



User's Manual

Internet IP Telephone

Model No.: SP5105/S

Website: <http://www.micronet.info>

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About this User's Manual

This user's guide gives hardware specifications and explains web configuration and command line configuration for the IP Phone

Online Upgrade

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

General Syntax Conventions

Mouse action sequences are denoted using a comma. For example, click start, Settings, Control Panel, Network means first you click Start, Click or move the mouse pointer over Settings the click or move the mouse pointer over Control Panel and finally click (or double-click) Network.

[Enter] means to press the Enter key from your keyboard

Predefined choices are in Bold Arial Font.

Button and field labels, link s and screen names in are in Bold Times New Roman font.

A single keystroke is in Arial font and enclosed in square brackets. [Enter] means the Enter. For brevity's sake, we will use "e.g.," as shorthand for "for instance", and "i.e.," for "that is" or "in other words."

Related Documentation

This user's guide provides hardware connection details and configuration and management instruction for the managements the IP Phone.

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

1. Introduction

Micronet SP5105/S VoIP Telephone provides unmatched levels of integrated business functionality and converged communications that go beyond today's conventional voice systems. It reduces costs of receiving local and long distance calls, and enables voice and data traffic over a single network connection. With internal voice/data switch, SP5105/S can prioritize traffic to ensure high-quality speech and reduce costs by conserving wiring closet ports and eliminating the need for separate cable drops to the desktop.

1.1. Features and Specification

Calling Features

- Call Hold
- Call Transfer
- Call Forward
- 5 configurable speed dials

Network Supported

- Fixed IP
- Dynamic Host Configuration Protocol (DHCP)
- PPPoE connection
- Behind NAT IP Sharing Device
- Support QOS by setting DSCP

Audio Features

- G.711 a/μ-Law, G.723.1, G.729, G.729a
- VAD, CNG
- G.165/G.168 compliant echo cancellation
- Programmable Dynamic Jitter Buffer
- Bad Frame Interpolation
- Gain/Attenuation Settings

Provisioning and Configuration

- SIP (RFC3261) compliance
- LCD configuration password protection
- Provide Proxy Mode or Peer-to-Peer Mode
- Ring tone, Speaker and Handset volume adjustable

Management Features:

- LCD Front Panel
- Web Browser
- TELNET

Environmental

- Operating and storage Humidity: 10 to 95 % (Non-condensing)
- Operational Temperature: 0 to +40 C
- Storage Temperature: -10 to 60°C
- Dimension & Weight : 200 x 210 x 79 mm, 830g

Certification

- CE

1.2. Appearance

Front View and Keypad function



❶ LED Indicator		When IP Phone didn't register to Proxy server or having incoming call, system indication LED will be blinking.
❷ LCD Display		Shows IP Phone status with 13 x 2 character Dot Matrix display.
❸ Function Keys	MENU	Press to enter Configuration Menu when in standby mode; if already in Configuration Menu, press this button can return to standby mode
	MUTE	Mute the voice of Microphone and let others can't hear from user in communication. Change input mode to be digit or character mode: When configuration in LCD menu can change input mode to be input digit only or input character.
	▲ / ▼	Up/Down, Left/Right, Increase/Decrease
	OK	To enter the sub menu or confirm the modification
	FLASH	Transfer a call. User A can press FLASH button when in communication with user B, then input phone number can make call to User C, after talk with C, A can hang up, User B and User C can communicate. Back to upper level of menu: when in Configuration mode
❹ Dialing Pad	Numeric	Digits or Characters input
	REDIAL /HOLD	Redial the last outgoing call or hold one call in communication.

Change the characters in Upper-case or Lower-case character mode.

- ⑤ **Speakerphone** Hands-free function
- ⑥ **Memory Dialing** Preset the numbers for speed dial

Back View



- ① **RJ45 LAN Port** Build-in two RJ45 LAN ports switch
- ② **Power Jack** DC 9V power input outlet

1.3. Corresponding list of keypad and symbol:

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"
#	"#"

Digit Mode:

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

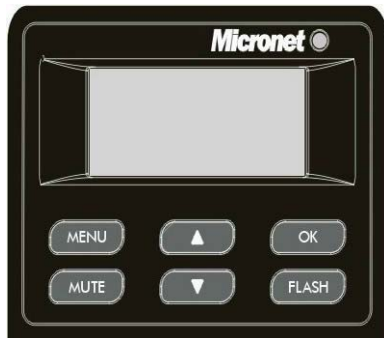
2. Setting Up the IP Phone

IP-Phone Default Network settings

IP : 10.1.1.3

Subnet : 255.0.0.0

Default Gateway : 10.1.1.254.



Function Buttons



Dialing Pad

2.1. Configure the IP network from LCD Menu

1. Static IP Mode

Press **MENU** and use **▲** or **▼** to select > Network and press **OK**

Select > Get IP Mode and press **OK**

Select >Fix and press **OK**

2. Change IP address

Press **MENU** and use **▲** or **▼** to select > Network and press **OK**

Select > IP Address and press **OK**

Use **▲** to clean up the preset IP address

Enter the new IP address by Dialing Pad and press **OK**

Example : IP = 192.168.0.3



3. Change the Subnet Mask

Follow the same procedures as change the IP address to configure Subnet Mask
Subnet Mask = 255.255.255.0 in most cases.

4. Change the Default GW

Follow the same procedures as change the IP address to configure Default Gateway
Default Gateway IP address is the LAN IP address of your NAT Router device.

5. Reboot the IP-Phone

After changed the IP network settings, you need to reboot the IP-Phone to take new settings effective.

Press  and select **> Reboot** by press  or  button and press 

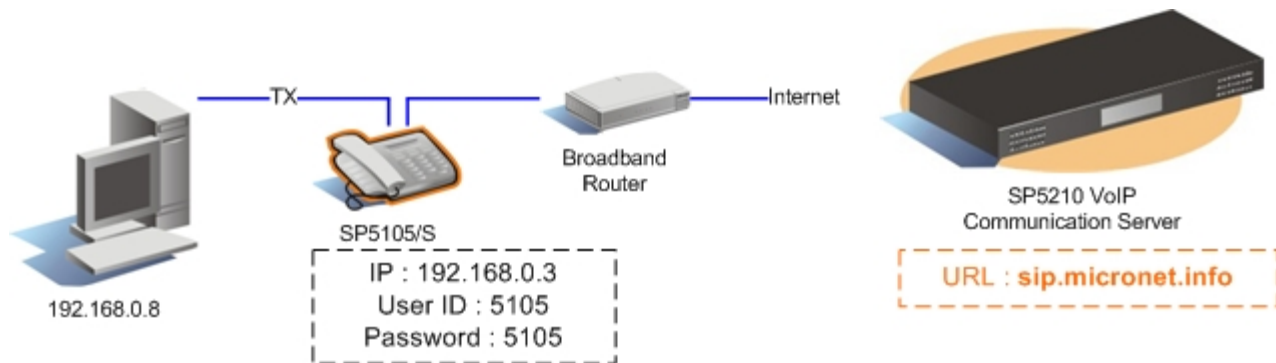
Select **>YES** and press  to reboot the IP-Phone

After IP-Phone re-start completed, you can now use web browser or telnet to configure your IP-Phone.

2.2. Configure the IP Phone with SIP service

After configure the IP-Phone IP network, now user can use the WEB browser to access the IP-Phone and change the settings.

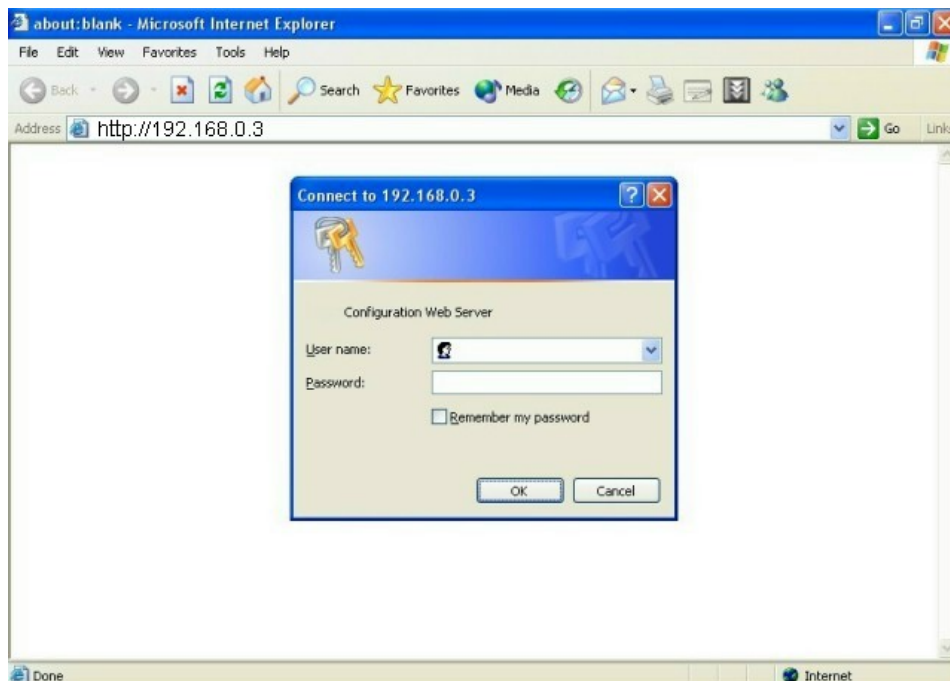
The following setup example shows how to configure the IP-Phone to connect to Micronet SIP server.



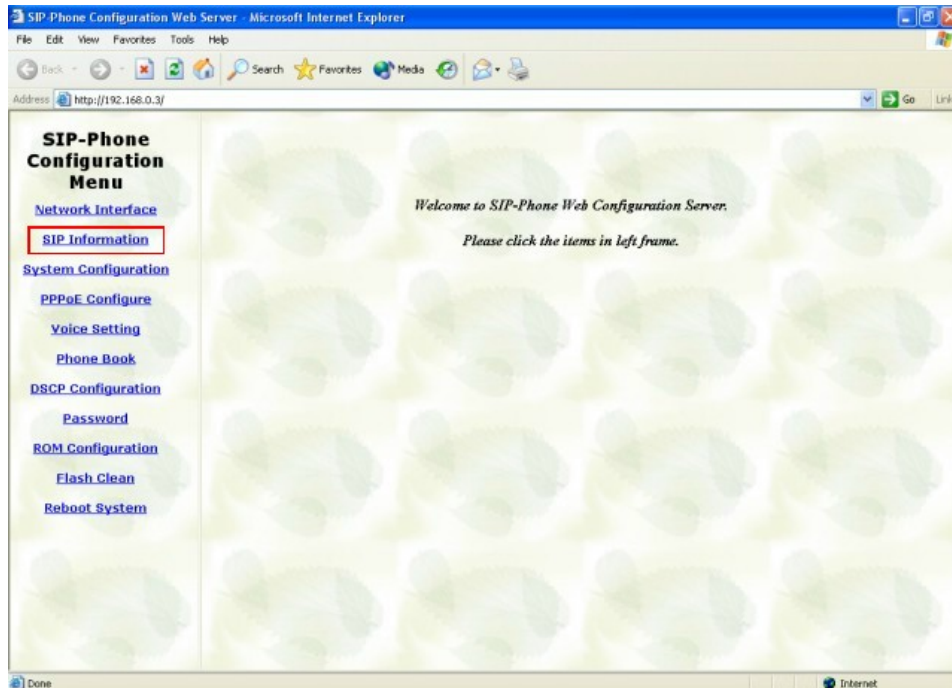
Note :

Access IP-Phone by WEB Interface, your PC and IP-Phone should be under the same subnet

1. Open WEB Browser from PC (ex: Internet Explorer), type your IP-Phone IP address into Address bar and press [Enter]



2. Type root as User name and no password as default, click OK button to enter the Configuration Menu.



3. Click SIP Information menu

4. Enter the Proxy IP Address, Line Number, Line Account and Line Password and click the OK button as following screen showed.

SIP Configuration	
Run Mode:	<input type="radio"/> Peer-2-Peer <input checked="" type="radio"/> Proxy
Proxy IP Address:	<input type="text" value="sip.micronet.info"/>
Outbound Proxy:	<input type="text" value="null"/>
Proxy port:	<input type="text" value="5060"/>
Phone Book Search:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Prefix String:	<input type="text" value="null"/>
Line Number:	<input type="text" value="5105"/>
Line Account:	<input type="text" value="5105"/>
Line Password:	<input type="text" value="5105"/>
SIP port:	<input type="text" value="5060"/>
RTP Port:	<input type="text" value="16384"/>
Expire:	<input type="text" value="60"/>
<input type="button" value="OK"/>	

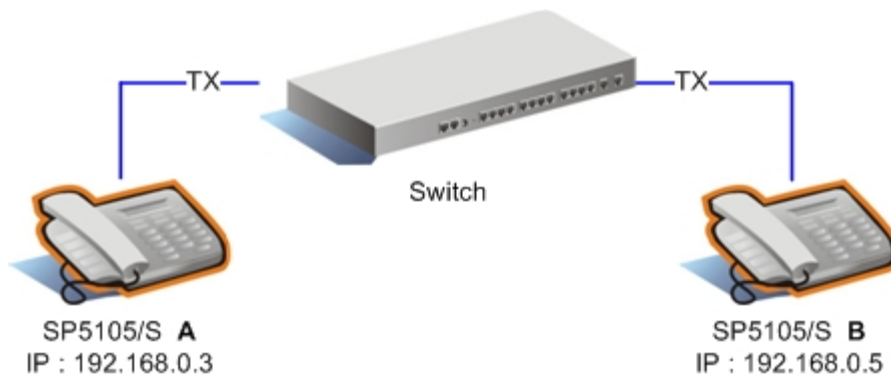
Note :

The DNS server address must be configured if enter the URL address instead of IP address on Proxy IP Address

5. Click the Reboot System and re-start the IP-Phone

2.3. Configure the IP-Phone in P2P Mode (LAN)

SP5105/S can work in Peer-to-Peer mode, the SIP server isn't necessary in this mode



1. Enter Configuration Menu by web browser

2. Click SIP Information, select Peer-2-Peer Mode, enter the Line number and click OK button.

SP5105/S A

SIP Configuration	
Run Mode:	<input checked="" type="radio"/> Peer-2-Peer <input type="radio"/> Proxy
Proxy IP Address:	220.130.173.70
Outbound Proxy:	null
Proxy port:	5060
Phone Book Search:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Prefix String:	null
Line Number:	5105
Line Account:	
Line Password:	
SIP port:	5060
RTP Port:	16384
Expire:	60
<input type="button" value="OK"/>	

SP5105/S B

SIP Configuration	
Run Mode:	<input checked="" type="radio"/> Peer-2-Peer <input type="radio"/> Proxy
Proxy IP Address:	220.130.173.70
Outbound Proxy:	null
Proxy port:	5060
Phone Book Search:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Prefix String:	null
Line Number:	5107
Line Account:	
Line Password:	
SIP port:	5060
RTP Port:	16384
Expire:	60
<input type="button" value="OK"/>	

3. Click Phone Book, create phone book table

SP5105/S A

Phone Book			
Index	Name	IP Address	E164
1	Paul	192.168.0.5	5107

New Record			
Index	Name	IP Address	E164 No.
1	Paul	192.168.0.5	5107
<input type="button" value="Add Data"/> <input type="button" value="Delete Data"/>			

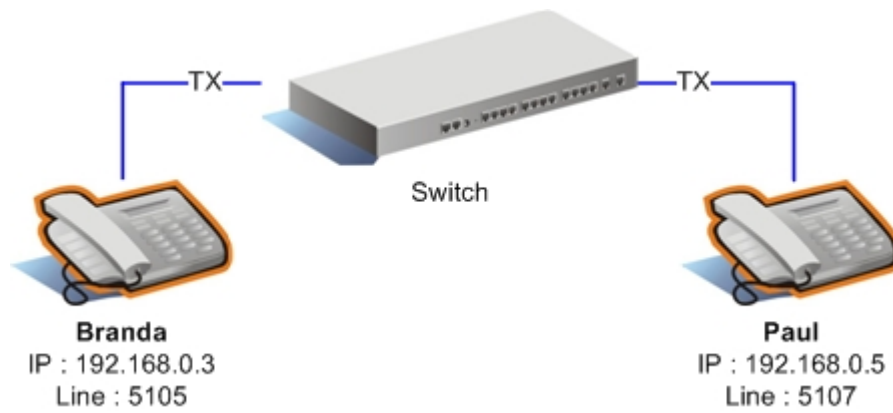
SP5105/S B

Phone Book			
Index	Name	IP Address	E164
1	Branda	192.168.0.3	5105

New Record			
Index	Name	IP Address	E164 No.
1	Branda	192.168.0.3	5105
<input type="button" value="Add Data"/> <input type="button" value="Delete Data"/>			

4. Reboot the IP-Phone

5. Placing the calls

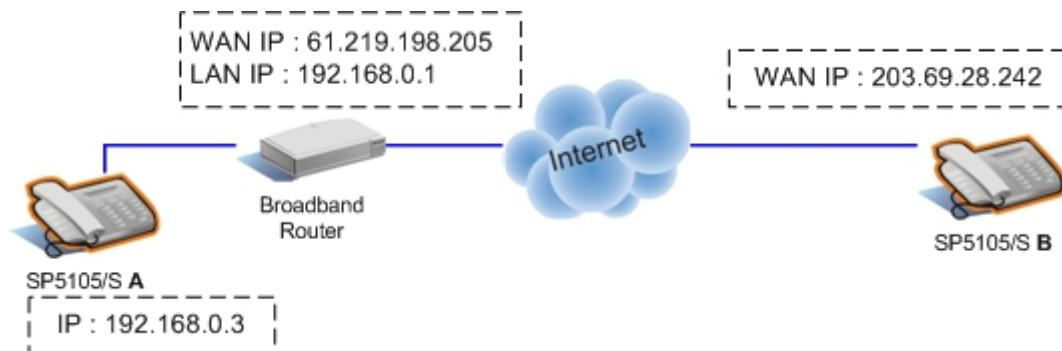


Branda dials **5107#** to make a call to Paul

Paul dials **5105#** to make a call to Branda

2.4. Configure the IP-Phone in P2P Mode (WAN)

Here shows how to configure the IP-Phone sets behind the NAT Router in Peer-to-Peer Mode. Your internet connection needs the static public IP address.

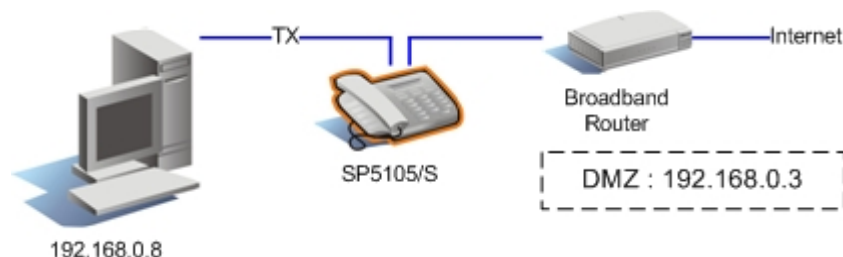


1. Enter Configuration Menu by web browser (SP5105/S A)

2. Click the Network Interface, enable the IP Sharing function, enter your Internet connection public IP address (ex. 61.219.198.205)

Network Interface	
IP Address:	192 . 168 . 0 . 3
Subnet Mask:	255 . 255 . 255 . 0
Default routing gateway:	192 . 168 . 0 . 1
Get IP Mode:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP <input type="radio"/> PPPoE
SNTP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	0
IP Sharing:	<input checked="" type="radio"/> enable <input type="radio"/> disable
IP Sharing Server Address:	61 . 219 . 198 . 205
Primary DNS Server:	168 . 95 . 1 . 1
Secondary DNS Server:	168 . 95 . 1 . 2
OK	

3. Assign the NAT Router DMZ as IP-Phone IP address (192.168.0.3)



4. Click SIP Information, select Peer-2-Peer Mode, enter the Line number and click OK button.

SP5105/S A

SIP Configuration	
Run Mode:	<input checked="" type="radio"/> Peer-2-Peer <input type="radio"/> Proxy
Proxy IP Address:	220.130.173.70
Outbound Proxy:	null
Proxy port:	5060
Phone Book Search:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Prefix String:	null
Line Number:	5105
Line Account:	
Line Password:	
SIP port:	5060
RTP Port:	16384
Expire:	60
OK	

SP5105/S B

SIP Configuration	
Run Mode:	<input checked="" type="radio"/> Peer-2-Peer <input type="radio"/> Proxy
Proxy IP Address:	220.130.173.70
Outbound Proxy:	null
Proxy port:	5060
Phone Book Search:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Prefix String:	null
Line Number:	5107
Line Account:	
Line Password:	
SIP port:	5060
RTP Port:	16384
Expire:	60
OK	

5. Click Phone Boot, create phone book table

SP5105/S A

Phone Book			
Index	Name	IP_Address	e164
1	Paul	203.69.28.242	5107

New Record			
Index	1	Name	Paul
IP Address	203.69.28.242		E164 No.
		5107	
Add Data Delete Data			

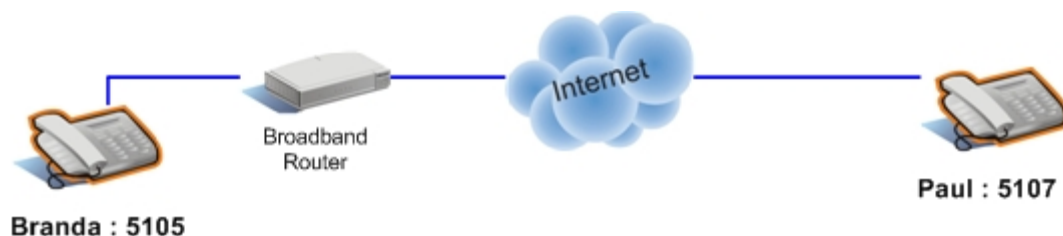
SP5105/S B

Phone Book			
Index	Name	IP_Address	e164
1	Branda	61.219.198.205	5105

New Record			
Index	1	Name	Branda
IP Address	61.219.198.205		E164 No.
		5105	
Add Data Delete Data			

6. Reboot the IP-Phone

7. Placing the calls



Branda dials **5107#** to make a call to Paul

Paul dials **5105#** to make a call to Branda

3. How To Upgrade the Firmware

The IP-Phone can update the software version by TFTP / FTP server, you can download the latest firmware from our WEB site.

1. Prepare the TFTP server and firmware

Firmware Section

<http://www.micronet.info/Download/driver/driver.asp#voip>

TFTP program

<http://www.micronet.info/Download/Driver/VoIP/utility/tftpd32m.zip>

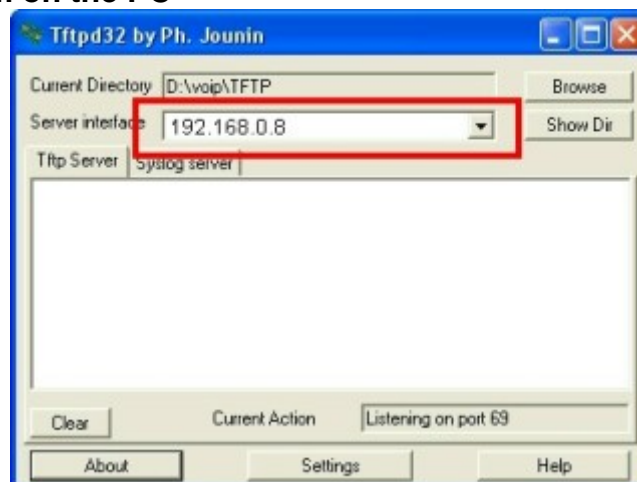
2. Connect your PC to the IP-Phone RJ-45 port



Note

Make sure the gateway and PC has same subnet, for example, 192.168.0.6 is your PC's IP, then the IP address of the gateway should be 192.168.0.xx

3. Run TFTP program on the PC



The TFTP server now is ready

4. Run the web browser, login your IP-Phone.

5. Click the ROM Configuration

6. Enter TFTP server IP address, ROM file name, select the upgrade method, ROM type and click OK button

ROM Configuration	
FTP/TFTP server IP Address:	<input type="text" value="192"/> . <input type="text" value="168"/> . <input type="text" value="0"/> . <input type="text" value="8"/>
Target File name:	<input type="text" value="lp302.101"/>
Method:	<input type="text" value="TFTP"/>
FTP Login:	name <input type="text"/> passwd <input type="text"/>
Target File Type:	<input type="text" value="Application Image"/>
<input type="button" value="OK"/>	




7. Now the IP-Phone is downloading the firmware and writes it into flash memory

4. LCD Menu Configuration

User can configure the IP Phone via keypad, press  to enter the system menu.

Note:

Press  before input data can switch characters to be capital or lowercase.

Press  before input data can switch input mode to be character mode or IP mode; for example, user wants to enter IP address, after pressing  can enter digits directly. When user is inputting data, press  will clear previous input data.

4.1. Call List

User can check all call records in this call list menu.

Missed Calls - to see all missed calls in message box.

Received - to see all received calls in message box.

Dialed No. - to see all dialed numbers in message box.

Exit - return to upper level of LCD Menu

4.2. Forward

Allow user to transfer incoming calls automatically to another number or the destination. The following types are available.

Busy

All incoming calls are forward to another number when the phone is busy.

Activate - Enter a forwarded phone number to activate busy forward function.

Deactivate - Deactivate Busy Forward function.

Exit - Return to upper level of LCD Menu

No Answer

All incoming calls are forward to another number when user does not answer within 10 seconds.

Activate - Enter a forwarded phone number to activate No Answer Forward function.

Deactivate - Deactivate No Answer Forward function.

Exit - Return to upper level of LCD Menu

Uncondition (Unconditional Forward)

All incoming calls are forward to another number.

Activate - Enter a forwarded phone number to activate Unconditional Forward function.

Deactivate - Deactivate Unconditional Forward function.

Exit - Return to upper level of LCD Menu

Delete All - Delete all forward activated data.

Exit - Return to upper level of LCD Menu.

4.3. Phone Book

List

List all records of name, telephone number, and IP address in the phone address book.

Edit/Del

Edit or delete a record of name, telephone number, and IP address of the phone address book.

New

Add a new record of name, telephone number, and IP address of the phone address book.




Exit - Return to upper level of LCD Menu

4.4. Ringer

Volume

User can adjust ring volume by press  or  on the keypad to decrease or increase ringer volume.

Style

There are three tone styles for IP Phone SP5105/S. Move the “>” symbol by press  or  on the keypad to select the tone style preferred, then press  to confirm it.

Exit - Return to upper level of LCD Menu

4.5. Network

Information

User can press  or  and  to check current network status:

Mode, IP, Mask, Gateway

Display current network connection mode of IP Phone to be Static (Fixed IP), DHCP, or PPPoE and display current IP information.

Get IP Mode

Set network mode of IP Phone to be **Fix** (Fixed IP), **DHCP**, or **PoE** (PPPoE).

IP address

Set SP5105/S IP address

Subnet Mask

Set SP5105/S subnet mask address

Default GW

Set SP5105/S.default gateway address

DNS (Domain Name Server)

Set IP address of Domain Name Server. Once IP Phone can connect to DNS server, user can set URL address for Proxy server or Phone book instead of IP address.

(1) Primary

Set Primary DNS server IP address

(2) Secondary

Set Secondary DNS server IP address

Exit - Return to upper level of LCD Menu

PoE Config (PPPoE Configuration)

If your IP-Phone is using PPPoE connection, you need to set the login name and password here which assigned by your Internet Service Provider

User Name - Set PPPoE connection authentication user name.

Password - Set PPPoE connection authentication password.

Reconnect

Select ON or OFF to enable or disable this function. If user enables this function, after PPPoE disconnected, IP Phone will automatically reboot to re-connect, and after reboot, if IP Phone still can't connect with server, 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.

Exit - Return to upper level of LCD Menu

IP-Sharing

If IP Phone is behind IP sharing or NAT device, and IP Phone is under Peer-to-Peer mode or Proxy mode (doesn't support endpoint behind NAT function), on IP sharing must enable "**DMZ**" function or set "**Virtual Server**" to open ports (UDP port: 5060 and 16384, 16385). IP Phone must enable this IP sharing function.

User must enter public IP address of IP sharing (Static Public IP connection)

Exit - Return to upper level of LCD Menu

4.6. Advanced Set

SIP

Mode

Select SIP connection mode to be **P2P** (Peer-to-Peer) or **Proxy**

Proxy

Proxy - Set Proxy IP address or Domain Name.

Outbound - Set Outbound Proxy IP address or Domain Name.

Px port - Set Proxy port for IP Phone to send messages.

Expire (in seconds)

Set expire time of registration, in the duration of 2/3 expire time, IP Phone will re-register to Proxy Server again.

Exit - Return to upper level of LCD Menu

User Info

User Name (Mandatory)

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.

Line No

Set Line Number of IP Phone to register to Proxy Server.

Password




Set User Password of IP Phone to register to Proxy Server. This configuration is not necessary, if Proxy server doesn't request client to set password, user only has to set User Name the same as Line Number.

Exit - Return to upper level of LCD Menu

Exit - Return to upper level of LCD Menu

SW Update

Method

There are two methods to download new version file, please move the ">" symbol by press  or  on the keypad to select **TFTP** or **FTP** method, then press  to confirm it.

Sever

User has to specify the TFTP/FTP server IP Address here

Account

User has to input user name for FTP server login .It is necessary for upgrading IP Phone new application rom file via FTP method.

Password

User has to input user password for FTP server login .It is necessary for upgrading IP Phone new application rom file via FTP method.

File Name

User has to specify the file name of new application rom file prepared for upgrading

Version

Show versions of all software and hardware. (**)

Upgrade

Select **YES** to start upgrade, or **NO** to cancel the action.

Exit

Return to upper level of LCD Menu

Menu Password

Set entry password of phone LCD menu.

> **Advanced Menu** can be protect by password if user set the Menu Password

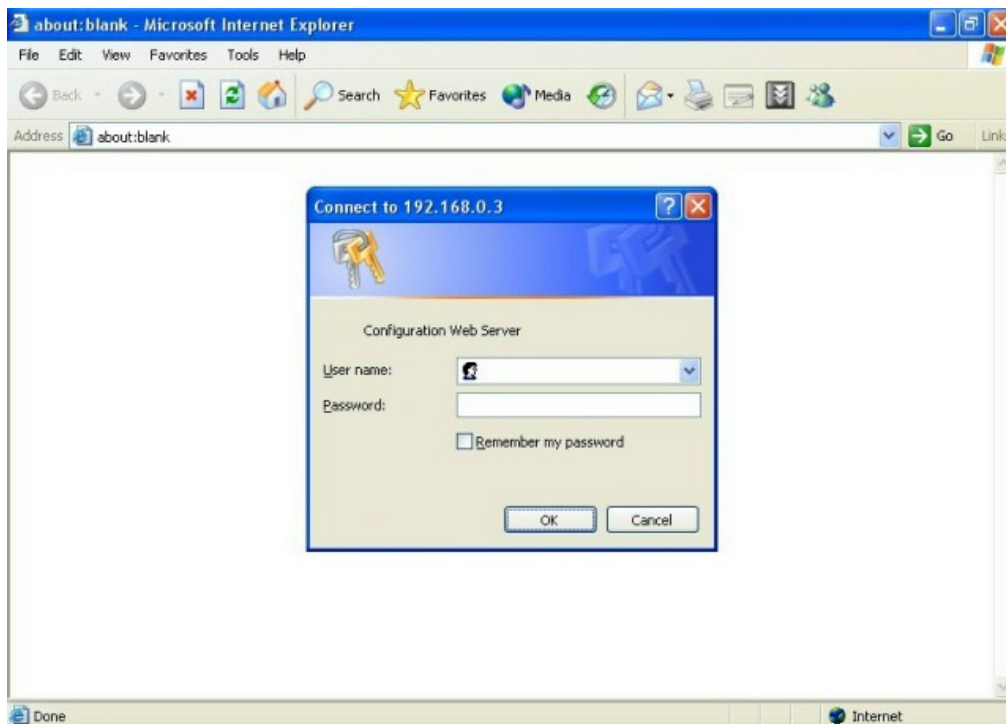
Exit - Return to upper level of LCD Menu

4.7. Reboot

Reboot machine. It is necessary and important for user to reboot IP Phone after any configurations has been made. IP Phone will ask user again before reboot.

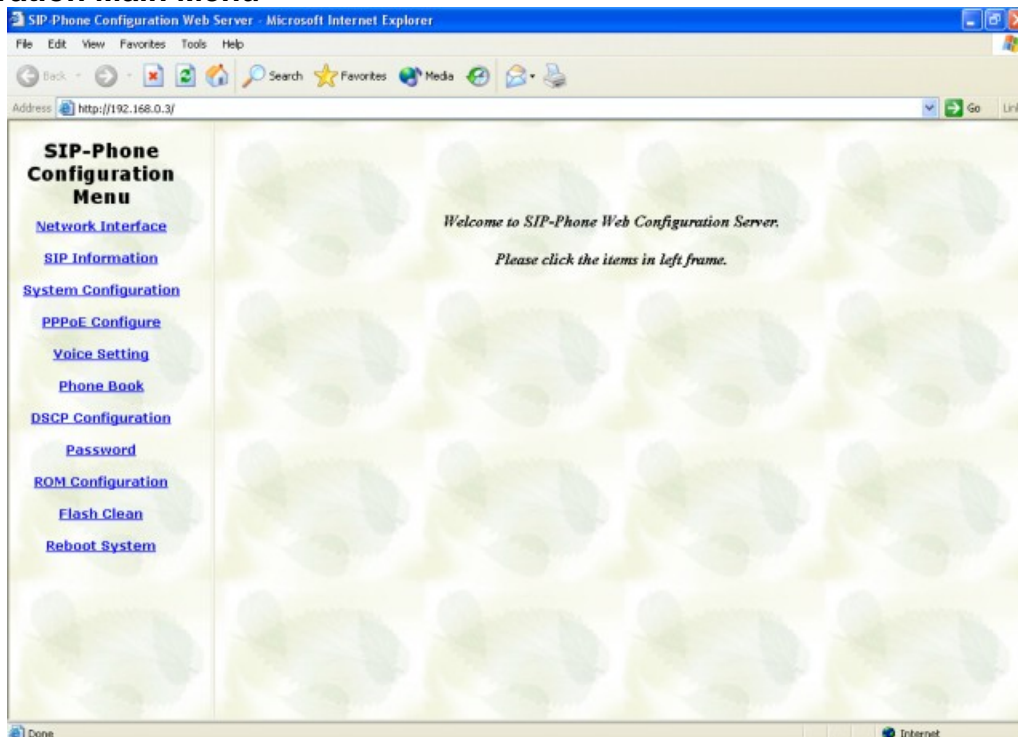
5. WEB Interface Configuration

The HTTPD web management interface provides user an easier way to configure. Run the web browser (ex. Internet Explorer or Netscape), type the IP-Phone IP address into Address bar and press [Enter]



Type **root** as User name, and no password as default.

Configuration Main Menu



5.1. Network Interface

Network Interface	
IP Address:	192 . 168 . 0 . 3
Subnet Mask:	255 . 255 . 255 . 0
Default routing gateway:	192 . 168 . 0 . 1
Get IP Mode:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP <input type="radio"/> PPPoE
SNTP:	<input type="radio"/> enable <input checked="" type="radio"/> disable
SNTP Server Address:	168 . 95 . 195 . 12
GMT:	0
IP Sharing:	<input type="radio"/> enable <input checked="" type="radio"/> disable
IP Sharing Server Address:	0 . 0 . 0 . 0
Primary DNS Server:	168 . 95 . 1 . 1
Secondary DNS Server:	168 . 95 . 1 . 2
OK	

IP Address Set IP Address of IP-Phone

Subnet Mask Set the Subnet Mask of IP-Phone

Default routing gateway Set Default routing gateway of IP-Phone

Get IP Mode User has to set IP-Phone to use which network mode

Fixed IP

User has to assign a fixed IP to IP-Phone

DHCP

When DHCP function enables, IP-Phone will automatically search DHCP server after reboot

PPPoE

If IP-Phone is working with PPPoE connection, user have to set related parameters in "PPPoE Configure" page

SNTP Enable / Disable the Simple Network Time Protocol function

SNTP Server Address Set SNTP Server Address
When SNTP server is available, enable IP-Phone SNTP function to point to SNTP server IP address so that IP-Phone can get correct current time

GMT Set time zone for SNTP Server time
User can set different time zone according to the location of

IP-Phone. For example, in Taiwan the time zone should be set as 8, which means GMT+8

IP Sharing	Enable it if IP-Phone is behind IP Sharing router
UPnP	Enable it if IP sharing or NAT device supports UPnP function so that no need to configure IP sharing or IP-Phone when IP-Phone is behind NAT device
IP Sharing Server Address	Set Public IP Address of IP Sharing router for IP-Phone to work behind IP sharing
Primary DNS Server	Set Primary Domain Name Server IP address 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 and phone book IP address
Secondary DNS Server	Set Secondary Domain Name Server IP address

Note:

If User set "Get IP mode" as DHCP or PPPoE, IP address, Subnet Mask, and Default routing gateway will become 0.0.0.0 and not allow to be configured

5.2. SIP Configuration

SIP Configuration	
Run Mode:	<input type="radio"/> Peer-2-Peer <input checked="" type="radio"/> Proxy
Proxy IP Address:	<input type="text" value="220.130.173.70"/>
Outbound Proxy:	<input type="text" value="null"/>
Proxy port:	<input type="text" value="5060"/>
Phone Book Search:	<input type="radio"/> Enable <input checked="" type="radio"/> Disable
Prefix String:	<input type="text" value="null"/>
Line Number:	<input type="text" value="5105"/>
Line Account:	<input type="text" value="5105"/>
Line Password:	<input type="text" value="5105"/>
SIP port:	<input type="text" value="5060"/>
RTP Port:	<input type="text" value="16384"/>
Expire:	<input type="text" value="60"/>
<input type="button" value="OK"/>	

Run Mode	Select IP-Phone to work under Peer-to-Peer mode or Proxy mode
Proxy IP Address	Set Proxy IP Address or URL address (Domain Name Server must be configured. Please refer to Network Interface)
Outbound Proxy	Set IP Address or URL address (Domain Name Server must be configured. Please refer to Network Configure) of outbound Proxy server
Proxy port	Set Proxy port for IP-Phone to send message, default value is 5060, if there is no special request of Proxy server, please don't change this value
Phone Book Search	enable/disable phone book search function. If user enabled this function, IP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, IP-Phone will make call to related IP address
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, 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	identify one number for the IP-Phone to register to the Proxy
Line Account	Set user name of IP-Phone 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 IP-Phone won't register successfully
Line Password	set password for registering
SIP Port	set SIP UDP port
RTP Port	set RTP port for sending voice data
Expire	Set expire time of registration. IP-Phone will keep re-registering to proxy server before expire timed out

5.3. System Configuration

System Configuration	
Keypad DTMF Type:	<input checked="" type="radio"/> In-Band <input type="radio"/> RFC2833
RFC2833 Payload Type:	<input type="text" value="96"/>
Inter Digit Time:	<input type="text" value="3"/>
End of Dial Digit:	<input type="radio"/> NONE <input type="radio"/> * <input checked="" type="radio"/> #
<input type="button" value="OK"/>	

Keypad DTMF Type set DTMF type. User can select DTMF type IP-Phone transmits

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

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, IP-Phone will dial out all number pressed

End of Dial Digit select end of dialing key, e.g. set end of dial key as * button, after finished pressing dialing number then press * will dial out

5.4. PPPoE Configuration

PPPoE Device Information and Configuration	
User Name:	<input type="text" value="pppoe"/>
Password:	<input type="password" value="*****"/>
Reboot After Remote Host Disconnection:	<input type="radio"/> On <input checked="" type="radio"/> Off
Authenticate:	<input type="text" value="PAP"/>
Protocol:	<input type="text" value="TCP/IP"/>
Device:	<input type="text" value="PPP/PPPoE"/>
<input type="button" value="OK"/>	

User Name Set PPPoE authentication User Name

Password Set PPPoE authentication password

Reboot After Remote Host Disconnection Enable/Disable auto reboot after PPPoE disconnection
If user enables this function, after PPPoE being disconnected, IP-Phone will automatically reboot to re-connect, and after reboot, if IP-Phone still can't get contact with server, IP-Phone will keep trying to connect. After re-connected, IP-Phone will also restart system. On the other hand, if user disables this function, IP-Phone won't reboot and keep trying to connect

Note:

The Authenticate, Protocol and Device section are for reference only, cannot allow to be configured.

5.5. Voice Setting

Voice Setting					
Codec Priority	1st G.723.1	2nd G.729	3rd G.711mu-Law	4th G.711A-Law	5th X
Frame Size	G.723.1 30ms	G.729a 20ms	G.729 20ms	G.711mu 20ms	G.711A 20ms
G.723 Silence Suppression:	<input type="radio"/> enable <input checked="" type="radio"/> disable				
Volume:	voice 4	ring 5	input 4	DTMF 5	
Echo Cancelor:	<input checked="" type="radio"/> enable <input type="radio"/> disable				
Jitter Buffer:	Min. Delay 90		Max. Delay 150		
Optimized Factor (Jitter):	9				
<input type="button" value="OK"/>					

- 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
- Frame Size** User can set different packet size for each codec
- 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” (sending out); “Input” (receiving); “DTMF” (DTMF sending out)
- Echo Cancelor** Enable / Disable (suggested always Enable this function)
- Jitter Buffer** Set Min. Delay and Max. Delay of Jitter Buffer for voice packets.
- Optimized Factor (Jitter)** Set Optimized Factor of voice, this is for advanced user only, please contact with your distributor before making any change

5.6. Phone Book

Phone Book			
Index	Name	IP_Address	e164

New Record			
Index <input type="text"/>	Name <input type="text"/>	IP Address <input type="text"/>	E164 No. <input type="text"/>
<input type="button" value="Add Data"/>		<input type="button" value="Delete Data"/>	

Add Data User can specify 20 sets of phone book via web interface. Please input index, Name, IP Address and E.164 number of the destination device

Delete Date User can delete any configured phone book data by index

Note : Phone Book works in Peer-to-Peer connection mode

5.7. DSCP (DiffServ Code Point) Configuration

DiffServ Code Point(DSCP) Configuration	
=== Signal Packet ===	
<input type="radio"/> Assured Forwarding(AF) PHB	Delay Priority : <input type="text" value="Class 1"/> Drop Precedence : <input type="text" value="Low"/>
<input type="radio"/> Expedited Forwarding(EF) PHB	
<input checked="" type="radio"/> Default	
<input type="radio"/> User Assign Special DSCP Code: <input type="text"/>	
=== RTP Packet ===	
<input type="radio"/> Assured Forwarding(AF) PHB	Delay Priority : <input type="text" value="Class 1"/> Drop Precedence : <input type="text" value="Low"/>
<input type="radio"/> Expedited Forwarding(EF) PHB	
<input checked="" type="radio"/> Default	
<input type="radio"/> User Assign Special DSCP Code: <input type="text"/>	
<input type="button" value="OK"/>	

Assured Forwarding (AF) PHB

Select Delay priority and Drop Precedence

Expedited Forwarding (EF) PHB

Select TOS value as EF

Default

Select TOS value as 0

User Assign Special DSCP Code

User can set other unspecified value here

Set Signal or RTP Packet DSCP value:

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 are shown as follows.

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

1. High priority with DS-field :

(1) Expected Forwarding (EF)	101110	====>	46 (Decimal System)
(2) 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)

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

5.8. Password

Password		
<input type="text" value="root"/>	Current Password:	<input type="text"/>
	New Password:	<input type="text"/>
	Confirm New Password:	<input type="text"/>
		<input type="button" value="CHANGE"/> <input type="button" value="ABORT"/>

Change First select login name as root or administrator, then enter current password, new password and confirm new password again to set new password

Abort Press abort will clean all inputs

Note:

Only root account can set few functions as following :

1. **Password: set password of login: "root" users.**
2. **Flash clean: clean all current configurations**
3. **Rom configuration: upgrade boot sector**
4. **Rom configuration: upgrade whole 2m software file**

5.9. ROM Configuration

ROM Configuration	
FTP/TFTP server IP Address:	<input type="text"/> . <input type="text"/> . <input type="text"/> . <input type="text"/>
Target File name:	<input type="text"/>
Method:	<input type="text" value="TFTP"/>
FTP Login:	name <input type="text"/> passwd <input type="text"/>
Target File Type:	<input type="text" value="Application Image"/>
<input type="button" value="OK"/>	

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

Target File name Set file name prepared to upgrade

Method Select download method as TFTP or FTP

FTP Login Set FTP login name and password
When IP-Phone use FTP mode to download the firmware, you need to set login name and password

Target File Type Select which type of ROM to upgrade for the IP-Phone APP, 2M ROM or BOOT ROM, etc.

Note:

After 2mb file download is finished, all configuration settings change to default values, it has to configure the parameters again.

After upgrade Application, please remember to execute Flash Clean, which will clean all configurations become factory values except IP address.

5.10. Flash Clean

Flash Clean
<i>SIP-Phone will be reseted to factory default values.</i>
<input type="button" value="CLEAN"/>

Click **CLEAN** will clear the programming data stored in the IP Phone, all configurations reset to factory default value.

Note

User must re-configure all commands all over again (except Network Configure) once execute this function

5.11. Reboot

Reboot SIP-Phone
It will take 40 seconds to reboot. (remember to COMMIT data before reboot!)
<input type="button" value="REBOOT"/>

Click **REBOOT** will re-start the 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 remember to do **Commit** Data before Reboot System.

6. Telnet Command

You can use telnet commands to configure the IP Phone, it has all the commands from web interface, but more

```
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
bureau        Bureau line information manipulation
rom           ROM file update
passwd        Password setting information and configuration

usage: help [command]
```

6.1. [help]

Type [help], [man] or [?] to display all the command lists

6.2. [quit]

Type [quit], [exit] or [close] will terminate the telnet connection

6.3. [debug]

This command is for engineers to debug system of IP-Phone. User can add debug flag via command debug -add “debug flags”, 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
```

6.4. [reboot]

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

Sometimes after user type reboot, on terminal screen will display: "Data modified, commit to flash rom?" which means IP-Phone will record call history or not. (Ex. REDIAL, outgoing and incoming call data)

6.5. [pbook]

This command is functional both in Proxy mode and Peer-to-Peer mode. In proxy mode, use speed dial or 10 DL button will dial out e.164 number in phone book. In the other hand, 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.
    -delete     Delete an record from phonebook.
    -modify     Modify an exist record.
Example:
pbook -print 1 10
pbook -print 1
pbook -print
pbook -add name Test ip 210.59.163.202 e164 1001
pbook -delete 3
pbook -modify 3 name Test ip 210.59.163.202 e164 1001
```

- | | |
|----------------|--|
| -print | display phone book data. User can print all data in phone book by command
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 called endpoint
(pbook -add name Paul ip 192.168.0.5 e164 33) |
| -delete | delete a record of certain listed index in phone book table
(pbook -delete 2) |
| -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 1 name Paul ip 192.168.0.6 e164 55) |

6.6. [commit]

Save any changes after configuring the IP-Phone

6.7. [ping]

You can use ping command to check the other end point connection

```
usr/config$ ping 192.168.0.8

PING 192.168.0.8: 56 data bytes
64 bytes from 192.168.0.8: icmp_seq=0. time=0. ms
64 bytes from 192.168.0.8: icmp_seq=1. time=0. ms
64 bytes from 192.168.0.8: icmp_seq=2. time=0. ms
64 bytes from 192.168.0.8: icmp_seq=3. time=0. ms
----192.168.0.8 PING Statistics----
4 packets transmitted, 4 packets received, 0% packet loss
round-trip (ms)  min/avg/max = 0/0/0
```

ping Command ping can test which the IP address is reachable or not
(usr/config\$ ping 192.168.0.8)

The message will display packets transmitting condition or no answer from the IP address.

6.8. [time]

It can show the IP-Phone current time and date when enable SNTP function and be able to connect with SNTP server, type time command will show the current time retrieved from SNTP server.

```
usr/config$ time

Current time is MON NOV 01 00:00:01 2004
```

6.9. [ifaddr]

Configure and display IP Phone network 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]]

    -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.
    -dns        specify IP address of DNS Server.
    -timezone   Set local timezone.
    -ipsharing  Specify usage of an IP sharing device and specify IP address.

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 1 168.95.1.1
```

-print Display the current configurations

-ip Set IP-Phone IP Address respectively

-mask Set IP-Phone Subnet mask respectively

-gate Set IP-Phone default gateway respectively

-ipmode Set IP-Phone network mode to be Fixed IP, DHCP or PPPoE
(0=Fixed, 1=DHCP, 2=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, IP-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

-sntp The Simple Network Time Protocol.
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 1 168.95.192.12)

-dns User can set primary and secondary 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.

(ifaddr -dns 1 168.95.1.1)

(ifaddr -dns 2 168.95.192.1)

-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.

(ifaddr -timezone -8)

-ipsharing

If IP-Phone is behind a IP-sharing , user must enable IP sharing function and specify public IP address.

Disable

(ifaddr -ipsharing 0)

Enable

(ifaddr -ipsharing 1 61.219.198.205)

Note:

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

6.10. [pppoe]

```
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.
```

- | | |
|----------------|---|
| -print | Display the current configurations |
| -id | to set PPPoE authentication user name |
| -pwd | to set PPPoE authentication password |
| -reboot | Select enable or disable this function. If user enables this function, after PPPoE disconnected, IP-Phone will automatically reboot to re-connect, and after reboot, if IP-Phone still can't connect with server, 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 |
- (usr/config\$ pppoe -reboot 1)

6.11. [flash]

This can reset all the configuration parameters back to default, except the network IP address.

```
usr/config$ flash

Flash memory information and configuration
Usage:
flash -clean

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

- | | |
|---------------|--|
| -clean | clear all the programming data except IP Network settings. |
|---------------|--|

6.12. [sysconf]

```
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.
-2833type       RFC2833 Payload Type (range:96~128 inter-used:100,102~105)
-eod            End of Dial Digit setting(0: NONE, 1: *, 2: #)
Example:
  sysconf -keypad 0 -eod 2
```

-print Display the current configurations


-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, IP-Phone will dial out all number pressed


-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.)

-2833type change RFC2833 Payload type

-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 directly.

sysconf -eod 0 for no end of dial key

sysconf -eod 1 for end of dial key by 

sysconf -eod 2 for end of dial key by 

6.13. [sip]

```
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)
    -outpx      OutBound Proxy server address. (Proxy IPv4 address or Proxy dns name)
    -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
```

- | | |
|----------------|--|
| -print | Display the current configurations |
| -mode | configure IP-Phone in Peer-to-Peer or Proxy Mode

sip -mode 0 for Peer-to-Peer mode
sip -mode 1 for Proxy mode |
| -px | set proxy server IP address or URL address
(sip -px 220.130.173.70). or (sip -px sip.micronet.info) |
| -pxport | set listening port of 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 |
| -prefix | set prefix string. If user ID contains alphabets, user can set it as prefix string here. 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, SIP-Phone will set out as abc123@”IP address of Proxy”. This function is for special proxy server |
| -line | identify one number for the IP-Phone to register to the Proxy (sip -line 5105) |

- pbsearch** enable/disable phone book search function under Proxy Mode. If user enabled this function, IP-Phone will search dialed number in phone book to see if there is any matched table before send to Proxy server, and if there is a matched data in phone book, IP-Phone will make call to related IP address
- expire** set expire time of registration. IP-Phone will keep re-registering to proxy server before expire timed out
- port** set listening UDP port or IP-Phone
(Default is 5060)
- rtp** set RTP port number. IP-Phone will use this port to send and receive voice
(Default is 16384)

Note:

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

6.14. [security]

```
usr/config$ security

Security 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
```

- | | |
|---------------|--|
| -print | Display the current configurations |
| -name | set user ID of IP-Phone 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 IP-Phone won't register successfully |
| -pwd | set account password for registering |

Most of SIP servers required SIP UA sends the authentication, it need the IP-Phone sends the login name and password to the SIP server. Check with your service provider for more details.

6.15. [voice]

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] ]
      [-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]

    -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)
    -priority   Priority preference of installed codecs.
                 G.723
                 G.729
                 G.711U
                 G.711A
    -volume     Specify the following levels:
                 voice volume (0~9, default: 5),
                 ring volume (0~9, default: 5),
                 input gain (0~8, default: 5),
                 dtmf volume (0~9, default: 5),
    -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
voice -volume voice 20 input 32 dtmf 27
voice -echo 1
```

-print Display the current configurations

-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)

-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 or four codecs

-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 IP-Phone
DTMF means DTMF transmitting volume

(voice -volume voice "value of volume")
(voice -volume input "value of volume")
(voice -volume dfmt "value of volume")

-nscng enable or disable sound compression and comfort noise generation.
It is only for codec G.723.1.

voice -nscng 0 means disable
voice -nscng 1 means enable

Note:

If value of volume set as 0 means -32db, 1 means -31db...etc.

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

VOIP Bandwidth consumption naturally depends on the codec used, it means if your IP-Phone set G.711 Codec at first priority, when it connected to the other IP-Phone with the same Codec priority settings, you'll get better voice quality, but also the higher bandwidth usage.

Codec	BR (Kbps)	NEB (Kbps)
G.723.1	5.3	20.8
G.723.1	6.4	21.9
G.729	8	31.2
G.711	64	87.2

6.16. [tos]

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.

```
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
```

- | | |
|-----------------|--|
| -print | Display the current configurations |
| -rtptype | set DSCP value of signaling packets from 0 to 63 |
| -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.

6.17. [bureau]

```
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
```

-print Display the current configurations

-hotline set hotline IP and remote phone number. If user has enable Hotline function, once IP-Phone been off-hook, it will automatically dial out to assigned IP and phone number.

(bureau -hotline "IP of destination" "Phone number of destination")

Note:

To set IP-Phone under hotline mode must set following configurations:

- 1. Peer-to-Peer mode: sip -mode 0**
- 2. Hotline service: sysconf -service 1**

6.18. [rom]

```
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
```

-print	Display the current configurations
-app	for upgrade the main application code Use APP ROM only to upgrade your IP-Phone in most cases, no need to upgrade another ROM, like BOOT, 2M, etc.
-boot	upgrade main boot code
-boot2m	2M ROM includes all the codes, APP, BOOT, DSP, etc.
-dsptest	DSP testing code (Optional)
-dspcore	DSP kernel code (Optional)
-dspapp	DSP application code (Optional)
-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(0 for TFTP.1 for FTP)
-ftp	specify user name and password for FTP download method

Upgrade the firmware through TFTP server example



1. Download the TFTP program and firmware, and save into the same folder

Firmware Section

<http://www.micronet.info/Download/driver/driver.asp#voip>

TFTP program

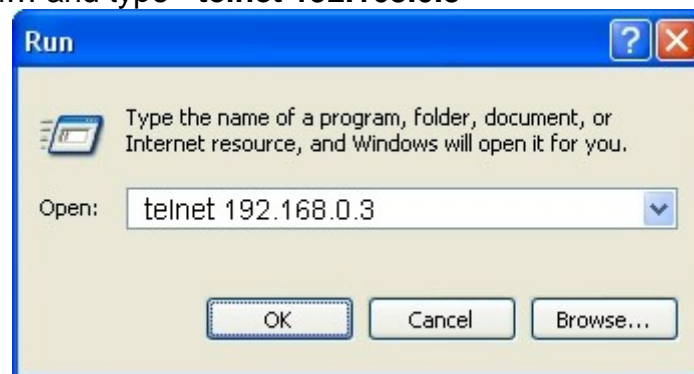
<http://www.micronet.info/Download/Driver/VoIP/utility/tftpd32m.zip>

2. Connect your PC to the IP-Phone RJ-45 port

3. Run TFTP program

4. Start the telnet and login to the IP-Phone

Click -> Start -> Run... and type "telnet 192.168.0.3"



5. Type "rom -app -s 192.168.0.8 -f lp302.101" and press [Enter]

Now the IP-Phone is downloading the ROM file and write into flash memory.

```
usr/config$ rom [-app] [-s 192.168.0.8] [-f lp302.101]
```

Diagram illustrating the command structure for the TFTP upgrade:

- Specify the ROM type** (points to `-app`)
- TFTP server IP** (points to `192.168.0.8`)
- ROM file name** (points to `lp302.101`)

6. Type "flash -clean" to reboot the IP-Phone

6.19. [passwd]

```
usr/config$ passwd

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

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.

- | | |
|---------------|--|
| -print | Display the current configurations |
| -set | set password of “root” users or “administrator” users
(passwd -set root/administrator “password”) |
| -clean | clean up password |

Note:

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

7. Appendix

7.1. LCD Menu Tree

