User's Manual

VoIP Telephone, SIP

Model No.: SP5100/S

(Application rom: v.104)

Website: http://www.micronet.info

About this User's Manual

This user's Manual gives users necessary steps for i

Please read this manual chapter by chapter.

Chapter 1. Introduction

Introduce the IP Telephony device to users in terms of feature, appearance.

Chapter 2. Startup

Help user complete basic configuration.

Chapter 3. Call Progress

Show user how to use the device to process phone call.

Chapter 4. Voice Quality

Show user how to improve voice quality.

Chapter 5. System Maintenance

Show user how to upgrade, recover the system.

Chapter 6. Web Administration

Provide command reference of Web Interface for advanced setting.

Chapter 7. Command Line Interface

Provide command reference of command-line interface via Telnet.

Chapter 8. Command Menu / LCD Interface

Provide instruction to configure the IP-Phone via Keypad on the phone set.

Chapter 9. Specification

List the specification of the device in detail.

Online Upgrade

Please refer to http://www.micronet.info/ for additional support.

Table of Content

1. Introduction

Micronet SP5100/S VoIP Phone provides unmatched levels of converged communications that go beyond present conventional voice system. With internal voice/data switch, SP5100/S can reduce costs by conserving wiring closet ports because separate cable drops to desktop are not required. It also prioritizes voice traffic against data to ensure high-quality voice communication.

1.1 Package Content

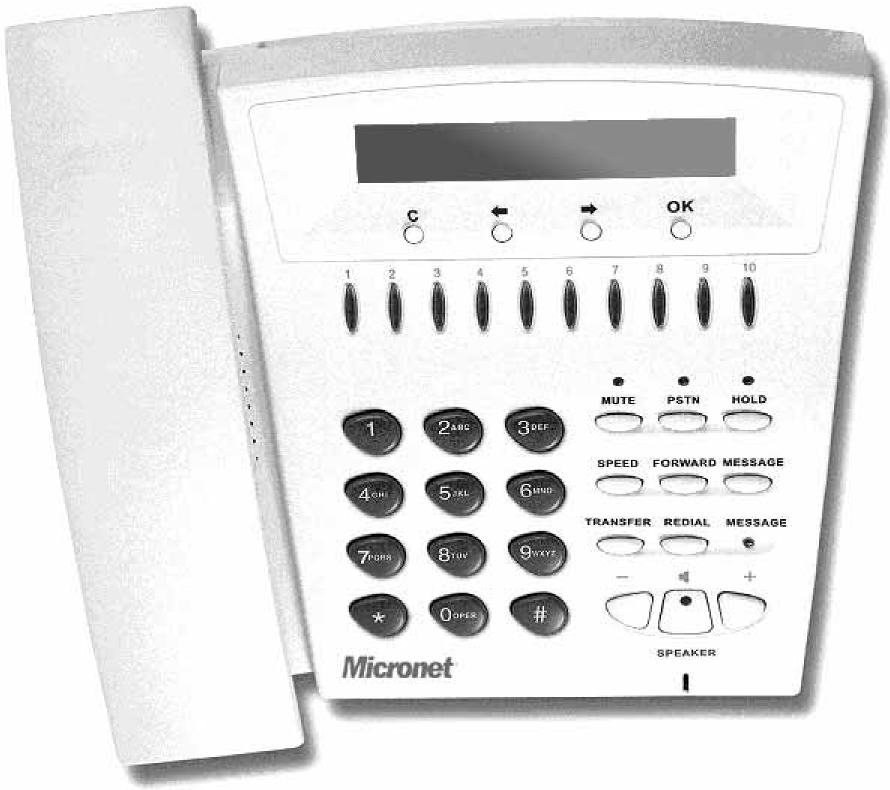
- SP5100/S VoIP Telephone
- Quick Installation Guide
- Manual CD
- Power Adapter

1.2 Key Features

- Compliant with IETF SIP standard
- Compliant with G.711A/µ law, G.723.1, G.729 CODEC standards
- 2 RJ-45 ports for 10/100 Mbps Ethernet for easy installation with your network
- Provide RJ-11 port for usual Telephone Network
- Support VAD, CNG, EC, Adaptive Jitter Buffer features for voice quality
- Support IP ToS / DSCP and 802.1q/p for Quality of Service
- Support both Peer to Peer dialing and central Proxy environment
- Support NAT traversal.
- Support Call forward / Call Hold / Call Transfer telephony features
- Support Quick Configuration through Keypad and LCD Display
- Support Remote Configuration through Telnet and browser interface
- Support static IP, PPPoE and DHCP
- Support DDNS for dynamic IP environment

1.3 Physical Description

1.3.1 Functional Keypads



Front Panel

Keypad	***Description***
C	Jump out current LCD menu / Cancel dialing digit.
←	Clear previous data / Move to previous selection of LCD menu
→	Move to right / next selection of LCD menu
OK	Press OK to confirm the modification
DL (1 – 10)	Direct Line Button. Press DL button to make a speed dial (index 1-10 of phone book) after off-hook. Number 1 –10, * and #: The function is the same as the general phone set.

(1). Character mode:

1	“1”
2	“A”; “B”; “C”; “2”
3	“D”; “E”; “F”; “3”
4	“G”; “H”; “I”; “4”

5	“J”; “K”; “L”; “5”
6	“M”; “N”; “O”; “6”
7	“P”; “Q”; “R”; “S”; “7”
8	“T”; “U”; “V”; “8”
9	“W”; “X”; “Y”; “Z”; “9”
*	“.”; “@”; “_”; ”-“; “,” ; “!” ; “?” ; “*” ; “’” ; “+” ; “$” ; “*”
0	“Space”; “0”
#	“#”

NOTE: *Switch characters to be capital or lowercase by pressing* *TRANSFER key.*

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(2) Digit mode:

1	“1”
2	“2”
3	“3”
4	“4”
5	“5”
6	“6”
7	“7”
8	“8”
9	“9”
*	“.”
0	“0”
#	

MUTE Mute the voice of MIC and let others can’t hear from user in communication.

PSTN Press PSTN to switch IP-Phone as PSTN or IP Phone Mode. In PSTN mode, “PSTN” characters will be displayed on LCD left bottom side, and then users can dial out as if he uses a traditional phone set. In IP Phone mode, “PROXY” characters will be displayed on LCD left bottom side.

NOTE:

- *When IP-Phone is in PSTN mode, only PSTN and SPEAKER function key can work.*
- *LCD will display "…Incoming Call..." to inform user of IP or PSTN incoming calls.*

HOLD — To hold a call, after press HOLD button, both sides will hear hold tone.

SPEED — Speed dial key. Press SPEED and number (index of phone book) after off-hook.
Switch input mode between character mode or digit mode.

FORWARD — Forward an incoming call to another IP device.

MESSAGE / indicated LED
When an incoming call is missed, the MESSAGE LED will be flashing. User can check the information of missed calls by pressing the MESSAGE button.

TRANSFER — Transfer a call to the third site.
Switch characters to be capital or lowercase.

REDIAL — Redial the last outgoing call.

+ And - — Adjust the voice volume heard of communication.

SPEAKER — Hand free mode. User can talk without picking up handset.

***NOTE:** All function keys mentioned above (except dialing keypad) are effective only in IP Phone mode.*

1.3.2 Connection

Rear Panel

DC 9V	DC 9V power input outlet
LAN	RJ-45 LAN port for connecting to router, modem, or local network (Hub / switch)
PC	RJ-45 PC port for connecting to the PC It needs only simple patch cord for PC, and no extra cabling any more. With 802.1q/p enabled, voice traffic can be insured higher priority over data traffic
Line	RJ-11 port for connecting to the PSTN analog line

***NOTE:** There are two LED indicated lights: LINK/ACT and 10/100 for LAN port and PC port. When network status is regular, LED of LINK/ACT will light on; when IP-Phone is transmitting or receiving data, LED will be flashing; when transmit rate is in 10 mbps or 100mbps, LED of 10/100 will light off or light on.*

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2. Startup

2.1 Initialize the IP-Phone

Turn on the IP-Phone, the LCD screen shows as below. Now, IP-Phone is running Boot sector program.

IP-Phone Board Start Booting

IP-Phone finishes boot program initialization. User can see flashing greeting as below:

System Initializing…………..

Then IP-Phone gets into standby mode:

	IP-Phone
Proxy	10:10:10 AM

The main LCD screen would be shown as similar as above. “Proxy” means the IP-Phone is in Proxy Mode. When IP-Phone connects to SNTP server, LCD will show current time captured from SNTP server.

When IP-Phone is in peer-to-peer mode, LCD will show “P2P”.

IP-Phone	
P2P	10:10:10 AM

Pressing the PSTN button, the “Proxy” or “P2P” will be replaced by “PSTN”. User must plug PSTN line in RJ-11 port when IP-Phone is in PSTN mode.

IP-Phone	
PSTN	10:10:10 AM

2.2 Login into the System

2.2.1 Keypad

User can set the following configurations by LCD keypad.

NOTE:

- *Press* *TRANSFER* *to switch characters to be capital or lowercase.*
- *Press* *SPEED* *to switch input mode as character mode or digit mode*
- *Press ← or → to enter keypad configuration mode, then press OK button to enter sub menus.*
- *Press **C** can jump out current menu to previous level. Press ← will clear previous input data.*

Main menu

1. Call List
2. Forward Type
3. Phone Book
4. Ringer Settings
5. Network
6. Advanced Settings (protected by password)
7. Reboot

First of all, you may modify the network setting to fit your existing network. Please follow the steps:

1. Press → to main menu "5.Network", and press **OK** to enter.
2. Get into sub-menu "2.Network Mode" to select Static, DHCP, or PPPoE.
3. Get into sub-menu "3.IP Address" to configure static assigned IP. When it's in PPPoE mode, the IP address is still working for local management.
4. Get into sub-menu "4.Subnet Mask" to configure subnet mask.
5. Get into sub-menu "5.Default Device" to configure default device.

2.2.2 Web Interface

The embedded web configuration allows you to use a web browser to manage the IP-Phone.

Step 1. Connect LAN port to modem / router or local network (hub, or switch).

Step 2. Launch your web browser and enter its IP address.

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*NOTE: If you don't change the default IP address in the preceding section, add an IP address "**10.1.1.x**" in your managing PC and use "**10.1.1.3**" (the default IP address of the IP-Phone) in the Location field of web browser.*

Step 3. The Password screen now appears. Type "***root***" in the user name field (it may display automatically for you) and your password (default "Null") in the password field.

Step 4. Click OK.

Step 5. After a successful login, you will see the welcome screen, as shown below, to start configuring.

Step 5. After completing configuration, please reboot the phone to take effect.

2.2.3 Command Line Interface

User can enter into Telnet command lines.

Step 1. Connect LAN port to modem / router or local network (hub, or switch).

Step 2. Click on Start and Run "telnet <ip_address>"

NOTE:

*If you don't change the default IP address in last section, add an IP address "**10.1.1.x**" in your managing PC and use "**10.1.1.3**" (the default IP address of the IP-Phone).*

Step 3. The Password screen now appears. Type "***root***" in the login field (it may display automatically for you) and your password (default "Null") in the password field.

```
login: root
Welcome to Terminal Configuration Mode
Please enter your configuration item

usr/config$
```

NOTE:

- *User must input lower-case command. For contents of configurations, such as SIP alias or user name, user can set them as capital case.*
- *After any change of configuration, please remember to do "commit" command to save changes, and then reboot "command" to restart system.*

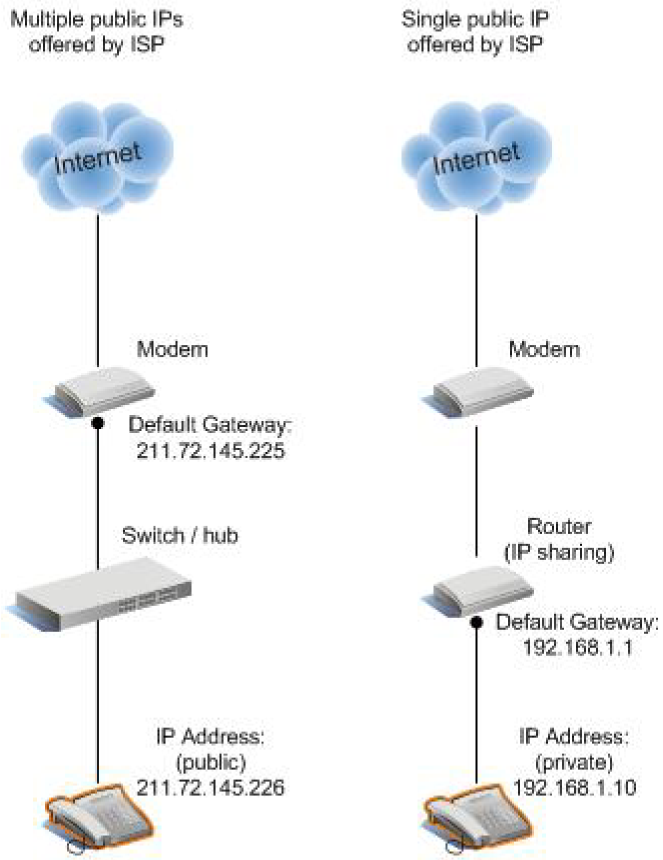
2.3 Network Configuration

First of all, set up the device connecting to Internet.

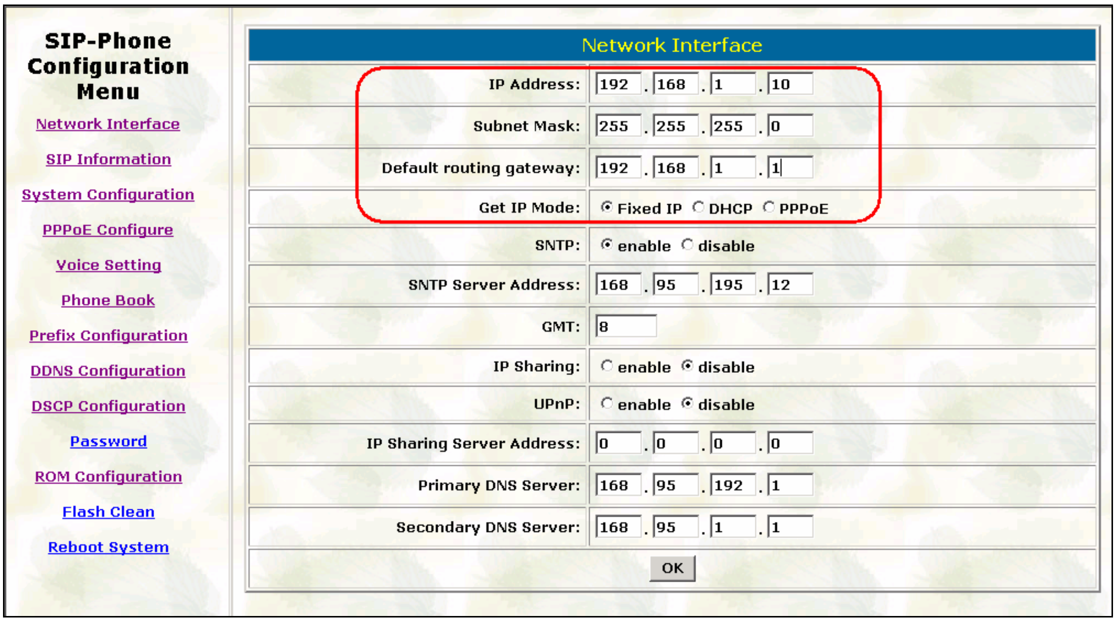
NOTE: Different ISPs require different methods of connecting to the Internet. Please consult your ISP to select right IP type (Fixed IP, PPPoE).

2.3.1 Static (fixed) IP Assign

Set the device with static IP mode when you get multiple public IP addresses or place it behind a NAT router.



Web Interface:



- Select "Fixed IP".
- Fill out the field "IP Address", "Subnet Mask", and "Default routing device".
- Click on **OK**

Command Line Interface:

---

usr/config$ ifaddr -mode 0

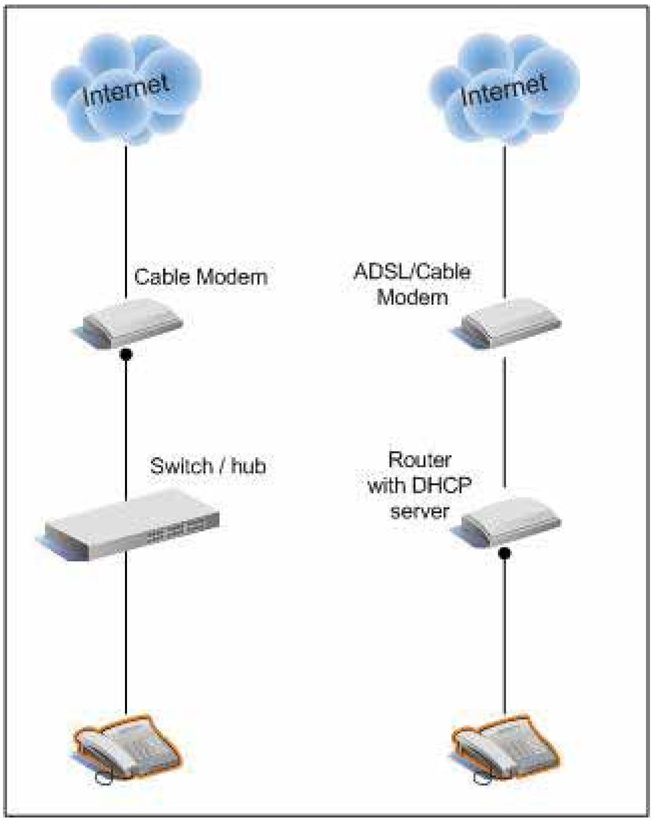
usr/config$ ifaddr -ip <ip_address> -mask <subnet_mask> -gate <default_device>

usr/config$ commit

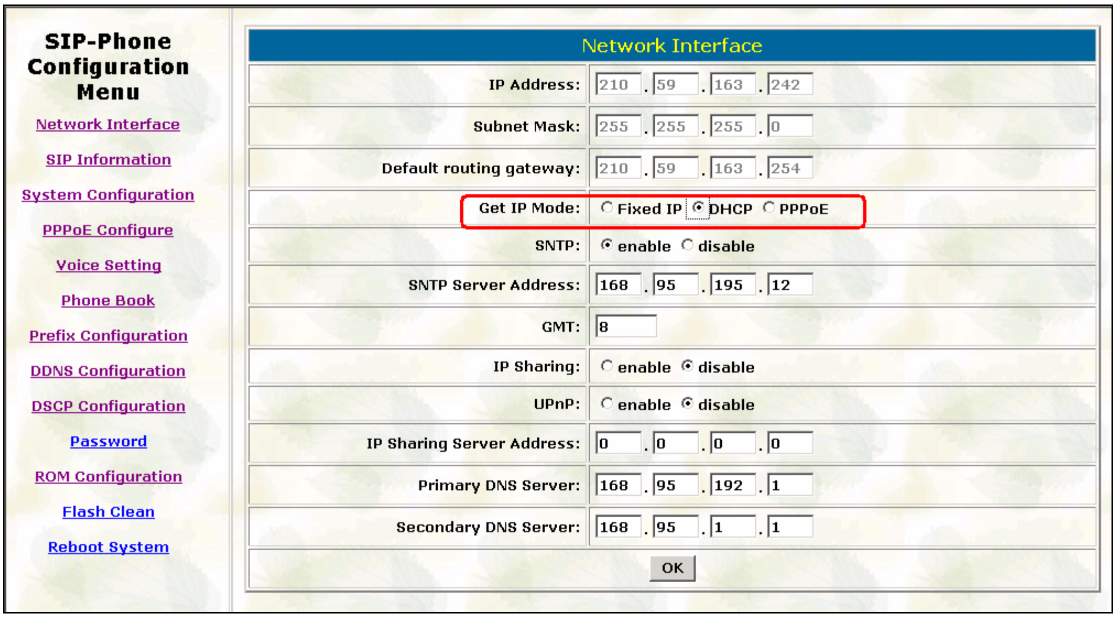
usr/config$ reboot

2.3.2 DHCP

Set the device with DHCP mode. The device can automatically get IP address, mask, default device, and DNS server IP from the NAT router or ISP.



Web Interface:



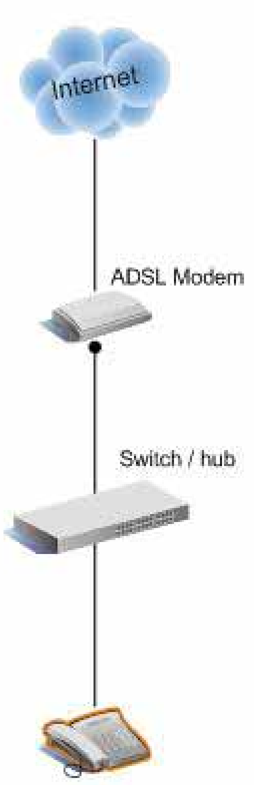
Select "DHCP" and then click on **OK**

Command Line Interface:

```
usr/config$ ifaddr -mode 1
usr/config$ commit
usr/config$ reboot
```

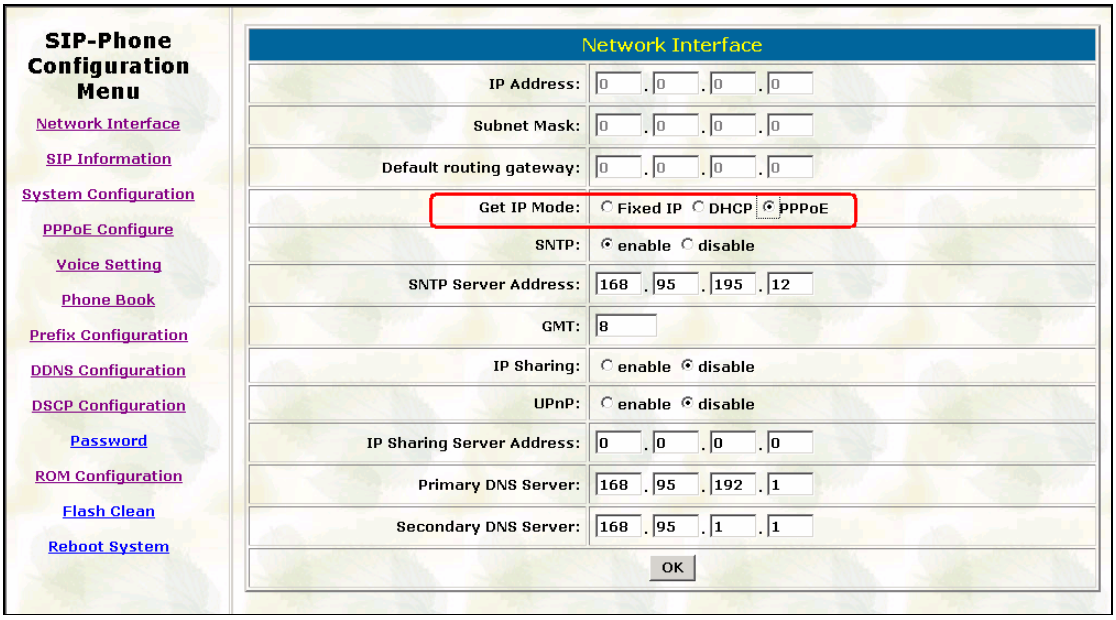
2.3.3 PPPoE

Set the device with PPPoE account offered by ISP. The device can get IP information automatically.



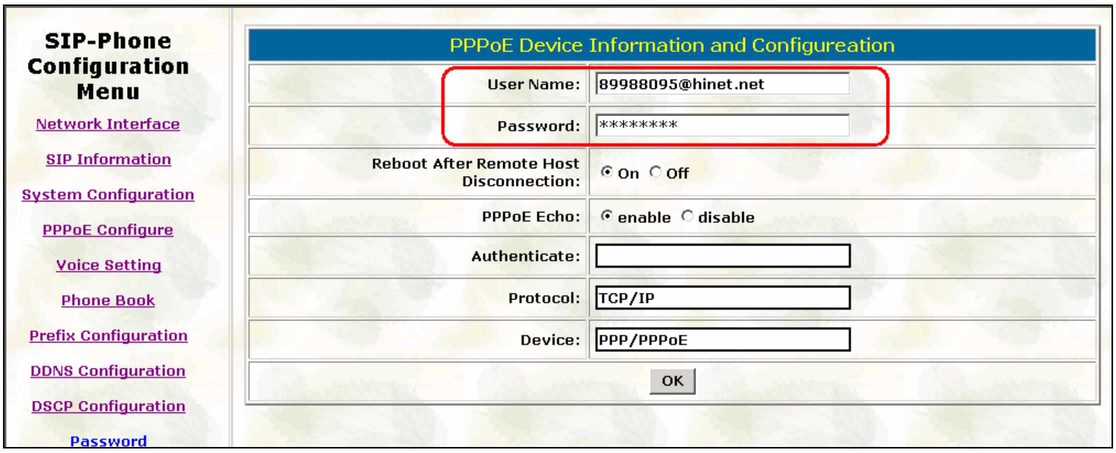
Web Interface:

Go to Network Interface,



Select “PPPoE”, and then click on OK

Go to PPPoE Configuration,



- Fill out the fields “User Name” and “Password” with the account offered by ISP.
- Click on **OK**

Command Line Interface:

```
usr/config$ ifaddr -mode 2
usr/config$ pppoe -id <username>
usr/config$ pppoe -pwd <password>
usr/config$ commit
usr/config$ reboot
```

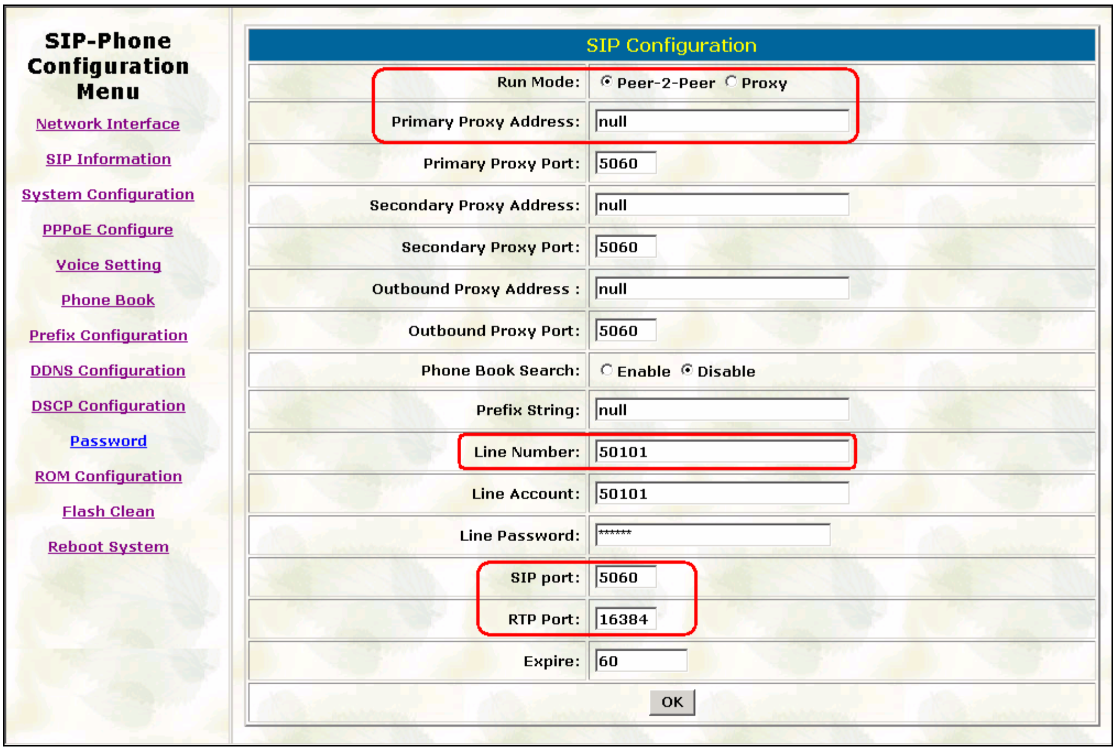
NOTE: *Local static Assigned IP is still valid for local management.*

2.4 SIP Configuration

Set operation mode (Peer-to-Peer" or "Proxy"), IP telephone number, SIP server's address/port, and authentication parameters for the IP-Phone.

2.4.1 Peer-2-Peer Mode

Web Interface:



Set the parameters as follows:

Parameter	***Description***
Run Mode	Select Peer-2-Peer
Line Number	Assign a number as its VoIP Telephone number
SIP/RTP Port	Just keep them default

NOTE: *Phone address book is necessary in P2P mode. Please refer to the section 3.1.1 Phone Address Book.*

Command Line Interface:

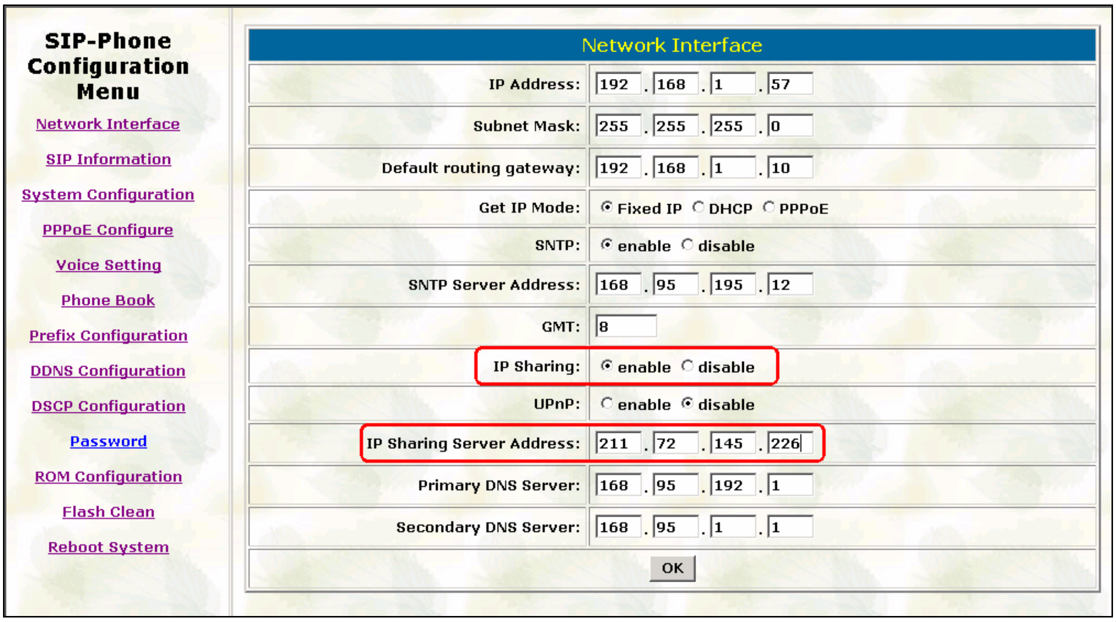
```
usr/config$ sip -mode 0
usr/config$ sip -line <tel_number>
usr/config$ commit
usr/config$ reboot
```

NOTE:

- *In P2P mode, it is better that one device is given with one public IP (dynamic IP at least).*
- *If only one dynamic IP address is offered by ISP, don't locate the phone behind NAT.*

The device can work behind NAT router with one fixed public IP address. Please set up Port mapping function in the NAT router, and enable "IP sharing" function in the IP-Phone.

Web Interface:



Select "enabled" for **IP sharing** and fill out the section **IP Sharing Server Address** with the WAN IP of NAT router.

Command Line Interface:

---

usr/config$ ifaddr -ipsharing 1 <router_wan_ip>

NOTE: *In the NAT router, DMZ or Virtual Server has to be enabled to re-direct traffic to the IP-Phone.*

DMZ: (re-direct all types of traffic to the IP-Phone)

Host IP: <lan_phone_ip>

Virtual Server: (re-direct traffic to the IP-phone for specific service/application)

Service: <service_name> (e.g. SIP, RTP)

Protocol: TCP or UDP

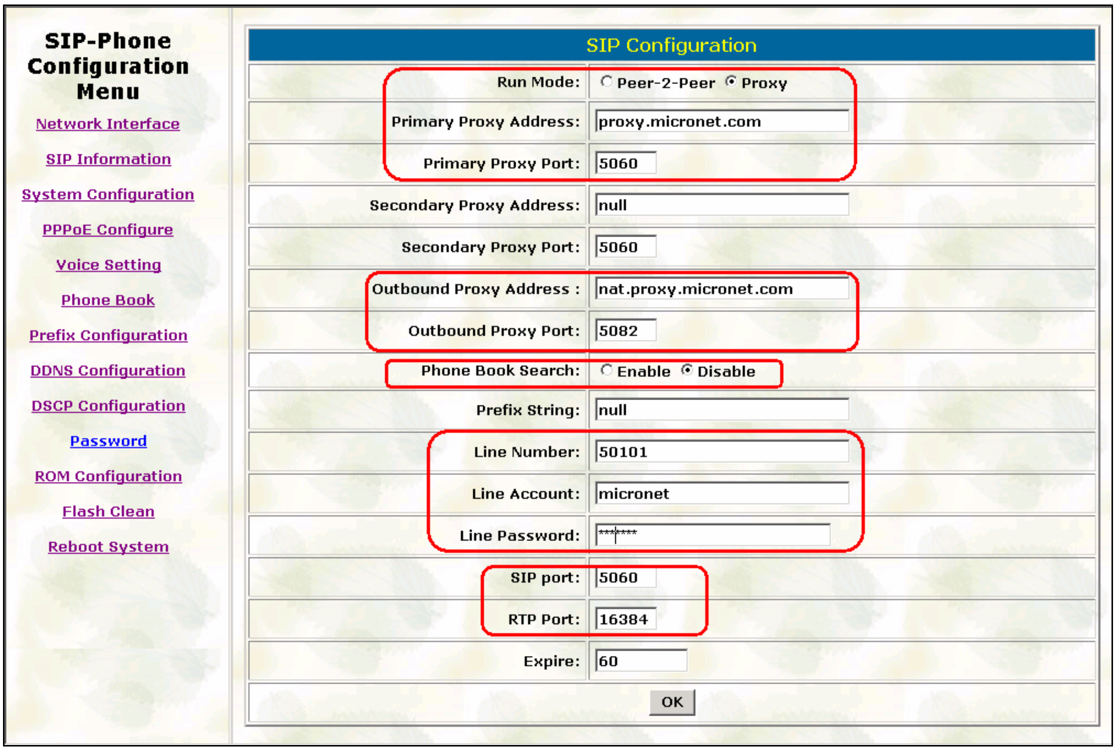
External Port: <ex_serv_port> (e.g. 5060, 16384~16385)

Internal Port: <in_serv_port> (e.g. 5060, 16384~16385)

Host IP: <lan_phone_ip>

2.4.2 Proxy Mode

Web Interface:



Set the parameters as follows:

Parameter	***Description***
Run Mode	Select Proxy
Primary Proxy Address/Port	Fill the field out with IP (or URL) / port of SIP proxy server.
Outbound Proxy Address/Port	Fill the field out with IP (or URL) / port of Outbound Proxy server for NAT traversal.
Line Number	Fill the field out with VoIP Telephone number, offered from ITSP or SIP server.
Phone Book Search	Enable it and pbook would be checked. **It allows P2P dialing when the phone is registered to server.**
Line Account / Password.	Set user name and password for registering

SIP / RTP Port Just keep them default (SIP: 5060, RTP: 16384)

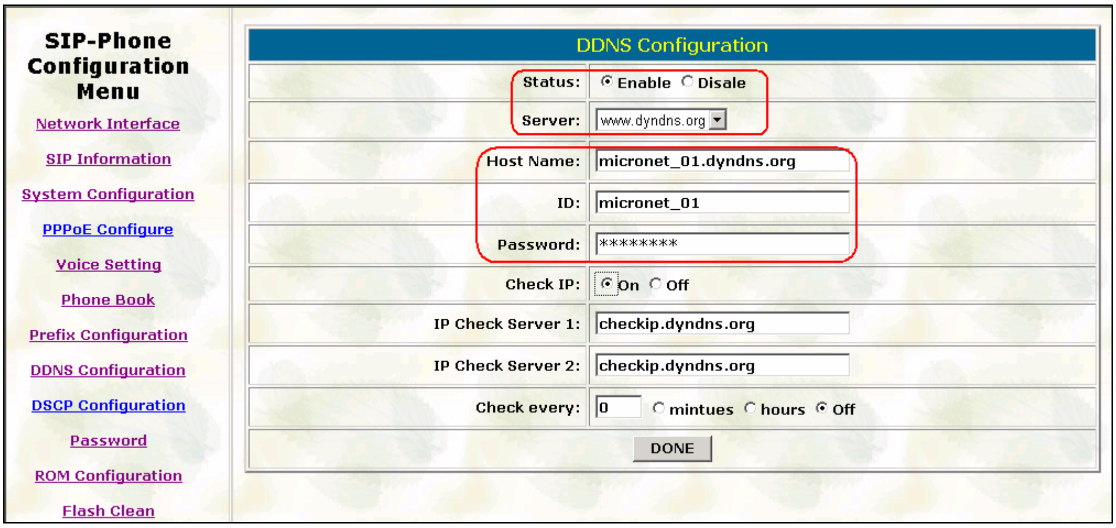
Command Line Interface:

```
usr/config$ sip -mode 1
usr/config$ sip -px <sip_server_ip>
usr/config$ sip -pxport <sip_server_port>
usr/config$ sip -outpx <outbound_proxy_ip>
usr/config$ sip -oupxport <outbound_proxy_port>
usr/config$ sip -pbsearch 1
usr/config$ sip -line <tel_number>
usr/config$ sip -security -name <username> -pwd <password>
usr/config$ commit
usr/config$ reboot
```

2.5 Dynamic DNS

DDNS allows you to map the static domain name to a dynamic IP address. DDNS makes the gateway accessible for other client to call in P2P (peer-to-peer) mode, when the IP address is dynamic.

Web Interface:



Command Line Interface:

----

usr/config$ ddns -enable 1

usr/config$ ddns -server members.dyndns.org -hostname <hostname>

usr/config$ ddns –id <id> -passwd <password>

usr/config$ ddns –checkip 1

usr/config$ commit

usr/config$ reboot

NOTE:

- *DDNS setting is not necessary when run mode is “Proxy”.*
- *If “Check IP” is enabled and the Phone is behind IP sharing, SIP Phone will check its public IP address by asking IP check server and send to DDNS server to update DDNS data.*
- *If this function is disabled and the Phone is behind IP sharing, it will send its private IP address to DDNS server.*

3. Call Progress

3.1 Make a Call

When the IP-Phone runs in Proxy mode, user can call out by just dialing IP telephone number of clients in the same SIP service domain.

Make IP Call	*Press **<ip_telephone_number>** + #*

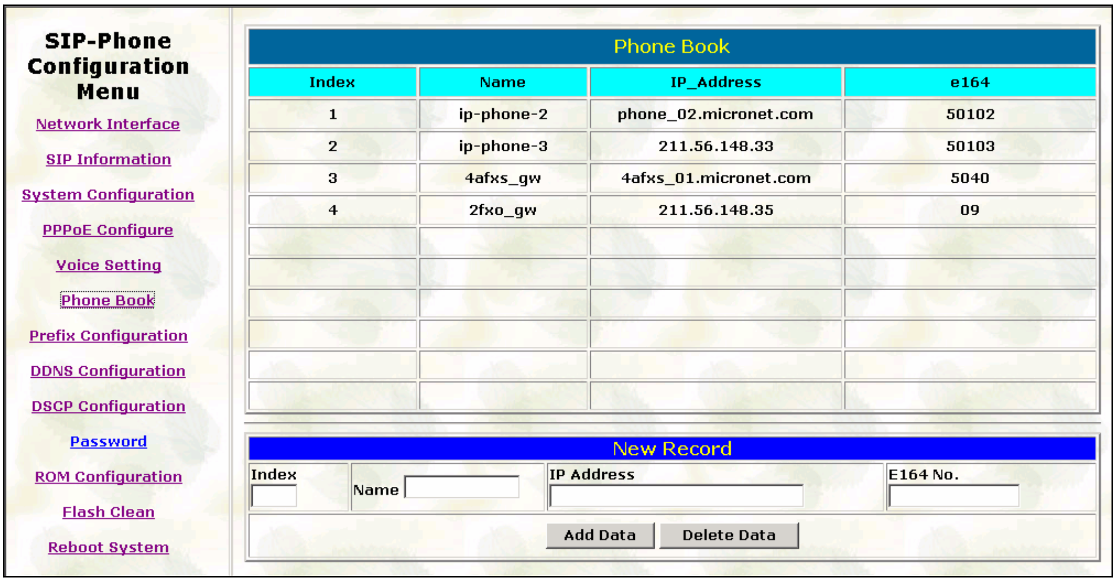
SP5100 provides 1 RJ-11 port for PSTN life line, and can tranceive PSTN calls even if VoIP fails.

Make PSTN Call	*Power off:* ● *Press **<pstn_number>*** *Power on:* ● *Press **PSTN** switching to PSTN mode* ● *press <pstn_number>*

When VoIP is deployed in P2P mode and in user-defined dialing rules, phone address book and prefix may be necessary.

3.1.1 Phone Address Book

Web Interface:



Please set the parameters as follows:

Parameter	Description
IP Address	Fill the field out with IP/URL address of peer VoIP device.
E.164	It can be… 1. Tel number 2. Prefix (eg. 5040) of 4-port FXS device (Tel number: 50401~50404). 3. Prefix (eg. 09) of PSTN number (0916-553468) out of FXO device.

** SP5100 provides 20 sets of phone address.*

Command Line Interface:

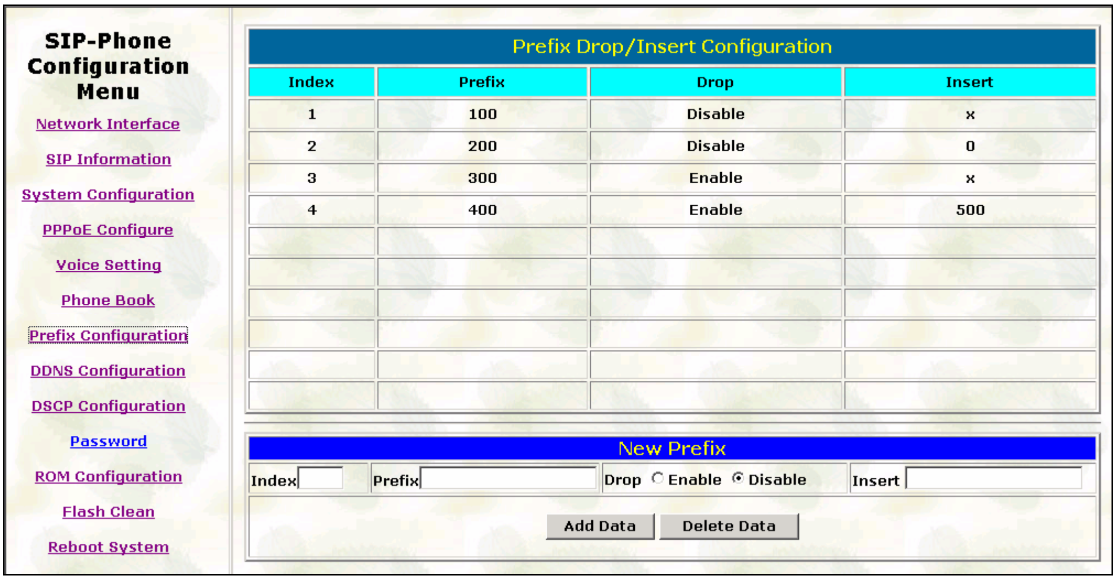
```
usr/config$ pbook -add name ip-phone-2 ip phone_02.micronet.com e164 50102
usr/config$ pbook -add name ip-phone-3 ip 211.56.148.33 e164 50103
usr/config$ pbook -add name 4afxs_gw ip 4afxs_01.micronet.com e164 5040
usr/config$ pbook -add name 2fxo_gw ip 211.56.148.35 e164 09
usr/config$ commit
usr/config$ reboot
```

Examples:

Destination	Dial Act
IP-Phone2	Press “50102#”
IP-Phone3	Press “50103#”
The 1st phone set of 4FXS	Press “50401#”
The 2nd phone set of 4FXS	Press “50402#”
The 3rd phone set of 4FXS	Press “50403#”
The 4th phone set of 4FXS	Press “50404#”
Cellular phone (0917771450)	Press “0915771450#”

3.1.2 Prefix

Web Interface:



Please set the parameters as follows:

Parameter	***Description***
Prefix	Set the prefix number of the whole numbers that could be incoming or what is dialed into this phone.
Drop	Set prefix dropped (enabled) or not (disabled).
Insert	Set the insert digits (or not) prior to the whole numbers that could be incoming or what is dialed into this phone.

NOTE:

- *For outgoing calls, prefix rules are checked after pbook checking.*
- *Prefix function may be not necessary in proxy mode. SIP server is in charge of digit manipulation for special dial rules.*

Command Line Interface:

usr/config$ prefix -add prefix 100 drop 0 insert x

usr/config$ prefix -add prefix 200 drop 0 insert 0

```
usr/config$ prefix -add prefix 300 drop 1 insert x
usr/config$ prefix -add prefix 400 drop 1 insert 500
usr/config$ commit
usr/config$ reboot
```

Examples:

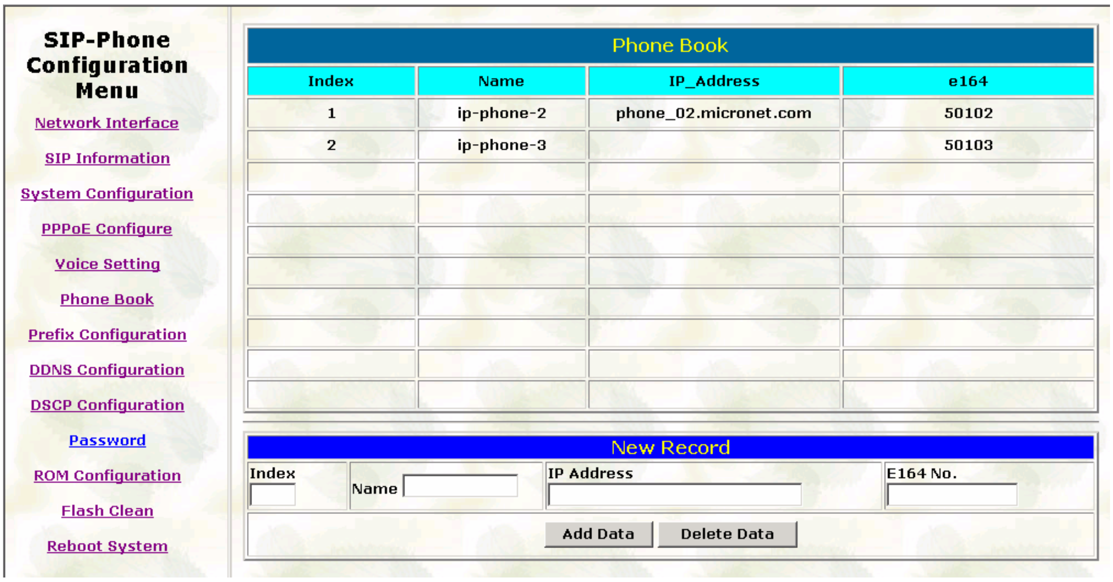
Index	Input (dialed number)	Output
1	100102	100102
2	200136112	0200136112
3	300736552	736552
4	40088152	500888152

3.2 Speed dial

SP5100 supports 10 sets of speed dial according to the first 10 sets in the phone address book. For proxy dialing peer, just fill out the field E.164 with registered TEL number of the device.

Speed dial	*Press **DL(1~10)***
	*Press **Speed** + **<index>** + #*
	*Press ＊ + **<index>** + #*

Web Interface:



Command Line Interface:

```
usr/config$ pbook -add name ip-phone-02 ip phone_02.micronet.com e164 50102
usr/config$ pbook -add name ip-phone-02 e164 50102
```

Examples:

Destination	Dial Act
IP-Phone2	*Press DL(1)*
	Press Speed + 1
	Press ∗1 #
IP-Phone3	*Press DL(2)*
	Press Speed + 2
	Press ∗2#

3.3 Hotline

It works only when it is in P2P mode. Pick up the handset, it directly dials out.

Command Line Interface:

```
usr/config$ sip -mode 0
usr/config$ sysconf -service 1 (hotline mode)
usr/config$ bureau -hotline <destination_ip> <destination_tel_number>
usr/config$ commit
usr/config$ reboot
```

3.4 Call Forward

Keypad Configuration:

There are 3 selections in Forward type. User can select the condition under which to forward calls.

**** Call Forward should be configured via keypads on the phone.***

Press **Forward**.

Busy: when the phone is in busy status, forward the call.

Select "1.Busy", and press **OK** to enter.
Select "1.Activate" and enter forward number.
Select "2.Deactivate" to disable busy forward.

No Respond: when the phone is not picked up for 10 seconds, forward the call.

Select "1.No Respond", and press **OK** to enter.
Select "1.Activate" and enter forward number.
Select "2.Deactivate" to disable busy forward.

Unconditional: forward the call in any conditions. It disables other two functions.

Select "1.Unconditional", and press **OK** to enter.
Select "1.Activate" and enter forward number.
Select "2.Deactivate" to disable busy forward.

3.5 Call Hold / Call Transfer

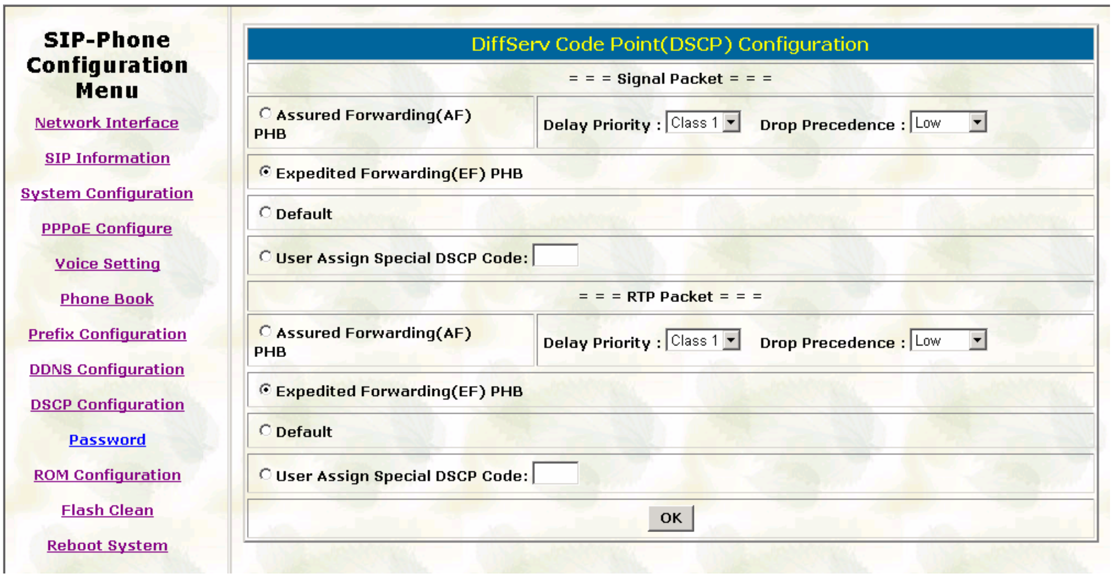
Call Hold	● *A talk is in progress…* ● *Press **Hold** to hold the call* *(Both sides can hear hold tone)* ● *Press **PSTN** for a PSTN call* ● *Press **Hold** again to resume*
Call Transfer	● *A talk is in progress…* ● *Press **Transfer** to hold the call* *(Press **Transfer** again to resume)* ● *Hear dial tone, and dial the third party's number* ● *Hang up after the third party picks up* ● *Call is transferred…*

4. Voice Quality

4.1 QoS

Enough bandwidth is critical to voice quality. When the IP-Phone is placed in the converged network of voice and data, DSCP is used to insure voice and sip traffic over data traffic. 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.

Web Interface:



Set it to DSCP value of assured forwarding (AF11, AF12…) or expedited forwarding (EF).

NOTE:

- *Within a converged network of voice and data, the junction devices (switch or router) should also support DSCP.*
- *The mapping of Code Point value of DS-field to egress traffic priority is shown as follows.*

Drop Precedence	Class 1	Class 2	Class 3	Class 4
Low	AF11 001010	AF21 010010	AF31 011010	AF41 100010

Medium	AF12 001100	AF22 010100	AF32 011100	AF42 100100
High	AF13 001110	AF23 010110	AF33 011110	AF43 100110

- High priority with DS-field.

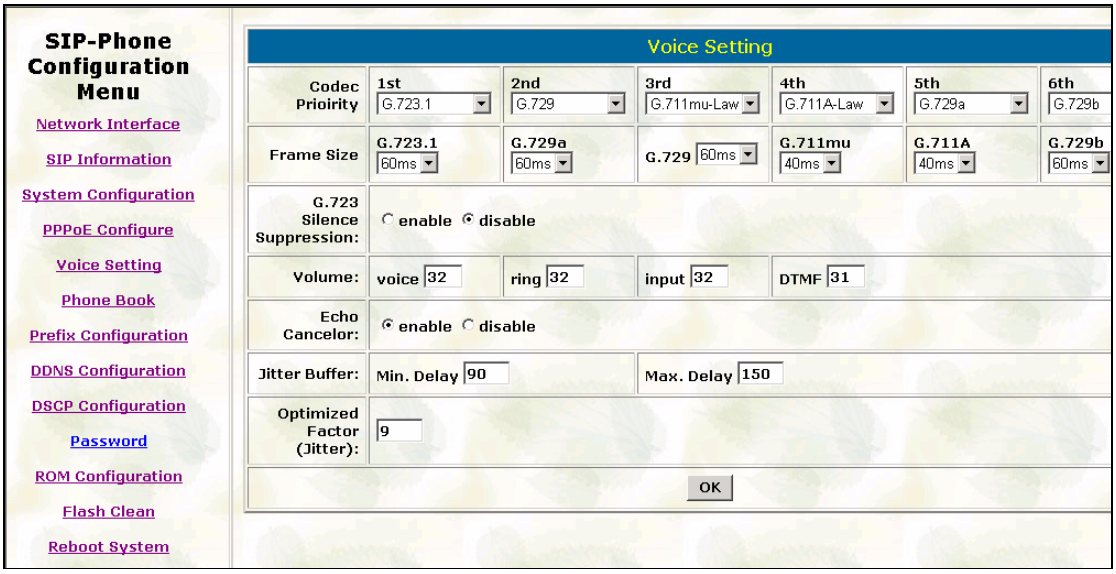
Expected Forwarding (EF)	101110	==>	46 (Decimal System)
Assured Forwarding (AF11)	001010	==>	10 (Decimal System)
Assured Forwarding (AF21)	010010	==>	18 (Decimal System)
Assured Forwarding (AF31)	011010	==>	26 (Decimal System)
Assured Forwarding (AF41)	100010	==>	34 (Decimal System)

- Low Priority with DS-field:

Assured Forwarding (AF12)	001100	==>	12 (Decimal System)
Assured Forwarding (AF22)	010100	==>	20 (Decimal System)
Assured Forwarding (AF32)	011100	==>	28 (Decimal System)
Assured Forwarding (AF42)	100100	==>	36 (Decimal System)
Assured Forwarding (AF13)	001110	==>	14 (Decimal System)
Assured Forwarding (AF23)	010110	==>	22 (Decimal System)
Assured Forwarding (AF33)	011110	==>	30 (Decimal System)
Assured Forwarding (AF43)	100110	==>	38 (Decimal System)
Best Effort (BE)	000000	==>	0 (Decimal System)

4.2 Voice Adjustment

Web Interface:



Please set the parameters as follows:

Parameter	***Description***
Codec Priority	To save bandwidth, set codec G.723.1 or G.729 as higher priority in order.
Frame size	Set different packet size for each codec. **With the larger size, the lower bandwidth is needed, but loss of frame can affect more when network quality is unreliable.**
Volume	Adjust the volume in “Voice” (for user to hear); “Input” (for peer side to hear); “DTMF” (DTMF transmitting volume), and ring.
Jitter Buffer	Set Min. Delay and Max. Delay of Jitter Buffer for voice packets. **With too high delay, voice quality is traded off. With too low delay, loss of frame can affect more when network quality is unreliable.**

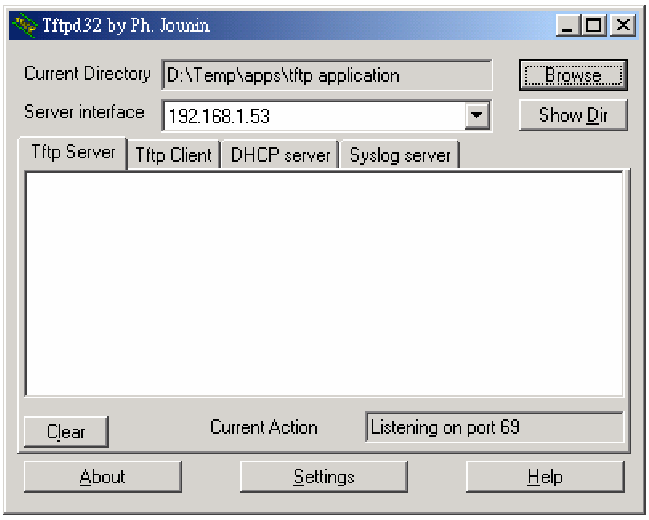
5. System Maintenance

5.1 System Upgrade

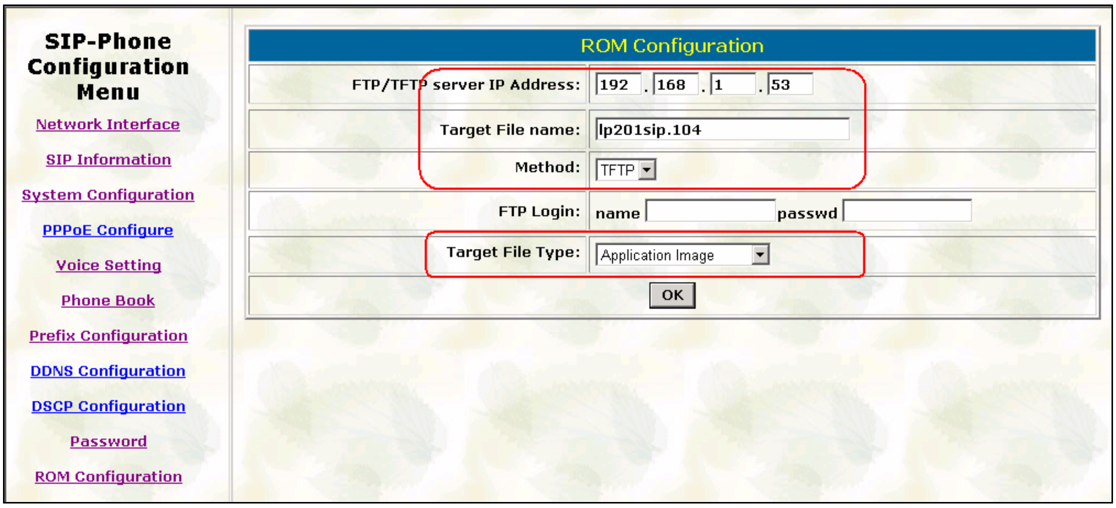
For most cases, use application rom file only to upgrade system. Please visit Micronet's web site http://www.micronet.info/ for update of firmware.

Firstly, prepare tftp server.

- Click Browse to location where the rom file is placed.
- Select the right server IP address.



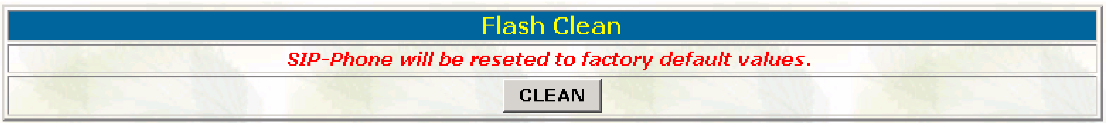
Web Interface:



Please set the parameters as follows:

Parameter	*Description*
FTP/TFTP server IP address	IP address of TFTP server (PC) or FTP server
Target File name	Name of rom file
Method	Choose TFTP or FTP to upgrade
FTP Login Name/Passwd	Username / Password for user to login FTP server
Target file type	Select rom file type (Application, Boot, 2M, DSP Application, DSP Core) to upgrade

Please reset the setting to default by “Flash Clean”, after upgrade is done.



Then, reboot the IP-Phone.

Command Line Interface:

usr/config$ rom –app –s <tftp_server_ip> -f <app_rom_name>
(or usr/config$ rom –boot2m –s <tftp_server_ip> -f <app_rom_name>
……
usr/config$ flash -clean

NOTE:

- *SP5100/s can be switched to SP5100 (H.323 model) by firmware upgrade.*
- *When update of boot rom is released, use 2M rom file to upgrade system. Be careful when using 2M rom to upgrade. Any carelessness might damage the unit.*

5.2 System Recovery

When the unit cannot boot completely, do this (the last resort) to recover the system is to upgrade firmware in boot mode. ***Make sure that the unit cannot boot normally after you try it many times.***

NOTE:

- *It needs special console kit to connect SP5100/s to PC.*
- *Please contact with local distributor for this repair fair.*
- *Any carelessness might crash the system.*

5.3 Password Recovery

Please contact with support@micronet.info. Send its MAC ID back for a unique password.

Telnet into the system by….

Login: mac

Password: <password_from_micronet>

Command to re-config password:

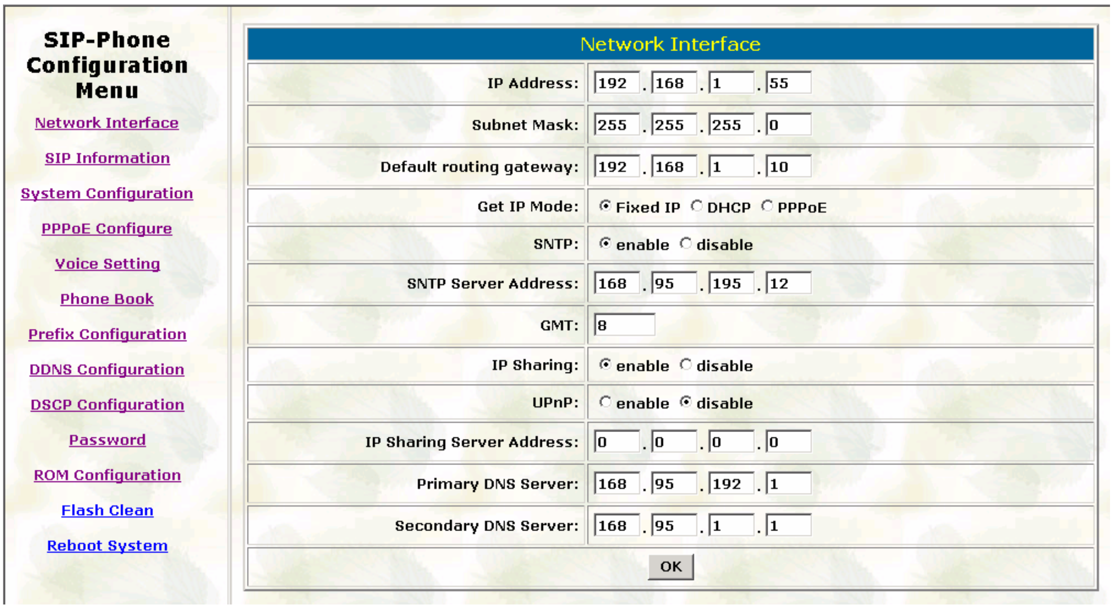
usr/config$ passwd –clean

usr/config$ passwd –set root <root_password>

usr/config$ passwd –set user <root_password>

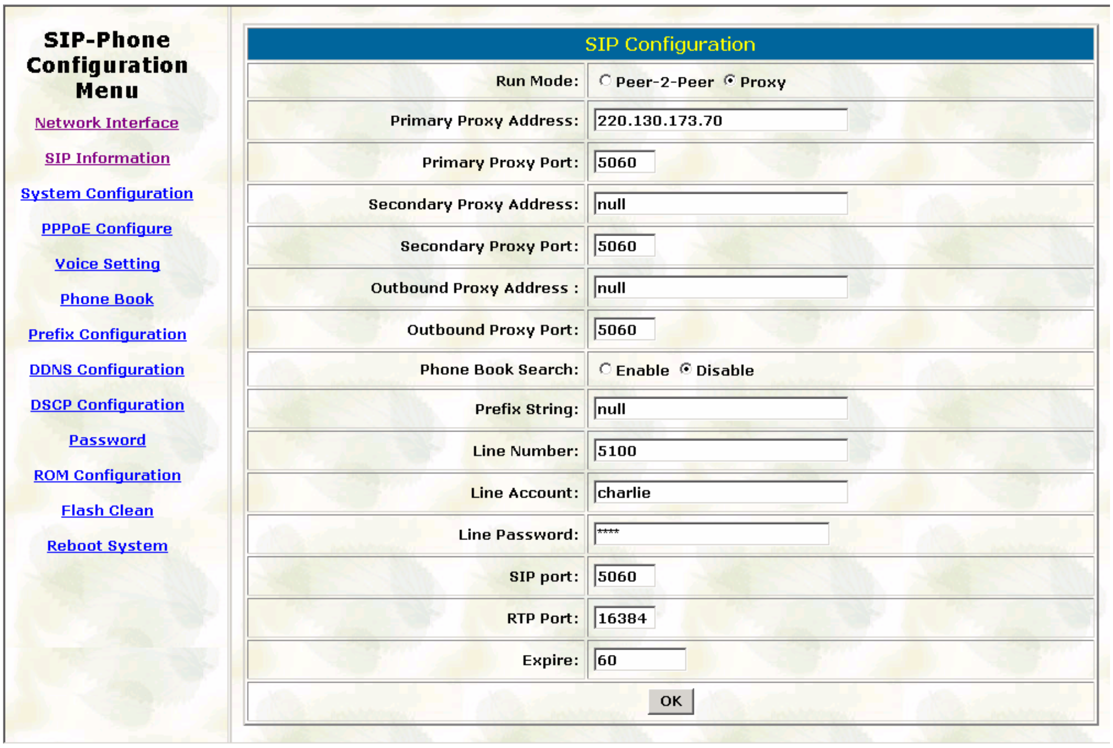
6. Web Administration

6.1 Network Interface



Parameter	***Description***
IP Address	IP Address of IP-Phone
Subnet Mask	Subnet Mask of IP-Phone
Default routing Gateway	Default routing gateway of IP-Phone
Get IP Mode	Network mode for IP-Phone to use
SNTP	Enable / Disable the Simple Network Time Protocol function
SNTP Server Address	SNTP Server Address
GMT	Time zone for SNTP Server time
IP Sharing	Enable it if IP-Phone is behind IP Sharing router.
UPnP	Enable it if IP sharing or NAT device supports UPnP.
IP Sharing Server Address	Public IP Address of IP Sharing router for IP-Phone.
Primary/Secondary DNS Server	Primary/ Secondary DNS IP address.

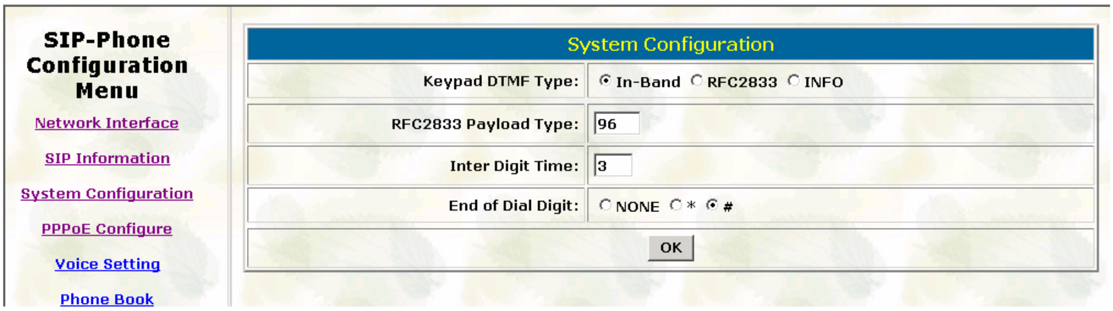
6.2 SIP Information



Parameter	***Description***
Run Mode	Select Peer-to-Peer or Proxy mode.
Primary/Secondary Proxy IP Address	Proxy IP / URL (domain name) address
Primary/Secondary Proxy port	Proxy port for IP-Phone to send message (default: 5060). If there is no special request of Proxy server, please don't change this value.
Outbound Proxy	IP / URL (domain name) address of outbound Proxy server.
Outbound Proxy Port	Port of outbound Proxy server.
Phone Book Search	Enable for IP-Phone to search dialed number in phone address book before sending call request to proxy server.
Prefix String	Prefix string to be added to dialed number. For example, if Account Name is 123, IP-Phone will sent out messages as Account Name @"IP address of Proxy", if user set prefix as abc, IP-Phone will set out as abc123@"IP

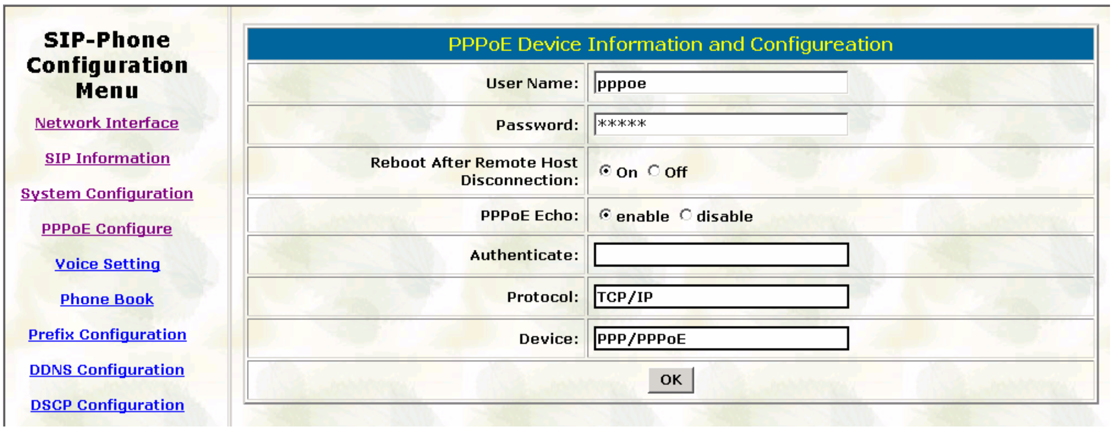
	address of Proxy”. This function is for special proxy server.
Line Number	IP telephony number for the IP-Phone.
Line Account	User name of IP-Phone for registering.
Line Password	User password for registering.
SIP Port	SIP UDP port.
RTP Port	RTP port for sending voice data.
Expire	Expire time of registration. IP-Phone will keep registering to proxy server before expire is time out.

6.3 System Configuration



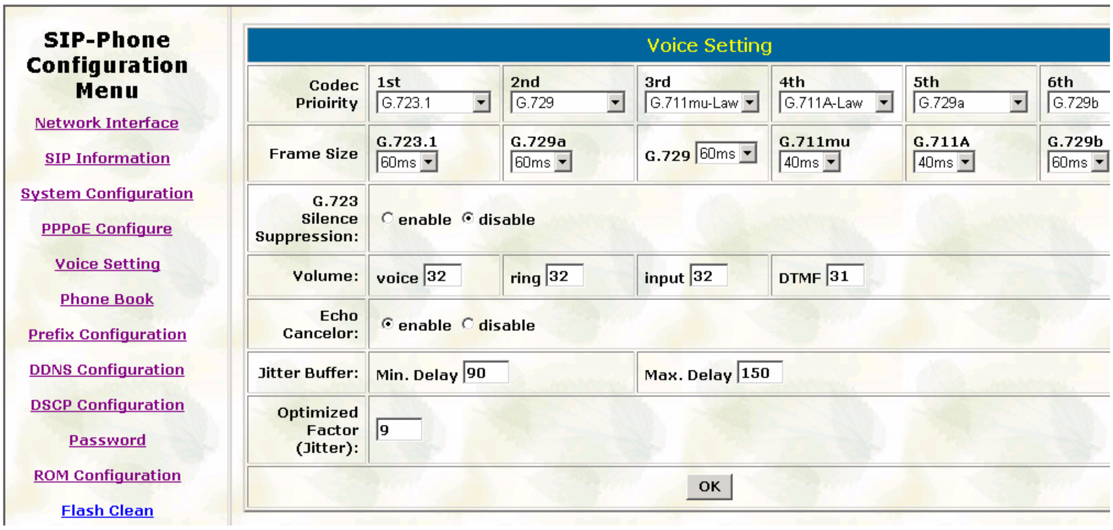
Parameter	***Description***
Keypad DTMF Type	DTMF type (In-band, RFC2833, SIP info) for IP-Phone to replay DTMF.
RFC2833 Payload	RFC2833 Payload type. This is for special request from the other site. **If RFC2833 payload types of 2 sites are different, it may cause some problem of connection.**
Inter Digit Time	The duration (in second) of two pressed digits in dial mode. If user hasn't pressed next number after the duration, IP-Phone will dial out all number pressed.
End of Dial Digit	End of dialing key. E.g. set end of dial key as * button. After pressing *, dialed number will be dialed out.

6.4 PPPoE Configuration



Parameter	***Description***
User Name	PPPoE authentication User Name.
Password	PPPoE authentication password.
Reboot After Remote Host Disconnection:	
	If user enables this function, IP-Phone will automatically reboot to re-connect after PPPoE disconnected. If IP-Phone still can't connect with server after reboot, IP-Phone will keep trying to connect. On the other hand, if user disables this function, IP-Phone won't reboot and keep trying to connect.
PPPoE Echo	With echo enabled, IP-Phone sends "Echo Request" message, and PPPoE server sends "Echo Reply". The RFC 2516 recommends both sides use this method to maintain the session. Disable it in case PPPoE server doesn't support it.
Other items	For reference only.

6.5 Voice Setting



Parameter	***Description***
Codec Priority	Set codec G.723.1 or G.729 as higher priority in order.
Frame size	Set different packet size for each codec. **With the larger size, the lower bandwidth is needed, but loss of frame can affect more when network quality is unreliable.**
G.723 Silence Suppression	
	Enable / disable sound compression and comfort noise generation. It is only for codec G.723.1
Volume	Adjust the volume in “Voice” (for user to hear); “Input” (for peer side to hear); “DTMF” (DTMF transmitting volume), and ring.
Echo Cancellation	
	Enable / disable echo cancellation function
Jitter Buffer	Min. Delay and Max. Delay of Jitter Buffer for voice packets. **With too high delay, voice quality is traded off. With too low delay, loss of frame can affect more when network quality is unreliable.**
Optimized Factor (Jitter)	
	Set Optimized Factor of voice. This is for advanced user only. Please contact with your distributor before making any change.

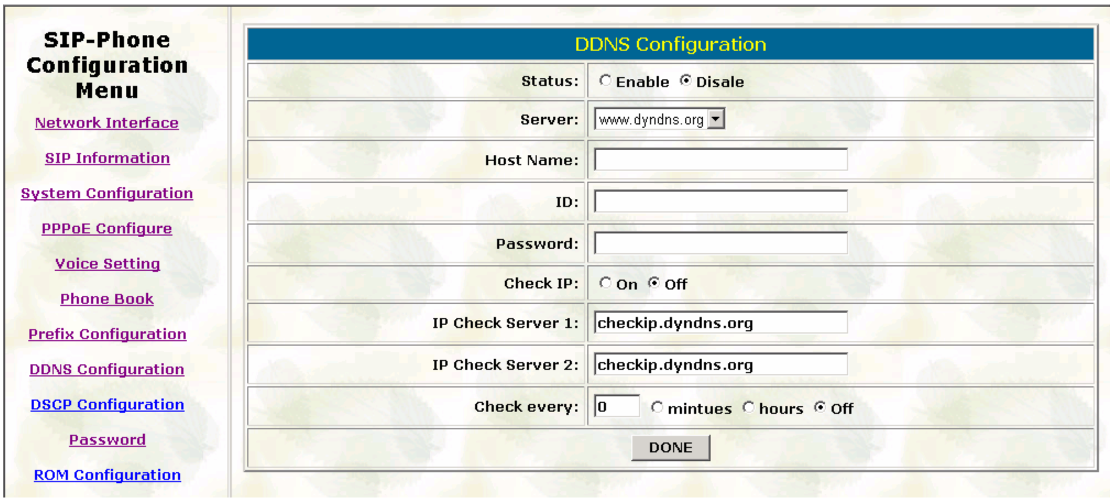
6.6 Phone Book

Please refer the section 3.3.1 Phone Address Book.

6.7 Prefix Configuration

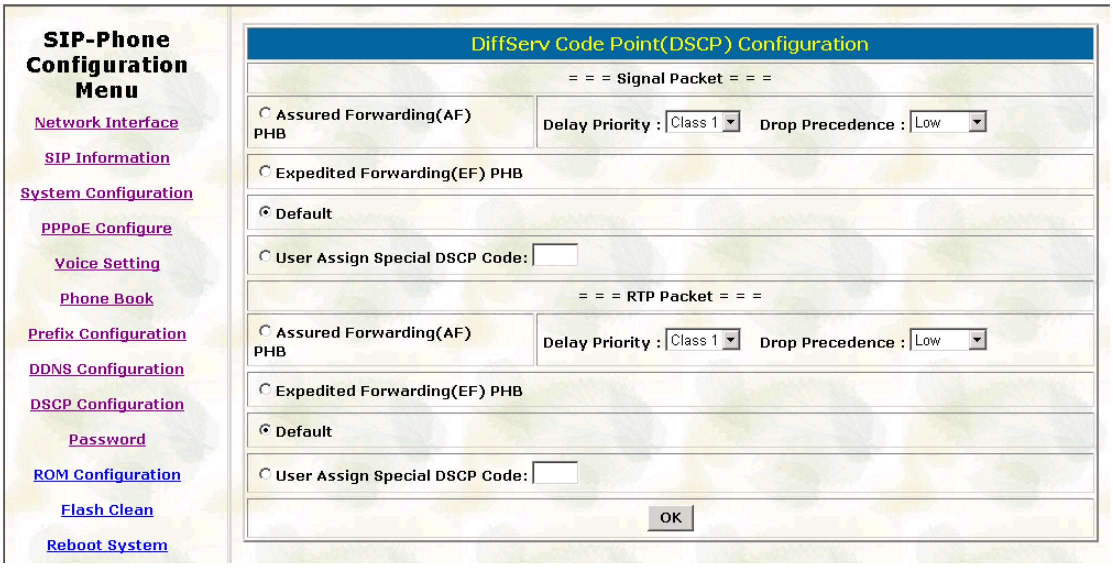
Please refer the section 3.3.1 Prefix.

6.8 DDNS Configuration



Parameter	*Description*
Status	Enable/disable DDNS function
Server	Choose one DDNS server on which user has already registered. Now only one DDNS server is available --- www.dyndns.org.
Host Name	The registered Domain Name of IP Phone
ID	Login ID of registered account to log in DDNS server
Password	Password of registered account to log in DDNS server
Check IP	Enable / disable check IP function. **If "Check IP" is enabled and the Phone is behind IP sharing, SIP Phone will check its public IP address by asking IP check server and send to DDNS server to update DDNS data. If this function is disabled and the Phone is behind IP sharing, it will send its private IP address to DDNS server.**
IP check Server 1/2	IP address of Primary / Secondry IP check server
Check every	Set the update interval time (in minutes / hours, off). IP-Phone will re-update its IP address in this time.

6.9 DSCP Configuration

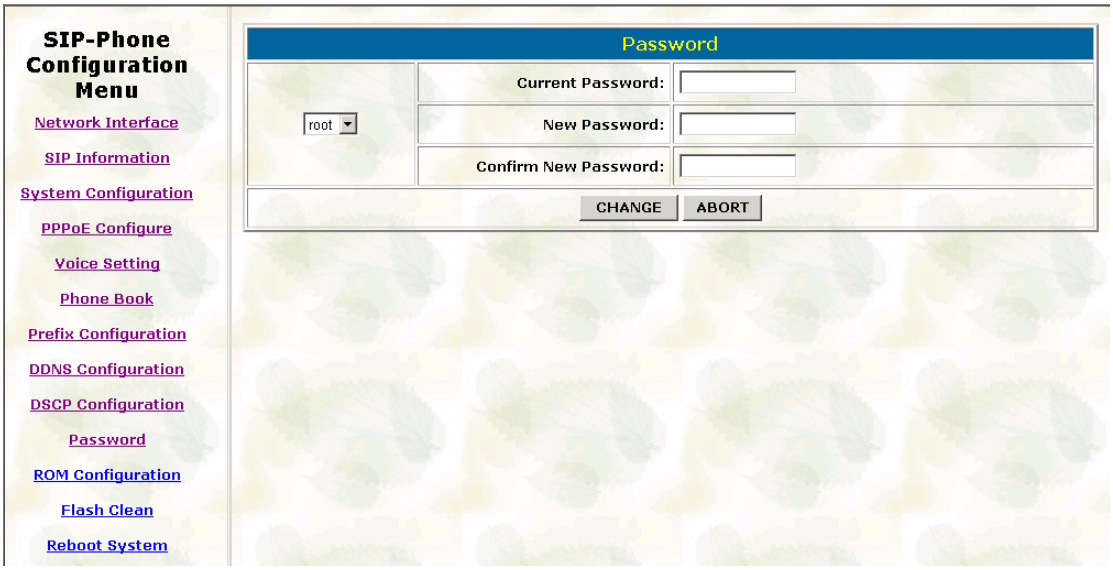


Parameter	***Description***
	Set Signal or RTP Packet DSCP value
Assured Forwarding (AF) PHB	Select Delay priority (Class 1-4) and Drop Precedence (low/medium/high)
Expedited Forwarding (EF) PHB	Select DSCP value as EF.
Default	Select DSCP value as 0
User Assign Special DSCP Code	User can set other unspecified value.

Please refer to RFC standard documents for more information about what is DSCP.

NOTE: *Within a converged network of voice and data, the junction devices (switch or router) should also support DSCP.*

6.10 Password

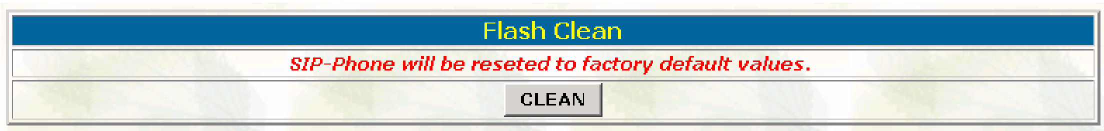


Parameter	*Description*
Root / User	Select login name as root or user
Current Password	
New password and	
Confirm new password	
Change	Apply the change
Abort	Clean all inputs.

6.11 ROM Configuration

Please refer the section 5.1 System Upgrade.

6.12 Flash Clean



Press CLEAN will clean all configurations of IP-Phone and reset to factory default value except for network setting.

6.13 Reboot

Press Reboot will reset IP-Phone.

NOTE: *To execute reboot via web browser, IP-Phone will automatically save all data before reboot. To execute reboot via TELNET command, please place a command "commit" before reboot.*

---

7. Command Line Interface

7.1 [help] command

Type **help**, **man**, or **?** to display all the command lists.

```
--------------------------------------------------------------------------------
usr/config$ ?

help        help/man/? [command]
quit        quit/exit/close
debug       show debug message
reboot      reboot local machine
pbook       Phonebook information and configuration
commit      commit flash rom data
ping        test that a remote host is reachable
time        show current time
ifaddr      internet address manipulation
pppoe       PPPoE stack manipulation
flash       clean configuration from flash rom
sysconf     System information manipulation
sip         SIP information manipulation
security    Security information manipulation
voice       Voice information manipulation
tos         IP Packet ToS (Type of Service)values
ddns        Dynamic DNS update manipulation
vlan        VLAN configuration and information
bureau      Bureau line information manipulation
prefix      Prefix drop/insert information manipulation
rom         ROM file update
passwd      Password setting information and configuration
auth        Set configuration items for "user".

usage: help [command]
--------------------------------------------------------------------------------
```

7.2 [quit] command

Type **quit** / **exit** / **close** will logout IP-Phone and close Telnet Program.

7.3 [debug] command

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

```
--------------------------------------------------------------------------------
    usr/config$ debug

    Debug message information and configuration
    Usage:
    debug [-add type1 [[type2]...]] | -open | -close | -status

        -status    Display the enabled debug flags.
        -add       Add debug flag.
        -delete    Remove specified debug flag.
        -open      Start to show debug messages.
        -close     Stop showing debug messages.
    Example:
        debug -add sip msg
        debug -open

    usr/config$
--------------------------------------------------------------------------------
```

7.4 [reboot] command

After typing commit command, type reboot to restart the IP-Phone.

7.5 [flash] command

This command "flash -clean" will clean the configuration stored in the flash rom to default value and reboot the IP-Phone.

```
--------------------------------------------------------------------------------
    Flash memory information and configuration
    Usage:
    flash -clean

    Note:
        This command will clean the configuration stored in
        the flash and reboot it.

    usr/config$
--------------------------------------------------------------------------------
```

After user upgrade new software version, please execute the command to make sure new software work well on IP-Phone.

NOTE: *To execute reboot via web browser, IP-Phone will automatically save all. To execute the command "flash –clean", all configuration of IP-Phone stored in flash will be cleaned. User's login name is "root" only.*

7.6 [commit] command

Save any changes after configuring the IP-Phone.

```
usr/config$ commit



This may take a few seconds, please wait....

Commit to flash memory ok!
usr/config$
```

7.7 [ifaddr] command

Configure and display the IP-Phone IP 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 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.
```

-ipsharing Specify usage of an IP sharing device and specify IP address.
-upnp Specify the upnp mode of ipsharing (0:Off/1:On)
-server Specify EMS Server IP address (x:disable)
-id Specify EMS Server ID
-pwd Specify EMS Server password
-emstime Specify EMS cycle time

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 -upnp 1
ifaddr -autodns 1
ifaddr -dns 1 168.95.1.1

usr/config$
```

- **-print**: print out all current configurations of ifaddr command.
- **-ip, -mask, -gate**: Set IP-Phone IP Address, subnet mask and default gateway respectively.
- **-ipmode**: Set IP-Phone IP mode to be Fixed IP, DHCP or PPPoE.
- **-sntp**: When SNTP server is available, enable IP-Phone SNTP function and assign SNTP server IP address so that IP-Phone can capture current time from SNTP server. (**ifaddr –sntp 1 "xxx.xxx.xxx.xxx"**)
- **-dns**: User can set Domain Name Server IP address. Once IP-Phone can connect with DNS server, user can specify URL address instead of IP address for Proxy Server and phone book IP address...etc.
- **-timezone**: User can set different time zone according to the location IP-Phone is. For example, in Taiwan the time zone should be set as 8,which means GMT+8. (GMT-8: **ifaddr –timezone -8**)
- **-ipsharing**: If IP-Phone is behind a IP-sharing device, user must

enable IP sharing function and specify public IP address. (ifaddr -ipsharing 1 <wan_ip>, 0 for disable and 1 for enable)

- **-upnp**: enable / disable UPnP function. If the IP sharing or NAT device supports UPnP, user can enable UPnP function. IP-Phone will automatically connect with NAT device without any configuration in IP Phone and NAT device.

7.8 [time] command

When IP-Phone enable SNTP function, it is able to connect with SNTP server. Type **time** command, it will show the current time retrieved from SNTP server.

```
usr/config$ time

Current time is SAT FEB 07 08:04:44 2006

usr/config$
```

7.9 [ping] command

Command ping can test which the IP address is reachable or not.

```
usr/config$ ping 192.168.1.152

PING 192.168.1.152: 56 data bytes
64 bytes from 192.168.1.152: icmp_seq=0. time=0. ms
64 bytes from 192.168.1.152: icmp_seq=1. time=0. ms
64 bytes from 192.168.1.152: icmp_seq=2. time=0. ms
64 bytes from 192.168.1.152: icmp_seq=3. time=0. ms
----192.168.1.152 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)    min/avg/max = 0/0/0
usr/config$
```

7.10 [pbook] command

This command is functional both in Proxy mode and Peer-to-Peer mode. In proxy mode, it's used for speed dial or 10 DL button to dial out e.164 number in phone book. In peer-to-peer mode, IP-Phone will dial out IP address.

```
--------------------------------------------------------------------------------
usr/config$ pbook

Phonebook information and configuration
Usage:
pbook [-print [start_record] [end_record]]
pbook [-add [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-search [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-insert [index] [ip ipaddress] [name Alias] [e164 phonenumber]]
pbook [-delete index]
pbook [-modify [index] [ip ipaddress] [name Alias] [e164 phonenumber]]

    -print      Display phonebook data.
    -add        Add an record to phonebook.
    -search     Search an record in phonebook.
    -delete     Delete an record from phonebook.
    -insert     Insert an record to phonebook in specified position.
    -modify     Modify an exist record.
Note:
    If parameter 'end_record' is omited, only record 'start_record' will be
lay.
    If   both parameters 'end_record' and 'start_record' are omited, all reco
will be display.
    Range of ip address setting (0.0.0.0 ~ 255.255.255.255).
    Range of index setting value (1~100),
Example:
    pbook -print 1 10
    pbook -print 1
    pbook -print
    pbook -add name Test ip 210.59.163.202 e164 1001
```

```
pbook -insert 3 name Test ip 210.59.163.202 e164 1001
pbook -delete 3
pbook -search ip 192.168.4.99
pbook -modify 3 name Test ip 210.59.163.202 e164 1001

usr/config$
```

- **-print**: display phone book data. User can print all data in phone book by command (**pbook -print**). Furthermore, user can also print only a section of data by indicate parameter "start index" and "end index" (**pbook -print "start index" "end index"**). If parameter "end index" is omitted, only record "start index" will be displayed. (**pbook -print "start prefix"**).
- **-add**: add a new record in phone book table by giving name, IP address, and e.164 number of callee endpoint. (**pbook –add name "X" ip "xxx.xxx.xxx.xxx" e164 "X"**).
- **-search**: search any record in the phone book according to IP address, name or e164 number.
- **-delete**: delete a record of certain listed index in phone book table. (**pbook –delete "index number"**).
- **-insert**: insert an record in specified index of phone book.
- **-modify**: modify record of a certain index in phone book. Please notice that the name, IP address and e164 number must be modified together; user cannot just modify one parameter only. (**pbook -modify "index" name "X" ip "xxx.xxx.xxx.xxx" e164 "X"**).

7.11 [pppoe] command

```
usr/config$ pppoe

PPPoE device information and configuration
Usage:
```

```
pppoe [-print]
pppoe [-id username][-pwd password]

    -print      Display PPPoE device information.
    -id         Connection user name.
    -pwd        Connection password.
    -reboot     Reboot after remote host disconnection.
    -echo       PPPoE Echo Request (0=disable, 1=enable).

usr/config$
```

- **-print**: display all current configurations and information.
- **-id**: set PPPoE authentication user name.
- **-pwd**: set PPPoE authentication password.
- **-reboot**: enabled by default. If user enables this function, IP-Phone will automatically reboot to re-connect after PPPoE disconnected. If IP-Phone still can't connect with server after reboot, IP-Phone will keep trying to connect. If user disables this function, IP-Phone won't reboot and keep trying to connect. (**pppoe -reboot 0** / **1**)
- **-echo**: IP-Phone sends "Echo Request" message, and PPPoE server sends "Echo Reply". The RFC 2516 recommends both sides use this method to maintain the session. Disable it in case PPPoE server does not support it.

7.12 [sysconf] command

```
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)
 -service          Specify lanphone service type. (0: Normal service,
                   1: HotLine service.)
 -keypad           Select DTMF type: 0=In-band, 1=RFC2833, 2=INFO.
 -2833type         RFC2833 Payload Type (range: 96~128 inter-used:100,102~105)
 -phone            Add the parameter user=phone (1: NO, 0:OFF)
 -eod              End of Dial Digit setting (0: NONE, 1: *, 2: #)
Example:
  sysconf -keypad 0 -eod 2

usr/config$
```

- **-print**: display all current configurations.
- **-idtime**: set the duration (in second) of two pressed digits in dial mode as timed out. If user hasn't pressed next number after the duration, IP-Phone will dial out all number pressed.
- **-keypad**: set type of DTMF replay. User can select DTMF type that IP-Phone receives / transmits in dial mode.
- **-2833type**: change RFC2833 Payload type.
- **-eod**: select end of dialing key. E.g. set end of dial key as "＊" button, and then dialing number will be immediately dialed out after pressing "＊".
- **-service**: set IP Phone to be normal mode or under hotline mode. In hotline mode, user has to set IP-Phone as P2P mode, and hotline table under bureau command.
- **-phone**: set the parameter user=phone enabled / disabled (default)

7.13 [sip] command

```
usr/config$ sip
```

```
SIP stack information and configuration
Usage:
sip [-mode    pxmode]
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.
        -mode           Configure as Proxy mode or Peer-to-Peer mode.
        -px             Proxy server address.(Proxy IPv4 address or Proxy dns name)
        -pxport         proxy server port.     (the port of proxy)
        -px2            Secondary proxy server address.(Proxy IPv4 address or dns name)
        -px2port        Secondary proxy server port.     (the port of Secondary proxy)
        -outpx          OutBound Proxy server address. (IPv4 address or dns name)
        -outpxport      OutBound Proxy server port. (the port of OutBound proxy)
        -prefix         Specify prefix string, use it when UserID contains alphabets
                        (if UserID uses numerals, specify as null)
        -line           TEL Phone number.
        -pbsearch       Search phone book       0: off / 1:on.
        -expire         The relative time after which the message expires (0 ~ (2^31-1))
        -port           SIP local UDP port number (5060~5070), Default: 5060
        -rtp            RTP receive port number (2326~65534), Default: 16384
Example:
          sip -mode 1
          sip -px 210.59.163.171 -line 70

usr/config$
```

- **-print**: display all current configurations.
- **-mode**: configure IP-Phone as Proxy or Peer-to-Peer Mode.
- **-px**: set primary proxy server's IP / URL address
- **-pxport**: set listening port of primary proxy server.
- **-px2**: set secondary proxy server's IP / URL address

- **-px2port**: set listening port of secondary proxy server.
- **-outpx**: set IP address of outbound proxy server. After user set outbound proxy, all packets form IP-Phone will be sent to outbound proxy server.
- **-outpxport**: set port of outbound proxy server.
- **-prefix**: set prefix string.
- **-line**: identify one number for the IP-Phone to register to the Proxy.
- **-pbsearch**: IP-Phone will make call according to phone address before sending call request to Proxy server.
- **-expire**: set expire time of registration. IP-Phone will keep re-registering to proxy server before expire is time out.
- **-port**: set listening UDP port or IP-Phone.
- **-rtp**: set RTP port number. IP-Phone will use this port to send and receive voice.

7.14 [security] command

```
--------------------------------------------------------------------------------
usr/config$ security

Secuirty information and configuration
Usage:
  security [-name username] [-password password]
  security [-print]

  -print            Display system account information and configuration.
  -name             Specify user name.
  -pwd              Specify password.
Example:
    security -name 1001 -pwd 1001

usr/config$
```

- **-print**: display all current configurations.
- **-name**: set user ID of IP-Phone for registering.
- **-pwd**: set account password for registering.

7.15 [voice] command

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

```
usr/config$ voice

Voice codec setting information and configuration
Usage:
voice [-send [G723 ms] [G729 ms] [G711U ms] [G711A ms] [G729A ms] [G729B ms] ]
        [-volume line [voice level] [ring level] [input level] [dtmf    level]]
        [-nscng [G711U used1] [G711A used2] [G723 used3]]
        [-echo used] [-mindelay t1] [-maxdelay t2] [-optfactor f]
voice -print
voice -priority [G723] [G729] [G711U] [G711A] [G729A] [G729B]

     -print       Display voice codec information and configuration.
     -send          Specify sending packet size.
                     G.723      (30/60/90 ms)
                     G.729      (20/40/60 ms)
                     G.711U     (20/40/60 ms)
                     G.711A     (20/40/60 ms)
                     G.729A     (20/40/60 ms)
                     G.729B     (20/40/60 ms)
     -priority    Priority preference of installed codecs.
                     G.723
                     G.729
                     G.711U
                     G.711A
                     G.729A
                     G.729B
```

```
    -volume        Specify the following levels:
                      voice volume (0~45, default: 32),
                      ring volume (0~45, default: 32),
                      input gain (26~45, default: 32),
                      dtmf    volume (0~31, default: 31),
    -nscng         No sound compression and CNG. (G.723.1 only, On=1, Off=0).
    -echo          Setting of echo canceller. (On=1, Off=0, per port basis).
    -mindelay      Setting of jitter buffer min delay. (0~150, default: 90).
    -maxdelay      Setting of jitter buffer max delay. (0~150, default: 150).
Example:
    voice -send g723 60 g729 60 g711u 60 g711a 60 g729a 60 g729b 60
    voice -volume voice 20 input 32 dtmf 27
    voice -echo 1
usr/config$
--------------------------------------------------------------------------------
```

- **-print**: display voice codec information and configuration.
- **-send**: packet size can be configured as 20 ms, 40 ms or 60 ms.(only 30 and 60 ms for G..723.1)
- **-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 g723 711a g711u g729 g729a” means IP-Phone can support only one codec up to five codecs.
- **-volume**: three types can be adjustable: **(0: -32db, 1: -31db)**
  - Voice means the volume user can hear,
  - Input gain means the volume the other side can hear from IP-Phone,
  - DTMF means DTMF transmitting volume.
- **-nscng**: enable / disable sound compression and comfort noise generation. It is only for codec G.723.1.
- **-echo**: enable / disable echo cancellation function.
- **-mindelay**: set minimum delay of jitter buffer(0~150)
- **-maxdealy**: set maximum delay of jitter buffer(0~150)

7.16 [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 priority is shown as follows.

Drop Precedence	Class 1	Class 2	Class 3	Class 4
Low	AF11 001010	AF21 010010	AF31 011010	AF41 100010
Medium	AF12 001100	AF22 010100	AF32 011100	AF42 100100
High	AF13 001110	AF23 010110	AF33 011110	AF43 100110

- High priority with DS-field.

Expected Forwarding (EF)	101110	==>	46 (Decimal System)
Assured Forwarding (AF11)	001010	==>	10 (Decimal System)
Assured Forwarding (AF21)	010010	==>	18 (Decimal System)
Assured Forwarding (AF31)	011010	==>	26 (Decimal System)
Assured Forwarding (AF41)	100010	==>	34 (Decimal System)

- Low Priority with DS-field:

Assured Forwarding (AF12)	001100	==>	12 (Decimal System)
Assured Forwarding (AF22)	010100	==>	20 (Decimal System)
Assured Forwarding (AF32)	011100	==>	28 (Decimal System)
Assured Forwarding (AF42)	100100	==>	36 (Decimal System)
Assured Forwarding (AF13)	001110	==>	14 (Decimal System)
Assured Forwarding (AF23)	010110	==>	22 (Decimal System)
Assured Forwarding (AF33)	011110	==>	30 (Decimal System)
Assured Forwarding (AF43)	100110	==>	38 (Decimal System)
Best Effort (BE)	000000	==>	0 (Decimal System)

```
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$
```

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

7.17 [bureau] command

```
usr/config$ bureau

Bureau line setting information and configuration
Usage:
bureau    [-hotline [Port DestIP TELnum]]
bureau -print

    -print        Display Bureau line information and configuration.
    -hotline      Set Hot line information.
Note:
    Hotline feature should be used together with:
        $sysconf -service 1 (HotLine service)
```

```
        $sip      -mode     0 (peer-to-peer mode)
Example:
    bureau -hotline 192.168.4.69 628

usr/config$
```

- **-print**: show all current configurations.
- **-hotline**: set hotline table. User can set hotline function to specify one IP address for IP-Phone to dial out directly.

7.18 [rom] command

```
usr/config$ rom

ROM files updating commands
Usage:
rom [-print][-app][-boot][-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)
    -boot2m      update 2M 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', 'boot',
    , 'dsptest', 'dspcore', and 'dspapp'.
Example:
    rom -method 1
```

```
    rom -ftp vwusr vwusr
    rom -app -s 192.168.4.101 -f app.bin

usr/config$
```

- **-print**: show all current configurations and version information.
- -app,-boot, -dsptest, -dspcore, -dspapp,: upgrade main boot code, main application code, DSP testing code, DSP kernel code, DSP application code, Ring Back Tone PCM file and Hold Tone.
- **-boot2m**: to upgrade 2mb rom file, which includes all firmware file mentioned in item 2.
- **-s**: it is necessary to prepare TFTP/FTP server IP address for upgrading firmware rom file.
- **-f**: the file name prepared for upgrading is necessary as well.
- **-method**: specify download method to be TFTP or FTP
- **-ftp**: specify user name and password for FTP download method.

7.19 [passwd] command

For security protection, user has to input the password before entering application user/config mode.

```
usr/config$ passwd

Password setting information and configuration
  Usage:
      passwd -set Loginname Password
      passwd -clean
  Note:
      1. Loginname can be only 'root' or 'user'
      2. passwd -clean will clear all passwd stored in flash,
            please use it with care.
```

```
Example:
    passwd -set root Your_Passwd_Setting

usr/config$
```

- **-set**: set password of “root” users or “administrator” users.
- **-clean**: clean up password restored before

7.20 [ddns] command

```
usr/config$ ddns

The dynamic DNS service information and configuration
Usage:
ddns [-print]
ddns [-enable 0/1]
ddns [-serve Address] [-hostname Name] [-id ID]
     [-passwd Password]
     [-checkip option] [-checkipsrv Address]
     [-delay time]
     [-force IP]

    -print          Display Dynamic DNS information and configuration.
    -enable         1:Enabled/0:Disable the dynamic DNS service.
    -server         Specify DDNS server address.
    -hostname       Registered domain name.
    -id             Registered account ID.
    -passwd         Registered account password.
    -checkip        1:Enabled/0:Disable check the host current IP address.
    -checkipsrv1    Specify IP address check server.
    -checkipsrv2    Specify secondary IP address check server.
    -delay          Setting the service delay time.(1~59 minutes or 1~24 horus)
    -force          Force execute the dynamic DNS service.
Example:
    ddns -print
```

ddns -enable 1
ddns -server member.dyndns.org -hostname ipphone.dyndns.org
ddns -delay 30 m (30 minutes)
ddns -delay 12 h (12 hours)
ddns -force 11.22.33.44
usr/config$

- **-print**: display DDNS overall information and configuration.
- **-enable**: to enable/disable DDNS function.
- **-server**: to set IP address of DDNS login server. (Now only one DDNS server is member.dyndns.org.)
- **-hostname**: to set the registered Domain Name of SIP Phone.
- **-id**: to set login ID of registered account to log in DDNS server.
- **-passwd**: to set password of registered account to log in DDNS server.
- **-checkip**: to enable/disable check IP function.
- **-checkipsrv1/2**: to set IP address of primary and secondary IP address check server.
- **-delay**: to set the update interval time. SIP Phone will re-update its IP address in this time.
- **-force**: to force to execute DDNS update. Once user excutes this command, SIP Phone will update DDNS data immediately.

7.21 [prefix] command

----
usr/config$ prefix

Prefix drop/insert information and configuration
Usage:
prefix -add [prefix number][drop number][insert digits]
prefix -delete index

prefix -modify index [prefix number][drop number][insert number]
prefix -print Prefix drop/insert information.

Prefix	The prefix of dialed number.
Drop	Drop prefix(Enable:1/Disable:0).
Insert	Insert digits.

Example:

prefix -add prefix 100 drop 1 insert 2000
prefix -add prefix 100 drop 1
prefix -add prefix 100 drop 0 insert 200
prefix -delete 1
prefix -modify 1 prefix 100 drop 0 insert 300

usr/config$

7.22 [vlan] command

----

usr/config$ vlan

VLAN information and configuration

Usage:

vlan [-dev enable] [-vid number] [-pcvid number]
[-priority number] [-pcpriority number] [-pcdroptag enable]

-print	Display VLAN information and configuration.
-dev	Enable/disable VLAN configuration. (0:Disable / 1:Enable.)
-vid	VID number.(1~4095, default:1)
-pcvid	PC port VID number.(1~4095, default:1)
-priority	Priority number.(0~3:Low priority, 4~7:High priority)
-pcpriority	PC port priority number.(0~3:Low priority, 4~7:High priority)
-pcdroptag	Drop PC port tag. (0:Disable / 1:Enable)

Example:

vlan -dev 1
vlan -vid 2
vlan -pcvid 2
vlan -priority 4
vlan -pcpriority 0

vlan -pcdroptag 1

usr/config$ vlan -print

(default value)

VLAN information

Device	: Disable
VID	: 100
Priority	: 5
PC port VID	: 102
PC port priority	: 0
Drop PC port tag	: Enable

usr/config$

- **-print**: display current VLAN configuration settings.
- **-dev**: enable/disable VLAN configuration.
- **-vid**: set CPU VLAN ID number.
- **-pcvid**: set PC port VLAN ID number.
- **-priority**: set CPU packets priority number.
- **-pcpriority**: set pc port priority number.
- **-pcdroptag**: enable/disable pc port drop tag function. If this function is enabled, SIP Phone will drop priority tag on packets sending out from PC port.

NOTE: *Only when vid numbers are the same, network can connect.*

7.23 [auth] command

The command is used to define authority that user has in each configuration section.

----

usr/config$ auth

```
Root control what command user can use.
Usage:
auth -print Display auth switch configuration.
     -auth    Caution:Set user have switch configuration.
              Use item name to do config name (0=Disable, 1=Enabled).
Example: auth -sip 1

usr/config$ auth -print

Root can control what command user can use.
    Auth       : Disable
    Flash      : Disable
    Sysconf    : Disable
    Sip        : Disable
    Security   : Disable
    Voice      : Disable
    Tos        : Disable
    Vlan       : Disable
    Bureau     : Disable
    Rom        : Disable
    Setmac     : Disable
    Debug      : Disable

usr/config$
--------------------------------------------------------------------------------
```

8. LCD / Command Menu Interface

User can set the following configurations by LCD keypad.

NOTE:

- *Press TRANSFER to switch characters to be capital or lowercase.*
- *Press SPEED to switch input mode as character mode or digit mode*
- *Press ← or → to enter keypad configuration mode, then press **OK** button to enter sub menus.*
- *Press **C** can jump out current menu to previous level. Press ← will clear previous input data.*

Main menu

1. Call List
2. Forward Type
3. Phone Book
4. Ringer Settings
5. Network
6. Advanced Settings (protected by password)
7. Reboot

8.1 Call List

Select 1.Call List to check call records.

Enter 1.Missed Calls to check the record of missed calls.
Enter 2.Received Calls to check the record of received calls.
Enter 3.Dialed Numbers to check the record of dialed calls.

Missed / received / dialed calls will be kept in message box. MESSAGE LED will be flashing until user presses MESSAGE to check missed call. Re-press MESSAGE to return to main screen.

8.2 Forward Type

Select 2.Forward Type to configure call forward.

Please refer to the selection 3.4 Call Forward.

8.3 Phone Book

Select 3. Phone Book to set entry of phone address book.

Enter 1.List to list all records of address book, as name, telephone number, and IP address.

Enter 2.Edit/Delete to edit/delete a record.

Enter 3.New to add a new record. Please input name, telephone number, and IP address in order.

8.4 Ringer Settings

Select 4.Ringer Settings to configure volume and style of ring.

Enter 1.Volume to adjust ring volume by pressing ← or →

Enter 2.Style to select a tone style of ring, as Low, Middle, and High.

8.5 Network

Select 5. Network to set networking parameters.

Enter 1.Information to check IP Mode, IP, Mask, and Gateway by pressing ← or →.

NOTE: *When IP-Phone is under DHCP mode, then change to Static mode, the following items: IP address, Subnet Mask, Default Gateway, will display empty,* *after reboot, user can see information again.*

Enter 2.Network Mode to set network mode of IP-Phone as Static (Fixed IP), DHCP, or PPPoE.

Enter 3.IP address to set IP address of IP-Phone.

Enter 4.Subnet Mask to set subnet mask address of IP-Phone.

Enter 5.Default Gateway to set default gateway address of IP-Phone.

Enter 6.Domain Name Server to set IP address of DNS Server.

Enter 7.PPPoE Configuration to set PPPoE parameters.

Enter 1.User Name to set PPPoE connection authentication user name.

Enter 2.Password to set PPPoE connection authentication password.

Enter 3.Auto Re-connect to choose ON or OFF to enable or disable this function.

NOTE: *If user enables this function, IP-Phone will automatically reboot to re-connect after PPPoE disconnected. If IP-Phone still can't connect with server after reboot, IP-Phone will keep trying to connect. On the other hand, if user disables this function, IP-Phone won't reboot and keep trying to connect.*

Enter 8.SNTP Configuration to capture current time from SNTP server

Enter 1.SNTP Mode to set SNTP function to be on or off. IP-Phone will capture current time from SNTP server or not.

Enter 2.SNTP Server to specify a SNTP server for IP-Phone.

Enter 3.Time Zone to set time zone via pressing ← or → according to the location IP-Phone is. For example, the time zone of Taiwan should be set as GMT+8:00.

Enter 9.Behind IP-Sharing to set IP sharing function On or Off, and enter public IP address of IP sharing.

8.6 Advanced Settings

Select 6.Advanced Settings to configure SIP settings, VLAN, LCD menu password, and upgrade firmware. Advanced Settings is protected by password. Please key in password to enter (Default: no password).

NOTE:

- *User can set password in 3.LCD Menu Password. Once password is set, user must input password to enter 6.Advanced Settings.*
- *If user forgets the password, please contact with support@micronet.info for a unique password. That needs MAC ID of the IP-Phone.*

Enter 1.SIP Settings to configure for SIP operation.

Enter 1.Connect Mode to choose P2P or Proxy mode.

Enter 2.Proxy to set SIP proxy parameters.

Enter 1.Proxy to set proxy IP address/domain name.

Enter 2.Outbound proxy to set outbound proxy IP address/domain name.

Enter 3.Proxy port to set Proxy port for IP-Phone to send messages.

Enter 4.Outbound port to set outbound proxy port.

Enter 5.Expire to set expire time of registration. In the duration of 2/3 expire time, IP-Phone will re-register to Proxy Server again.

Enter 3.User Info to configure for SIP authentication.

Enter 1.User Name to set user name of IP-Phone to register to proxy server. If Proxy server doesn't request specific user name, please enter line number here.

Enter 2.Line Number to set Line Number of IP-Phone to register to proxy server.

Enter 3.Password to set password of IP-Phone to register to Proxy Server.

Enter 2.Firmware Update to proceed with firmware upgrade.

Enter 1.Download method to choose TFTP or FTP by pressing ← or →.

Enter 2.FTP/TFTP Sever to set TFTP/FTP server IP Address.

Enter 3.FTP Account to input user name for FTP server login.

Enter 4.FTP Password to input user password for FTP server login.

Enter 5.File Name to press the file name of new application rom file prepared for upgrading.

Enter 6.Application Version to display current version of application.

Enter 7.Start to Upgrade to start upgrade by pressing yes..

NOTE: *Download via LCD command can only upgrade new application rom file.*

Enter 3.LCD Menu Password to set entry password of phone LCD Menu.

Enter 4.VLAN to set VLAN parameters and priority for voice and data.

Enter 1.VLAN Configuration to enable VLAN or not.

Enter 2.VID Number to set VLAN ID (1-4095) for SIP/voice traffic (Default: 100).

Enter 3.PC port VID Number to set VLAN ID (1-4095) for data from PC.

Enter 4.Priority Number to set priority (L: 0-3, H: 4-7) for SIP/voice traffic.

Enter 5.PC Port Priority Num. to set priority (L: 0-3, H: 4-7) for data from PC.

Enter 6.Drop PC Port tag to drop priority tag when transmitting data out from PC port.

8.7 Reboot

Select 5.Reboot to reboot the phone. It is necessary to reboot IP-Phone after any configurations has been made.

9. Specification

Model	SP5100/S
Standard	IETF SIP (RFC3261)
Interface	2 x RJ-45 ports for linking to Ethernet LAN and PC 1 x RJ-11 port for usual Telephone Network
Display	2x16 dot-matrix LCD display
Keypad	10 Memory Keys 8 Function Keys 4 Menu control Keys
Voice	G.711A/μ law, G.723.1, G.729, G.729A Codec CNG (comfort noise generation) EC (Echo Cancellation), G.168 Dynamic Jitter Buffer Bad Frame Interpolation Gain (Voice Volume) Settings Provide Call Progress Tone
DTMF	In-band, SIP Info, RFC2833
Telephony	Redial (last 10 numbers) Speed dial (10 sets) Call Transfer / Call Forward / Call Hold Caller ID Display Phone address book
QoS	VAD IEEE802.1q/p, VLAN & Port prioritization IP ToS / DSCP
NAT Traversal	IP sharing function Outbound proxy (SP5100/S only)
Networking	Static assign, PPPoE, DHCP
Management	Web / Telnet / Keypad
Environment	Operating temperature: 0 - 40 degree C Storage temperature: -10 - 60 degree C Operating Humidity: 10% to 90%
Dimension & Weight	215 x 198 x 71 mm (W x D x W), 850g
Power Supply	9VDC, 1A
Emission	FCC part 15 Class A, CE

Date: 2006 / 04 / 05