IP Phone

User Manual

IP0120 (SIP)

IP0121 (H.323)

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1 Overview

1.1 Introduction

VanAccess believes that the **next generation networks** based on VoIP technologies will change the way people communicate with each other and people will benefit a lot from the technologies. The usage of the technologies should be simple to most users, the users should only find the benefits without facing the technology barriers. So, our mission is to make the usage of VoIP technologies just as simple as the usage of normal telephone set. “Simple is the best” is our goal in developing the IP Phone products.

The Ethernet Phone IP0120/IP0121 is a next generation IP Phone that provides a cost-saving solution for small business/home users on their telecommunication needs. The IP0120/IP0121 is following the open standard SIP/H.323 protocol to make sure that the user can easily install this IP phone on most of the existing VoIP(Voice over IP) services. By using the state-of-the-art DSP(Digital Signal Processing) technology, the IP0120/IP0121 delivers outstanding voice quality that is very comparable to the PSTN voice quality.

With the built in LCD display, the user can easily configure the IP0120/IP0121 for first time installation in a few minutes. And besides the advanced VoIP functions in easy installation, IP0120/IP0121 also provides rich telephone features such as the last number redial, speed dial, phone book, call forward/transfer, call history, volume adjustment and speakerphone. IP0120/IP0121 is your best VoIP solution in the new generation communication.

The differences between the different IP Phone models are listed in the following table:

	Ethernet Interfaces	SIP	H.323
IP0120	O(WAN+PC)	O	X
IP0121	O(WAN+PC)	X	O

Table 1. Models table

1.2 Key Features

- Following H.323 version 4 standard (IP0121) :
  - Support Fast start, Tunneling and H.245 DTMF relay.
- Following RFC-3261 SIP standard (IP0120) :
  - Support password authentication using MD5 digest and RFC-2833 for DTMF relay.
- Dynamic IP support (DHCP and PPPoE) :
  - Getting IP from DHCP server using DHCP protocol or through ADSL modem using PPPoE protocol, automatically reconnect when PPPoE lost connection.
- Passing through NAT devices:
  - Can make outgoing and incoming calls under any NAT devices (even under two layer NAT devices) when working with the specific gatekeeper/proxy devices.
- Remote software upgrade capability (via ftp):
  - FTP protocol provides reliable remote upgrade through Internet.
- Advanced Digital Signal Processing (DSP) technology to ensure superior audio quality:
  - Hardware System on a Chip solution with built in DSP processor makes sure the perfect voice quality.
- Support G.723.1, G.729A/B, G.711 (A-law/U-law) voice codecs :
  - Following ITU-T standard to support best compatibility.
- Support supplementary services, including immediate (unconditional) call forwarding, busy call forwarding, no answer call forwarding and call hold/transferring.
- Provide call history:
  - Record incoming call history, outgoing call history, missed (not accepted) call history, and let users make direct call from the call history.
- Phone Book : 50 sets
- Speed dial : 10 sets

- Support Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation) :
    - Silence suppression can save about half of the network bandwidth needed during normal VoIP conversation.
- Ping function supported :
    - Ping other device in the Internet from IP0120/IP0121 to make sure the Internet connection is ok.
- System status display on the LCD panel :
    - User can easily know if the IP0120/IP0121 is working normally and monitor the system status(network status, registering status) from the LCD panel display.
    - A “PKT Trace” function is supported to display the packets received on the LCD panel to let administrator find the network problem on line.
- Call with or without gatekeeper / proxy(direct IP dialing) :
    - Following standard SIP / H.323 protocol and is compatible with most of existing SIP proxy / H.323 gatekeeper.
- Provide easy configuration methods:
    - Very easy setting by keypads on the phone set.
    - Setting by web browser.
- Support RFC-3261, H.323, TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, FTP, PPP, PPPoE protocols.
- Interoperable with most of the existing SIP/H.323 VoIP devices (IP-phone, gateway, gatekeeper, proxy, soft-switch, IP-PBX), including Microsoft NetMeeting, Cisco gateways / gatekeepers:
    - Please refer to the section 6.2/6.3 Interoperability List for the complete listing.
- The WAN Port automatically works for paralleled Ethernet cable and crossed Ethernet cable.

1.3 Hardware Specification

HARDWARE SPEC	
Spec\Model	**IP0120 / IP0121**
PC Port	1xRJ45 10/100 Base-T Ethernet, line auto-sensing/switching.
WAN Port	1xRJ45 10/100 Base-T Ethernet, line auto-sensing/switching.
	(Option)Power Over Ethernet 802.3af function.
LCD display	2x16 characters
Phone Case	36-button keypad
Universal Switching Power Adaptor	Input: 100-240V AC
	Output: +7V DC, 800mA
Speaker	8 Ohm/0.2 Watt speaker for speakerphone operation
Dimension	19cm(W)x23cm(D)x9cm(H)
Weight	870 g
Operating Temperature	32 - 104°F (0 – 40°C)
Humidity	10%-95% (non-condensing)
EMI Compliance	UL/EN/FCC Class B

Table 2. Hardware Specification of IP0121 and IP0120

2 Basic Installation

2.1 Appearance Introduction

2.1.1 Key parts of the IP0120/IP0121

The key parts of the IP0120/IP0121 series IP Phone are shown in figure 1, including the followings :

1. 2x16 LCD Display
2. Keypad
3. Indicator of usage
4. User Manual
5. Handset
6. Ethernet Cable
7. Power Adaptor

Figure 1. Key parts of the IP0120/IP0121 IP Phone

2.1.2 Rear and Back side Panel of the IP0120/IP0121

The rear and back side panel illustration is shown in figure 2 and 3, main parts include:

1. RJ-45 Ethernet Port
2. RJ-45 Ethernet Port
3. Power Adaptor Jack
4. Jack of line to handset

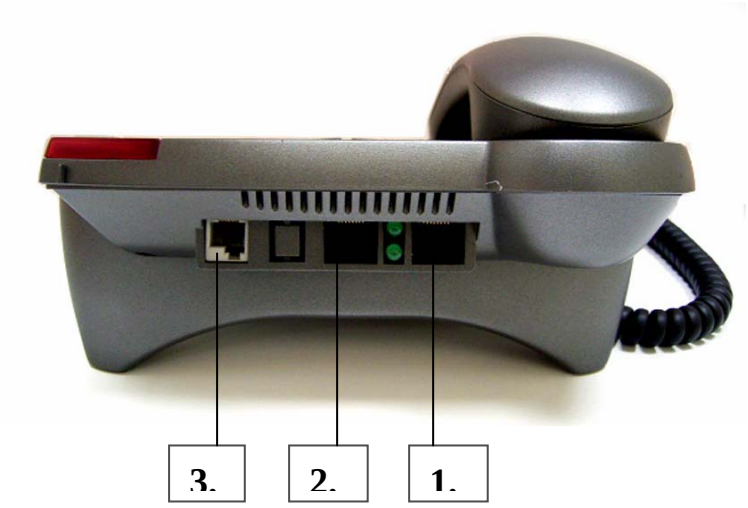


Figure 2. Rear Panel of the IP Phone

Figure 3. Back Side Panel of the IP Phone

2.1.3 Keypad Definition

Figure 3. Keypad illustration

1. **Light**
   The red light goes on-off when there is an incoming call.

2. **LCD Display**
   Menu and all status shall be displayed for users.

3. **"🡐 UP" Key**
   When the IP phone is entered into the menu selection, this key is used to scroll up the menu items.

   And when the IP phone is editing some menu item's contents, this key is used as " left delete" to delete a digit per each key press.

   When the IP phone is on dial mode, the "🡐" key is used as "delete" key.

"→ DOWN" Key

When the IP phone is entered into the menu selection, this key is used to scroll down the menu items.

And when the IP phone is editing some menu item's contents, this key is used as " right shift" to shift the cursor right a digit per each key press.

4. - IN VOL+

When the IP phone is at idle state or talking on handset or speaker, this key is used to increase/decrease the volume of the voice sound.

The volume of speaker, handset and ring are separately adjusted according to the mode of current usage. When in idle, "+" key increase the volume of ring tone, "-" key decrease the volume of ring tone; in hand-free mode, "+" key increase the volume of speaker, "-" key decrease the volume of speaker; when in handset mode, "+" key increase the volume of handset, "-" key decrease the volume of handset.

5. +OUT VOL-

Users are able to increase and/or decrease the volume which they send out to remote party.

6. SPEAKER

This key is pressed to switch between the usage of the handset and the speaker devices.

7. HOLD

To hold the conversation.

This key can also be pressed to do consultant-transfer of an active call to another IP phone. When the IP phone is active(incoming call answered or outgoing call accepted), by pressing this key, a dial tone will be heard, then the user can key in another IP phone's number to call to another party and have a conversation with him. Then, by pressing "TRANSFER" key, the call will be transferred to this new party.

8. NET

When users are not successfully registered to their service provider, this

button light will turn on-off. User can press this button to try to register again to their service provider.

9. MESSAGE

This is for future use of functions upgrade.

10. Menu / OK

When the IP phone is at idle state, this key is used as “Menu” key to bring out the menu selection on the LCD display.

When inside the menu selection/setting on the LCD display, this key is used as “OK” key to enter into a lower layer of menu selection or to accept the edited item’s contents.

When the IP phone is on dial mode, the “OK/Menu” key is used as “Dial Out” key.

11. Cancel

When the IP phone is entered into the menu selection, this key is used to escape to an upper layer of the menu selection.

And when the IP phone is editing some menu item’s contents, this key is used to cancel current edit and escape to an upper layer of the menu selection.

12. TXT ⟷ NUM

When users need to press information in characters, press this button and alphabets shown on the keypads will be displayed.

13. TRANSFER

This key is pressed to transfer an active call to another IP phone. When the IP phone is active(incoming call answered or outgoing call accepted), by pressing this key, a dial tone will be heard, then the user can key in another IP phone’s number to transfer the call to another party. If the call transfer is successful, a busy tone will appear to notice the user to on-hook the phone. If the call is failed, the user can press this “Transfer” key again to retrieve back this original call connection.

When a call is incoming and this IP phone is ringing, by pressing this "Transfer" key and then another IP phone's number can transfer the call immediately to another party without answering the call.

14. Redial / Dial

When the IP phone is off-hooked and dialed some number, this key is pressed to call out. The **"#"** key has this same function to send out the called number.

When the IP phone is off-hooked and this key is pressed immediately, the last dialed number will be called out right away.

15. Speed Dial M1 – M10

Users are able to store 10 specific phone numbers in the slots of M1 - M10. Users are able to make a speed dial call to the specific party by pressing the speed dial key from M1 – M10.

16. Phone Book

Users are able to store up to 50 phone numbers by pressing the "PhoneBook" button. For each item of the 50 phone book numbers, the user can store both the number and the name for display.

2.2 LCD Menu List

The following is the roadmap of the menu on your IP phone.
SIP & H.323 share similar menu, different parts are written separately.

- View
  - Network Value
    - IP Address
    - Network Mask
    - Default Route
    - DNS Server
  - Ping
  - Quota
  - Restart
  - Image Version
  - (Yes / No) PKT Trace

- Configure (Password: 135)
  - Network
    - (Yes / No) Dynamic IP
    - (Yes / No) PPPoE
      - If yes:
        - PPPoE User Name
        - PPPoE Password
    - Static IP
      - IP Address
      - Network Mask
      - Default Route
      - DNS Server
    - Time Zone

  - H323
    - Numfber
    - Password
    - H.323 ID
    - (Yes / No) Reg to GK
    - (Yes / No) RTPLowBW

- SIP
  - Number
  - Password
  - Login ID
  - (Yes / No) Proxy On
    - If Yes:
      - Proxy Address
      - Proxy Port
  - (Yes / No) Outbound Proxy
    - If Yes:
      - Outbound Proxy IP
      - Outbound Proxy Port
  - (Yes / No) RTPLowBW

- Forward Mode
  - (Yes / No) Immediate
    - If Yes:
      - Immediate Number
  - (Yes / No) Busy
    - If Yes:
      - Busy Number
  - (Yes / No) No Answer
    - If Yes:
      - No Answer Number
      - No Answer Time

◎ Advanced

- System
  - DSP Version
  - Upgrade / DnLoad
    - FTP Server IP
    - Image File Name
    - Upgrade Image
    - Upgrade Loader
    - Config Profile
  - Debut Mode
    - Dump Address
    - Dump Size
    - Dump!

- Network
  - MAC Address
  - NTP Server
- Restart Count
- RTP Process
- (Yes / No) Bypass Server
- (Yes / No) Jitter Buffer
- (Yes / No) Auto Upgrade

- Phone Advanced
  - Codec
    - G.711u
    - G.711a
    - G.729
    - G.723
  - Voice
    - (Yes / No) VAD
    - BG Noise Level
  - Volume
    - Ring Volume
    - Handset Volume
    - Hand free Volume
    - Codec Tx Gain
  - Scrn Con (0-9)
  - Ring Type (1-10)
  - UI Mode
    - Console
    - Lcd
    - Both

- SIP Advanced
  - Protocol
    - Min Media Port
    - Max Media Port
    - Reg From
    - Reg To
    - Reg Expire
    - Reg Action
  - Local Port
  - User Setting

- Platform
- Billing Server

- Login
  - User Name
  - Password
  - Confirm Password
- Statistics
  - User Statistics
    - User Statistic
    - Phone Stastic
  - Phone Statistics
    - Call Missed
    - Call Received
    - Call Dialed
- Additment
  - International
  - My Country Code
  - Area Prefix Code

2.3 Installation Environment

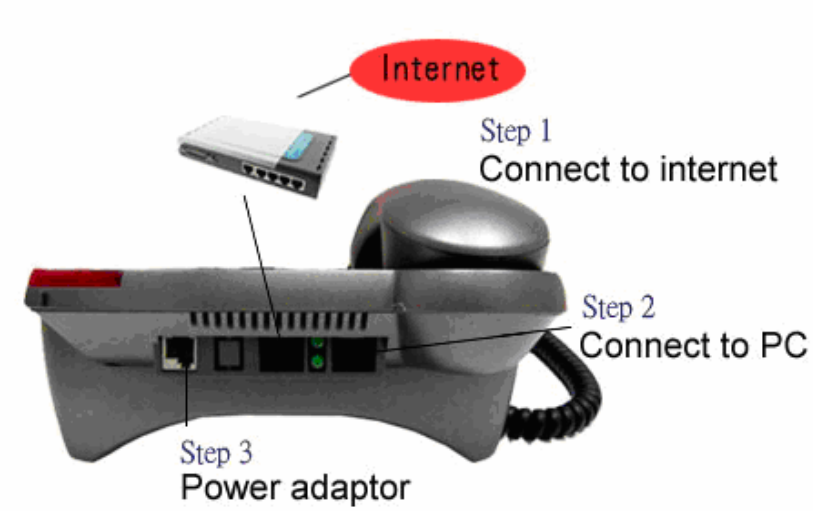


Figure 4. Installation Environment

Step 1：

Please take out your IP phone and its handset. Then connect your handset to IP phone by plugging the left jack of handset. If you use RJ-45 broadband connection, please connect your network cable from your broadband modem to WAN port of your IP phone, such as NAT、HUB、ADSL、CABLE.

Step 2：

If you would like to have PC online at the same time, please connect this port to your PC.

Step 3：

Please plug in your power adaptor to your IP phone and power source. LCD of your IP phone will display “Starting…….” and then “SIP / Hi xxx” menu within approximately 4 seconds.

3 Configuration from Keypads

The IP0120/IP0121 IP Phone is designed to be installed and used very friendly and easily, so, almost all the configurations can be done through the keypads and LCD screen display on the phone set in a few minutes. In order to make a VoIP call, please do the configurations through keypads as described in the following few sections.

Notice:

(1). *When need to input an English character in any menu item, please press that key button quickly to switch between different characters to set the correct one needed.*

(2). *When the input mode is in digit mode(only can input '0'~'9' and '*','#'), if want to input English character, please press "***TXT ⟷ NUM***" key first to toggle to "character" input mode.*

3.1 Network Configuration

The first thing in using the IP0120/IP0121 IP Phone is to set the network configuration to let the IP Phone connect to internet. Depends on your network environment and phone models, please use the proper method to configure the IP Phone to connect to Internet.

3.1.1 Dynamic IP Method (DHCP)

Most of the network environment at office or at hotel room or at home is under a NAT (IP sharing device)/router device, under this environment, the easiest way to connect to internet is using the DHCP method. The IP Phone when configured using the following method, it will get the IP parameters dynamically, and connect to internet automatically.

Configure ► Password：135 ► Network ► (Yes/No) Dynamic IP

Please press "ok" key to set Yes on (Yes / No) Dynamic IP when using DHCP method.

Most of the cable modem connections also use this DHCP method.

3.1.2 PPPoE Method

Most of the broadband network environment provided now by ISP is the ADSL connection, under this environment, the IP Phone can directly connect to the ADSL modem by setting the PPPoE account (user name and password) provided by ADSL service provider, please follow below.

Configure ► Password：135 ► Network ► (Yes/No) PPPoE

Please press “ok” key to set Yes on (Yes/No) PPPoE when using PPPoE method. And then key in the account from the ISP vendor.

PPPoE Username – please input the user name of the account given by ADSL ISP.

PPPoE Password – please input the password of the account given by ADSL ISP.

3.1.3 Static IP Method

For the other network environment, the users will need to set the static IP provided by ISP or from the MIS person at office.

Configure ► Password：135 ► Network ► Static IP

Under the “Static IP” submenu, please key in the IP address, network mask and default router settings provided by your ISP or private IP address.

Once all the network settings are completed, please restart your IP phone. Then you are able to check whether your internet connection is working properly or not by going to View → Ping, then key in a public IP address (e.g. 168.95.1.1) to ping it, if the response is ok, then the network settings are completed.

3.2 Registration to GK / Proxy Server

After the network environment is set and connected to Internet, you can register the IP Phone to the H.323 gatekeeper or SIP Proxy server by the account from your VoIP vendor/operator. Please choose one of the following methods to register to a gatekeeper or proxy server depends on your IP Phone model.

3.2.1 Registration to a gatekeeper (IP0121)

When the VoIP vendor/operator is running the H.323 system, use the IP0121 to register to the gatekeeper. Configure the following parameters to do the registration.

Configure ► Password：135 ► H323

A. Number – please input the phone number to register to the gatekeeper.

B. Password – please input the password to register to the gatekeeper.
This password is carried in the H.235 message for authentication purpose. Not every gatekeeper needs this field, if not needed, keep it empty.

C. H323_ID – please input the H.323 ID to register to the gatekeeper.
Not every gatekeeper needs this field, if not needed, keep it empty.

D. (Yes/No) Reg To GK – please select Yes and register to the gatekeeper.
The gatekeeper IP address is not configurable at the keypad if this IP Phone device is provided by the vendor/operator. If really want to change this setting, please goto web settings and change it under the H.323 web configuration page.
If the IP Phone does not register to any gatekeeper, it still can call to other IP phone by calling the IP address directly.

E. (Yes/No) Full RRQ – whether to register with full RRQ(* Default (No))。
In normal situation, the IP Phone will automatically re-register to the gatekeeper periodically with minimum information to keep the connection alive. But for some gatekeeper, it will need the devices to keep re-registering to it using the full information, in this special case, please set

the “FullRRQ” to “Yes”.

***Notice**: The IP0121 can pass the NAT/router devices if the gatekeeper supporst this function. The IP Phone will automatically detect if the gatekeeper support this and do the proper control to work under this NAT environment.*

3.2.2 Registration to a proxy server (IP0120)

When the VoIP vendor/operator is running the SIP system, use the IP0120 to register to the proxy server. Configure the following parameters to do the registration.

Configure ► Password：135 ► SIP

A. Number – please input the phone number (username) to register to the proxy server.

B. Password – please input the password to register to the proxy server.

This password is carried in the SIP Proxy-Authorization field using MD5 digest method for authentication purpose. Not every proxy server needs this field, if not needed, keep it empty.

IP0120 follows the standard RFC-2617 to do the authentication.

C. Login ID – please input the username for registration authorization.

When the IP Phone is doing Registration to the proxy server, the Authorization field will have the username=”xxxxx” field, if the “Login ID” entry is empty, the username will use the “Number” input value, but when the “Login ID” field has some value, the username will use this “Login ID” value for authorization.

D. (Yes / No) Proxy On – please select Yes and register to the proxy server.

After this item is enabled, two more menu items will appear. One is proxy server address, to set the IP address or domain name address of the proxy server. Another is the proxy server port, to set the port of the proxy server, its value is usually 5060 unless specified by the service vendor.

If the IP Phone does not register to any proxy server, it still can call to other IP phone by calling the IP address directly.

E. (Yes / No) Outbound Proxy – please set this item to Yes if the registration needs to pass through the Outbound Proxy server.

***Notice**: Most of the proxy servers now have the built-in ability to let IP Phone pass through the NAT/router devices, and this pass-through function does not need the IP Phone to change anything. This method is also more reliable and easier and successful than the other NAT pass-through method like STUN or others. The Outbound Proxy method is just very like the Proxy server built-in NAT pass-through solution, except that the packets need to pass through the Outbound proxy server. And of course, this pass-through method does not need the IP Phone to change anything.*

3.3 Registration / Startup message

When the network and registration configurations are set, please restart the IP Phone, the LCD display on the IP phone will show one of the following message depends on whether the registration is ok or not.

(a). When the IP Phone is registering to a H.323 gatekeeper successfully, the LCD screen will display the following message:

Hi (number) Date Time

Means that the IP0121 is working OK and ready for outgoing/incoming calls. The number inside braces () is the IP phone's number.

Hi (GK Off) Date Time

Means that the IP0121 is working OK and ready for outgoing/incoming calls. But the "Reg to GK" flag is "No", i.e. the IP Phone not need to register. In this case, can call the IP address of other IP Phone directly.

(b). When the IP Phone is registering to a SIP proxy server successfully, the LCD screen will display the following message:

SIP (number) Date Time

Means that the IP0120 is working OK and ready for outgoing/incoming calls. The number inside braces () is the IP phone's number.

SIP (Proxy Off) Date Time

Means that the IP0120 is working OK and ready for outgoing / incoming

calls. But the “Proxy On” flag is “No”, i.e. the IP Phone not need to register. In this case, can call the IP address of other IP Phone directly.

(c). When the IP Phone is configured to use a gatekeeper or proxy server, but not yet registered successfully or failed, the LCD screen will display the following message:

Registering (number) Date Time

When the “Registering” message displayed, the IP phone can not make any calls now. But the menu selection and the on-hook / off-hook function can work OK.

(d). When the IP Phone is failed in registering to a H.323 gatekeeper or SIP proxy server, the LCD screen will display the following message:

RegFail (failed message) Date Time

The failed message could be one of the followings:

(1). Duplicate: means that the registering number is duplicated with others, or the IP Phone’s previous registration information is still kept in the gatekeeper/proxy server if not unregistered last time(this could happened when the IP Phone is powered off instead of restarted from the menu item). If the previous registration information is not cleared, may need to wait about 4 minutes to let IP Phone register successfully again.

(2). Security: means that the account (username / password / H.323 ID) is not correct, please check your account again.

When the RegFail message displayed, the IP phone can not make any calls now. But the menu selection and the on-hook / off-hook function can work OK.

(e). When the IP Phone is failed in registering to a H.323 gatekeeper or SIP proxy server, the LCD screen will display the following message :

DHCPFail (number)
Date Time

Means that the IP phone is configured to use DHCP to get IP address, but the DHCP procedure failed (can not find a DHCP server or the DHCP server rejected the IP assignment). The IP phone can not make any calls and the on-hook / off-hook functions can not work, can not hear any sound on the handset and the speaker. Only the menu selection can work, now.

The IP phone will continuously try to get the IP address from the DHCP server, so, if the DHCP server is responding, the DHCP Fail () display will change to Hi() display to notify that the IP phone is working OK.

(f). When the IP Phone is set to use the PPPoE method for network connection, but have some problem in finding the PPPoE server (ADSL modem), the LCD screen will display the following message :

PPPoE FindFail(number)
Date Time

Means that the IP phone is configured to use PPPoE to get IP address, but the PPPoE procedure failed (can not find a PPPoE server or the network connection has some problem). The IP phone can not make any calls, but the menu selection and the on-hook / off-hook function can work OK.

(g). When the IP Phone is set to use the PPPoE method for network connection, but have some problem in the PPPoE account, the LCD screen will display the following message:

PPPoE AuthFail (numer)
Date Time

Means that the IP phone is configured to use PPPoE to get IP address, but the PPPoE server refuse the connection because the username/password is not correct. The IP phone can not make any calls, but the menu selection and the on-hook / off-hook function can work OK.

(i). When the IP Phone is enable the “Immediate Forward”, the LCD screen will display the following message :

FWD(number) Date Time

Means that the IP0121 is working OK and ready for outgoing/incoming calls. But for any incoming call, it will be forwarded to the “Immediate Forward Number”, this IP Phone will not ring.

3.4 Configurations under “View” item

Under the “View” menu item, there are many submenu items, these items are mainly for quick information about the current status of the IP Phone and also provide some simple functions to check the network status of the IP Phone, described below:

3.4.1 View current network settings

By selecting “View”→”Network” item, you could see your current IP Address network mask, default router and DNS server address. Depends on the network method configured under the Configure→Network submenu, the IP Phone could get its IP address through modem, DHCP, PPPoE or static method. Viewing these values can know current real IP address settings.

3.4.2 Ping another device

“Ping” function is one of the most often used tools in PC to check if the network connection is ok or check if another device is there. By selecting “View”→”Ping” item and key in the IP address or domain name of another device, the IP Phone can check if the connection to that device is ok. This could also be used to check if the connection to Internet is ok by pinging to another device that is already on the public Internet area. E.g. you can ping 168.95.1.1 or www.hinet.net to check the public connection status.

3.4.3 View current quota

The IP0120/IP0121 IP Phone could display remaining quota of the account by selecting “View”→”Quota” item. But the “view quota” ability is not available for every VoIP service provider, please check this with your vendor to see if they support this function. The IP0120/IP0121 IP Phone could get the quota information from gatekeeper/proxy or from other billing server by many different methods depends on different platform.

3.4.4 “Warm restart” the IP Phone

By selecting the “View”→”Restart” item, the IP Phone will do “warm restart” immediately. This “warm restart” is different from the power down/up (cold

restart) action in that the “warm restart” will do the “un-registration” process to gatekeeper or proxy server if the IP Phone is registered, and the “warm restart” will also disconnect the modem connection if modem is connected.

3.4.5 Display current software version

By selecting the “View”→”Image Version” item, the users can view current software version in the IP Phone. The software version is identified by the date of the release of the software image.

3.4.6 Packet Trace for signal monitoring

Some times, the IP Phone could have some problems registering to the gatekeeper or proxy server, or the IP Phone could have some problems making or receiving VoIP calls. Under these situations, by setting the “View”→”PKT Trace” to (Yes), the LCD screen will display all the signaling messages received, and this could let users monitor the signals to identify the problem.

3.5 “Forward” Configurations

The IP0120/IP0121 series IP Phone supports three different kinds of call forward functions, please select “Configure”→”Forward mode” for these three kinds of selection :

3.5.1 Immediate Forward

Under “Forward mode” submenu, users are able to setup for immediate forward by selecting Yes on (Yes / No) Immediate and input the number he/she would like the call to be forwarded to. For example, when A calls B, and B’s phone has Yes on “(Yes / No) Immediate” with number 555-5555, then, under any circumstance, this phone call from A will be forwarded to number 555-5555.

When the “immediate forward” is enabled, the “busy forward” and “no answer forward” can not be enabled at the same time.

3.5.2 Busy Forward

Under “Forward mode” submenu, users are able to setup for busy forward by selecting Yes on “(Yes / No) Busy” and input the number he/she would like the call to be forwarded to when this phone is busy (active). For example, when A calls B, and B’s phone is busy and has Yes on “(Yes / No) Busy” with number 555-5555, then, this phone call from A will be forwarded to number 555-5555.

3.5.3 No Answer Forward

Under “Forward mode” submenu, users are able to setup for no-answer forward by selecting Yes on “(Yes / No) No Answer” and input the number he/she would like the call to be forwarded to when this phone is not answered for the specified seconds. For example, when A calls B, and B’s phone is not answered and has Yes on “(Yes / No) No Answer” with number 555-5555 and no-answer time equals 10, then, this phone call from A will be forwarded to number 555-5555 after ringing for 10 seconds no answered.

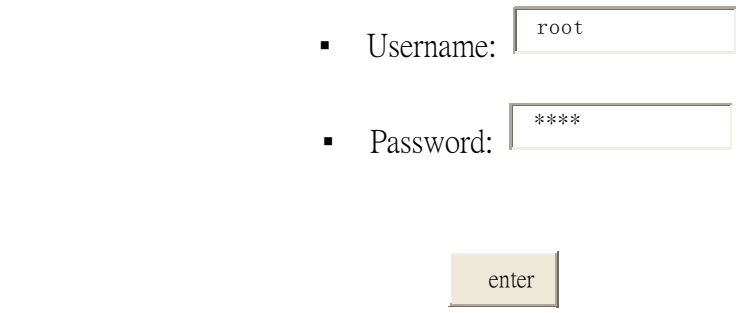
3.6 Software Upgrade

The IP0120/IP0121 series IP Phone provide very simple software upgrade method, following the hot key definitions in section 6.1(table 3) to do the upgrade of different models. The upgrade is from the FTP server configured inside the IP Phone, can be changed from the Web configuration described in section 4.5.

During the software upgrade process, do not power down the IP Phone until the upgrade is done.

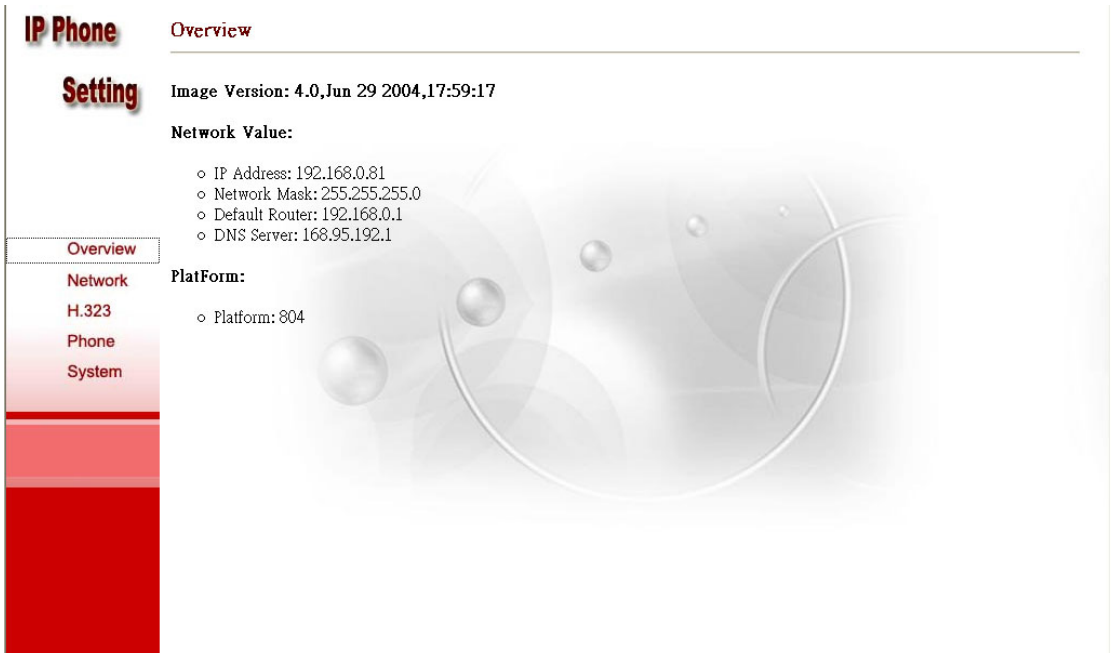
4 Configurations on Web

The IP0120/IP0121 series IP Phone also provide the web interface for configuration, just key in the IP Address of the IP Phone using any web browser (IE5 Explorer or Netscape), you will see the following login window :



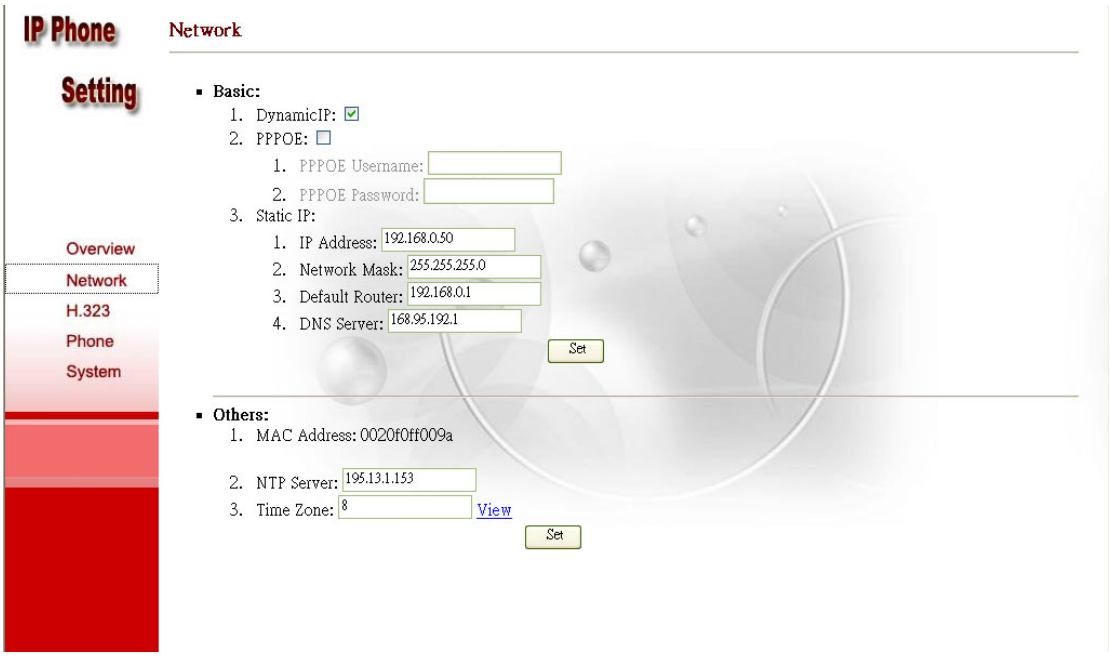
Please key in the username “root” and password “1234”, then press “enter”, you will go to the home page of the IP Phone as shown below.

From this first page, you will get the basic information of the IP Phone, also can set the configurations by selecting the specific pages:



4.1 Network Configuration on web

By click on the “Network” icon on the left banner, the following page will display to allow you to set all the network related configurations.



4.1.1 Basic

Please see section 3.1 for the explanation of each field.

4.1.2 Others

4.1.2.1 MAC Address –

View the MAC Address of the IP Phone. Not changeable here.

4.1.2.2 NTP Server –

Set the NTP (Network Time Protocol) server’s IP Address for the IP Phone to get current date/time and display it on the LCD screen.

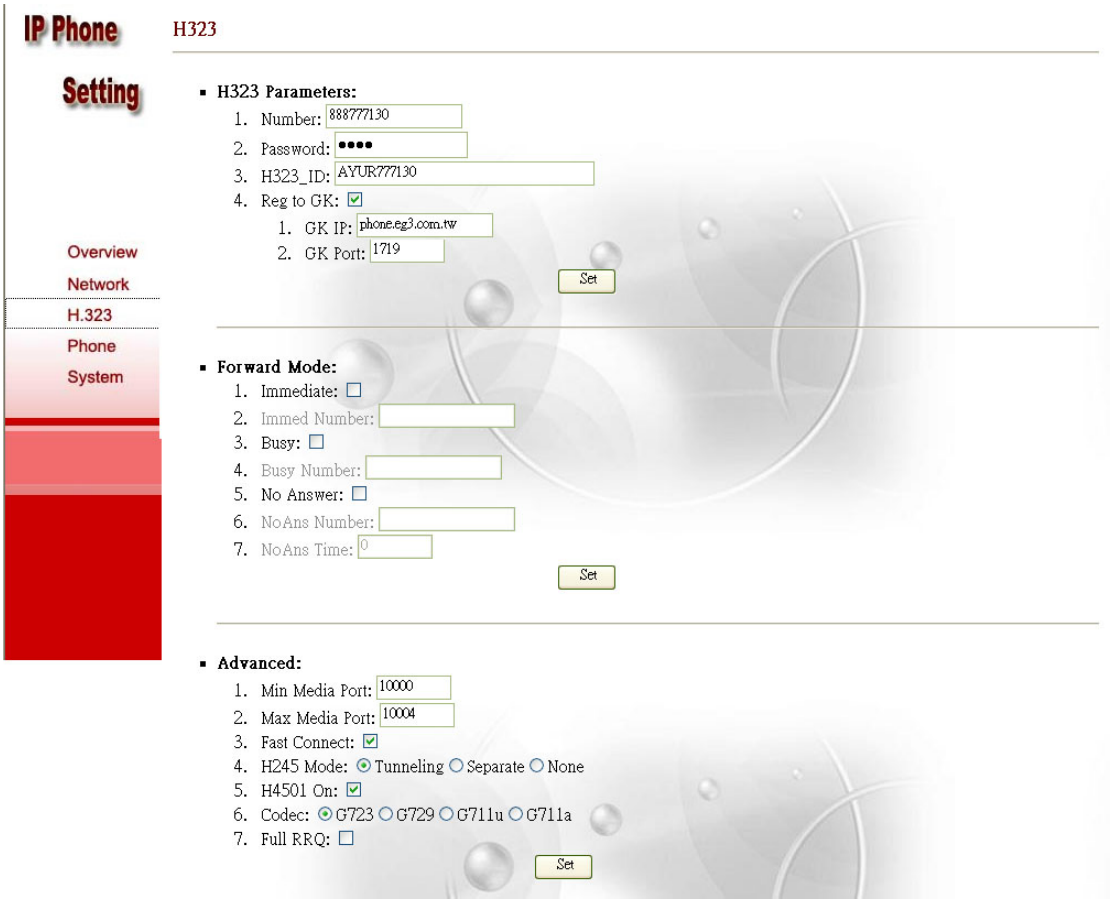
4.1.2.3 Time Zone –

Specify the time zone of your area, you could click the nearby “View” icon to see the time zone of your area. E.g. set the value

to 8 for Taiwan area

4.2 H.323 Configuration on web

By click on the “H.323” icon on the left banner, the following page will display to allow you to set all the H.323 related configurations.



4.2.1 H.323 Parameters

Please see section 3.2 for the explanation of each field.

4.2.2 Forward Mode

Please see section 3.5 for the explanation of each field.

4.2.3 Advanced

4.2.3.1 Min Media Port :

The minimum value of the range of the transmitted RTP packet’s port.

4.2.3.2 Max Media Port :

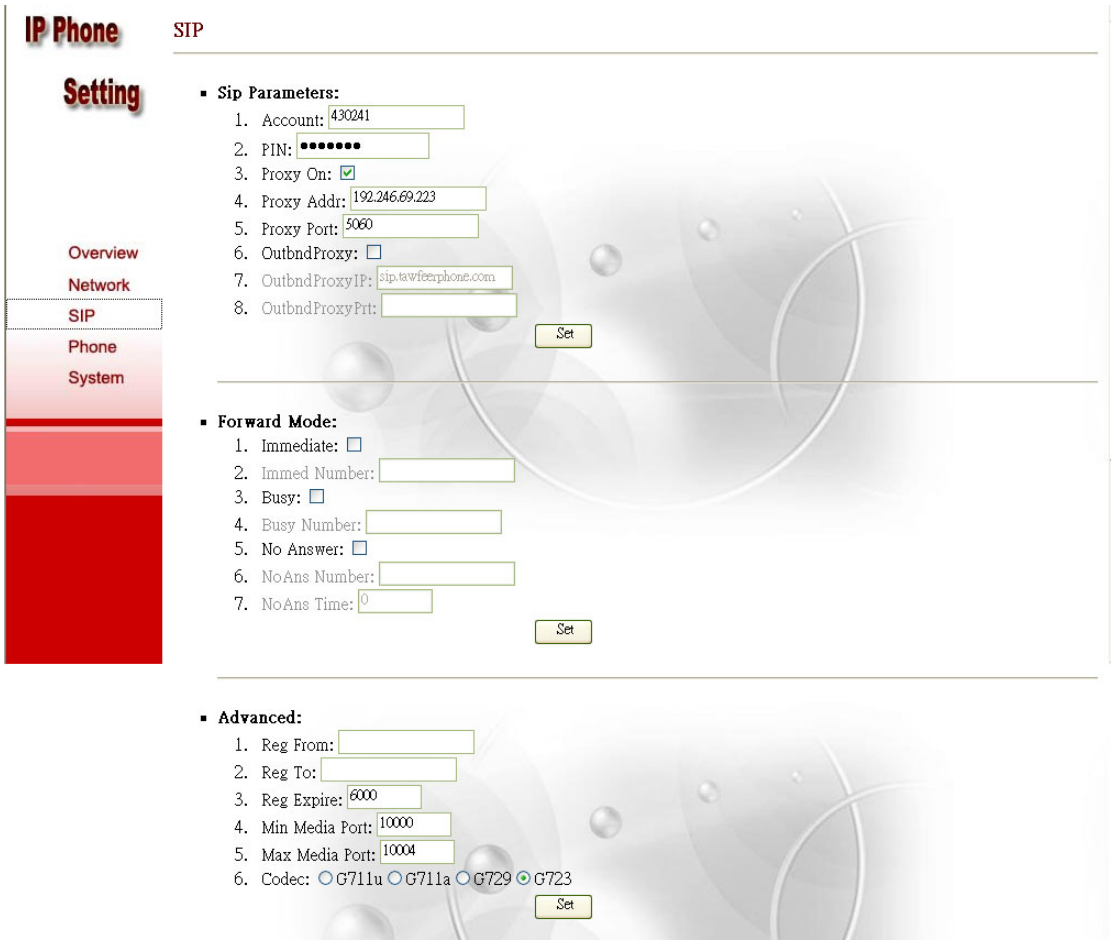
The maximum value of the range of the transmitted RTP packet's port.

4.2.3.3 Codec:

The type of the Codec of the transmitted RTP packets. In SIP call process, the type of Codec negotiated is determined by the called party.

4.3 SIP Configuration on web

By click on the “SIP” icon on the left banner, the following page will display to allow you to set all the SIP related configurations.



4.3.1 SIP Parameters

Please see section 3.2 for the explanation of each field.

4.3.2 Forward Mode

Please see section 3.5 for the explanation of each field.

4.3.3 Advanced

4.3.3.1 Reg From –

Modify the “From:” field in the SIP / SDP messages. Should left this empty unless for very specific system.

4.3.3.2 Reg To:

Modify the “To:” field in the SIP / SDP messages. Should left this empty unless for very specific system.

4.3.3.3 Reg Expire:

Modify the value “Expires:” field in the SIP/SDP messages. This will control the re-registration period.

4.3.3.4 Min Media Port:

The minimum value of the range of the transmitted RTP packet’s port.

4.3.3.5 Max Media Port :

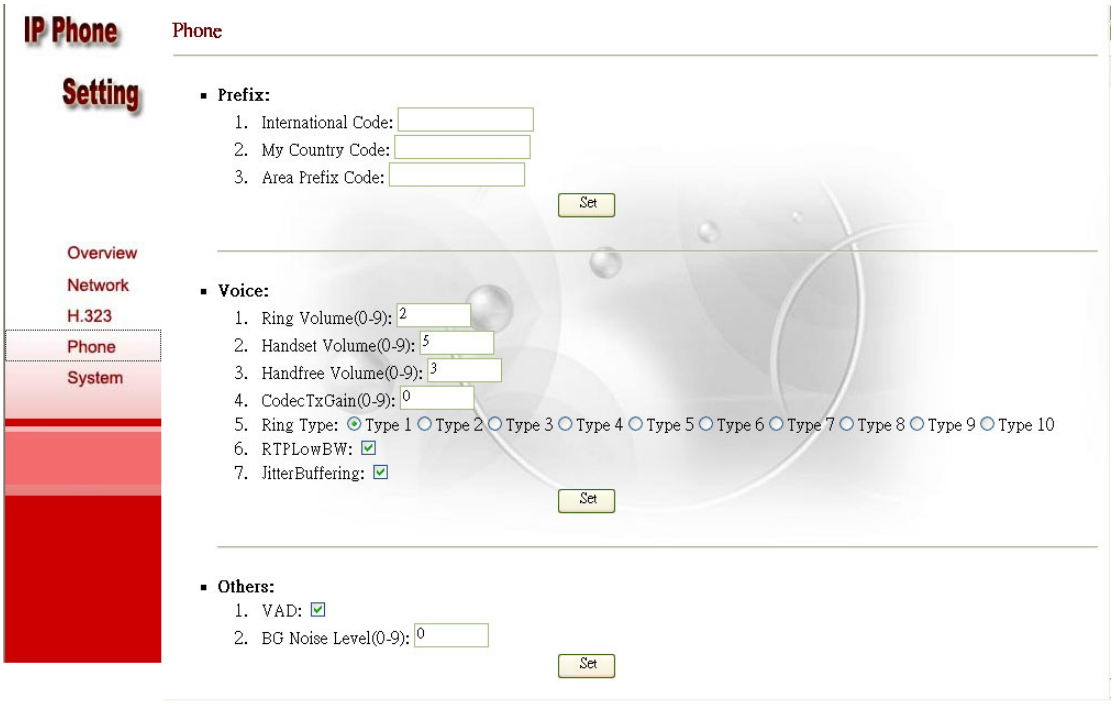
The maximum value of the range of the transmitted RTP packet’s port.

4.3.3.6 Codec:

The type of the Codec of the transmitted RTP packets. In SIP call process, the type of Codec negotiated is determined by the called party.

4.4 Phone Configuration on web

By click on the “Phone” icon on the left banner, the following page will display to allow you to set all the phone related configurations.



4.4.1 Prefix

For some specific VoIP system, the IP Phone can call to PSTN / mobile number and the called number must include the country code, e.g. the country code of Taiwan is 886, so if want to call a Taipei city call 02-25621234, the called number needed to be 886-2-25621234. But this is not consistent with the normal dialing behavior. In order to use the normal dialing behavior, you can set the “My Country Code” value to 886, and set the “Area Prefix Code” value to 0.

As for the “International Code”, when you want to dial an international call, the International Code will be removed. E.g., suppose want to make a call to China, the dialed number will be 002-86-2112341234, the IP Phone will remove the “International Code” 002, and send the real number of 86-2112341234.

This prefix code implementation is only needed for some specific VoIP system. Normally, this is not needed, just let all the prefix values be empty.

4.4.2 Voice

4.4.2.1 Ring / handset / handfree Volume –

Adjust the volume of receiving voice.

4.4.2.2 CodecTxGain –

Adjust the output voice volume.

4.4.2.3 Ring Type –

Adjust the ring type(ring pattern) of an incoming call.

4.4.2.4 RTPLowBW –

This ability is to decrease the voice packet bandwidth of the transmit direction in order to improve the voice quality of transmit direction if the internet bandwidth is not good enough.

4.4.2.5 Jitter Buffering –

Enlarge the jitter buffering ability of receiving voice packet direction from normal 150ms to 400ms. When in some specific internet connection, the traffic is arriving is with large jitter, i.e. with bad voice quality, you can improve the voice quality by enabling the “Jitter Buffering” function.

4.4.3 Others

4.4.3.1 VAD –

Voice Activity Detection, the IP Phone will detect if the user is talking or not, and avoid to send voice packet when the user is silent to decrease the bandwidth requirement.

4.4.3.2 BG Noise Level –

This value controls the sensitivity of the VAD detection.

4.5 System Configuration on web

By clicking on the “System” icon on the left banner, the following page will display to allow you to set all the system related configurations.

In this page, you can do the software upgrade, change username/password of the web login account, and reboot (restart).

5 Call Functions

5.1 Making Calls

To make a call, users can pick up the handset, dial the party's number that wish to call, and end with "#" key to complete the dial. An alternative way to make a call is to press the light blue button ("SPEAKER" key) at the down left corner, dial the party's number that wish to call, and end with "#" key to complete the dial.

***Notice**: This "handfree" operation can only hear the remote party's voice, there is no handfree microphone to send out voice.*

5.2 Receiving Calls

When a call is incoming and the IP Phone is ringing, to receive this call, just pick up the handset.

5.3 Check call history (incoming / outgoing / missed calls)

The call history can be displayed on the LCD screen by pressing "←" key or "→" key when the IP Phone is in idle state.

There are three kinds of call history:

"Incoming" is the record of numbers and time of last 10 incoming calls when answered

"Outgoing" is the record of numbers and time of last 10 successful outgoing calls.

"Missed" is the record of numbers and time of last 10 incoming calls when not answered

When viewing the number of any call history, you can press "MENU/OK" key to do one of the following three actions:

1. Dial – to call out to that number by directly pressing the "MENU/OK" key again.
2. Delete – to delete this item from the call history.
3. To PhoneBook – add this item and the name of this item to phone book, the added item is automatically sorted alphabetically in the phone book.

5.4 Auto Redial

Users are able to call out with last dialed number (redial) by pressing "RE/DIAL" key after off-hooked or in handfree mode.

5.5 Call Forward

You can set the call forward functions through keypad or web configurations. Please refer to section 3.5 for the detailed settings and function descriptions.

5.6 Call Transfer

This IP phone supports two types of call transfer function. They are blind transfer and consultant transfer.

5.6.1 Blind Transfer

With blind transfer, users transfer the caller / callee to third party without informing the third party who is transferring the call. For example, A calls B and A want B to transfer the call to C. B will transfer the call to C by pressing "TRANSFER" button first, B hear dial tone, B dial the C's number, B hang up.

The flowchart of the blind transfer is:

A call to B and talk to B
A ask B to transfer to C
B press "TRANSFER" key
B hear dial tone
B dial the number of C
C is ringing
C is answered
B disconnected automatically

5.6.2 Consultant Transfer

With consultant transfer, users transfer the caller/callee to third party by informing the third party who is transferring the call. For example, A calls B and

A want B to transfer the call to C. B will transfer the call to C by pressing "HOLD" button first, B hear dial tone, B dial the C's number, B talk to C, B hang up, then A can talk to C.

The flowchart of the consultant transfer is:

A call to B and talk to B
A ask B to transfer to C
B press "HOLD" key
B hear dial tone
B dial the number of C
C is ringing
C is answered
B talk to C
B hang up or press "TRANSFER" key
A can talk to C

5.7 Call Hold

Users are able to press HOLD to do consultant transfer, but if the number of third party is not dialed in 8 seconds, the IP Phone will go to "Hold" state. By pressing HOLD again, both ends are able to continue the conversation.

5.8 Phone Book and Speed Dial

By simply pressing Phone Book button at the right bottom corner, users are able to restore totally 50 phone numbers with this function. When users press Phone Book button, two options shown in LCD display are Phone Book and Speed Dial.

After selecting Phone Book option, users will see Add, Edit, Delete, and Delete All. To add person's name and numbers, users may select Add function. To edit or delete phone book information, users may simply select edit or delete function in this category. Users may delete all information in the entire phone book by selecting Delete All.

To set up with speed dial function, users should select Speed Dial and M1-M10 will show in the menu. By selecting each slot, two options are Current Info and Change Setting. Current Info displays the phone number for this slot. By selecting Change Setting, users are able to change the person of the phone book list for this individual slot.

After the speed dial items M1-M10 are set, user can make speed dial call by directly pressing the M1-M10 key when the phone is off-hooked.

6 Attachment

6.1 Hot Key Definitions

The hot key is a sequence of keys pressed when the IP Phone is in idle state.

The following table defines the hot key sequence and their corresponding functions.

Hot Key	Function
#*101*	Upgrade MAC
#*102*	Test LCD
#*103*	Test LCD and Keypad
#*104*	Loop back test (call out)
#*105*	Loop back test (call in)
#*109*	MAC Check and modify
#*110*	Set to default
#*111*	Upgrade software of IP0121
#*112*	Upgrade software of IP0120
#*800*	Clear Gatekeeper IP
#*801*	Platform #801
#*802*	Platform #802
#*803*	Platform #803
#*804*	Platform #804
#*805*	Platform #805

Table 4. Hot key definitions

6.2 H.323 Interoperability List

The followings are the list of some of the devices that have been tested to be interoperable with IP0121 series IP Phone.

- Client/Terminal

Microsoft NetMeeting, Version 3.01
OpenH323 Phone, Version 1.6.0
WellTech, LANPhone 101
Sagitta IP Phone
RadVision H.323v4 Protocol Stack (IMTC 2002, From Software Vendor)
France Telecom R&D, eConf (software: RasVision v3.09) (IMTC 2002, From Software Vendor)
Leadtek Research Inc., BVP8770 Video Phone (IMTC 2002, From Software Vendor)
Sony Electronics, PCS-6000 / PCS-1600 (IMTC 2002, From Software Vendor)
Aethra Srl (Italy), Vega Pro (IMTC 2002, From Software Vendor)
IBM Corporation, Java-based Voice Server (IMTC 2002, From Software Vendor)
ST Microelectronics, H.323 Phone (IMTC 2002, From Software Vendor)
First Virtual Communications, Click To Meet Web Endpoint (IMTC 2002, From Software Vendor)
Innovaphone, IP200 (IMTC 2002, From Software Vendor)
UniData iW200
Cisco7905 IP Phone

- GateKeeper

Cisco AS5350 Gatekeeper function
Citron Network Inc. Gatekeeper
Radvision ProLab GateKeeper Simulator, Version 1.0, October 2001
OpenH323 GateKeeper, Version 1.3.2
GnuGK
III (Institute, Institute for Information Industry) GateKeeper, Version 1.0
NexTone Communications Inc., MSW (IMTC 2002, From Software Vendor)
MediaDigm Technology Inc. (Canada), SureKeeper (IMTC 2002, From Software Vendor)
Lucent iMerge gatekeeper
Wintel gatekeeper
MKY gatekeeper

EZTone gatekeeper
Tellink gatekeeper
Hivocal gatekeeper
GCard gatekeeper

- Gateway

D-Link 4-port gateway
Vanguard Managed Solutions, Vanguard (IMTC 2002, From Software Vendor)
AudioCodes Ltd., MP108 (IMTC 2002, From Software Vendor)
Cisco Systems Inc., Test Bed#1 (IMTC 2002, From Software Vendor)
Polycom Israel, GW-25 (IMTC 2002, From Software Vendor)
antek Tech., 2~16 port gateway
WellTech gateway
Quintum gateway

- MCU

OpenH323 MCU, Version 1.1.3
Polycom Israel, MGC-100 (IMTC 2002, From Software Vendor)

- SoftSwitch

ZTE Telcom. Softswitch
WalkerSun softswitch

6.3 SIP Interoperability List

The followings are the list of some of the devices that have been tested to be interoperable with IP0120 series IP Phone.

Client/Terminal

-SCS-Client V1.00
-SIPS 2.0.43.11
-SJ-Phone 1.10.187c
-X-Lite – FWD V2.0
-Softphone – Simems V0.90Bata27
-Estara V3.0.0.15
-Cisco ATA-186
-Cisco 7905
-BCM SIP Phone
-GrandStream SIP Phone

Proxy Server

-Wparty SIP V0.5.0
-Wparty SIP V0.5.5.2
-ZTE SOFT SWITCH
-Asterisk 0.5.0
-Linux – Vovida V1.5
-Linux – Vovida V1.5
-SER 0.8.10
-WalkerSun softswitch
Gentrice proxy server
Inphonex platform
DeltaThree platform
NEC SV7000 IP PBX

Trunking Gateway

-Cisco 5300
-Quintum D2400