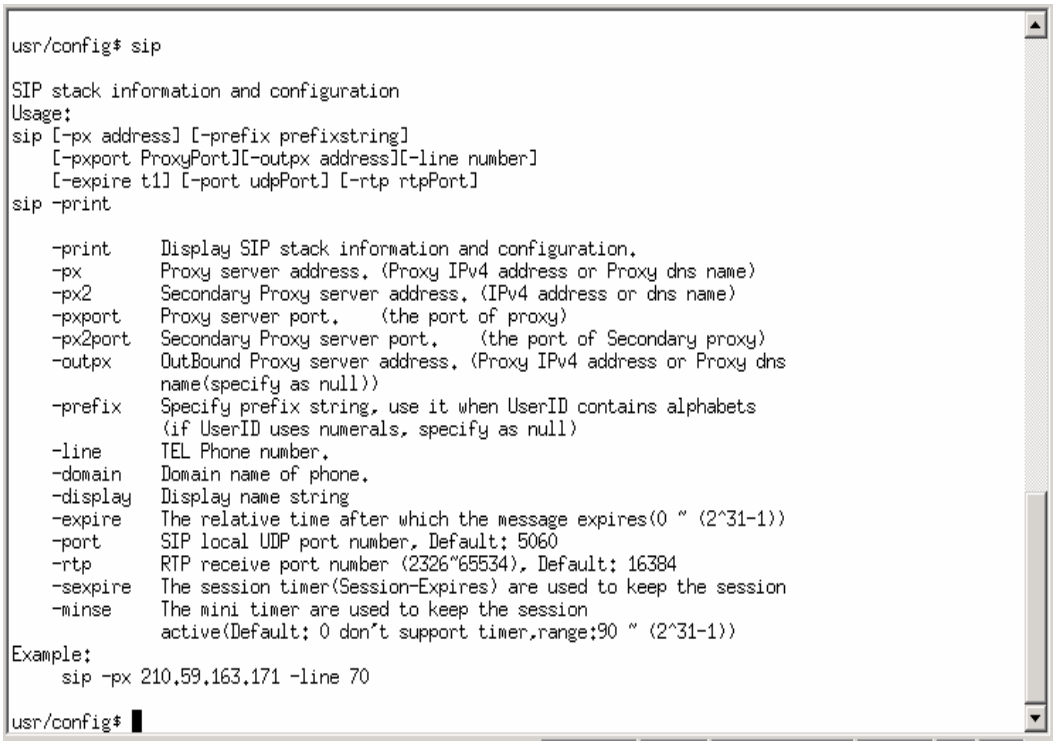
```
usr/config$ sysconf

System information and configuration
Usage:
 sysconf [-idtime digit] [-keypad dtmf]
         [-2833type type] [-eod digit]
 sysconf -print

 -print        Display system overall information and configuration.
 -idtime       Inter-Digits time.(1~10 sec)
 -keypad       Select DTMF type: 0=In-band,1=RFC2833,2=INFO.
 -2833type     RFC2833 Payload Type (range:96~128 inter-used:100,102~105)
 -eod          End of Dial Digit setting(0: NONE, 1: *, 2: #)
 -privacy      Privacy Number Type(0: NONE, 1: Japan)
 -phone        Add the parameter user=phone(0: OFF, 1: ON)
 -waiting      Enable the Call Waiting function(0: OFF, 1: ON)
 -transfer     Enable the Call Transfer function(0: OFF, 1: ON)
 -pick         Call Group Pick up Code
 -spick        Call Specific Pick up Code
 -park         Call Park Code
 -mwi          Enable the MWI subscription function
               (0: OFF, 1: Client(Subscribe),2: Proxy(NOTIFY))
 -mwiexpires   MWI Subscription Keep Alive for Client(default:1200 sec)
 -mwinumber    Voice MailBox Number
 -mwikey       Voice MailBox Hot Key(default:*69)
 -atanswer     auto answer incomming call(0: OFF, 1: ON)
Example:
  sysconf -keypad 0 -eod 2

usr/config$
```

6.13 [sip] command

1. **–print:** display all current configurations.
2. **–px**: set proxy server IP address or URL address (**sip –px "IP address or URL of Proxy server"**).
3. –px2: set alternative proxy server IP address or URL address. If phone failed to register first proxy, it will try to register this alternative proxy.
4. **–pxport:** set listening port of Proxy server.
5. **–outpx:** set IP address of outbound proxy server. After user set outbound proxy, all packets form SIP Phone will be sent to outbound proxy server.
6. **–prefix:** set prefix string. If user ID contains alphabets, user can set it as prefix string here. For example, if Account Name is 123, SIP Phone will sent out messages as Account Name @"IP address of Proxy", if user set prefix as abc, SIP Phone will set out as abc123@"IP address of Proxy". This function is for special proxy server.
7. **–line**: identify one number for the SIP Phone SP5102 to register to the Proxy **(SIP –line "line number").**

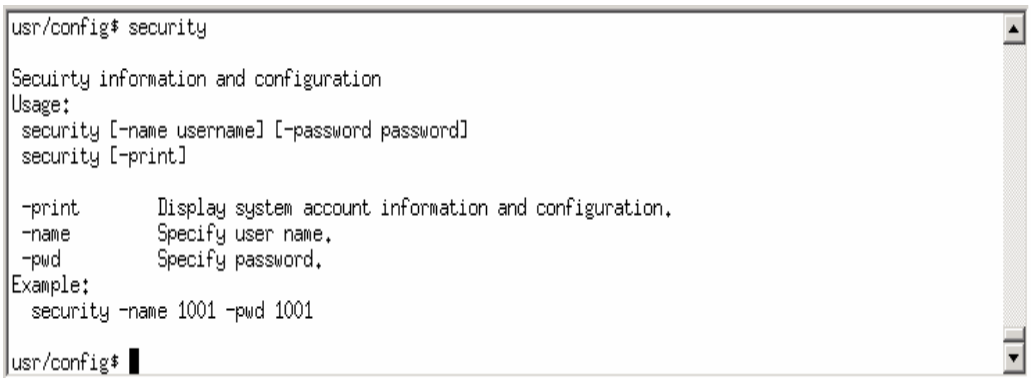
   Note: In proxy mode please remember to set user account information under security command.

8. –domain: set Domain of phone.
9. **–expire:** set expire time of registration. SIP Phone will keep re-registering to proxy server before expire timed out.
10. **–port:** set listening UDP port or SIP Phone.
11. **–rtp:** set RTP port number. SIP Phone will use this port to send and receive voice.
12. –sexpire: set the session timer.
13. –minse: set the mini session timer.


usr/config$ sip
SIP stack information and configuration
Usage:
sip [-px address] [-prefix prefixstring]
[-pxport ProxyPort][-outpx address][-line number]
[-expire t1] [-port udpPort] [-rtp rtpPort]
sip -print
-print Display SIP stack information and configuration.
-px Proxy server address. (Proxy IPv4 address or Proxy dns name)
-px2 Secondary Proxy server address. (IPv4 address or dns name)
-pxport Proxy server port. (the port of proxy)
-px2port Secondary Proxy server port. (the port of Secondary proxy)
-outpx OutBound Proxy server address. (Proxy IPv4 address or Proxy dns
name(specify as null))
-prefix Specify prefix string, use it when UserID contains alphabets
(if UserID uses numerals, specify as null)
-line TEL Phone number.
-domain Domain name of phone.
-display Display name string
-expire The relative time after which the message expires(0 ~ (2^31-1))
-port SIP local UDP port number, Default: 5060
-rtp RTP receive port number (2326~65534), Default: 16384
-sexpire The session timer(Session-Expires) are used to keep the session
-minse The mini timer are used to keep the session
active(Default: 0 don't support timer,range:90 ~ (2^31-1))
Example:
sip -px 210.59.163.171 -line 70
usr/config$

6.14 [security] command

1. **–print**: display all current configurations.
2. **-name**: set user ID of SIP Phone SP5102 for registering. User can set user name and password for registering. If password is no need, please set user name the same as line number or SIP Phone won't register successfully.
3. **–pwd**: set account password for registering.



6.15 [line] command

1. **–print**: display all current configurations.
2. **–fwd:** set forward function.

There are 3 selections in Forward type, user must select under which condition to forward calls.

6.15.1 Busy Forward

When SIP-Phone is in busy status, the incoming call will be forwarded to the assigned phone number.

6.15.2 No Response

When SIP-Phone hasn't been picked up for around 10 seconds, the incoming call will be forwarded to the assigned phone number.

6.15.3 Unconditional

It is included the above two types. Whether the SIP-Phone is in which status, calls will be automatically forwarded to the assigned phone number.

–hotline: set hotline number.('X' for disable)

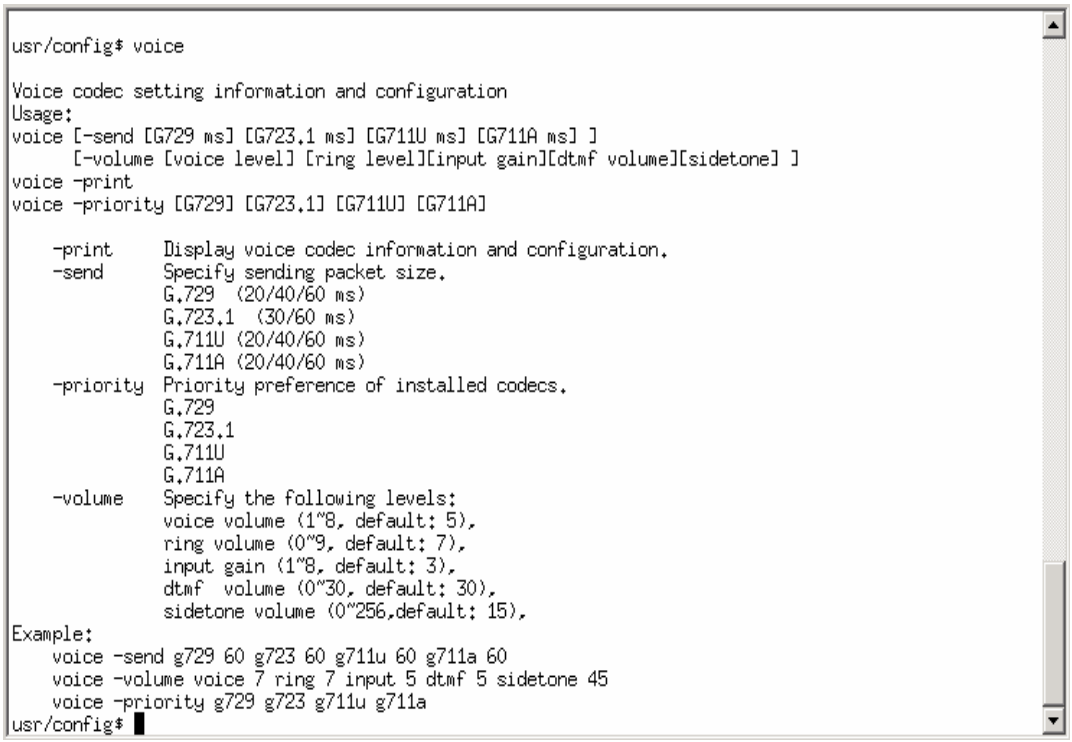
```
usr/config$ line

Gateway line information and configuration
Usage:
line -print      Gateway line information.
line -fwd        Forward type and number.
                  1:unconditional
                  2:busy
                  3:no answer
line -hotline    Set Hot line information('x' for disable).
Example:
     line -print
     line -fwd 1 1001
     line -fwd 2 x
     line -hotline 628

usr/config$
```

6.16 [voice] command

The voice command is associated with the voice codec setting information.

1. **-print**: display voice codec information and configuration.
2. **-send**: three voice packet size can be configured as 20 ms, 40 ms or 60 ms.(only 30 and 60 ms for G..723.1)
3. **-priority**: set codecs priority in order. Please notice that user can set from 1 to 5 codecs as their need, for example, **voice –priority g723** or **voice –priority g729 711a g711u** means SIP Phone can support only one codec or four codecs.
4. **-volume**: There are three types can be adjustable, voice volume, input gain and DTMF volume. Voice volume means the volume user can hear, input gain means the volume the other side can hear from SIP Phone, DTMF means DTMF transmitting volume. (**voice –volume voice "value of volume", voice –volume input "value of volume", voice –volume dfmt "value of volume"**)

Note: It is for advanced administrator use only. Please ask your distributor before changing any settings of this command.



6.17 [phone] command

1. **-print:** Print current call progress tone configurations (ring: ring tone, rbt: ring back tone, bt: busy tone, dt: dial tone). This parameter should be accompanied with tone type.
2. **-ring:** To set RING tone value. The played tone type, when Phone ha incoming call.
3. **-rbt:** To set Ring Back Tone value. The played tone type, when Phone dials out a call.
4. **-bt:** To set Busy Tone value. The played tone type, when Phone dials to a destination that is busy.
5. **-dt:** To set Dial Tone value. The played tone type, when hook off the Phone.

```
usr/config$ phone

Phone ringback tone , busy tone , dial tone setting and notes
Usage:

phone [-ring [freq   ][ringON  ][ringOFF ][ringLevel]]
phone [-rbt  [freqHi ][freqLo  ][freqHiLev][freqLoLev]
            [Tone1ON][Tone1OFF][Tone2ON  ][Tone2OFF ]]
      [-bt   [freqHi ][freqLo  ][freqHiLev][freqLoLev]
            [Tone1ON][Tone1OFF][Tone2ON  ][Tone2OFF ]]
      [-dt   [freqHi ][freqLo  ][freqHiLev][freqLoLev]
            [Tone1ON][Tone1OFF][Tone2ON  ][Tone2OFF ]]
      [-flash [freqLo ][freqHi ]]
phone [-print [rbt]|[bt]|[dt]|[flash]]

      -print  Display phone ringing/tone configuration.
              rbt  :  ringback tone
              bt   :  busy tone
              dt   :  dial tone
      -ring   ringing configuration set .
      -rbt    ringback tone configuration set .
      -bt     busy tone configuration set .
      -dt     dial tone configuration set .
      -flash  flash configuration set .
Note:
      tone    frequency   : 0  ~ 65535 (Unit : Hz)
      tone    freqLevel   : 0  ~ 65535 (Unit : mVrms)
      tone    Tone ON/OFF : 0  ~ 8000  (Unit : ms)
Example:
      phone -print rbt
      phone -ring 20 2000 4000 94
      phone -rbt 480 440 8 8 2000 4000 2000 4000
      phone -bt 620 480 8 8 500 500 500 500
      phone -dt 440 350 8 8 500 1023 1023 1023
      phone -flash 100 300

usr/config$
```

6.18 [tos] command

TOS/DiffServ (DS) priority function can discriminate the Differentiated Service Code Point (DSCP) of the DS field in the IP packet header, and map each Code Point to a corresponding egress traffic priority. As per the definition in RFC2474, the DS field is Type-of-Service (TOS) octet in IPv4. The recommended DiffServ Code Point is defined in RFC2597 to classify the traffic into different service classes. The mapping of Code Point value of DS-field to egress traffic priorities is shown as follows.

1. **High priority with DS-field.**

Expected Forwarding (EF)	101110	====>	46 (Decimal System)
Assured Forwarding (AF)	001010	====>	10 (Decimal System)
	010010	====>	18 (Decimal System)
	011010	====>	26 (Decimal System)
	100010	====>	34 (Decimal System)

2. **Low Priority with DS-field:**

Assured Forwarding (AF)	001100	====>	12 (Decimal System)
	010100	====>	20 (Decimal System)
	011100	====>	28 (Decimal System)
	100100	====>	36 (Decimal System)
	001110	====>	14 (Decimal System)
	010110	====>	22 (Decimal System)
	011110	====>	30 (Decimal System)
	100110	====>	38 (Decimal System)
	000000	====>	0 (Decimal System)

1. **-print** : display all current configurations.
2. **–rtptype:** set DSCP value of signaling packets from 0 to 63
3. **–siptype:** set DSCP value of RTP packets from 0 to 63

Note: This command won't be functional until whole network environment support DSCP function, e.g. all routers or switches in your network have enabled DSCP feature.

usr/config$ tos
IP Packet ToS(type of Service)/Differentiated Service configuration
Usage:
tos [-rtptype dscp]
tos [-sigtype dscp]
tos -print
[-rtpreliab mode]
tos -print
Example:
tos -rtptype 7 -sigtype 0
usr/config$

6.19 [prefix] command

This command is for digit manipulation.

1. -add: Add a rule to drop or insert prefix digits of incoming call.
    - prefix: Set which prefix number to implement prefix rule.
    - drop: Enable or disable drop function. If this function is enabled, Gateway will drop prefix number on incoming call.
    - insert: Set which digit to insert on incoming call.

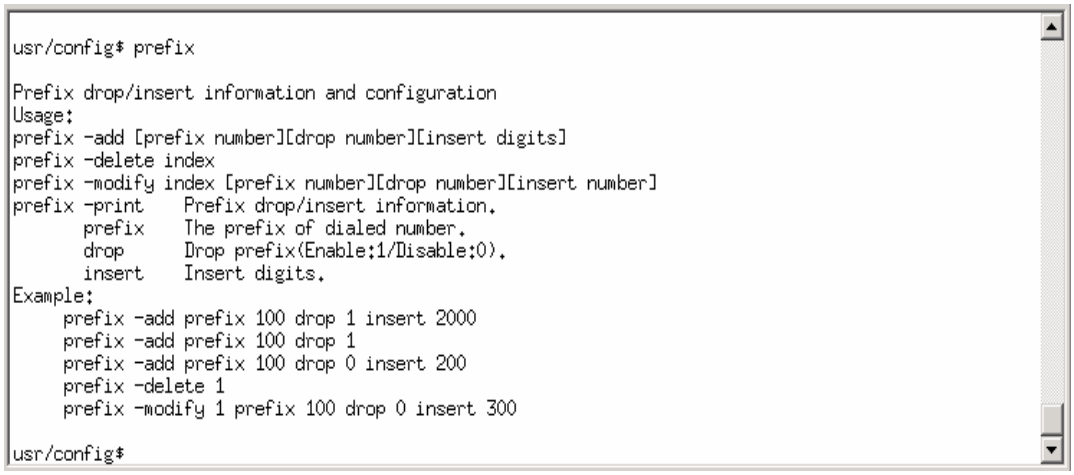
Example: prefix -add prefix 100 drop 1 insert 2000

2. -modify: Modify a rule to drop or insert prefix digits of incoming call.

Example: prefix –modify 100 drop 0 insert 200

3. -delete: Delete a rule to drop or insert prefix digits of incoming call.

Example: prefix –delete modify 100 drop 0 insert 200



6.20 [rom] command

1. **-print**: show all current configurations and version information.
2. **-app**: upgrade main application code.

   > ***Note:***
   >
   > After upgrade Application, please remember to execute **flash –clean** command, which will clean all configurations become factory values except IP address.

3. **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading firmware rom file.
4. **-f**: the file name prepared for upgrading is necessary as well.
5. **–method**: specify download method to be TFTP or FTP(0 for TFTP.1 for FTP)
6. **–ftp**: specify user name and password for FTP download method.

   For example: User prepares to upgrade the latest app rom file – lpsip.100, the TFTP server is 192.168.1.1, User has to input command as below:

   rom –app –s 192.168.1.1 –f lpsip.100

```
usr/config$ rom

ROM files updating commands
Usage:
rom [-print][-app][-dsptest][-dspcore][-dspapp]
    -s TFTP/FTP server ip -f filename
rom  print
    -print     show versions of rom files, (optional)
    -app       update main application code(optional)
    -boot      update main boot code(optional)
    -dsptest   update DSP testing code(optional)
    -dspcore   update DSP kernel code(optional)
    -dspapp    update DSP application code(optional)
    -s         IP address of TFTP/FTP server (mandatory)
    -f         file name(mandatory)
    -method    download via TFTP/FTP (TFTP: mode=0, FTP: mode=1)
    -ftp       specify username and password for FTP
Note:
    This command can run select one option in 'app',
    , 'dsptest', 'dspcore', and 'dspapp'.
Example:
    rom -method 1
    rom -ftp vwusr vwusr
    rom -app -s 192.168.4.101 -f app.bin
    rom -boot -s 192.168.4.101 -f boot.bin

usr/config$
```

   Command **rom –print** can show current version installed in SIP Phone SP5102.

```
usr/config$ rom -print


Download Method  :  TFTP

Application Rom  :  inca388_060913.bin
usr/config$
```

6.21 [auth] command

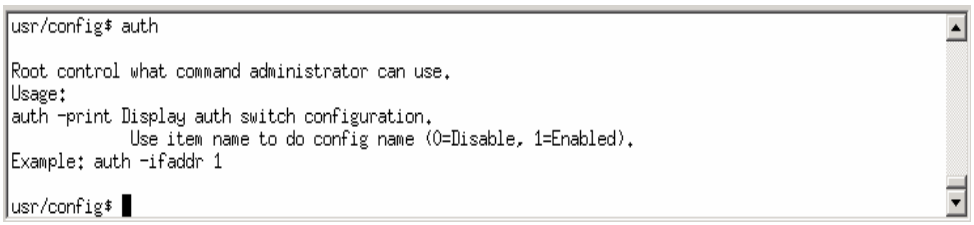
For security concern, the “root” user can customize some configurable items for “administrator” user.

1. **-“item name”:** Assign the configurable item for “administrator” user.

Example: **auth -ifaddr 1** **auth -sip 1** **auth -voice 1**

Now the Administrator can use these command which Root user assign to them.

2. **-print:** Display the configurable items for “administrator” user.

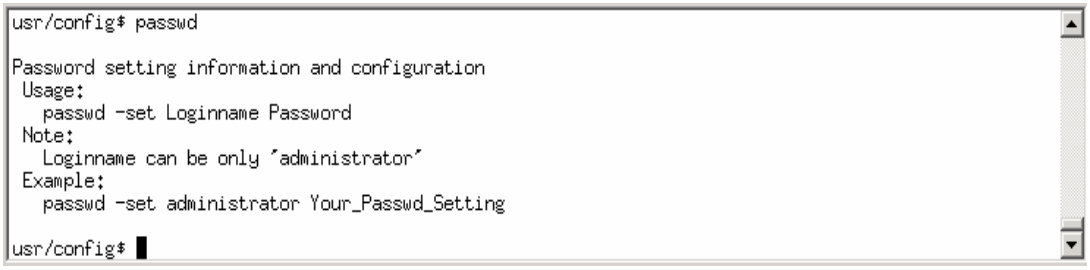


6.22 [passwd] command

For security protection, user has to input the password before entering ***application user/config mode***. Two configurations of login name/password are supported by the system.

1. **–set**: set password of “root” users or “administrator” users.

   (passwd –set root/administrator “password”)

2. **–clean**: clean up password restored before, and user can login :”root/administrator”, password: ”press enter”.

   User who requests authorization to execute **all** configuration commands needs to login with “root”. If a user login with “administrator”, commands below are not functional: