

The Users' Manual of SIP Phone



Specification Note

In this manual, the “ ” mark indicates that the specification or feature described there may be not available yet and is reserved for the future.

FCC Statement

This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions in this manual, may cause interference to radio communications. This equipment as been tested and found to comply with the limits for a Class A computing device pursuant to Subpart J of Part 15 of FCC rules, which are designed to provide reasonable protection against radio interference when operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference, in which case the user, at is own expense, will be required to take whatever measures are necessary to correct the interface.

CE Declaration of Conformity

This equipment complies with the requirements relating to electromagnetic compatibility, EN55022 class A for ITE and EN 50082-1. This meets the essential protection requirements of the European Council Directive 89/336/EEC on the approximation of the laws of the Member States relating to electromagnetic compatibility.

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Table of Contents

1. Introduction	5
1.1 Package Contents	5
2. The Basic Application Concept.....	6
2.1 Phone-to-Phone Direct Dial.....	6
2.2 Phone-to-Gateway Direct Dial	6
2.3 Phone-thru-SIP Server Dial	6
2.4 Typical Application in Enterprise Office	7
2.5 Advanced Application in Enterprise Office.....	8
3. The Appearance of Product.....	10
4. The Features and Specifications.....	13
4.1 Features of Product.....	13
4.2 Specifications of Product	14
4.2.1 Software Specifications	14
4.2.2 Hardware Specifications	16
5. Installation.....	18
5.1 How to Verify/Check if IP Phone Works Initially	18
5.2 How to Place a Simple Call	18
5.2.1 IP Phone To IP Phone Call	19
5.2.2 PC To IP Phone Call	20
5.2.3 How To Connect The IP Phone To Office LAN	21
5.3. How to Place A Call Directly to A Gateway	22
5.4 How to Register A IP Phone To A SIP Server	23
5.5 How to Call to Remote Points	25
6. The Definitions of Parts	27
6.1 The Definition of Keys	錯誤! 尚未定義書籤。
6.2 The Definition of LED Status.....	錯誤! 尚未定義書籤。
6.3 The Definition of Programmable Features	29
7. The Setting With Menus.....	32
7.1 Turn On the Phone.....	32

7.2. Menus Operation	34
8. The Operation of Product.....	51
8.1 Introduction and Basic Phone Setting	51
8.1.1 To Prepare Typical IP Phones	51
8.1.2 Connecting To The SIP Server Before Operation.....	52
8.2 The Basic Operation With The IP Phone	53
8.3 The Operation With Fixed Function Keys	55
8.4 The Operation With Programmable Feature Buttons	59
Appendix A World GMT Time Zone Table.....	71

Chapter 1

1. Introduction

This IP Phone (Internet Protocol Telephone) is a desktop network telephone that can act like a traditional telephone and deliver good quality voice. Basically, it will be connected to the LAN (Local Area Network) of enterprise as an office VoIP (Voice over Internet Protocol) phone. Besides, it also can be connected to broadband access devices (like XDSL/Cable Modem) for SOHO and residential users.

A VoIP phone can transfer and receive voice in packet type via IP network instead of traditional PSTN network. Besides, VoIP calls can go thru existing global Internet to an endpoint anywhere around the world without paying any charge to the telephony company. So, it becomes more and more popular and convenient for toll call application.

H.323, SIP, MGCP, and Megaco are four major types of VoIP protocol. And SIP is getting popular and well-proven protocol in global enterprise VoIP and multi-media networking communications. The IP phone introduced in this manual is a SIP protocol based product. So, all protocols and applications illustrated in this manual will be limited to the scope of SIP technologies.

1.1 Package Contents

At first, we are thankful very much to you for using our IP Phone product. Upon opening the package, you may find the following items. Be sure that no damage is found on these items and the plug specification of Power adaptor meets the standard of your country. If you found any problem with them, please contact the supplier or distributor for assistance.

1. The main unit
2. Handset
3. Wall-mounting kit
4. Power adapter
5. Quick guide
6. CD: Containing Manuals and Firmware
7. Warranty Card



The main unit



Handset



Wall-mounting kit



Power adapter

Chapter 2

2. The Basic Application Concept

This IP phone supports 3 ways of calling method as described below.

2.1 Phone-to-Phone Direct Dial

This is peer-to-peer dialing method. You can place a VoIP call to a remote IP phone with dialing its URL (Universal Resource Locator) address directly via keypad. No SIP Server or IP-enabled PBX will be involved here. But, be sure that the calling and the called remote IP phones must have fixed real IP addresses. (Please refer to Fig. 2.1 for this call way ①).

2.2 Phone-to-Gateway Direct Dial

This is one-to-group dialing method. You can place a VoIP call to a remote analog phone with dialing its URL (Universal Resource Locator) address directly via keypad. The remote analog phone shall be connected directly to a VoIP Gateway or can be recognized by the Gateway device, such like an Extension line of traditional PBX system. No SIP Server or IP-enabled PBX will be involved here. But, be sure that both the IP phone and the called VoIP Gateway have fixed real IP addresses. (Please refer to Fig. 2.1 for this call way ②).



Figure 2.1. The Calling Ways of Phone-to-Phone and Phone-to-Gateway

2.3 Phone-thru-SIP Server Dial

This is the most common dialing method. You can place a VoIP call to any IP phone(s) or even analog phones with dialing its Extension phone number directly via keypad. A SIP Server or IP-enabled PBX system will be involved for the centrally control

of all call activities, like a traditional way. In this case, the IP phone doesn't need a fixed real IP address. Enterprise internal DHCP server can automatically assign a virtual IP address to it. But, the SIP Server must have a fixed real IP address anyway. This way allows IP phone to IP phone call, IP-phone to Gateway's phone call, and IP-phone to PBX's phone call. (Please refer to Fig. 2.2 for this call way ③).



Figure 2.2. The Calling Way of Phone-thru-SIP Server

2.4 Typical Application in Enterprise Office

Figure 2.3.illustrates typical network architecture of office VoIP system. The IP-enabled PBX system is equipped with VoIP Gateway module(s) and associated with a SIP Server (at least having Proxy and Registrar functions). As usually, it supports internal extension lines to traditional analog phones and external trunks to the PSTN telephone network. At the same time, it may be equipped with a Voice Mail System (VMS) for message recording service. And the SIP Server will act as a call manager to control the routes of VoIP calls. Therefore, all users can take the advantages of IP-enabled PBX features as well as VoIP's benefits.

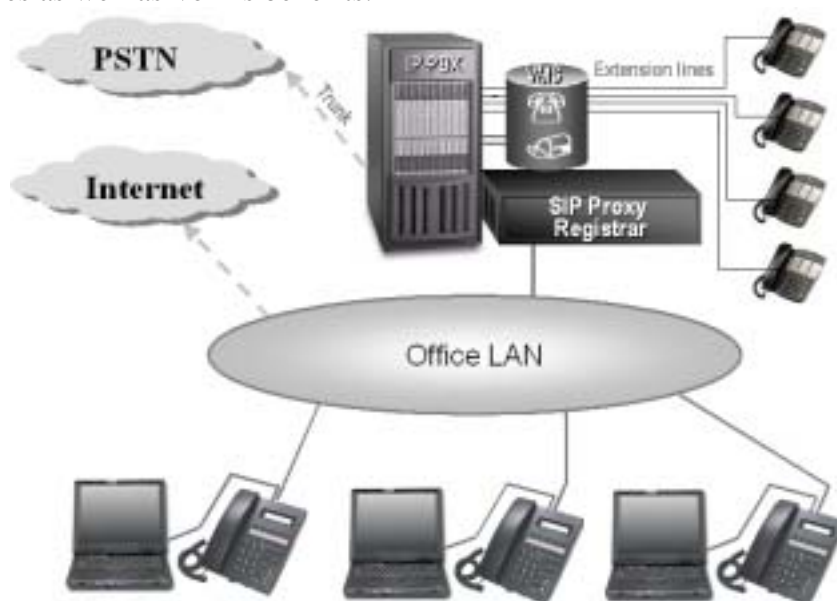


Figure 2.3: The Network Architecture of Office VoIP System

With an additional PC-uplink port, user also can connect notebook or PC to the phone. It will keep voice quality of VoIP calls as usually.

If you are an Enterprise user, your IP phone is suggested to be properly pre-configured by your Administrator (or the supplier) at first. Thus, you can connect it to the Local Area Network (LAN) and register to your SIP Server or IP-enabled PBX system. Then, you will be easy with the use of this IP phone. If you are an Administrator or system installer, please read Administrator Guide at first in order to do necessary and detailed settings.

2.5 Advanced Application in Enterprise Office

For the purpose of management, some enterprises would utilize Intranet /Internet Servers to improve administration capabilities. The following Figure 2.4 shows that some kinds of server are often involved with the operation of VoIP communications. This approach is becoming popular today in modern office and can improve the productivity and efficiency of management.



Figure 2.4. The Advanced System Architecture For VoIP Application

Basically, this IP phone supports cooperation with some servers, such like DHCP server, FTP server, SNTP server DNS server and so on.

Different server can offer different function. However, NOT every server is essential. But they are useful for administrator in management. Regarding the function and protocol of each server, the following table will give you clear idea for understanding their application.

Table 2.1. The Relative Servers Used For Administration

Server Name	Description of Server Function
PC runs browser to access IP phone's	To do Web-based Management with IP phone And upload new firmware.

Web page.	
SIP Server	User can register the IP phone to the SIP Server as an extension line.
DHCP Server	IP Phone will get an assigned IP address from DHCP server
DNS Server	With the help of DNS server, IP phone uses the domain name to discover the IP address of SIP Server for registration.
SNTP Server	IP phone can contact to SNTP server and adjust the system time according to the time zone setting. .

Chapter 3

3. The Appearance of Product

The Figure 3.1 illustrates the top view of the IP phone. With the point number, you can find its name and simple description of the part in the following table.

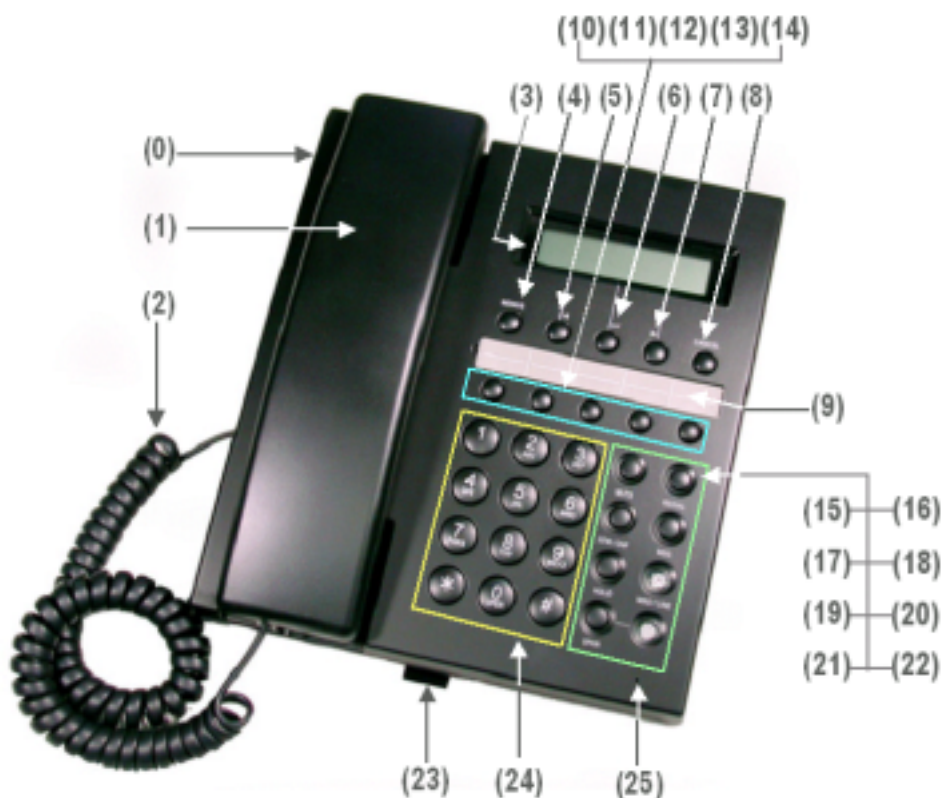


Figure 3.1. The Panel of The IP Phone

Table 3.1 The Name & Description of Parts on The IP Phone

Part #	Feature	Description
0	Main unit	The IP Phone main unit
1	Handset	For placing/receiving calls or retrieving voice message.
2	Handset detachable cable	For connecting the main unit with the handset
3	LCD Screen	The LCD Screen is for displaying your settings, phone number, call status and so forth.
4	[MENUS]	Menus key, to enter the Menus mode for configuration

		purpose.
5	Soft key F1, or [◀]	One of soft key: used for item selection on the screen. Can be used as shift control key for Backward search or used as (volume) Decrease control key.
6	Soft key F2, or [↵]	One of soft key, used for item selection on the screen. Can be used as Enter /Confirm control key.
7	Soft key F3, or [▶]	One of soft key, used for item selection on the screen. Can be used as shift control key for Forward search or used as (volume) Increase control key.
8	[CANCEL]	To quit from the current page and go back to the upper page without changing the settings.
9	Hot/Programmable Memory label	Paper label for user to write down Hot Dial number or Programmable Feature name
10	[P1]	Programmable Feature Button #1, called P1. Default is set to Hot Speed Dial key 1.
11	[P2]	Programmable Feature Button #1, called P2. Default is set to Hot Speed Dial key 2.
12	[P3]	Programmable Feature Button #1, called P3. Default is set to Hot Speed Dial key 3.
13	[P4]	Programmable Feature Button #1, called P4. Default is set to Hot Speed Dial key 4.
14	[P5]	Programmable Feature Button #1, called P5. Default is set to Hot Speed Dial key 5.
15	[MUTE]	To mute the current call on or off.
16	[REDIAL]	To redial the last number.
17	[XFR/CNF]	To transfer a line or to drop multiple lines into a Conference party.
18	[MSG]	Message Retrieval key to access to the Voice Mail System of SIP Server (or IP-enabled PBX).
19	[HOLD]	To hold on the current call and get another new call.
20	MSG / LINE	LED to indicate status of message waiting that the SIP Server notified.
21	[SPKR]	To activate the hand-free call with Speakerphone
22	SPKR	LED to indicate the Line 1 status.
23	Speed Dial Memory Phone Sheet	The Phone sheet paper can be pulled out and written down your 10 sets of Speed dial numbers for remember.
24	[1], [2]...[9], [*], [0], [#]	The numeric dialing pad for dialing number.
25	Microphone Hole	A small hole for Microphone input

The Figure 3.2 illustrates the rear panel of the IP phone. With the names indicated, you can find its simple description of the part in the following table.

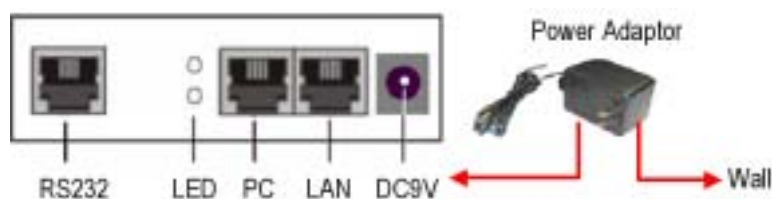


Figure 3.2. The Rear Panel of The IP Phone

Table 3.2 The Parts on The Rear Panel of IP Phone

26	RS-232 Port	A RJ-45 connector on rear panel for RS-232 serial connection to testing PC. It is reserved for engineering test or emergency download only. To connect with this port, you need a special convert cable (RJ-45 to DB-9). [Note: To use this port, please contact your supplier.]
27	LED indicators	2 link active LEDs for PC and LAN Ethernet ports
28	PC Port	A RJ-45 Ethernet 100/10M port for connecting PC
29	LAN Port	A RJ-45 Ethernet 100/10M port for connecting Office LAN or to ADSL/XDSL/Cable Modem
30	DC 9V	9V DC Power Adaptor Jack to connect to an external power adaptor.

Chapter 4

4. The Features and Specifications

With highly integrated chip and sophisticated design, this IP phone offers a variety of features. And it can provide high performance and reliable VoIP communication for the user. The key features are list as follows. For detailed description of them, please refer to the next session of product specifications.

4.1 Features of Product

- Provides LCD screen with 32 characters
- Provides 28 keys Keypad
- Provides 2 LED indicators for call status and message waiting
- Provides dual 10/100Mbps Fast Ethernet ports for LAN and PC
- Supports voice priority function on built-in Ethernet switch and VLAN function

- Supports SIP (RFC 3261) standard of VoIP communications
- Supports two line access call appearances
- Supports Call Hold, Call Transfer, Call Forward feature
- Supports 3-Way Conference.
- Supports standard Message Waiting Indication
- Supports SIP Server Registration and Digest Authentication function
- Supports Phone-to-phone and Phone-to-Gateway direct calls
- Compliant with general SIP phones, Gateways and Servers

- Supports multiple Audio Codecs: G.711 a-law/ μ -law, G.729A, G.723.1 6.3K, G.723.1 5.3K.
- Preferred order and auto-switching of Codec supported
- VAD and CNG supported
- Supports AGC and volume adjustments for voice input and output
- Supports AEC Echo Cancellation
- Adaptive Jitter Buffering supported
- Side tone and good voice quality supported
- Envelope sending supported for dialing digits
- Local Dial, Local Ring, Ring back, and Busy tones supported
- On-hook dialing supported
- In-band DTMF tone generation supported
- Out-band DTMF (RFC-2833) supported
- Support sending DTMF through SIP INFO method
- Up to 10 sets Programmable ringer tone for selection

- Supports last number redial.
- Supports 5 Hot Speed Dial numbers
- Supports up to 10 sets of Speed Dial numbers

- Support 5 programmable buttons for call features
- Supports Soft lock and Pin protection function
- Support Call record for 10 sets outgoing and 10 sets answered calls
- Support a Phone book with 100 sets of phone number

- Supports local configuration with Keypad and LCD display
- Supports DHCP client function
- Supports Web browser-based management supported
- Supports FTP server download
- Supports Web page upload function
- Supports SNTP to synchronize network clock
- Support PPPoE protocol to connected to ADSL/XDSL/Cable modem for dial-up users.
- Supports Telnet server for engineering support and tracing log
- Supports local and remote warm boot function
- Supports STUN NAT traversal protocol
- Supports UDP Heartbeat NAT traversal protocol
- Supports Fix PORT Mapping
- Supports SIP PING method
- Supports auto-provisioning through FTP

4.2 Specifications of Product

4.2.1 Software Specifications

- **Protocols supported:**
 - IETF SIP, SDP
 - TCP/IP, UDP, RTP, RTCP
 - DHCP client (for getting a assigned IP address)
 - SNTP client (for retrieving network time)
 - Web Http Server (Web-based management)
 - FTP client (for download)
 - Telnet Server (for Engineering use)
 - Ethernet IEEE 802.3 (10 Base-T), IEEE 802.3u (100Base-TX),
 - PPPoE

- **SIP: Compliant with IETF RFC3261 standards**

- **SDP: Compliant with IETF RFC2327 standards:**

- **Special dialing methods:**
 - Sending methods: Envelope sending
 - Supports on-hook dialing (Speakerphone) function
 - Redial key for last dialed number
 - Speed dial keys up to 10 sets
 - URL address direct dial
 - Direct IP dial

- **Calling ways:**
 - Phone-to-Phone IP address dialing with URL address or IP address
 - Phone-to-Gateway-to-Phone direct dialing with URL address
 - Phone-thru-SIP Server indirect dialing with E.164 number

- **Supplementary Call Services:**
 - 2 Line appearances with Hold key control
 - Call Hold /Release
 - Call Forward
 - Call Transfer
 - Call Waiting
 - Call Park/Call Pickup
 - Call Conference bridged by IP phone

- **Message Waiting:**
 - Supports message display LED (MSG /LINE) on phone.
 - Provides [MSG] retrieval key to access message (may need the help of IP-enable PBX or PBX).

- **Voice Handling:**
 - Supports multiple Audio Codecs: G.711 a-law/ μ -law, G.729A, G.723.1 6.3K, G.723.1 5.3K.
 - Preferred order and auto-switching of Codec supported.
 - Supports VAD (Voice Auto Detection) and CNG (Comfort Noise Generation)
 - Supports AGC (Automatic Gain Control)
 - Volume adjustable for Handset Microphone, Handset Receiver, Handset DTMF output, Handset Ring output, Speakerphone output, Speakerphone Microphone, Speakerphone DTMF output and Speakerphone Ring output.
 - Support G.165 16ms line Echo Cancellation (AEC).
 - Adaptive Jitter Buffering function supported.

- **Tone Function:**
 - In-band DTMF tone generation.
 - Out-band DTMF relay (RFC-2833) support.
 - Local tone support (Dial, Ring, Ring back, and Busy)
Up to 10 different sets Programmable tone ringer function

- **SIP Server Support:**
 - Can register to the server automatically
 - Support periodically registration by user programmable time
 - Digest authentication Support with MD5 encryption
 - Support Outbound Proxy

- **SIP Server Discoverable way:**
 - By Manually setting
 - By DNS SRV searching

- **System Date & Time supported:**

- SNTP client function
- Automatically re-adjust to the standard time every hour.
- **Supports Convenient system booting methods**
 - Support local and remote warm boot functions
 - Can be done from Menus or Web-page
- **Web-based management support:**
 - Basic Web server built-in
 - Support password control and change capability
 - Most parameters can be configured thru browser easily
 - Phone book can be easily maintained thru browser.
 - Supports uploading configuration file from PC local file
 - Supports issuing download command to FTP server and IP phone
- **Upgrade firmware supported:**
 - User can initiate download through Menus on phone
 - From FTP sever to download firmware and configuration file.
 - Allows user to check firmware version in the phone.
- **Different servers support:**
 - Able to download from FTP server
 - Able to get standard network time from SNTP server
 - Able to get IP address from DHCP server
 - Able to register to SIP Server
- **Engineering support:**
 - Supports RS-323 port for emergency download
 - Supports Telnet server function for command control
 - Telnet interface thru Ethernet port or RS-232 port
 - Allows to collect trace log information via Telnet

4.2.2 Hardware Specifications

- **Main Unit:**
 - Dimension: 170 * 190 * 46 (mm)
 - Plastic material: ABS type and anti-fire
 - Key definition (28-Key Keypad)
- **Detachable Handset:**
 - Connects with the main unit with a 4P4C Phone Jack
 - Plastic material: ABS type and anti-fire
- **LAN and PC up-link Ports:**
 - Supports 2 ports Ethernet switch
 - 2 standard RJ-45 connectors
 - IEEE 802.3 10BaseT / 802.3u 100BaseTx compliant
 - Supports voice priority function on the Ethernet Switch
 - Auto-negotiation with link speed and full/half duplex mode

- Auto MDI/MDIX for both downlink and uplink auto-swapping
- 2 LED indicators for monitoring network link and activities
- Support VLAN function with QOS function.

- **RS-232 Serial Port:**
 - Supports 1 serial port
 - One standard RJ-45 connector (need a RJ-45/DB-9 convert cable)
 - Mainly for engineering support use

- **LED Indicators for phone operation:**
 - 2 dual-color LED indicators supported
 - One [SPKR] for Line 1 status
 - One [MSG / LINE] for Line 2, Message Waiting indication and Download

- **LCD Screen:**
 - Dot matrix 16 characters by 2 lines (total 32 characters)
 - To support Menus operation

- **Keypad:**
 - Support 28 keys
 - 3 soft keys and 5 programmable buttons

- **Power supply:**
 - 9VDC 850mA linear power adapter
 - With 5 types of different country /area for option.

- **Wall-mounting accessory:**
 - Can be used as a stand of the phone.
 - Can be used to mount the phone on the wall (screws provided)
 - Plastic material: ABS type and anti-fire

Chapter 5

5. Installation

Before the installation, please attach the Handset to the main unit of IP phone.

5.1 How to Verify/Check if IP Phone Works Initially

Step 1: Plug the power cord to the rear panel of IP Phone as shown below. Then plug the power adaptor to the wall socket. On the LCD screen, you will see that the phone is “**Booting...**”.

[NOTE]: During booting the IP phone, both **MSG/LINE** and **SPKR** LED will go red steady. When phone start loading program from flash ROM these two LEDs will all turn to green steady. So, please don't interrupt the booting until it gets ready. Then do the following steps.

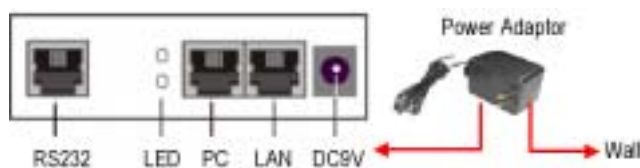


Figure 5.1. Install the IP phone With Power Adaptor

Step 2: Check if the IP phone works by lifting up the handset. If there is a dialing tone, then dial several numbers to see whether the LCD screen is also displaying the dialed numbers or not. If it works, basically the IP phone is fine.

Besides, you may do the above step by pressing the [SPKR] button. When this key is activated, the [SPKR] LED shall go red steady and dial tone will be given, too.

5.2 How to Place a Simple Call

If this IP phone has already been checked and preset well by your Administrator, you just follow his/her instruction and may skip this Chapter. If the IP phone is not yet configured, you may try the following Installation and testing. For Administrator, please read the Administrator Guide completely before you starts the presetting for all IP phones.

There are several settings that need your understanding before you start to place a simple call. Basically, the factory default settings of the phone are listed as follows: (Be sure that it has not preset by your Administrator yet)

- (1) DHCP is "**Disabled**".
- (2) The default IP address of each IP phone is "192.168.1.10".
- (3) The IP address of **SIP Server** is "0.0.0.0".
- (4) The **phone number** is "100".
- (5) So, the URL address of IP phone is 100@192.168.1.10
- (6) SIP Client port is at "5060" and RTP/RTCP port is at "8006/8008".
- (7) The default Codec is "G.711 u-Law".
- (8) **No server** is assumed, including DHCP, DNS, FTP, Web download, STNP, or others.

5.2.1 IP Phone To IP Phone Call

Step 1: Now, if you have two identical IP Phones, please connect them together with a regular (not cross-over) Ethernet Cable as shown in Figure 5.2. Remember to plug the power adapters to the city power.

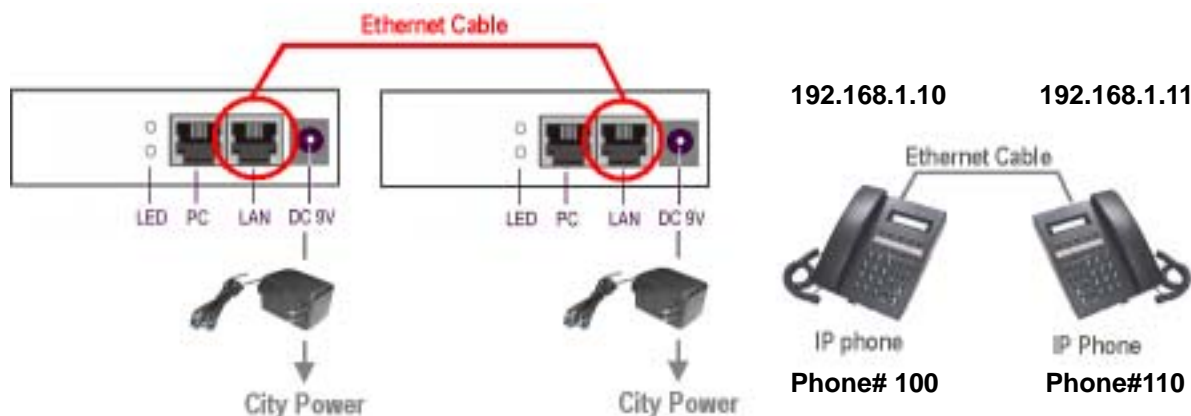


Figure 5.2. To Connect Two IP Phones together For Testing

Step 2: Please choice one IP Phone, press the [MENUS] key and then reach to [IP] Address setting page. Set its IP address to [192.168.1.11] (or any other one before [192.168.1.255]). Then save the setting and reboot the phone to ready state.

Step 3: Now, you can use keypad to directly dial the URL address of the other phone, like:

On IP phone [192.168.1.10]: dial "110*192*168*1*11#" (means 110@192.168.1.11) or

On IP phone [192.168.1.11]: dial "100*92*168*1*10#" (means 100@192.168.1.10)

When the called phone is ringing, you may pick up the handset to answer the phone.

[Remarks] With this URL address direct dialing method, you may dial to any IP phone that has already known real public IP address. For both the calling and the called IP phones, they shall have a reachable IP address respectively. If the called party is using a dynamic IP address, you have to check him to know its currently dynamic IP address before dialing.

5.2.2 PC To IP Phone Call

(**The following test is reserved and please skip this Section).**

It is not necessary to use a PC for testing the IP phone. So, you may skip this step. But, if you just have only one IP phone, it is a good way for you to try testing a simple VoIP call.

You may use a PC for testing the IP phone. The eStara SoftPhone delivers SIP-based communications and services in a software-based phone, it is like Microsoft's NetMeeting™ in H.323, you may choose other SIP User Agent software for testing. For eStara, you may get a free trial version of eStara SoftPhone from web site:

<http://www.estara.com/softphone/index.jsp>

Step 1: Please connect the Administrator PC to an IP phone with an Ethernet cable as shown in Figure 5.3.



Figure 5.3 To Connect An IP phone to a PC For Call Testing

Step 2: To adjust the IP address of PC to the same subnet of IP phone. For example, you want to set the IP address to “**192.168.1.2**”. Please enter to the Console of PC and set the IP address of TCP/IP network. Then check the IP address of your PC with “ipconfig” command if it is really at “**192.168.1.2**” or not.

```
C:\>ipconfig
Windows IP Configuration
Ethernet adapter LAN Connection:
    Connection-specific DNS Suffix.:
    IP Address . . . . . : 192.168.1.2
    Subnet Mask . . . . . : 255.255.255.0
    Default Gateway . . . . . : 192.168.1.1
```

Step 3: To check if the communications between PC and IP phone is ok?

In PC's DOS mode try with “**ping 192.168.1.10**” command. If the IP phone replies well, it means the TCP/IP connection between PC and the IP Phone is good. Where “192.168.1.10” is the new IP address configured for the IP phone.

```

C:\>ping 192.168.1.10
Pinging 192.168.1.10 with 32 bytes of data:

Reply from 192.168.1.10: bytes = 32 time < 2 ms TTL = 64
Reply from 192.168.1.10: bytes = 32 time < 2 ms TTL = 64
Reply from 192.168.1.10: bytes = 32 time < 2 ms TTL = 64
Reply from 192.168.1.10: bytes = 32 time < 2 ms TTL = 64

Ping statistics for 192.168.1.10:
    Packets: Sent = 4, Received = 4, Lost = 0 (0% loss),
    Approximate round trip time in milli-seconds:

```

Step 4: Then install and run eStara SoftPhone in your PC. (Please prepare a Microphone headset for use at PC end.)

Step 4: From the menu of SoftPhone, select File -> Settings -> General, you will see the screen for setting the phone number. You could set it at “200”.

Step 3: Then enter IP address the destination IP phone to be called in the eStara softPhone, for example “100@192.168.1.10#” and press DIAL to initiate the call as well. When the IP phone receives the call, it will ring. You may pick up the handset to start the conversation.

Step 4: On reverse, you may pick up the handset of IP phone, and dial in the number of eStara softPhone, “200*192*168*1*2#”. When the eStara softPhone is ringing, please press “LINE 1” to accept the call.



Figure 5.4. The software SIP Phone for testing

5.2.3 How To Connect The IP Phone To Office LAN

In general application, a SIP server will be invoked in the test and real application. Please setup a SIP server and construct the VoIP system in the test network as shown below (Please find a reference SIP Server in the CD). Please configure the IP phones properly and connect them to the office LAN for operation. (If your Administrator didn't properly configure the phone for you,

please refer to Administration Guide for necessary settings for the IP phone. If it has been well configured, just execute the following steps to install it.)

Step 1: Please connect your notebook or PC (if any) to IP phone at PC RJ-45 port with an Ethernet cable. Then connect your IP phone to the Ethernet Hub with another Ethernet cable. Besides, Administrator needs to connect the Ethernet Hub to Office LAN in order to connect all IP phone users to office VoIP communication environment. Remember to plug the power adapter to the wall and hang on the handset correctly.



Step 2: When the user's NB (or PC) is connected to the IP phone, please check if the lower Link/Activity LED beside the RJ-45 "PC" port is lit or not. Moreover, when the IP phone is connected to the Ethernet Hub, please check if the upper Link/Activity LED beside the RJ-45 "PC" port is lit or not.

Step 3: If the accounts of IP phones have been properly configured already in the SIP Server, normally, the IP phones will automatically logged onto the Server and get ready for calls. Thus, IP phones can call each other thru the SIP Server any time, whatever where they are.

5.3. How to Place A Call Directly to A Gateway

(The following test is reserved and please skip this Section).

Now, we would like to test a second call method with a VoIP Gateway. Please prepare a SIP VoIP Gateway with (at least one FXS port). And connect analog phone(s) to the FXS port(s) of VoIP Gateway. Within your private LAN, you may assign fixed virtual IP address (say, 192.168.5.100 and 192.168.5.101 respectively) and phone numbers (say "100" and "101" respectively) to two IP phones. At the same time, assign another IP address to SIP Gateway (say, 192.168.5.10). (Please refer to the Figure 5.6.)

Besides, you have to configure your Gateway properly. Assign extension phone numbers to the test analog phones (say extension number 800 to 804 individually). Now, with [Menus] setting of the IP phones, please perform the following setting and call tests.

Step 1: Please press [Menus] button to enter the menu mode on IP phone. Then find and enter the IP setting selection; and set the address (default is 192.168.1.10) to a new one ("192.168.5.100" and "192.168.5.101" respectively). Then press [↵] to confirm and get out of this selection.

Step 2: In Menu mode, please enter [Admin] mode with keying in the Password. (Please check with your administrator or the Administrator Guide to learn the default Password). After getting in, please press [▶] key to search [phone number] selection then change the phone number to "100" and "101" for two SIP phones.

Step 3: Now, press [Cancel] to get out to Menu settings Save page. Please select [Save] and [Yes] to confirm save & reboot. Later, you will see the IP phone immediately reboots it and get to ready mode.

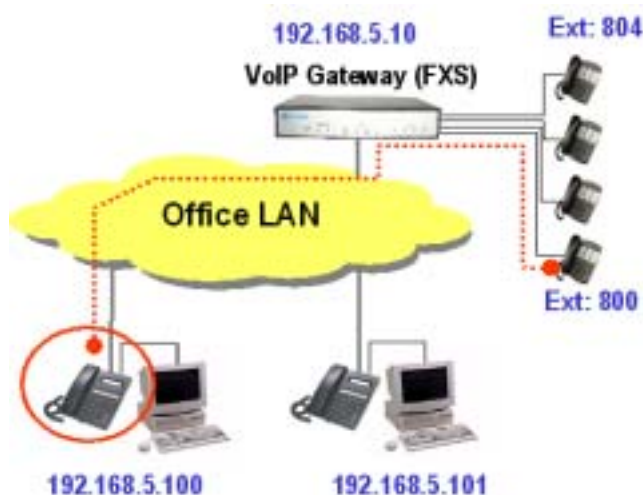


Figure 5.6 To Connect IP phones To A SIP Gateway

Step 4: Now, you can start to test the calls. At first, you may dial the URL address of the first extension phone attached to the SIP Gateway. The phone#800 will ring and you may pick it up for conversation. Please:

Dial : "800*192*168*5*10" (means "800@192.168.5.10")

Don't dial "192*168*5*10" or "192*168*5*10*800. It doesn't work.

Step 5: Besides, you can use other IP phone to place a call to Gateway's extension #4 by dialing "804@192.168.5.10". The phone#804 will ring and you may pick it up for conversation.

Step 6: For Analog phone to dial back to IP phones, you can dial the IP address of IP phone, such like "100*192*168*5*100" or "101*192*168*5*101".

(Note: some commercial Gateways don't support IP address direct dial; so this maybe doesn't work).

[Remarks] With this URL address direct dialing method, you may dial to any IP phone or gateway that has already known real public IP address. For both the calling and the called IP devices, they shall have a reachable IP address respectively. If the called party is using a dynamic IP address, you have to check him to know its currently dynamic IP address before dialing.

5.4 How to Register A IP Phone To A SIP Server

If you have no Administrator to help with the configuration and registration to the SIP Server or you are a technical staff just doing SIP Server registration test, please follow the following steps for testing.

At first, you have to install a SIP Server that supports Digest Authentication function. You may use a commercial one or download a free test copy from the following sites: **SER server at <http://www.ipstel.org> or VOCAL SIP Server at <http://www.vorida.org/>.**

And then build up your SIP test system as shown in the following figure. You need to configure the SIP Server properly. And if you have a (FXS) SIP Gateway, with it, you may bridge the VoIP network and PBX phone system together and configure it properly. At the same time, please register this Gateway to the SIP Server (Say its ID number: “*10” that has been registered to SIP Server successfully already.) (Please refer to Figure 5.7.)

Then configure two IP phones (Say its phone number “10” and “11”) as the following steps.

Step 1: Please use the PC's browser to view the web page of IP phone by key in: <http://192.168.1.10> and <http://192.168.1.11> respectively.

Step 2: With the Web page, please configure the phone especially on these field marked with a circle.

IP: 192.168.1.10, the other 192.168.1.11
Subnet Mask: 255.255.255.0
DHCP: Disable
SIP Server: 192.168.1.250 (assumed)
Authorized User: Test1 and Test2 (assumed)
Authorized Password: test1pass and test2pass (assumed)
Phone number: 10, the other 11.



Figure 5.7. To build a SIP Test Bed in Office Environment

Step 3: Then press “UPDATE” button of the web page to reboot the phone. When Registration

is successful, you will see “Registration OK” message on the LCD display after booting. Then the IP phone will get into ready state.

Step 4: Now you can place a call at phone #10 by pressing

- (1) “11#”: Extension number dialing method, or
- (2) “11*192*168*1*11#”:(for URL:11@192.168.1.11) dialing method(Reserved)
- (3) “101@mycompany.com#”: domain name dialing method (Reserved)

Then the IP phone #11 will ring. You may pick up the phone for conversation.

Step 5: Also you may place a call to Gateway at IP phone by pressing “20#”. Few seconds later, a ring tone will be heard from Gateway. Then you may dial the extension number of the PBX system thru the Gateway. (Note: This needs you to configure the Gateway well before the testing).

Step 6: Of course, after proper setting, office extension phones can dial to the IP phone #10 or #11 thru the Gateway. For example: the administrator may set “8” for VoIP calls and “9” or “0” for external PSTN calls. So, the analog phone user can dial”810” to ring the IP phone #10.

5.5 How to Call to Remote Points

After the above local call tests, you may be eagerly to try a long VoIP call that really goes over the world wide Internet. The following procedures will offer reference for your trials. But, however, you have to prepare the remote testing environment by yourself. If you have no such equipments or broadband lines, you may try the following location that the supplier offers. But it is NOT available always.

In the following figure, you can find a remote SIP Server, IP phone, Gateway, and an analog phone number under testing at the remote site. Before testing, please assure that you can get two real public fixed IP addresses for your two IP phones. (say 200.1.1.100 and 200.1.1.101).

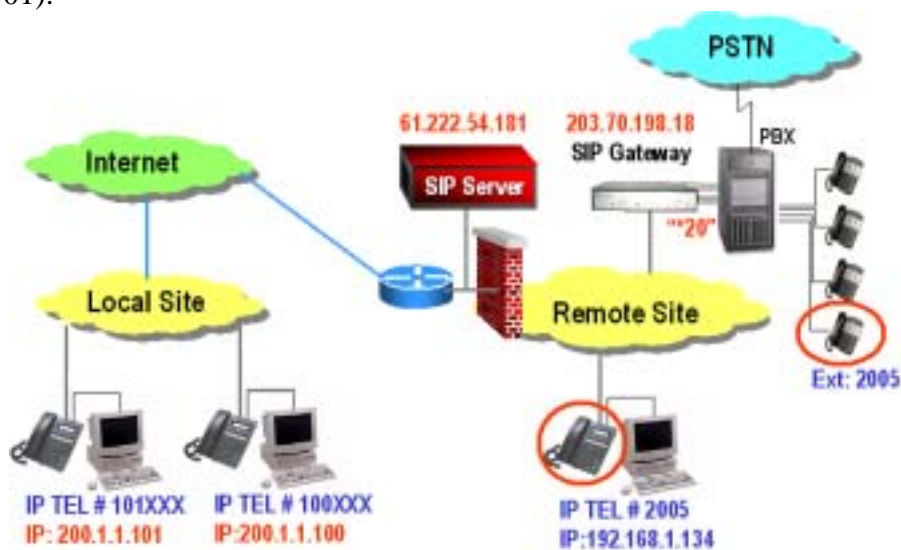


Figure 5.8. A Test With Remote Site

The SIP Server, Gateway, IP phone and PBX system have been well configured for the tester. Suppose we have assigned an ID number “*20” already for VoIP call to directly

accessing to the internal extension phone number.

Then configure two IP phones (Say its phone number “100XXX” and “101XXX”) as the following steps. The XXX is 3 digits and is up to your random choice. If you got a “SIP Server Fail” message as contacting SIP Server, please change it to another random number (Maybe it is used by another party already.)

Step 1: Please use the PC's browser to view the web page of IP phone by key in:
<http://200.1.1.100> and <http://200.1.1.101> respectively.

Step 2: With the Web page, please configure the phone especially on these field marked with a circle.

IP: 200.1.1.100, the other 200.1.1.101
Subnet Mask: 255.255.255.0
DHCP: Disable
SIP Server: “61.222.54.181”
Phone number: 100886, the other 101886.
(where 886 is XXX, a random number).
Authorized User: (null)
Authorized Password: (null).

Step 3: Then press “UPDATE” button of the web page to reboot the phone. You will see “Contacting SIP Server” message on the LCD display after booting. When Registration is successful, the IP phone will get into ready state.

Step 4: Then you can place a call at phone #100886 by pressing “101886#”. The phone #101886 at remote site will ring. Then pick up the phone for conversation.

Step 5: Also you can place a call at phone #100886 by pressing “2005#”. The phone #2005 at remote site will ring. Then somebody may pick up the phone for conversation with you.

Step 6: Of course, you may please the remote person at “IP Tel: 2005” to dial back to your IP phone #100886 or #101886 thru the SIP Server support.

[Note]: The remote Gateway doesn't register to the SIP Server. Here we would like to test the direct call between IP phone and Gateway. (Because maybe you have no Gateway on hand, but you still utilize (your friend's) remote Gateway for a trail.

For more trials or trails with different types of IP address (such like dynamic real IP or virtual IP), or more complicated environment application, please read this manual, and administration guide for learning more detailed information and setting skill, as well as the operational principles of the phone.

Chapter 6

6. The Definitions of Parts

6.1 Definition of Keys

MENUS



[MENUS]

Press this key to enter Menu mode for configuring the IP Phone. Press this key again to Save and/or Exit from the Menu mode.



Soft keys

There are 3 soft keys with vertical lines connecting to the LCD screen. With these soft keys, user may select the desired corresponding item from which shows on the screen. For next group of selectable items, user may press the corresponding key of "Next".



[◀] [▶]

These 2 keys are located inside the Soft key array. Usually, these two keys are used in menu mode as the navigation keys to browse previous / next available options. Sometimes, it will be used as increase/decrease volume in menu mode. During editing

string or IP address, the  will be used as backspace.

Outside Menu mode, they can be used to adjust the volume of speakerphone or handset, when the handset has been picked up or using speakerphone.



[↵] (Same as ENTER)

This key is also located in soft key array. Press this key to Enter edit mode or confirm the setting in current page.

CANCEL



[CANCEL]

Press this key to Cancel current setting(s) and go back to the upper level menu without change. During dialing phase, this key is also used as a backspace.

MUTE



[MUTE]

Press this key to mute the microphone.

XFR/CNF



[XFR / CNF]

Press this key to transfer the current call to another one. Currently this button is for transfer function only.

HOLD



[HOLD]

Press this key to hold current call and pick up another line.

SPKR



[SPKR]

Press this key to pick up the phone in hand-free mode. If the key is already in hand-free mode, pressing this key will hang up the phone.

REDIAL



[REDIAL]

Press this key to invoke the last number dialed and make call immediately.

MSG



[MSG>(* Reserved)

Press this key to retrieve the voice mailbox.







Programmable Feature buttons

There are 5 buttons located above the numeric keypad and without any printing above. Starting from the most left one, these buttons are referred to Key P1, Key P2, ... to Key P5 respectively. The detail description of available functions will be shown in section 2.3.

6.2 The Definition of LED Status

On the panel of phone, there are two LED indicators. One is associated with SPKR key (referred as LED1) and the other is marked "MSG / LINE"(referred as LED2). The first one LED is mainly for the status display of Line # 1. And the second LED is for the status display of Line # 2 and Message Waiting. The following table describes the detailed definition of LED status.

Table 2.1. The Definitions Of LED Status

	RED		Green	
				
	Red Steady	Red Flashing	Green Steady	Green Flashing
MSG / LINE (LED2)	<ul style="list-style-type: none"> ◆ Downloading ◆ Booting 	<ul style="list-style-type: none"> ◆ Line 2 has a holding call or incoming call. (When engage line 1) ◆ Message Waiting (When IDLE) 	<ul style="list-style-type: none"> ◆ Line 2 is engaging 	N/A
SPKR (LED1)	<ul style="list-style-type: none"> ◆ Downloading ◆ Booting 	<ul style="list-style-type: none"> ◆ Line 1 has a holding call or incoming call. (When engage line 2) 	<ul style="list-style-type: none"> ◆ Line 1 is engaging 	<ul style="list-style-type: none"> ◆ Ringing ◆ DND, Call Forward

[Remark] Line #1 status is always shown on the “SPKR” LED while the Line #2 status is always displayed on the “MSG / LINE” LED. And when a line is held, the LED is turned to flashing. When the line is resumed again, the LED will light steadily again.

6.3 The Definition of Programmable Features

The programmable features can be enabled and mapped to the programmable feature buttons on the phone panel. The Administrator (or user) may configure these features via Menus, Web-management, Telnet, or Administrator tool. The following list shows basic programmable features for selection.

(1) Speed Dial:

This function aim to provide a shortcut for users to access most frequently used numbers or important numbers. Up to 10 sets of numbers mapping to the number keys (0,1...9). User may invoke predefined numbers by a speed dial key plus a number key. For example, the programmable button P1 is programmed to perform Speed Dial function and the predefined number for keypad 1 is defined as “100”. User can click P1 then click keypad 1 to invoke the speed dial number “100”. User will see a number “100” show on the screen. The following operation is exactly the same as he just dialed the three digits (“100”) one by one.

(2) Call Record:

This function provides a way to review the call history stored in memory. There are 3 different types of call history the outgoing calls, the incoming calls and missed calls. The call history records the latest 10

sets of numbers for each type. User may review the record, maintain the record (delete single number or delete entire record) and dial the number in the record.

(3) Phone Book:

When pressing this programmable button, the phone book is prompted to the Menus display. A maximum of 100 sets of phone number can be stored in the phone book. User may view, modify and delete the phone number (or IP address) and its associated Name.

(4) Call Forward:

With this feature, the IP phone will execute the extension service of "Call Forward". The call-forward function will be invoked base on 3 different conditions. First, unconditional forward, user may enable this and all incoming call will forward to another number defined by the user immediately. Second, forward on busy, as the name explained, the incoming call will be forward to another number defined by the user as long as all lines on this phone is occupied. Third, forward on no answer, user may enable this function and all incoming call will forward to another number defined by the user when the incoming call was not answered for a certain period of time (Also defined by the user). With this call-forward feature, user can implement a so-called "follow me" on this phone without the SIP server support.

(5) Conference:

By selecting this function this phone can bridge two existing calls and do a 3-Way conference through this button.

(6) Lock IP-Phone:

Upon pressing this key, the phone will jump to lock state. User will need to provide his password to unlock the phone before he can access the service. This function aim to provide a way to protect the phone form unauthorized user.

(7) Use DND:

By selecting this function user can turn on/off Do Not Disturb (DND) function by a single click on this button.

(8) HOT Speeddial:

By selecting this function user can immediately place a call to predefined number by a single click on this button. The predefined numbers are the first 5 entries of the speed dial numbers. (Speed dial 0 to Speed dial 4) The predefined hot-speed-dial number for the

programmable button 1 is the speed dial 0 and the predefined hot-speed-dial number for the programmable button 5 is the speed dial 4.

Chapter 7

7. The Setting With Menus

The Screen of LCD provides 2 lines information with a total of 32 characters for the operation of Menus. User and Administrator may use it to configure the IP phone. The following descriptions will guide you how to operate and configure the IP phone.

7.1 Turn On the Phone

[Power on the phone]

When the phone is powered on, the system will boot up and the following information will be displayed on the LCD screen.

<u>Index</u>	<u>LCD Screen</u>	<u>Description of Action</u>
Booting	Booting...	Upon power-on, the LCD will show "Booting" and the phone start to load the firmware into memory. The system will be initialized and both LED will go red steadily. And all setting in the memory will be loaded, too.
	Loading Program Please Wait.....	
Fix IP	Registering...	After loading the settings from the flash, the company name and model number will shows on the screen. In static IP case, the phone will use the fix IP address stored in flash ROM to start the network stack. The phone will try to register to the SIP server now.
	DHCP	If DHCP is enabled, the IP-Phone will try to discover the DHCP server and get network settings from the DHCP server.
DHCP	CHCP Connecting...	If the DHCP success, the IP address will shows on the screen. Later you can examine the current IP address through WEB or MENU.
	DHCP Success 192.168.5.23	
PPPoE	PPPoE Connecting...	If using PPPoE, the IP-Phone will try to connect to the ISP.

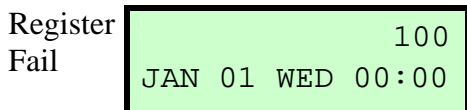
	<pre>PPPoE Success 210.63.16.37</pre>	If the PPPoE successfully establishes connection to the ISP you will see the IP address on the screen.
Registering	<pre>Registering...</pre>	After successfully construct the network stack, the phone will try to register to the SIP server.
	<pre>Registering... JAN 01 WED 00:00</pre>	If the SNTP is enabled, the phone will try to synchronize the system time with NTP server. When the time is successfully synchronized with the NTP server, the time will shows on the second line.
Ready	<pre>TEL: 100 JAN 01 WED 00:00</pre>	When the phone successfully register to a SIP server, the left of the first line will shows "TEL:" indicating you can make call through SIP server now. The current phone number will shows on the right hand side of the first line. If the "TEL:" is now seen on the screen that means the phone is not registered to any server. You have to check the SIP related setting and network settings.
	<pre>L1 :</pre>	When you pick up the phone, the upper left corner will tell you which line you are using.
Make a call	<pre>2150</pre>	After dialing the first digit, the line indicator will disappear, and the current number dialed will shows on the right hand side of the screen.

[Register to SIP Server]

Before this, you have to make sure that Administrator has already built your account information (phone number, name and password) into the Account database of SIP Server. And configure the account number and password, as well as SIP Server IP address on your IP phone correctly, too.

During Power-on or rebooting the phone, if the IP phone will attempt to register to the SIP server after system initialization is done.

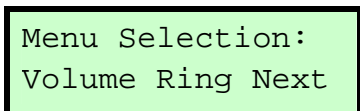
Register	<pre>Registering... JAN 01 WED 00:00</pre>	If the SIP Server IP address has been specified correctly, during booting, the IP phone will try to register to the appointed SIP Server.
Register Success	<pre>TEL: 100 JAN 01 WED 00:00</pre>	When IP-Phone successfully register to a SIP server, the upper left corner will shows "TEL:" to indicate the phone is ready to make



call or receive call through SIP server. If IP-Phone fails to register to the SIP server, the upper left corner will keep blank.


7.2. Menu Operation






Components involved in menu system:



These are the major components in menu system. Surely the keypad (0~9) will be involved for editing specific settings.

General Operation:

User may press  [MENUMS] key to enter Menu mode for configuration. Besides, after setting, user may press this key again to save and/or quit from the Menu.

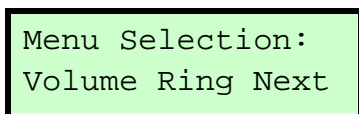
There are 5 buttons involved in menu selection. The  [MENUMS],  [Left Arrow],  [Left Arrow with Down Arrow],  [Right Arrow] and  [CANCEL].

[MENUMS]: Invoke the menus or leave the menu selection mode.

[CANCEL]: Leave the current selection item and back to upper layer menu item.

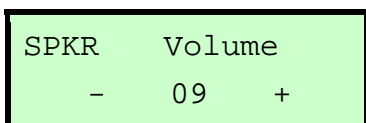
[Left Arrow], [Left Arrow with Down Arrow], [Right Arrow]: These three keys are SOFT Keys. Their functions will depend on what is shown on the LCD panel. The following is some general case.

Browsing the menu:

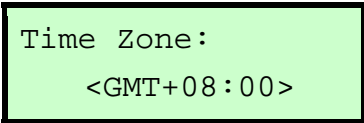


In this case, press [Left Arrow] will invoke the Volume Control menu. Press [Left Arrow with Down Arrow] will invoke Ring Selection menu. Press [Right Arrow] will simply jump to next menu.

Volume adjustment:

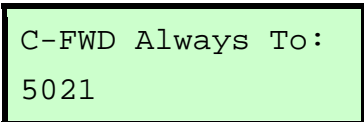


In this case, press [Left Arrow] will decrease the volume. Press [Right Arrow] will increase the volume. Press [Left Arrow with Down Arrow] will confirm and save the current setting, and go back to upper layer menu.

Select desired settings:


Time Zone:
<GMT+08:00>

In this case, there is a fixed list of values. User can use [◀] and [▶] to navigate previous/next available value and then press [↵] to confirm and save the current setting, and go back to upper layer menu.


Editing:



C-FWD Always To:
5021

When editing some string, the [◀] will be the backspace key, and [↵] will confirm and save current setting. ([▶] Key has no function during editing.)

Alphabetic Input:

User may need to input non-numeric settings for various purposes such as the domain name of SIP server, the name field in phonebook. This phone provide a widely used mechanism to input alphabetic characters via the number pad. The following is a list of how the characters and symbols map to the number pad.

 '0' @:+-_

 1 .,(Note the second character is a blank)

 2ABC

 3DEF

 4GHI


 5JKL


 6MNO

 7PQRS

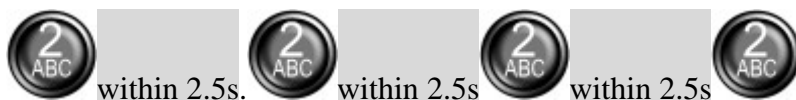
 8TUV

 9WXYZ

If you want to input a character 'C' you can press the key  4 times before the cursor move to the next position. The cursor will move to next position if you didn't press any key for 2.5 seconds. The phone will automatically confirm the current selection if you didn't press the same key for 2.5 seconds or you press a different key.

You can speed up to confirm the current selection by pressing the key  during alphabetic editing. The current selection will be confirm immediately and move the cursor to next input position.

So, here is a detail sequence of how you input a 'C':



The 'C' should have been shown on the screen, now you can wait for 2.5 seconds the cursor will automatically move to next position. Or, you can press any other keys to perform next input the cursor will move to next position immediately.

Menu System:

The following is a list of available menu items and their functions.

<u>Index</u>	<u>LCD Screen</u>	<u>Description of Action</u>
0	Menu Selection: Volume Ring Next	After pressing the [MENUS] key, the first menu selection will shows on screen. User may press [MENUS] again during the menu operation, and the 0.0 menus will shows on screen.
0.0	Menu Selection: Save Quit Back	Press: Save: to Save all settings to Flash permanently and Re-Boot the phone directly. Quit: to abort the settings and quit from Menus mode. Back or [CANCEL]: to go to the upper page without change.
0.1	Save & Reboot Yes No	Press: Yes: to Save all settings to Flash permanently and Re-Boot the phone directly. No or [CANCEL]: to go to the upper page without change.
0.2	Quit & No Save Yes No	Press: Yes: to abandon all settings and quit from Menus. No or [CANCEL]: to go to the upper page without change.

When the phone is in standby mode, user may press [MENUS] key to enter Menus mode. There are many selections of function in Menus. They are listed as follows:

Index: 1 2	Menu Selection: Volume Ring Next	Volume: to adjust Speaker, Microphone and Handset. Ring: to select a type of ring. Next: to proceed to the next selection. [CANCEL]: to go to the upper page without change.
3 4	Menu Selection: Tone Time Next	Tone: Set preferred tone type. Time: Set NTP related settings. [CANCEL]: to go to the upper page without change.
5 6	Menu Selection: Info Progrm Next	Info: Phone specific information. Progrm: There are 5 programmable buttons on this phone; you can use this menu to select their functions. [CANCEL]: to go to the upper page without change.
7 8	Menu Selection: DND C-FWD Next	DND: Turn on/off the do not disturb (DND). C-FWD: Turn on/off various call forward functions and set the call forward numbers. [CANCEL]: to go to the upper page without change.
9 A	Menu Selection: A-Ans Lock Next	A-Ans: Turn on/off auto-answer. Lock: Lock the phone to prevent unauthorized using. [CANCEL]: to go back to the upper page.
B C	Menu Selection: Record Ping Next	Record: Examine the call record for last 10 sets of incoming, outgoing and missing calls. Ping: Perform a PING request to a specific address. You can use this to verify the network or to verify whether some devices are on line or not. [CANCEL]: to go back to the upper page.
D E	Menu Selection: P.Book Spd Next	P.BOOK: Accessing the phone book. Spd: Accessing the speed dial settings. [CANCEL]: to go back to the upper page.
F G	Menu Selection: TCP/IP Admin Next	TCP/IP: TCP/IP network settings. Admin: The advance setting goes here. [CANCEL]: to go back to the upper page.
H I	Menu Selection: Block Save Home	Block: Accessing the call-blocking list. Save: Save the change to phone. Home: Back to the first menu.

[Volume Control]

After entering the selection of Volume Control, user may use [◀]/ [▶] key to decrease (◻) / increase (■) the volume of voice input and output.

- | | | |
|------------|-------------------------------------|--|
| 1 | Volume Control:
SPKR HS-MIC Next | SPKR: Adjust the speakerphone volume.
HS-MIC: Adjust microphone input level of the handset.
[CANCEL]: to go back to the upper page. |
| 1.1
1.2 | Volume Control:
Handst Ring Next | Handst: Adjust handset output level.
Ring: Adjust ring volume.
[CANCEL]: to go back to the upper page. |
| 1.3
1.4 | Volume Control:
SPK-MIC Back | SPK-MIC: Adjust input level for hand-free microphone
Home: will go back to the first item of this selection.
[CANCEL]: to go back to the upper page. |
| 1.5
1.6 | SPKR Volume
- 09 + | Adjust the hand-free speaker volume:
[■]: Increase the volume.
[▣]: Decrease the volume.
Current volume value: say "09", is shown on the screen. (Range 01~14).
[CANCEL]: to go back to the upper page. |
| 1.1 | HS-MIC Volume
- 09 + | Adjust the microphone input level of handset.
[■]: Increase the volume.
[▣]: Decrease the volume.
Current volume value: say "09", is shown on the screen. (Range 01~12).
[CANCEL]: to go back to the upper page. |
| 1.2 | Handset Volume
- 09 + | Adjust the output volume of handset.
[■]: Increase the volume.
[▣]: Decrease the volume.
Current volume value: say "09", is shown on the screen. (Range 01~12).
[CANCEL]: to go back to the upper page. |
| 1.3 | Ring Volume
- 09 + | Adjust the ring volume.
[■]: Increase the volume.
[▣]: Decrease the volume.
Current volume value: say "09", is shown on the screen. (Range 01~14).
[CANCEL]: to go back to the upper page. |
| 1.4 | SPK-MIC Volume
- 09 + | Adjust the input level of hand-free microphone.
[■]: Increase the volume.
[▣]: Decrease the volume.
Current volume value: say "09", is shown on the screen. (Range 01~12).
[CANCEL]: to go back to the upper page. |
| 1.5 | SPK-MIC Volume
- 09 + | Adjust the input level of hand-free microphone.
[■]: Increase the volume.
[▣]: Decrease the volume.
Current volume value: say "09", is shown on the screen. (Range 01~12).
[CANCEL]: to go back to the upper page. |

[Ring Type]

There are 10 different ring type are provided. Basically, they are the ring pattern of different countries. User may browse and hear the different ring types.

- 2
- ```
Ring Type:
 < USA >
```
- Select the Ring Type.  
 [←]: Examine the previous ring type.  
 [→]: Examine the next ring type.  
 [↵]: Confirm and save the selection.  
 [CANCEL]: to go back to the upper page.

### [Tone Type]

There are 10 different tone type are provided. Basically, they are the DTMF tone pattern of different countries.

- 3
- ```
Tone Type:
  <   USA   >
```
- Select the Tone Type.
 [←]: Examine the previous tone type.
 [→]: Examine the next tone type.
 [↵]: Confirm and save the selection.
 [CANCEL]: to go back to the upper page.

[Time]

By entering this menu, user can set the network time related settings.

- 4
- ```
Time Setting:
Zone DayLS
```
- 4.1
- ```
Time Zone:
<GMT+08:00>
```
- 4.2
- ```
Daylight Saving:
Disable Back
```
- Select the Tone Type.  
 [←]: Examine the previous tone type.  
 [→]: Examine the next tone type.  
 [↵]: Confirm and save the selection.  
 [CANCEL]: to go back to the upper page.
- Select the Tone Zone.  
 [←]: Select previous time zone.  
 [→]: Select next time zone.  
 [↵]: Confirm and save the selection.  
 [CANCEL]: to go back to the upper page.
- Select the Tone Zone.  
 [◀]: Turn on/off daylight saving. The screen shows the current setting.  
 [↵]: No action.  
 [▶]: Back to upper layer.  
 [CANCEL]: to go back to the upper page.

### [Information]

Examine the status and miscellaneous information of the phone.

- 5
- ```
Info:Comany
Company Name
```
- The company name.
 [◀]: Examine the previous information.
 [▶]: Examine the next tone information.

- | | | |
|-----|-----------------------------------|--|
| 5.1 | Info:Model
SIP Phone | [↵]: Examine the next tone information.
[CANCEL]: to go back to the upper page.
The model name.
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information.
[CANCEL]: to go back to the upper page. |
| 5.2 | Info:Firmware
1.0.08 | The firmware version.
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information. |
| 5.3 | Info:MAC Address
0003C9:3043E8 | [CANCEL]: to go back to the upper page.
The MAC address.
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information.
[CANCEL]: to go back to the upper page. |
| 5.4 | Info:SIP Proxy
192.168.2.17 | The SIP Proxy server.
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information.
[CANCEL]: to go back to the upper page. |
| 5.5 | Info:OutboundSrv
192.168.2.18 | The outbound proxy server.
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information.
[CANCEL]: to go back to the upper page. |
| 5.6 | Info:RegistrarSrv
192.168.2.17 | The REGISTRAR server.
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information.
[CANCEL]: to go back to the upper page. |
| 5.7 | Info:RegOutbound
192.168.2.18 | The outbound REGISTRAR server.
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information.
[CANCEL]: to go back to the upper page. |
| 5.8 | Info:Reg Status
Register OK | Registration status:
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information.
[CANCEL]: to go back to the upper page. |
| 5.9 | Info:Local IP
192.168.1.10 | The current local IP address.
[⏪]: Examine the previous information.
[▶]: Examine the next tone information.
[↵]: Examine the next tone information. |

5.10	Info:Subnet Mask 255.255.255.0	<p>[CANCEL]: to go back to the upper page. The current subnet mask.</p> <p>[◀]: Examine the previous information. [▶]: Examine the next tone information. [↵]: Examine the next tone information.</p>
5.11	Info:Router IP 192.68.1.1	<p>[CANCEL]: to go back to the upper page. The current router IP address.</p> <p>[◀]: Examine the previous information. [▶]: Examine the next tone information. [↵]: Examine the next tone information.</p>
5.12	Info:DHCP Disable	<p>[CANCEL]: to go back to the upper page. The DHCP enable or not.</p> <p>[◀]: Examine the previous information. [▶]: Examine the next tone information. [↵]: Examine the next tone information.</p>
5.13	Info:PPPoE Disable	<p>[CANCEL]: to go back to the upper page. The PPPoE function is enabled or not.</p> <p>[◀]: Examine the previous information. [▶]: Examine the next tone information. [↵]: Examine the next tone information.</p>
5.14	Info:PPPoE ID pppoeid	<p>[CANCEL]: to go back to the upper page. The PPPoE user ID.</p> <p>[◀]: Examine the previous information. [▶]: Examine the next tone information. [↵]: Examine the next tone information.</p>
5.15	Info:DNS IP 168.95.1.1	<p>[CANCEL]: to go back to the upper page. The Primary DNS server address.</p> <p>[◀]: Examine the previous information. [▶]: Examine the next tone information. [↵]: Examine the next tone information.</p>
5.16	Info:Service Num 886-2-26551000	<p>[CANCEL]: to go back to the upper page. The Service telephone number.</p> <p>[◀]: Examine the previous information. [▶]: Examine the next tone information. [↵]: Examine the next tone information.</p>

[Programmable Button]

There are 5 programmable buttons on the phone. (Refer as P1~P5). Each of them can be programmed to invoke various predefined functions. User can examine the current settings of these buttons.

6	Program Key [P1] < Speed Dial >	<p>Examine and select programmable button.</p> <p>[=]: Examine the previous programmable button setting.</p>
---	------------------------------------	--

- ▶]: Examine the next programmable button setting.
- ↵]: No action.
- [CANCEL]: to go back to the upper page.

[Do Not Disturb]

User can invoke Do Not Disturb function to reject all incoming calls. In the other hand, user still can make call to others.

```
7
Do Not Disturb
Disable      Back
```

- Turn on/off the Do Not Disturb (DND) function. The status shows on the second line is the current setting.
- ◀]: Toggle between enable and disable.
- ▶]: Back: Back to previous menu.
- ↵]: No action.
- [CANCEL]: to go back to the upper page.

[Call Forward]

User can invoke Call Forward function base on three different conditions.

Forward always: If this function is enabled, all incoming call will be automatically forward to user-defined number.

Forward on busy: If this function is enabled, when all the lines are occupied by existing calls, the incoming call will be automatically forwarded to user-defined number.

Forward on no answer: If this function is enabled, the incoming call will automatically forward to user-defined number if the user didn't answer the phone in certain period of time defined by the user.

```
8
8.1
8.2
8.3
Call Forward:
Always Busy NoAns
```

- Select type of call forward function to be configured.
- ◀]: Configure the forward always function.
- ↵]: Configure the forward on busy function.
- ▶]: Configure the forward on no answer function.
- [CANCEL]: to go back to the upper page.

```
8.1
8.1.1
8.1.2
8.1.3
C-FWD Always:
Disable Num Back
```

- Configure the forward always function. The function is enabled or disabled is shown on the screen.
- ◀]: Toggle between enable/disable.
- ↵]: Num: To edit the number where incoming call is forwarded to.
- ▶]: Back: Back to previous menu.
- [CANCEL]: to go back to the upper page.

```
8.1.2
C-FWD Always To:
5021
```

- Edit the forward always number.
- ◀]: Backspace.
- ↵]: Confirm the current editing result.

8.2	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> C-FWD OnBusy: Disable Num Back </div>	[▶]: No action.
8.2.1		[CANCEL]: to go back to the upper page.
8.2.2		Configure the forward on busy function. The function is enabled or disabled is shown on the screen.
8.2.3		[◀]: Toggle between enable/disable.
		[↵]: Num: To edit the number where incoming call is forwarded to.
		[▶]: Back: Back to previous menu.
8.2.2	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> C-FWD OnBusy To: 5021 </div>	[CANCEL]: to go back to the upper page.
		Edit the forward on busy number.
		[◀]: Backspace.
		[↵]: Confirm the current editing result.
		[▶]: No action.
		[CANCEL]: to go back to the upper page.
8.3	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> C-FWD NoAnswer: Disable Num Back </div>	Configure the forward on no answer function. The function is enabled or disabled is shown on the screen.
8.3.1		[◀]: Disable: Toggle between enable/disable.
8.3.2		[↵]: Num: To edit the number where incoming call is forwarded to.
8.3.3		[▶]: Back: Back to previous menu.
		[CANCEL]: to go back to the upper page.
		Edit the forward on no answer number.
		[◀]: Backspace.
		[↵]: Confirm the current editing result.
		[▶]: No action.
		[CANCEL]: to go back to the upper page.
8.3.2	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> C-FWD NoAns To: 5021 </div>	Configure the forward on no answer function. The function is enabled or disabled is shown on the screen.
		[◀]: Disable: Toggle between enable/disable.
		[↵]: Num: To edit the number where incoming call is forwarded to.
		[▶]: Back: Back to previous menu.
		[CANCEL]: to go back to the upper page.
		Edit the forward on no answer number.
		[◀]: Backspace.
		[↵]: Confirm the current editing result.
		[▶]: No action.
		[CANCEL]: to go back to the upper page.

[Auto Answer]

User can invoke Auto Answer function to automatically answer incoming call.

9	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> AutoAns Setting: DisableTime Back </div>	Turn on/off the Auto answer function. The status shows on the second line is the current setting.
9.1		[◀]: Disable: Toggle between enable and disable.
9.2		[↵]: Time: Set the time before auto answer the call.
9.3		[▶]: Back: Back to previous menu.
		[CANCEL]: to go back to the upper page.
9.2	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> AutoAns Timeout: 1 </div>	Configure the timeout value before the phone to be auto-answered. (in seconds)
		[◀]: Backspace.
		[↵]: Confirm and save current setting.
		[▶]: No action.
		[CANCEL]: to go back to the upper page.

[Lock the Phone]

User can lock the phone to prevent unauthorized user to use this phone.

A
A.1
A.2

```

Lock Setting:
Lock          Back
    
```

Turn on/off the Phone Lock function. The status shows on the second line is the current setting.

[◀]: Lock: Lock the phone, there will be a confirm menu (9.2) shows up to make sure the user really want to lock the phone. Later on, user needs to provide the password to unlock the phone.

[↵]: No action.

[▶]: Back: Back to previous menu.

[CANCEL]: to go back to the upper page.

Confirm to lock the phone.

[◀]: Yes: Confirm to lock the phone. The phone will show a locked screen. Any one wants to use this phone has to input the pin number to unlock the phone.

[↵]: No: Back to previous menu.

[▶]: No action.

[CANCEL]: to go back to the upper page.

A.1

```

Confirm Lock:
    Yes No

Phone Locked...
Pin:
    
```

[Call Record]

User can retrieve the call history by examine the three different call records. IP-Phone will record last 10 sets of incoming calls, outgoing calls and missed calls.

B
B.1

```

Call Record:
Dialed Next
    
```

Examine the call record.

[◀]: Dialed: Examine the outgoing call records.

[↵]: Next: Next menu.

[▶]: No action.

[CANCEL]: to go back to the upper page.

B.1

```

Dialed Calls:1
5021
    
```

Examine the record of dialed calls. If the second line shows "Empty" that means the call record is empty.

[◀]: Examine the previous entry in this record.

[↵]: Make call to this number directly

[▶]: Examine the next entry in this record.

[CANCEL]: Perform a delete in this record.

Call record delete.

B.1.1
B.1.1.1
B.1.1.2
B.1.1.3

```

Call Record Del:
Del DelAll Back
    
```

[◀]: Delete current entry in this record.

[↵]: Clear the entire history.

[▶]: Back: Back to previous menu.

B.2 B.3	Call Record: Answer Missed	Call record second menus. [◀]: Answer: Examine the incoming call record. [↵]: Missed: Examine the missed call record. [▶]: No action. [CANCEL]: to go back to the upper page.
B.2	Answer Calls:1 5021	Examine the record of answered calls. If the second line shows "Empty" that means the call record is empty. [◀]: Examine the previous entry in this record. [↵]: Make call to this number directly [▶]: Examine the next entry in this record. [CANCEL]: to go back to the upper page.
B.2.1 B.2.1.1 B.2.1.2 B.2.1.3	Call Record Del: Del DelAll Back	Call record delete. [◀]: Del: Delete current entry in this record. [↵]: DelAll: Clear the entire history. [▶]: Back: Back to previous menu.
B.3	Missed Calls:1 5021	Examine the record of missed calls. If the second line shows "Empty" that means the call record is empty. [◀]: Examine the previous entry in this record. [↵]: Make call to this number directly [▶]: Examine the next entry in this record. [CANCEL]: to go back to the upper page.
B.3.1 B.3.1.1 B.3.1.2 B.3.1.3	Call Record Del: Del DelAll Back	Call record delete. [◀]: Del: Delete current entry in this record. [↵]: DelAll: Clear the entire history. [▶]: Back: Back to previous menu.

[PING]

User can issue an ICMP ECHO request by issue a PING to specific address. Usually, this will help to verify whether the network settings are correct or not. Also, this function might help to verify whether the network is connected.

C	Ping Setting: 192.168.1.1	Send a PING to specific address. [◀]: Backspace. [▶]: No action. [↵]: Ping this address. [CANCEL]: to go back to the upper page.
C.1	Ping Setting: Ping Success...	Receive the response of this address. [◀]: No action [▶]: No action. [↵]: No action. [CANCEL]: Go on to ping another address.

- C.2
- | |
|---------------|
| Ping Setting: |
| Ping Fail... |
- No response from this address.
 - [◀]: No action
 - [▶]: No action.
 - [↵]: No action.
 - [CANCEL]: Go on to ping another address.

[Phone Book]

This IP-Phone provides a user phone book with 100 entries. Each entry has a name field and number field. The name field is aim to help user to recognize the entry. The number field can be phone number (A valid number registered on SIP server.), IP address or SIP URL.

Phone Book Setting page:

- D
- | |
|--------------|
| Phone Book: |
| Goto[1-100]: |
- Select the phone book entry:
 - [◀]: Backspace.
 - [▶]: No action
 - [↵]: Confirm the selection.
 - [CANCEL]: to go back to the upper page.
- D.1
- | |
|------------|
| 001:George |
| 5021 |
- Showing the entry:
 - [◀]: Examine previous entry.
 - [▶]: Examine next entry.
 - [↵]: Action for this entry.
 - [CANCEL]: to go back to the upper page.
- D.1.1
- | |
|------------------|
| 5021 |
| Dial Delete Edit |
- Action to be taken on this entry.
 - [◀]: Dial: Dial this number.
 - [↵]: Delete: Delete this entry. There will be a confirm screen to make sure user really want to delete this entry.
 - [▶]: Edit: Edit this entry.
 - [CANCEL]: to go back to the upper page.
- D.1.1.1
- | |
|----------------|
| 5021 |
| Delete? Yes No |
- Confirm the deletion of a phone book entry
 - [◀]: No action.
 - [↵]: Yes: Confirm to delete this entry.
 - [▶]: No: Back to previous menu.
 - [CANCEL]: to go back to the upper page.
- D.1.1.2
- | |
|-----------------|
| Edit Phone: 001 |
| No. Name |
- Editing to be taken on this entry. The number shows on the upper right corner is the entry number.
 - [◀]: No.: Edit the phone number.
 - [↵]: Name: Edit the name.
 - [▶]: No action.
 - [CANCEL]: to go back to the upper page.
- D.1.1.3
- | |
|----------------|
| Edit Phone No. |
| 5021 |
- Editing the phone number of this entry.
 - [◀]: Backspace.
 - [↵]: Confirm the current editing result.
 - [▶]: No action.
 - [CANCEL]: to go back to the upper page.

D.1.1.3.2

```

Edit Name:
George

```

Editing the name field of this entry.

[◀]: Backspace.

[↵]: Confirm the current editing result.

[▶]: No action.

[CANCEL]: to go back to the upper page.

[Speed Dial]

The Speed Dial allows user to invoke 10 sets of pre-defined numbers by using the programmable button (which is programmed to be perform speed dial function) and an additional 0~9 number pad key. User enters this menu to perform the speed dial number configuration.

E

```

Speed Dial: [0]
5021

```

Examine the current setting of the speed dial numbers. The upper right corner shows which number-pad key will invoke this number.

[◀]: Previous number.

[▶]: Next number.

[↵]: Edit this number.

[CANCEL]: to go back to the upper page.

E.1

```

Edit Speed: [0]
5021

```

Edit the speed dial number.

[◀]: Backspace.

[↵]: Confirm the current editing result..

[▶]: No action.

[CANCEL]: to go back to the upper page.

[TCP/IP]

The network related settings.

F

F.1

F.2

```

TCP/IP Setting:
IPTyp SMask Next

```

First menu of TCP/IP settings.

[◀]: IPTyp: Configure how the phone gets IP address.

[↵]: SMask: Configure the Sub Net Mask when using static network settings.

[▶]: Next: Next TCP/IP menu.

[CANCEL]: to go back to the upper page.

F.3

F.4

```

TCP/IP Setting:
Router DNS Next

```

Second TCP/IP menu.

[◀]: Router: Router (Default Gateway) setting when using static network settings.

[↵]: DNS: Domain Name Server setting.

[▶]: Next: Next TCP/IP menu.

[CANCEL]: to go back to the upper page.

F.5

F.6

```

TCP/IP Setting:
SNTP          Back

```

Third TCP/IP menu.

[◀]: SNTP: Timeserver setting.

[↵]: No action.

[▶]: Back: Back to previous menu.

[CANCEL]: to go back to the upper page.

- | | | |
|-----|---------------------------------|---|
| F.1 | IP Type:
< DHCP > | Select the option of how this phone gets IP address. The available options are 1.Static IP, 2.DHCP and 3.PPPoE.
[=]: <Previous option.
[↵]: Confirm this selection.
[▶]: Next option.
[CANCEL]: to go back to the upper page. |
| F.2 | TCP/IP S-MASK:
255.255.255.0 | Set the Subnet Mask for static address.
[⏪]: Backspace.
[↵]: Confirm the current editing result.
[▶]: No action.
[CANCEL]: to go back to the upper page. |
| F.3 | TCP/IP Router:
192.168.1.1 | Set the Router for static address.
[⏪]: Backspace.
[↵]: Confirm the current editing result.
[▶]: No action.
[CANCEL]: to go back to the upper page. |
| F.4 | Primary DNS IP:
168.95.1.1 | Set the Primary DNS server IP for static address.
[⏪]: Backspace.
[↵]: Confirm the current editing result.
[▶]: No action.
[CANCEL]: to go back to the upper page. |
| F.5 | TCP/IP SNTP:
140.112.2.189 | Set the SNTP server address.
[⏪]: Backspace.
[↵]: Confirm the current editing result.
[▶]: No action.
[CANCEL]: to go back to the upper page. |

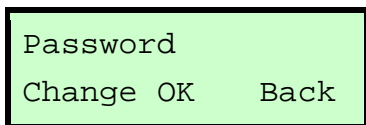
[User Administration]

User can do some administration of this phone especially his own user name and password to access this phone.

- | | | |
|-----------------|----------------------------------|---|
| G.0 | Admin Password: | Before user can entering the administration menu, he has to pass the authentication procedure by entering correct user password. The password is an alphanumeric string.
[⏪]: Backspace.
[↵]: Sending the password.
[▶]: Confirm the current character setting.
[CANCEL]: to go back to the upper page. |
| G
G.1
G.2 | User Setting:
Phone Acct Next | First Administration menu.
[⏪]: Configure the phone number of this phone.
[↵]: Acct: Configure the account of this phone.
[▶]: Next: Next User Administration menu. |

G.3 G.4	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> User Setting: Boot Home </div>	[CANCEL]: to go back to the upper page. Second Administration menu. [◀]: No action. [↵]: Boot: Reboot the phone. [▶]: Home: Back to the first Administration menu.
G.1	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> Edit Phone No: 100 </div>	[CANCEL]: to go back to the upper page. Edit the phone number of this phone. [◀]: Backspace. [↵]: Confirm the current editing result. [▶]: No action.
G.2 G.2.1	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> Account Setting: User </div>	[CANCEL]: to go back to the upper page. There is only one account to be configured. [◀]: Configure the User account [↵]: No action. [▶]: No action.
G.2.1 G.2.1.1 G.2.1.2	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> User Setting: Name Password </div>	[CANCEL]: to go back to the upper page. Configure the user account setting. [◀]: Configure the user name. [↵]: Configure the user password. [▶]: No action.
G.2.1.1	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> Name Setting: user </div>	[CANCEL]: to go back to the upper page. Edit the user name of user account. [◀]: Backspace. [↵]: Confirm the current editing result. [▶]: No action.
G.2.1.2	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> Change Password: New: </div>	[CANCEL]: to go back to the upper page. Change the password of user account. [◀]: Backspace. [↵]: Confirm the current editing result. There will be a confirm screen follow this selection. [▶]: No action.
G.2.1.3	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> Confirm Again: </div>	[CANCEL]: to go back to the upper page. Input the new password again to confirm the change. [◀]: Backspace. [↵]: Confirm the current editing result. [▶]: No action.
G.2.1.4	<div style="border: 1px solid black; background-color: #e0ffe0; padding: 5px;"> Saving... </div>	[CANCEL]: to go back to the upper page. If the new password has been confirmed, the password will save to phone. [◀]: No action [↵]: No action. [▶]: No action. [CANCEL]: No action

G.2.1.5



After saving the password this screen will show password change ok.

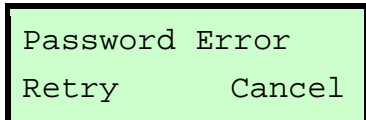
[◀]: No action

[↵]: No action.

[▶]: Back to previous menu.

[CANCEL]: No action

G.2.1.6



If the new password and confirmed password not match, this screen will show up. User can choose to retry or just cancel this operation.

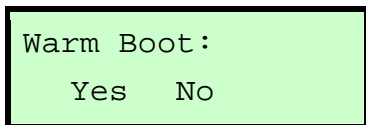
[◀]: Retry: Retry the password change again.

[↵]: No action.

[▶]: Cancel password change, back to previous menu.

[CANCEL]: No action

G.3



Confirm to reboot the system.

[◀]: Yes: Reboot the system.

[↵]: No: Not to reboot the system and back to previous menu.

[▶]: No action.

[CANCEL]: to go back to the upper page.

Chapter 8

8. The Operation of Product

8.1 Introduction and Basic Phone Setting

User may use handset and keypad to operate the IP phone. The corresponding message will be displayed on the LCD screen. And status of call(s) will be shown on the LED indicators, too. The operation of the IP phone will be described in the following distinct groups.

(i) Basic Operation

Standby Mode

Making Calls

With handset

Dial a Phone Number

Dial URL Address

With Speakerphone

Dial a Phone Number

Dial URL Address

Adjusting Voice Volume

Under idle state for Speakerphone

During a call for handset

Answering a Call By handset

(ii) The Operation Fixed Function Keys

Mute a Call

Hold a Call

Use Speakerphone

Redial a Call

(iii) To Operate Programmable Feature Buttons

Place a Hot Speed Dial

Place a Speed Dial

Place a Call Record Dial

Place a Phone Book Dial

Lock the Phone

8.1.1 To Prepare Typical IP Phones

For practicing general operation of the IP phone, please prepare several IP phones in advance. But, please consult to your Administrator or read Administrator Guide to understand how to configure the phones. Upon opening the package, each IP phone has

factory “Default” settings. However, Administrator may use a PC to browse the Web page of IP phone and do some changes accordingly. The typical settings are expected to be:

- (1) To **Enable** DHCP flag.
- (2) The IP address will be auto-assigned by DHCP server.
- (3) To give the **IP address of SIP Server**.
- (4) To give the **(Extension) phone number**.
- (5) The default order of Audio Codec is **G.711 u-law> G.711 a-law> G.723.1 (Optional)> G.729 (Optional)**.
- (6) No any server is MUST, including FTP, Backup SIP, SNTP or other Servers.

Step 1: At first, please isolate the IP phone and power it on to ready mode.

Step 2: Administrator shall press [MENUS] key and get into the [TCPIP/IPTyp] setting menu. Then select [DHCP] to enable DHCP function.

Step 3: Go to [SIP Setting] /[SIP Server]/[IP] selection to set the IP address of SIP Server. (For example, say “10.10.10.10).

Step 4: To go to [Phone]/ [Edit Phone] selection to input extension number for SIP Server. (For example, say “1005”).

Step 5: Then go to [Save] to write all settings to Flash memory and reboot the phone.

Step 6: To disconnect it and put it alone.

Step 7: Repeat the above step1 to step 7 for all IP phones. Remember to write down their IP addresses and Phone number. (For example: “192.168.100.6” for “1006” and “192.168.100.7” for “1007”).

Step 8: Finally, please connect each IP phones to the hub of office LAN. During power on, each IP phone shall get assigned IP address from the DHCP server.

[Note]: Please check if all handsets are on-hook and LCD screens show ready or not. If something wrong, please check if the network link LEDs are all ok or not.

8.1.2 Connecting To The SIP Server Before Operation

Figure 8.1 illustrates a typical example for basic VoIP application. All users will be asked to log-on to the SIP Server. With this following Figure, users may try typical operations, like:

1. To place a phone to phone direct IP address call
2. To place a phone thru SIP Server call with E.164 numbers.
3. To hold a call
4. To place a 2nd line call
5. To dial with Speed dial function
6. To dial within Call Record function

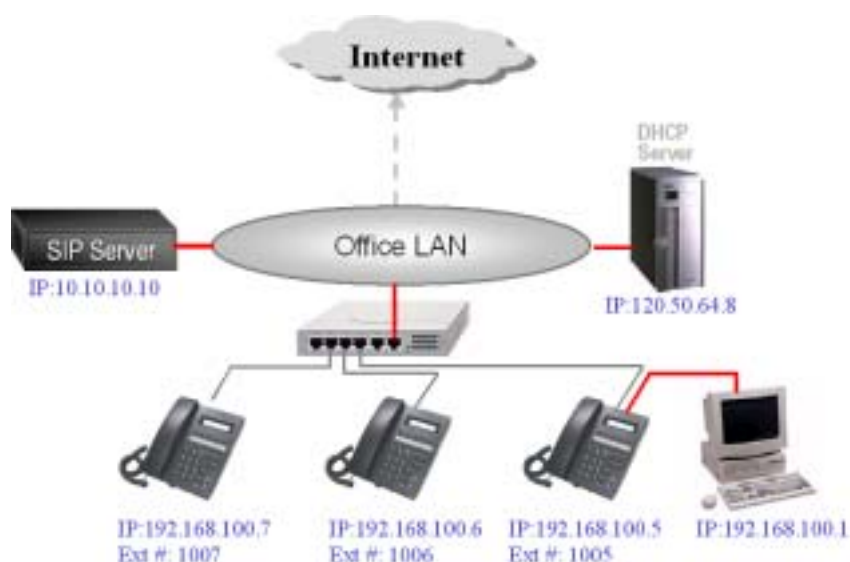


Figure 8.1 Typical Operation Example of VoIP phones

8.2 The Basic Operation With The IP Phone

The following illustrations are given for understanding each basic operation of IP phone:

[Standby Mode]

When the phone is powered on, it will show the following information on the LCD screen.

LEDs LCD Screen

```
TEL:           100
JAN 01 WED 00:00
```

Description

When the system gets ready and stays in Standby mode, the current Date and Time is displayed.

[Placing A Call using Handset]

LEDs LCD Screen

```
L1 :
```

LED2:
Off

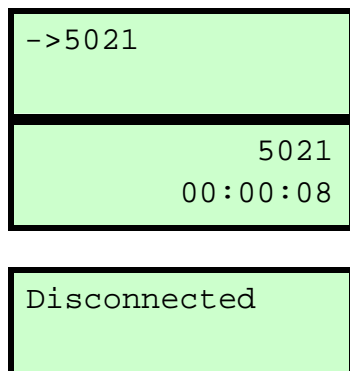
```
5021
```

Description

Upon picking up the handset, Dial tone is given to the user. The screen is cleaned up and the upper left corner will shows the current line.

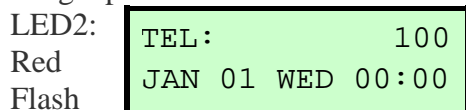
Use may now dial the number of the called party. The number you dial will shows on the upper right corner. You can use [CANCEL] as backspace to correct wrong numbers. The number of the called party might be a valid phone number register to the SIP server or might be a SIP URL, or even an IP address. For the number part, just using the keypad number keys. If for example, dialing the SIP

URL, you need some thing like '@' and '.'. During dialing phase, you can use [MENUS] key to toggle between alphabetic input and pure number input. Also, you can use '*' to represent the '.' in SIP URL or IP address. After you enter all the numbers or characters you need, press [↵] or "#" to invoke the call. If you didn't do anything after 20 seconds IP-Phone will call the current entered number automatically.



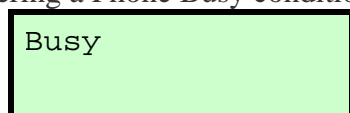
During calling phase, the screen will show the number of whom you are calling to. You should hear ringback now. After the conversation established. Your called party will show on the right side of the first line and in the second line will be the connection time. If the remote party hang up the phone, you will hear busy tone to indicate that this call has been dropped and the screen will show "Disconnected"

Hang Up the Call:



If you are using handset for this conversation, just put that handset on hook the phone will hang up this call and back to ready state. The standby mode message is displayed again.

Encountering a Phone Busy condition:



If the calling party is unavailable, you will hear a busy tone and the screen will show "Busy".

Encountering a Network /SIP Server Congestion condition:



When the SIP Server doesn't response to the call request or the IP phone already logged off the SIP Server, a busy tone will be heard. And the screen will display "Busy".

[Placing a Call using Speakerphone]

User may place a call using Speakerphone (Hand-free dialing mode). Upon pressing the [SPKR] key on the phone, the dial tone is given to the speakerphone. The procedure and the response of IP-Phone is exactly the same as using handset. Except, the input output is using the hand-free microphone and speaker on the phone. There is another difference that is if the remote party hang up the call, the phone will automatically back to standby mode. You don't have to hang up a disconnected call.

[Adjusting Voice Volume]

After pick up the handset or enter hand-free dialing by pressing the [SPKR] key you can use [◀] to decrease the volume and use [▶] to increase the volume. Also, you can use [↵] to save current volume setting as default. The volume adjustment is based on the current operation mode. If you are using handset, the handset volume is modified. If you are using speakerphone, the hand-free volume is modified. If the phone is ringing, the ring volume is modified.



VOLUME: 09

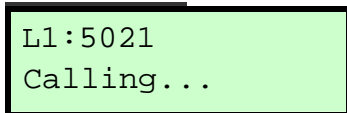
All volume control has the same interface. You will see the volume value in the first line and a horizontal bar will indicate the volume scale.

[Answering a Call]

When a call is coming, user may answer the call using handset.

LEDs LCD Screen

LED1:
Green
flash

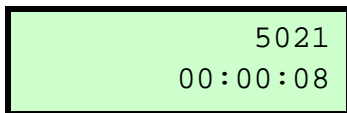


L1: 5021
Calling...

Description


When there is an incoming call present, the line ID and the phone number of the caller will be displayed.

LED2:
Off
LED1:
Off



5021
00:00:08

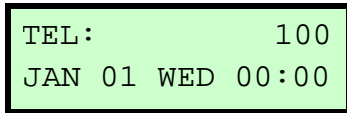
You can either pick up the handset or press the [SPKR] key to accept this call. If the caller sends his DisplayName you will see his name instead of his number.



George
00:00:08

Hang Up the Call:

LED2:
Red
Flash



TEL: 100
JAN 01 WED 00:00

If you are using handset for this conversation, just put that handset on hook the phone will hang up this call and back to ready state. The standby mode message is displayed again. If you are using speakerphone for this conversation, just press the [SPKR] key to terminate current call.

8.3 The Operation With Fixed Function Keys**[MUTE]**

Mute is used to temporarily prevent the remote party hear from you for any reason. The line is still kept there but your voice will not send to the other end of line.

<u>LEDs</u>	<u>LCD Screen</u>	<u>Description</u>
LED1: Green Flash	<div style="text-align: right;">5021</div> MUTE 00:00:08	When pressing the [MUTE] key on the phone during conversation, the voice is muted and not sending to the calling party. LED1 goes green flashing.
LED1: off	<div style="text-align: right;">5021</div> <div style="text-align: right;">00:00:08</div>	When a [MUTE] key is pressed again, the voice is recovered. The LED1 goes off.

[HOLD a Call]

Call Hold is used to hold a line and get connection to other line by dialing a new phone number or connecting to voice channel.

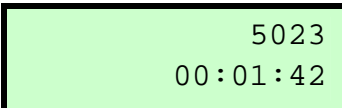
<u>LEDs</u>	<u>LCD Screen</u>	<u>Description</u>
LED1: Red Flash	L2:	During conversation, when pressing the [HOLD] key on the phone, the voice channel with current party is suspended.
LED2: Green Steady	<div style="text-align: right;">5023</div>	A dial tone will be provided to user for dial to a secondary line. Or the voice channel of the line 2 will be heard.

[Hold Lines & Switching Between Lines]

For dual line support, user may use continuous Call Hold keys to switch between 2 lines. The following scenario shows that Call Hold to switch between 2 lines (Line#1: xxx.xxx.xxx.xxx and Line#2: yyy.yyy.yyy.yyy) which have already been established

<u>LEDs</u>	<u>LCD Screen</u>	<u>Description</u>
LED2: Green Steady	<div style="text-align: right;">5023</div> <div style="text-align: right;">00:01:08</div>	Currently, Line 2 is engaging and Line 1 is held there. Note: the original is for Line 1.
LED1: Red Flash		Now, the new timer is for Line 2.
Switch back to Line 1:		
LED2: Green Flash	<div style="text-align: right;">5021</div> <div style="text-align: right;">00:02:48</div>	When pressing the [HOLD] key on the phone, the line 2 is held. Now, the phone is switched back to Line 1 for talking. Timer is for Line 1.
LED1: Off		

Switch back to Line 2:

LED2:  5023
 Green 00:01:42
 Steady
 LED1:
 Red
 Flash

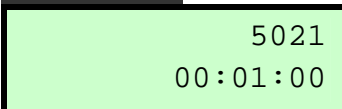
When pressing the [HOLD] key on the phone, the Line 1 is held again. Now, the phone is switched to Line 2 again for talking. Timer is for Line 2.

[XFR/CNF: Transfer a Call]

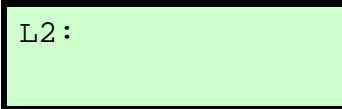
There are two different ways to transfer an existing call to third party. One is named "Attended Transfer" or "Consultation Transfer". The other is named "Blind Transfer". As the name explained, the first way will need to consult the third party, that means you have to establish a connection with the third party first, then transfer the transferor to him. The second way will do it blindly. That means you just issue the transfer without consult the third party. The following is the procedure of how to do attended transfer and blind transfer.


[Attended Transfer] (Consultation Transfer)

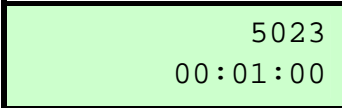
LEDs **LCD Screen**

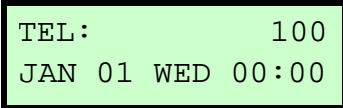
 5021
 00:01:00

LED1:
 Red
 Flash
 LED2:
 Off

 L2:

 5023

 5023
 00:01:00

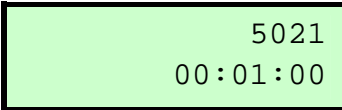
 TEL: 100
 JAN 01 WED 00:00

Description

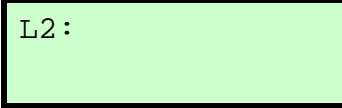
During talking with A (5021),

Press [HOLD] key to access second line (Line 2 in this case)
 Dial to B (5023)
 After call with B established, press [XFR] key To transfer A to B.
 After the call has been transferred, your phone will back to standby mode. A and B are now talking to each other.

[Blind Transfer]

 5021
 00:01:00

LED1:
 Red
 Flash
 LED2:

 L2:

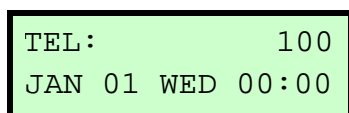
During talking with A (5021),

Press [XFR/CNF] key to access second line (Line 2 in this case)
 Dial the number of B (5023) and press '#' to send this number. A will be transferred to B.

Off



Your phone will back to standby mode. In blind transfer operation; your phone won't establish connection with the transfer target.



[Call Conference]

Our phone supports a special way to do a 3-Way conference. One of the three communication parties will act as a bridge of the voice. This phone will require establishing communication with the other two and doing the voice mixing and exchanging. **Be sure you have set one of the programmable button to perform conference function.**

[3-Way Conference]

<u>LEDs</u>	<u>LCD Screen</u>	<u>Description</u>
		Establish a call with A (5021),
LED1: Red Flash		Press [HOLD] key to access second line (Line 2 in this case) Dial to B (5023)
LED2: Off		After call with B established, press programmable button which is programmed to perform conference function.
		Conference Break: If any one of these 3 parties hangs up the phone, the other two will still keep conversation. For example, A leave the conference by hanging up the phone, you will talk to B at line 2 only.

[REDIAL a Call]

There is a set of dialed (outgoing) number stored in the memory. When this [REDIAL] key is pressed, the system will place a call using the stored dialed phone number.

<u>LEDs</u>	<u>LCD Screen</u>	<u>Description</u>
-------------	-------------------	--------------------

Press Redial key, the IP phone will send the last dialed number directly. If the phone haven't pickup yet, it will automatically pickup as hand-free mode and dial directly.

[Quick dial]

23

Whenever you saw a “Q-Dial” when dialing numbers, you can press [↵] to retrieve the phone book entry (Using the dialed number as index). In this example, you will invoke the phone book entry number 23. If the number of phone book entry 23 is 26551000 it will be shown on the screen. You can press [↵] or [#] to dial this number.

[Special Phonebook Dial through Web Page]

User can dial the number stored in the phonebook through web page. Refer to the Chapter 9 phonebook setting page for the detail operation.

8.4 The Operation With Programmable Feature Buttons

After a phone feature is enabled and assigned to a specific Programmable button [Pn], user may directly touch it to launch the feature. In general, the user can launch it without hanging up the phone.

[Speed Dial]

You may get in the Menus mode to check out the Speeddial setting:

For example:

Programmable Button: set at [P1]

Feature: set as Speed Dial

Speed Dial 0: set as 1006

Speed Dial 1: set as 1007

Speed Dial 2: set as 5023

Testing: directly press [P1] button, then dial a number “2”. The speed dial number mapped to keypad 2 (5023) will be invoked just as you dial it from keypad. Then press [↵] or #: to call the number.

[Call Record]

If the programmable button is programmed to perform “Call Record” function, it will be a short cut to get into the B.1 section of the menu operation. You can examine 3 different

types of call history and delete some record and even dial directly using the number in call record.

[Phone Book]

```
Phone Book:  
Goto[1-100]:
```

If the programmable button is programmed to perform "Phone Book" function, it will be a short cut to get into the D.1 section of the menu operation. You can examine 100 sets of phone book entries and make modification and invoke number to dial from phone book.

[Call Forward]

```
Call Forward:  
Always Busy NoAns
```

If the programmable button is programmed to perform "Call Forward" function, it will be a short cut to get into the 8.1 section of the menu operation. You can maintain the call-forward function settings of the phone.

[Conference]

```
3-Way Conference  
00:03:40
```

As described in the section 8.3. The programmable button will establish a 3-way conference through a special operation.

[Lock IP-Phone]

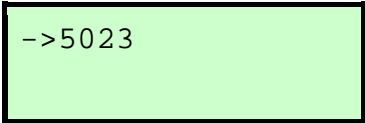
```
Phone Locked...  
Pin:
```

Lock the IP-Phone immediately.

[Use DND]

```
DND: 100  
JAN 01 WED 00:00
```

Turn on/off the DND function. If DND is enabled, the upper left corner of the phone will show "DND:"

[HOT Speeddial]

->5023

If the programmable button is programmed to perform “HOT Speeddial” function, the button will map to the speed dial number according to its position. P1 will map to speed dial 0, P2 will map to speed dial 1.... When you press the programmable button (say P1) the speed dial number 0 will be invoked and dial directly.

Chapter 9

9. Configure the phone through web page

Accessing the phone through web browser, just simply enter the “http://192.168.1.10” in the location field of the browser. (If you are not sure about the IP address, you can examine the current IP address through Info menu.)

Login:

The following dialog box will pop up and prompt you to provide the user name and password in order to prevent unauthorized user access the phone.



Default Page:

The following is the default page you will see when you login the phone's web page.

SIP Phone IP Phone

Phone Personal PhoneBook TCP/IP System Reboot

Version: 1.0.08

Phone Settings			
Handset Mic	7	Handset Speaker	8
Speaker Mic	7	Speakerphone	12
Ring Tone Volume	13	Side Tone Volume	2
Enable Echo Canceller	<input checked="" type="checkbox"/>	Enable VAD+CNG	<input type="checkbox"/>
Tone Type	User Defined	Ring Type	User Defined
DTMF Relay	RFC2833+Tone	Enable DND	<input type="checkbox"/>
Enable Auto Answer	<input type="checkbox"/>	Enable Phone Lock	<input type="checkbox"/>
Programmable Button Settings			
Programmable Button 1	Speed Dial		
Programmable Button 2	Speed Dial		
Programmable Button 3	Speed Dial		
Programmable Button 4	Speed Dial		
Programmable Button 5	Speed Dial		

User's Menu:

The upper area of the web page is the menu of all configuration categories. You can click on the menu items to select different group of settings.

SIP Phone IP Phone

Phone Personal PhoneBook TCP/IP System Reboot

Version: 1.0.08

When mouse pointer move over the menu item, the menu item will highlighted and the cursor will change to a hand indicating this item can be clicked.

SIP Phone IP Phone

Phone Personal PhoneBook TCP/IP System Reboot

Version: 1.0.08

When you click on the menu item, the web page will change to different setting page.

Settings Pages:

There are 6 different settings pages.

Phone:

Phone Settings			
Handset Mic	7	Handset Speaker	8
Speaker Mic	7	Speakerphone	12
Ring Tone Volume	13	Side Tone Volume	2
Enable Echo Canceller	<input checked="" type="checkbox"/>	Enable VAD+CNG	<input type="checkbox"/>
Tone Type	User Defined	Ring Type	User Defined
DTMF Relay	RFC2833+Tone	Enable DND	<input type="checkbox"/>
Enable Auto Answer	<input type="checkbox"/>	Enable Phone Lock	<input type="checkbox"/>
Programmable Button Settings			
Programmable Button 1	Speed Dial		
Programmable Button 2	Speed Dial		
Programmable Button 3	Speed Dial		
Programmable Button 4	Speed Dial		
Programmable Button 5	Speed Dial		

Here is the description of each field.

Field Name	Function
Handset Mic	Set the input level of handset microphone.
Handset Speaker	Set the output level of handset speaker.
Speaker Mic	Set the input level of hand-free microphone.
Speakerphone	Set the output level of hand-free speaker.
Ring Tone Volume	Set the output level of ring.
Side Tone Volume	Set the output level of side tone.
Tone Type	Select the tone type.
Ring Type	Select the ring type.
DTMF Relay	Select the way to send DTMF through in-band or various out band mechanism.
Enable DND	Turn on DND function.
Enable Auto Answer	Turn on auto answer function.
Enable Phone Lock	Lock the phone.
Programmable button	Select the function of 5 programmable buttons.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.
Logout	Logout and close the browser window.

Personal:

Call Forward Settings			
Enable Always Forward	<input type="checkbox"/>	Forward to user@[host]	<input type="text"/>
Enable On Busy Forward	<input type="checkbox"/>	Forward to user@[host]	<input type="text"/>
Enable No Answer Forward	<input type="checkbox"/>	Forward to user@[host]	<input type="text"/>
No Answer Timeout(sec)	<input type="text" value="60"/>		
Speed Dial Entry Settings			
Speed Dial Entry 0	<input type="text" value="5021"/>	Speed Dial Entry 1	<input type="text" value="5023"/>
Speed Dial Entry 2	<input type="text"/>	Speed Dial Entry 3	<input type="text"/>
Speed Dial Entry 4	<input type="text"/>	Speed Dial Entry 5	<input type="text"/>
Speed Dial Entry 6	<input type="text"/>	Speed Dial Entry 7	<input type="text"/>
Speed Dial Entry 8	<input type="text"/>	Speed Dial Entry 9	<input type="text"/>
Caller Blocking Settings			
Caller Blocking Entry 0	<input type="text"/>	Caller Blocking Entry 1	<input type="text"/>
Caller Blocking Entry 2	<input type="text"/>	Caller Blocking Entry 3	<input type="text"/>
Caller Blocking Entry 4	<input type="text"/>	Caller Blocking Entry 5	<input type="text"/>
Caller Blocking Entry 6	<input type="text"/>	Caller Blocking Entry 7	<input type="text"/>
Caller Blocking Entry 8	<input type="text"/>	Caller Blocking Entry 9	<input type="text"/>

Here is the description of each field.

Field Name	Function
Enable Always Forward & Forward to user@[host]	Turn on unconditional forward, the right hand side is where the will be forwarded to.
Enable On Busy Forward & Forward to user@[host]	Turn on the forward on busy function.
Enable No Answer Forward & Forward to user@[host]	Turn on the forward on no answer function.
No Answer Timeout (sec)	Set the timeout for forward on no answer.
Speed Dial Entry	10 Sets of speed dial entries. Each entry map to the 10 keypad number.
Caller Blocking Entry	10 Sets of caller blocking settings. User can set 10 numbers to reject their call automatically.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.
Logout	Logout and close the browser window.

PhoneBook:

Phone Book VIP(001-050)			
Dial Name	Phone number	Dial Name	Phone number
001	5021	002	
003	5023	004	
005		006	
037		038	
039		040	
041		042	
043		044	
045		046	
047		048	
049		050	

(Part of the page is not shown here.)

Save Setting Next Page Logout

Here is the description of each field.

Field Name	Function
Name	The name of the entry.
Phone Number	The number to call, it could be number, IP address or SIP URL.

Buttons:

Save Settings	Save changes in this page to the phone.
Next Page	Go to examine next 50 entries. (This button is in first page)
Prev Page	Go to examine previous 50 entries. (This button is in second page)
Logout	Logout and close the browser window.

Special Phonebook dial: You can click on the index of the phonebook entries to invoke a phone book dial. When the mouse pointer moves over the index, it will be heighten. (Change to white word with blue background) By clicking the index (For example, index 003), the following dialog box will pop up. (If the number is not empty)



By clicking the [Dial] button, the phone will call 5023 as requested.

TCPIP:

TCP/IP Settings			
MAC	00:03:09:30:43:E8	Current IP	192.168.5.22
IP Type	Static IP ▼	IP Address	192.168.5.22
Subnet Mask	255.255.255.0	Default Gateway	192.168.5.1
Primary DNS	168.95.1.1	Secondary DNS	63.250.206.138
PPPoE User Name		PPPoE Password	
PPPoE Reconnect	<input type="checkbox"/>		
Time Server	140.112.2.189	Daylight Saving	<input type="checkbox"/>
Time Zone	GMT (Greenwich Mean Time, London, ...) ▼		

Here is the description of each field.

Field Name	Function
Mac	Show the current MAC address.
Current IP	Show the current IP address.
IP Type	Select how this phone gets IP address. Use DHCP, static IP or PPPoE.
IP Address	Static IP address.
Subnet Mask	Static Subnet Mask.
Default Gateway	Static Default Gateway.
Primary DNS	Static Primary DNS.
Secondary DNS	Static Secondary DNS.
PPPoE User Name	User name for PPPoE session.
PPPoE Password	Password for PPPoE session.
PPPoE Reconnect	Reconnect when PPPoE session disconnected.
Time Server	NTP server address.
Daylight Saving	Using daylight saving time.
Time Zone	Time Zone.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.
Logout	Logout and close the browser window.

System:

Account Settings			
User Name	<input type="text" value="user"/>	User Password	<input type="password" value="****"/>
<input type="button" value="Save settings"/>		<input type="button" value="Logout"/>	

Here is the description of each field.

Field Name	Function
User Name	User's account name for accessing web page.
User Password	User's password for accessing web page and the admin section in the menu.

Buttons:

Save Settings	Save changes in this page to the phone.
Cancel	Discard all changes in this page.
Logout	Logout and close the browser window.

Reboot:**System Reboot**

- Please Read:**
1. Please make sure all settings are correct before rebooting.
 2. When enabling DHCP or PPPoE, please remember that the current IP address will be invalid. You may use the [MENUS] option on the device to see the current IP address of the IP2006.
 3. Rebooting may take 30 seconds to one minute.

A rectangular button with a light blue background and a thin border, containing the word "Reboot" in a simple, sans-serif font.

Issue a software reboot.

Conclusion

This IP phone is a feature-rich product. More functions and applications could be upgraded in the future when a new firmware is released. For advanced settings and operations, please consult to your Administrator or the Administrator Guide, Application Reference Book or your supplier for assistance. Finally, we would like to express our much appreciation again for your using this product and cooperation.

Appendix A

Appendix A World GMT Time Zone Table

The Greenwich meridian Time is the standard for worldwide area and countries. The zone code in the format of “±hh:m (hh:00~12, m:0 or 5,). It is listed by alphabet of each country.

Country	GMT	Country	GMT	Country	GMT	Country	GMT
A		E		L		S	
Afghanistan	+ 4.5	Ecuador	- 5.0	Laos	+ 7.0	Saba	- 4.0
Albania	+ 1.0	Egypt	+ 2.0	Latvia	+ 2.0	Samoa	- 11.0
Algeria	+ 1.0	El Salvador	- 6.0	Lebanon	+ 2.0	San Marino	+ 1.0
American Samoa	- 11.0	Equatorial Guinea	+ 1.0	Lesotho	+ 2.0	Sao Tome	+ 0.0
Andorra	+ 1.0	Eritrea	+ 3.0	Liberia	+ 0.0	Saudi Arabia	+ 3.0
Angola	+ 1.0	Estonia	+ 2.0	Libya	+ 2.0	Senegal	+ 0.0
Antarctica	- 2.0	Ethiopia	+ 3.0	Liechtenstein	+ 1.0	Seychelles Islands	+ 4.0
Antigua and Barbuda	- 4.0	F		Lithuania	+ 2.0	Sierra Leone	+ 0.0
Argentina	- 3.0	Faeroe Islands	+ 0.0	Luxembourg	+ 1.0	Singapore	+ 8.0
Armenia	+ 4.0	Falkland Islands	- 4.0	M		Slovakia	+ 1.0
Aruba	- 4.0	Fiji Islands	+ 12.0	Macedonia	+ 1.0	Slovenia	+ 1.0
Ascension	+ 0.0	Finland	+ 2.0	Madagascar	+ 3.0	Solomon Islands	+ 11.0
Australia North	+ 9.5	France	+ 1.0	Malawi	+ 2.0	Somalia	+ 3.0
Australia South	+ 10.0	French Antilles (Martinique)	- 3.0	Malaysia	+ 8.0	South Africa	+ 2.0
Australia West	+ 8.0	French Guinea	- 3.0	Maldives	+ 5.0	Spain	+ 1.0
Australia East	+ 10.0	French Polynesia	- 10.0	Mali Republic	+ 0.0	Sri Lanka	+ 5.5
Austria	+ 1.0	G		Malta	+ 1.0	St Lucia	- 4.0
Azerbaijan	+ 3.0	Gabon Republic	+ 1.0	Marshall Islands	+ 12.0	St Maarten	- 4.0
B		Gambia	+ 0.0	Mauritania	+ 0.0	St Pierre & Miquelon	- 3.0
Bahamas	- 5.0	Georgia	+ 4.0	Mauritius	+ 4.0	St Thomas	- 4.0
Bahrain	+ 3.0	Germany	+ 1.0	Mayotte	+ 3.0	St Vincent	- 4.0
Bangladesh	+ 6.0	Ghana	+ 0.0	Mexico Central	- 6.0	Sudan	+ 2.0
Barbados	- 4.0	Gibraltar	+ 1.0	Mexico East	- 5.0	Suriname	- 3.0
Belarus	+ 2.0	Greece	+ 2.0	Mexico West	- 7.0	Swaziland	+ 2.0
Belgium	+ 1.0	Greenland	- 3.0	Moldova	+ 2.0	Sweden	+ 1.0
Belize	- 6.0	Grenada	- 4.0	Monaco	+ 1.0	Switzerland	+ 1.0
Benin	+ 1.0	Guadeloupe	- 4.0	Mongolia	+ 8.0	Syria	+ 2.0
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Country	GMT	Country	GMT	Country	GMT	Country	GMT
Bermuda	- 4.0	Guam	+ 10.0	Morocco	+ 0.0	T	
Bhutan	+ 6.0	Guatemala	- 6.0	Mozambique	+ 2.0	Taiwan	+ 8.0
Bolivia	- 4.0	Guinea-Bissau	+ 0.0	Myanmar	+ 6.5	Tajikistan	+ 6.0
Bosnia erzegovina	+ 1.0	Guinea	+ 0.0	N		Tanzania	+ 3.0
Botswana	+ 2.0	Guyana	- 3.0	Namibia	+ 1.0	Thailand	+ 7.0
Brazil West	- 4.0	H		Nauru	+ 12.0	Togo	+ 0.0
Brazil East	- 3.0	Haiti	- 5.0	Nepal	+ 5.5	Tonga Islands	+ 13.0
British Virgin Islands	- 4.0	Honduras	- 6.0	Netherlands	+ 1.0	Trinidad and Tobago	- 4.0
Brunei	+ 8.0	Hong Kong	+ 8.0	Netherlands Antilles	- 4.0	Tunisia	+ 1.0
Bulgaria	+ 2.0	Hungary	+ 1.0	New Caledonia	+ 11.0	Turkey	+ 2.0
Burkina Faso	+ 0.0	I		New Zealand	+ 12.0	Turkmenistan	+ 5.0
Burundi	+ 2.0	Iceland	+ 0.0	Nicaragua	- 6.0	Turks and Caicos	- 5.0
C		India	+ 5.5	Nigeria	+ 1.0	Tuvalu	+ 12.0
Cambodia	+ 7.0	Indonesia Central	+ 8.0	Niger Republic	+ 1.0	U	
Cameroon	+ 1.0	Indonesia East	+ 9.0	Norfolk Island	+ 11.5	Uganda	+ 3.0
Canada Central	- 6.0	Indonesia West	+ 7.0	Norway	+ 1.0	Ukraine	+ 2.0
Canada Eastern	- 5.0	Iran	+ 3.5	O		United Arab Emirates	+ 4.0
Canada Mountain	- 7.0	Iraq	+ 3.0	Oman	+ 4.0	United Kingdom	+ 0.0
Canada Pacific	- 8.0	Ireland	+ 0.0	P		Uruguay	- 3.0
Canada Newfoundland	- 3.5	Israel	+ 2.0	Pakistan	+ 5.0	USA Central	- 6.0
Cape Verde	- 1.0	Italy	+ 1.0	Palau	+ 9.0	USA Eastern	- 5.0
Cayman Islands	- 5.0	J		Panama, Republic Of	- 5.0	USA Mountain	- 7.0
Central African Rep	+ 1.0	Jamaica	- 5.0	Papua New Guinea	+ 10.0	USA Pacific	- 8.0
Chad Rep	+ 1.0	Japan	+ 9.0	Paraguay	- 4.0	USA Alaska	- 9.0
Chile	- 4.0	Jordan	+ 2.0	Peru	- 5.0	USA awaii	- 10.0
China	+ 8.0	K		Philippines	+ 8.0	Uzbekistan	+ 5.0
Christmas Is.	- 10.0	Kazakhstan	+ 6.0	Poland	+ 1.0	V	
Colombia	- 5.0	Kenya	+ 3.0	Portugal	+ 1.0	Vanuatu	+ 11.0
Congo	+ 1.0	Kiribati	+ 12.0	Puerto Rico	- 4.0	Vatican City	+ 1.0
Cook Is.	- 10.0	Korea, North	+ 9.0	Q		Venezuela	- 4.0
Costa Rica	- 6.0	Korea, South	+ 9.0	Qatar	+ 3.0	Vietnam	+ 7.0
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Country	GMT	Country	GMT	Country	GMT	Country	GMT	
Croatia	+ 1.0	Kuwait	+ 3.0	R		W		
Cuba	- 5.0	Kyrgyzstan	+ 5.0	Reunion Island	+ 4.0	Wallis And Futuna Islands	+ 12.0	
Cyprus	+ 2.0	↶ next column		Romania	+ 2.0	Y		
Czech Republic	+ 1.0			Russia West	+ 2.0	Yemen	+ 3.0	
D				Russia Central 1	+ 4.0	Yugoslavia	+ 1.0	
Denmark	+ 1.0			Russia Central 2	+ 7.0	Z		
Djibouti	+ 3.0			Russia East	+ 11.0	Zaire	+ 2.0	
Dominica	- 4.0			Rwanda	+ 2.0	Zambia	+ 2.0	
Dominican Republic	- 4.0			↶ next column		Zimbabwe	+ 2.0	
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