Edge-corE

Powered by Accton

VP3302
IP Phone

User Manual

www.edge-core.com

Contents

1 Overview

1.1 Introduction

The Ethernet Phone VP3302 is a next generation IP Phone that provides a cost-saving solution for small business/home users on their telecommunication needs. The VP3302 follows the open standard SIP protocol to make sure that users can easily install this IP phone with most existing VoIP (Voice over IP) services. By using state-of-the-art DSP (Digital Signal Processing) technology, the VP3302 delivers outstanding voice quality that is comparable to PSTN voice quality.

With the built-in LCD display, the user can easily configure the VP3302 for first time installation in a few minutes. And besides the advanced VoIP functions and easy installation, the VP3302 also provides rich telephone features such as last number redial, speed dial, phone book, call forward/transfer, call history, volume adjustment and speakerphone. The VP3302 is the best VoIP solution in the new generation of communication.

1.2 Key Features

- Follows RFC-3261 SIP standard:
  - Supports password authentication using MD5 digest and RFC-2833 for DTMF relay.
- Dynamic IP support (DHCP and PPPoE):
  - Gets IP from DHCP server using DHCP protocol or through ADSL modem using PPPoE protocol, and automatically reconnects when PPPoE loses connection.
- Passing through NAT devices:
  - Can make outgoing and incoming calls under any NAT device (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 upgrades through the Internet.
- Advanced Digital Signal Processing (DSP) technology to ensure superior audio quality:
  - Chip solution with built in DSP processor ensures perfect voice quality.
- Supports G.723.1, G.729A/B, G.711 (A-law/U-law) voice codecs:
  - Follows ITU-T standard to support best compatibility.
- Supports supplementary services, including immediate (unconditional) call forwarding, busy call forwarding, no answer call forwarding and call hold/transfer.
- Provides call history:
  - Records incoming call history, outgoing call history, missed (not accepted) call history, and let users make direct calls from call history.
- Phone Book: 50 sets
- Speed dial: 10 sets
- Supports Silence Suppression, VAD (Voice Activity Detection), and CNG (Comfort Noise Generation):
  - Silence suppression can save about half of the network bandwidth needed during normal VoIP conversation.

- Ping function supported:
    - Pings other device in the Internet from the VP3302 to make sure the Internet connection is ok.
- System status display on the LCD panel:
    - User can easily know if the VP3302 is working normally and monitor the system's 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 network problems online.
- Calls with or without proxy server (direct IP dialing):
    - Follows standard SIP protocol and is compatible with most existing SIP proxy servers
- Provides easy configuration methods:
    - Very easy settings by using the keypad on the phone set.
    - Settings by web browser.
- Supports RFC-3261, TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, DHCP, NTP/SNTP, FTP, PPP, PPPoE protocols.
- Interoperable with most existing SIP VoIP devices (IP-phone, gateway, proxy, soft-switch, IP-PBX), including Microsoft NetMeeting, Cisco gateways /gatekeepers:
    - Please refer to section 6.2/6.3 Interoperability List for the complete listing.
- The WAN Port automatically works for parallel Ethernet cables and crossed Ethernet cables.

1.3 Hardware Specifications

HARDWARE SPECIFICATIONS	
Spec\Model	**VP3302**
PC Port	1 x RJ45 10/100 Base-T Ethernet, line auto-sensing/switching.
WAN Port	1 x RJ45 10/100 Base-T Ethernet, line auto-sensing/switching.
	(Optional) 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) x 23cm(D) x 9cm(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 VP3302

2 Basic Installation

2.1 Appearance Introduction

2.1.1 Key parts of the VP3302

The key parts of the VP3302 series IP Phone, shown in figure 1, include the following:

1. 2 x 16 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 VP3302 series IP Phone

2.1.2 Rear and Back side Panel of the VP3302

The rear and back side panel illustrations are 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

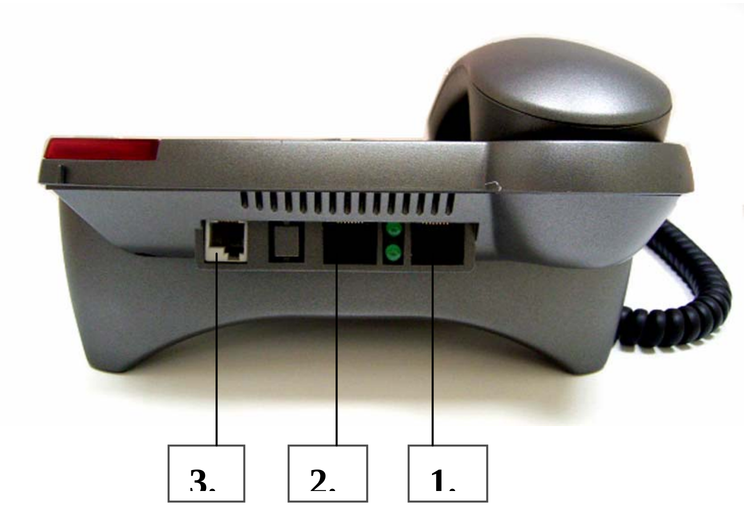


Figure 2. Rear Panel of the IP Phone

Figure 3. Back Side Panel of the IP Phone

2.1.3 Keypad Definition

Figure 3. Keypad illustration

1. **LIGHT**
   The red light flashes when there is an incoming call.

2. **LCD DISPLAY**
   Menu and status are displayed for users.

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

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

   When the IP phone is in dial mode, the "← UP" key is used as the "delete" key.

“DOWN →” KEY

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

And when the IP phone is editing 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 in an idle state or the handset or speaker is being used, this key is used to increase/decrease the volume of the voice sound.

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

5. **+ OUT VOL. -**

Users are able to increase/decrease the volume which is sent out to the remote party.

6. **SPEAKER**

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

7. **HOLD**

Holds the conversation.

This key can also be pressed to do a consultant-transfer of an active call to another IP phone. When the IP phone is active (incoming call is answered or outgoing call is accepted), by pressing this key, a dial tone will be heard, then the user can key in the IP phone number of another party and have a conversation with him. Then, by pressing the “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 flash. Users can press this button to try to register again to their service provider.

9. MESSAGE

This is for future use with functions upgrade.

10. MENU/OK

When the IP phone is in an idle state, this key is used as “Menu” 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 the “OK” key to enter into a lower layer of menu selection or to accept the edited item’s contents.

When the IP phone is in dial mode, the “OK/Menu” key is used as the “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 the current edit and escape to an upper layer of the menu selection.

12. TXT ⟷ NUM

When users need to enter characters, press this button and the 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 is answered or outgoing call is 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 be heard to notify the user to hang up the phone. If the call fails, the user can press the “Transfer” key again to retrieve the original call connection.

When a call is incoming and the IP phone is ringing, by pressing the “Transfer” key and then another IP phone’s number, a user can transfer the call immediately to another party without answering the call.

14. RE/DIAL

When the IP phone is off the hook and some number has been dialed, this key is pressed to call out. The **“#”** key does this same function to send out a called number.

When the IP phone is taken off the hook and this key is pressed immediately afterward, 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. PhoneBook

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.

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

◎ Configure (Password: **135**)
- Network
    - (Yes/No) Dynamic IP
    - (Yes/No) PPPoE
        - If yes:
            - PPPoE Username
            - PPPoE Password
    - Static IP
        - IP Address
        - Network Mask
        - Default Route
        - DNS Server
    - Time Zone
- SIP
    - Login ID
    - Number
    - Password
    - (Yes/No) Proxy On
        - If Yes:
            - Proxy Address
            - Proxy Port
    - (Yes/No) Outbound Proxy
        - If Yes:

- Outbound Proxy IP
- Outbound Proxy Port
- SIP Domain Name
- Frame Size

- 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 (Password: **1230**)

- System
  - DSP Version
  - Upgrade/DnLoad
    - FTP Server IP
    - Image File Name
    - Upgrade Image
    - Upgrade Loader
    - Config Profile
  - Debug Mode
    - Dump Address
    - Dump Size
    - Dump!
  - Network
    - MAC Address
    - NTP Server
  - (Yes/No) APS Enable
    - APS 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
  - DTMF
    - DTMF Relay
    - Payload Type
  - **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
  - **(Yes/No) JmpPxyOn**
    - Jmp Number
    - Jmp Password
    - Jmp Proxy Address
    - Jmp Proxy Port
  - Protocol
    - (Yes/No) STUN Server
    - Media Port
    - Reg From
    - Reg To
    - Reg Expire
    - Reg Action

    - Local Port
    - User Setting
        - Platform
        - Billing Server
- Login
    - User Name
    - Password
    - Confirm Password
    - Admin Username
    - Admin Password
- Statistics
    - User Statistics
        - Call Missed
        - Call Received
        - Call Dialed
    - Phone Statistics
        - Call Missed
        - Call Received
        - Call Dialed
- Additament
    - 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 the IP phone by plugging it into the left jack of the handset. If you use RJ-45 broadband connection, please connect your network cable from your broadband modem to the WAN port of your IP phone, such as NAT, HUB, ADSL, or CABLE.

Step 2:

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

Step 3:

Plug in your power adaptor to your IP phone and power source. The LCD of your IP phone will display "Starting……" and then the "SIP" menu will appear within approximately 4 seconds.

3 Configuration from The Keypad

The VP3302 series IP Phone installation design is very friendly. 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 the keypads as described in the following few sections.

Note:

(1). *When inputting an English character in any menu item, please press that key button quickly to switch between the different characters until the correct one is found.*

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

3.1 Network Configuration

When using the VP3302 IP Phone, first set the network configuration to let the IP Phone connect to the internet. , The proper method to configure the IP Phone to connect to Internet depends on the network environment and phone model.

3.1.1 Dynamic IP Method (DHCP)

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

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

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

Most cable modem connections also use the DHCP method.

3.1.2 PPPoE Method

Most of the broadband network environments provided now by ISPs are ADSL connections Under this environment, the IP Phone can directly connect to the ADSL modem by entering the PPPoE account information (user name and password) provided by the ADSL service provider. Please follow the directions below.

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

Press the "OK" key to set Yes on "(Yes/No) PPPoE" when using the PPPoE method. And then key in the account information from the ISP vendor.

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

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

3.1.3 Static IP Method

For other network environments, users will need to set the static IP provided by the ISP or from a MIS person in the 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. You are now able to check whether your internet connection is working properly. Go to View → Ping, then key in a public IP address (e.g. 168.95.1.1) to ping. If the response is ok, then the network settings are completed.

3.2 Registration to Proxy Server

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

3.2.1 Registration to a proxy server

When the VoIP vendor/operator is running a SIP system, configure the following parameters to register the VP3302 to the proxy server.

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 the MD5 digest method for authentication purposes. Not every proxy server needs this field. If not needed, keep it empty.

The VP3302 follows the standard RFC-2617 to do authentication.

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

When the IP Phone is registering to the proxy server, the Authorization field will have the username = “xxxxx”, 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 the proxy server address; it sets 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 a proxy server, it still can call another 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.

***Note**: Most proxy servers now have the built-in ability to let IP Phone pass through NAT/router devices. This pass-through function does not need the IP Phone to change anything. This method is also more reliable, easier and successful than the other NAT pass-through methods like STUN. The Outbound Proxy method is similar to 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 messages depending on whether the registration is ok or not.

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

SIP (number)	
Date	Time

This means that the VP3302 is working and ready for outgoing/incoming calls. The number inside braces () is the IP phone's number.

SIP (Proxy On)	
Date	Time

This means that the VP3302 is working and ready for outgoing/incoming calls. But if the LCD displays the "Proxy Off", then the IP Phone did not register. In this case, to malke calls, dial the IP address of other IP Phone directly.

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

Registering (number)	
Date	Time

When the "Registering" message is displayed, the IP phone can not make any calls, but the menu selection and the on-hook/off-hook function can work.

(c). When the IP Phone has failed in registering to a SIP proxy server, the LCD screen will display the following message:

RegFail (failed message)
Date Time

The failed message could be one of the following:

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

(2). Security: means that the account (username/password) is not correct. Please check your account again.

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

(e). When the IP Phone fails in registering to a SIP proxy server, the LCD screen will display the following message:

DHCPFail (number)
Date Time

This means that the IP phone is configured to use DHCP to get an 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, the on-hook/off-hook functions can not work, and no sound is heard on the handset and the speaker. Only the menu selection can work.

The IP phone will continuously try to get an IP address from the DHCP server. So, if the DHCP server is responding, the DHCPFail() display will change to SIP() to notify the user that the IP phone is working.

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

PPPoE FindFail(number)
Date Time

This means that the IP phone is configured to use PPPoE to get IP address, but the PPPoE procedure has 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.

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

PPPoE AuthFail (number)
Date Time

This means that the IP phone has been configured to use PPPoE to get an IP address, but the PPPoE server has refused the connection because the username/password are not correct. The IP phone can not make any calls, but the menu selection and the on-hook/off-hook function can work.

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

FWD(number)
Date Time

This means that the VP3302 is working and ready for outgoing/incoming calls. But for any incoming call, it will be forwarded to the “Immediate Forward Number”. The IP Phone with the “Immediate Forward” function enabled 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 to provide some simple functions to check the network status of the IP Phone, as described below:

3.4.1 View current network settings

By selecting "View" → "Network", you can see your current IP Address, network mask, default router and DNS server address. Depending 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 allows you to know the current, real IP address settings.

3.4.2 Ping another device

The "Ping" function is one of the most often used tools for PCs to check if the network connection is ok or to check if another device is there. By selecting "View" → "Ping" and keying 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 the Internet is ok by pinging another device that is already in the public Internet area. For example, you can ping 168.95.1.1 or www.hinet.net to check the public connection status.

3.4.3 "Warm restart" the IP Phone

By selecting "View" → "Restart", the IP Phone will "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 the gatekeeper or proxy server if the IP Phone is registered. The "warm restart" will also disconnect the modem connection if a modem is connected.

3.4.4 Displays current image version

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

3.4.5 Packet Trace for signal monitoring

If the IP Phone has some problem registering to the gatekeeper or proxy server, or there is a problem making or receiving VoIP calls, then, select "View" → set "PKT Trace" to (Yes). The LCD screen will then display all the signaling messages received, and this allows users to monitor the signals to possibly identify the problem.

3.5 "Forward" Configurations

The VP3302 series IP Phone supports three different kinds of call forward functions, please select "Configure" → "Forward mode" for these three kinds of selections:

3.5.1 Immediate Forward

Under "Forward mode" submenu, users are able to setup for immediate forward by selecting Yes on the "(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 set for "(Yes/No) Immediate" with number 555-5555, then, under any circumstance, a phone call from A will be forwarded to number 555-5555.

When the "immediate forward" is enabled, "busy forward" and "no answer forward" can not be enabled. Only one can be enabled at a time.

3.5.2 Busy Forward

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

3.5.3 No Answer Forward

Under the "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 the phone is not answered for the specified amount of seconds. For example: B's Phone has Yes set for "(Yes/No) No Answer" with number 555-5555 and no-answer time equals 10. When A calls B, and B's phone is not answered, then, the call from A will be forwarded to number 555-5555 after ringing for 10 seconds with no answer.

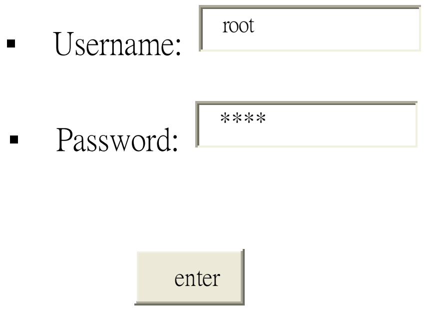
3.6 Software Upgrade

The VP3302 series IP Phone provides a very simple software upgrade method. Follow the hot key definitions in section 6.1 (table 3) to upgrade the different models. The upgrade is done from the FTP server configured inside the IP Phone, or 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.

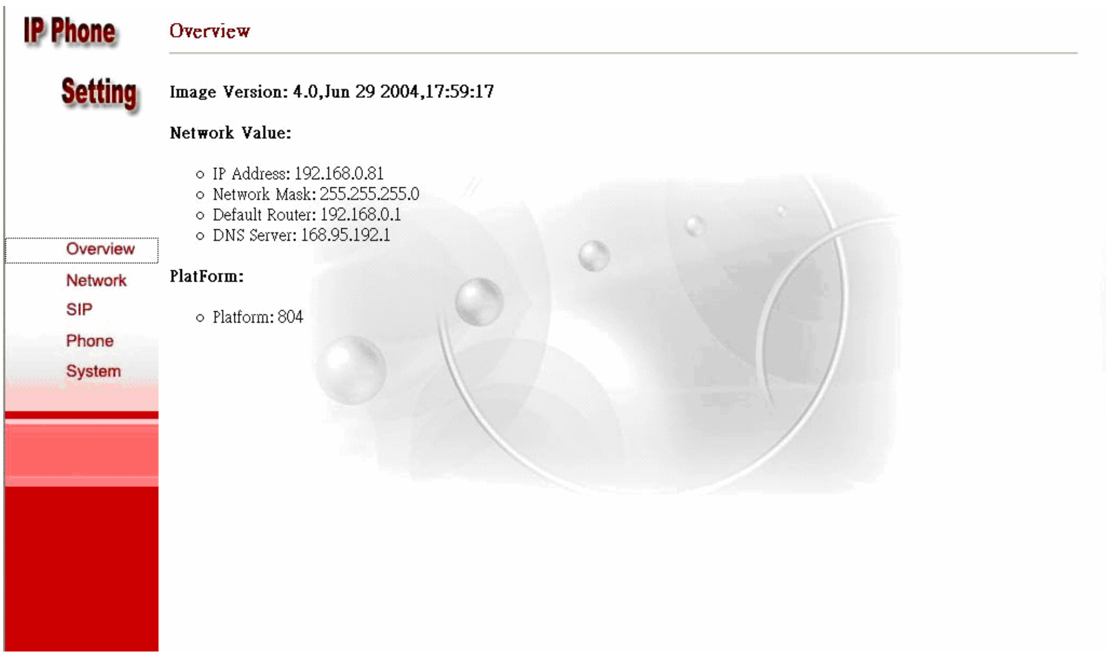
4 Configurations on The Web

The VP3302 series IP Phone also provides web interface for configuration. Key in the IP Address of the IP Phone using any web browser (IE5 Explorer or Netscape). You will see the following login window:



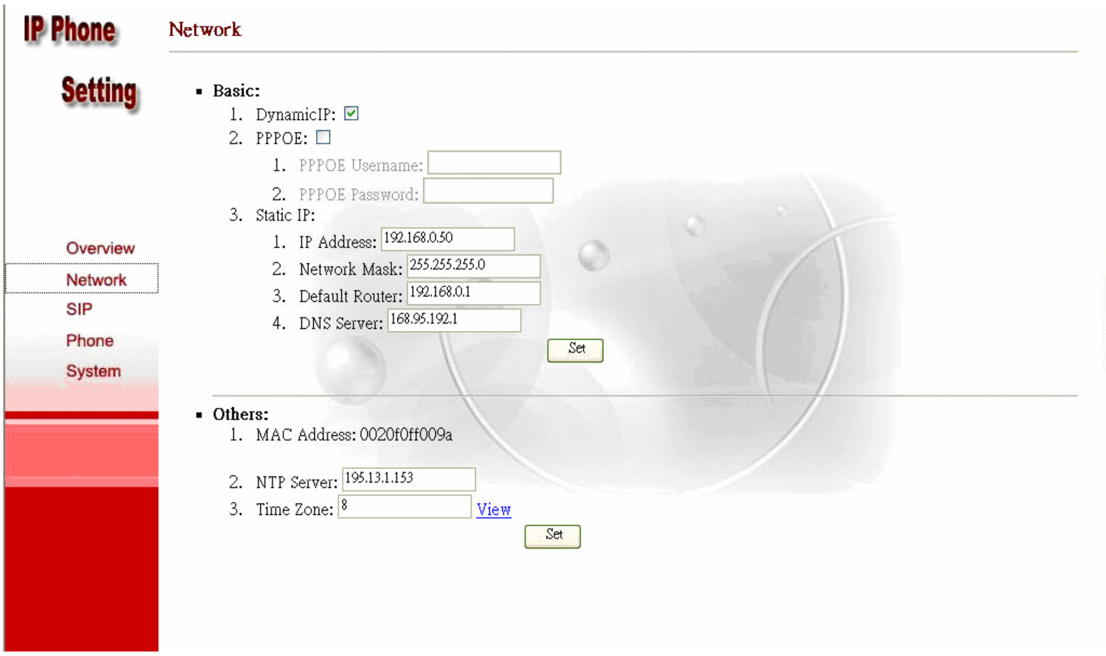
Please key in the username "**root**" and password "**1234**", then press "enter". You will enter the home page of the IP Phone as shown below.

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



4.1 Network Configurations on the Web

By clicking on the "Network" icon on the left banner, the following page will be displayed 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 –

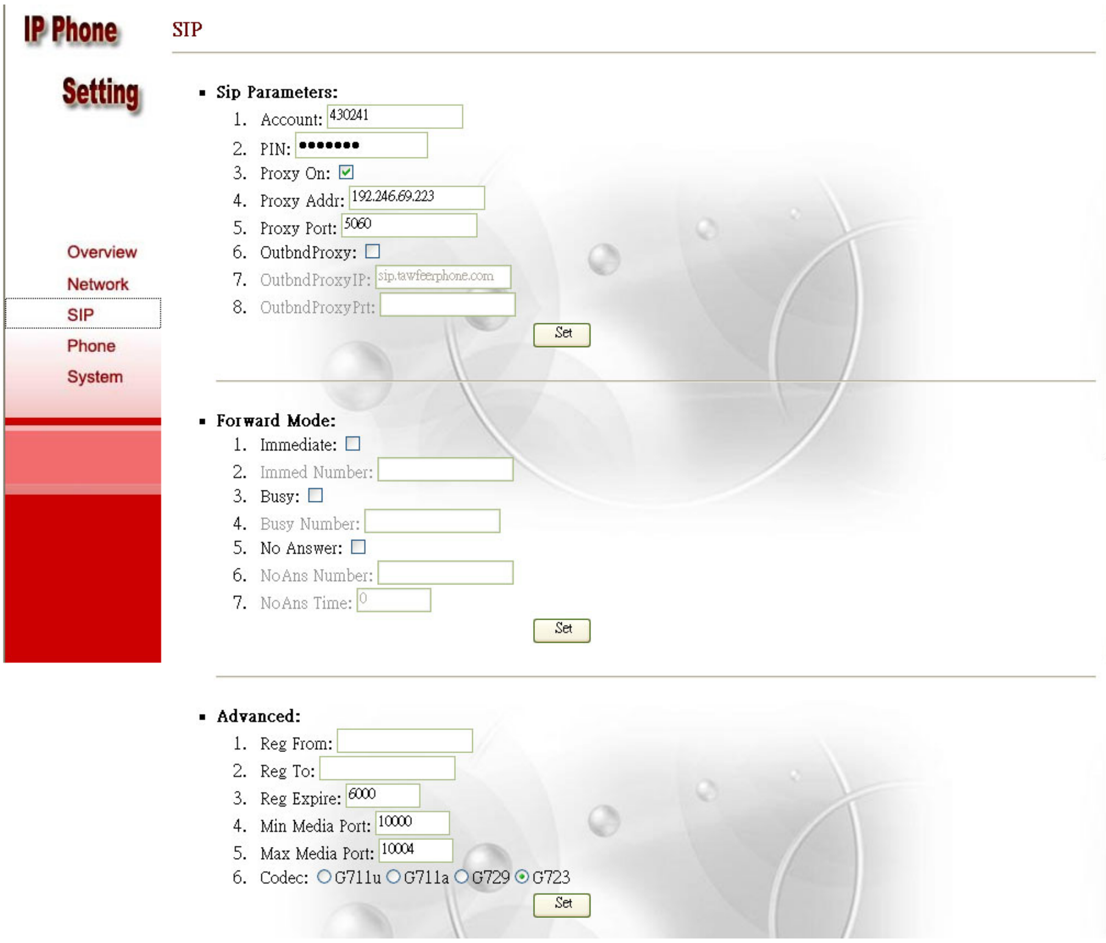
Set the NTP (Network Time Protocol) server's IP Address for the IP Phone to get current date/time and display it on the LCD screen.

4.1.2.3 Time Zone –

Specify the time zone of your area. You can click the nearby "View" icon to see the time zone of your area. For example, set the value to 8 for the Taiwan area.

4.2 SIP Configurations on the Web

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



4.2.1 SIP 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 Reg From:

Modify the "From:" field in the SIP/SDP messages. This should be left empty unless for a specific system.

4.2.3.2 Reg To:

Modify the "To:" field in the SIP/SDP messages. This should be left empty unless for a specific system.

4.2.3.3 Reg Expire:

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

4.2.3.4 Min Media Port:

This is the minimum value for the range of the transmitted RTP packet's port.

4.2.3.5 Max Media Port:

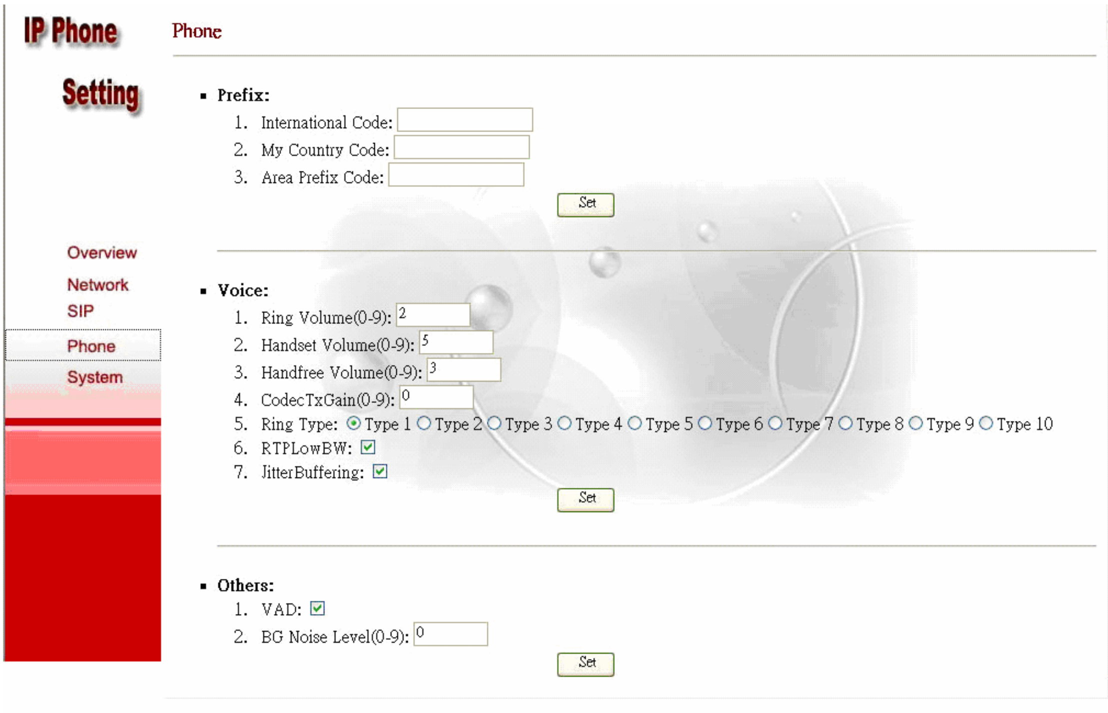
Thhis is the maximum value for the range of the transmitted RTP packet's port.

4.2.3.6 Codec:

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

4.3 Phone Configurations on the Web

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



4.3.1 Prefix

For some specific VoIP systems, when the IP Phone calls the PSTN or a mobile number, the called number must include the country code. For example, the country code of Taiwan is 886. So, if the user wants to call 02-25621234 in Taipei city, then the called number needs to be 886-2-25621234. But this is not consistent with normal dialing behavior. In order to use normal dialing behavior, 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. For example, suppose a user wants to make a call to China, the dialed number will is 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 systems. Normally, this is not needed, just let all the prefix values be empty.

4.3.2 Voice

4.3.2.1 Ring/handset/handfree Volume –

Adjusts the volume of the receiving voice.

4.3.2.2 CodecTxGain –

Adjusts the output voice volume.

4.3.2.3 Ring Type –

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

4.3.2.4 RTPLowBW –

This ability improves voice quality of the transmited direction if the internet bandwith is not sufficient by decreasing the voice packet bandwidth.

4.3.2.5 Jitter Buffering –

This adjusts the jitter buffering ability of receiving voice packets from the normal 150ms to 400ms. In some specific internet connections, the traffic arrives with large jitter, i.e. bad voice quality. You can improve the voice quality by enabling the "Jitter Buffering" function.

4.3.3 Others

4.3.3.1 VAD –

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

4.3.3.2 BG Noise Level –

This value controls the sensitivity of the VAD detection.

4.4 System Configurations on the 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 the username/password of the web login account and reboot (restart).

5 Call Functions

5.1 Making Calls

To make a call, pick up the handset, dial the desired party's number, and end with the "#" key to complete the dialing. An alternative way to make a call is to press the light blue button ("SPEAKER" key) at the lower-left corner. Dial the desired party's number, and end with the "#" key to complete the dialing.

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

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 Checking call history (incoming/outgoing/missed calls)

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

There are three kinds of call history:

"Incoming" is the record of numbers and time of last 10 incoming calls that were 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 that were 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 the number by directly pressing the "MENU/OK" key again.
2. Delete – to delete the item from the call history.
3. Add To PhoneBook – adds the name and number of the item to the phone book. The added item is automatically sorted alphabetically in the phone book.

5.4 Auto Redial

Users are able to call out using the last dialed number (redial) by pressing "RE/DIAL" key after taking the handset off-hook or in handfree mode.

5.5 Call Forward

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

5.6 Call Transfer

The IP phone supports two types of call transfer functions. They are blind transfer and consultant transfer.

5.6.1 Blind Transfer

With blind transfer, users transfer the caller/callee to a 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. When B hears a dial tone, B dials C's number. B then hangs up.

The flowchart of a blind transfer:

A calls and talks to B
A asks B to transfer the call to C.
B presses the "TRANSFER" key.
B hears a dial tone.
B dials the number of C.
C is ringing.
C answers.
B is 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. A wants B to transfer the call to C. B will transfer the call to C by doing the following. B presses the “HOLD” button, and after hearing a dial tone, B dials C’s number. B talks to C, B hangs up and then A can talk to C.

The flowchart of the consultant transfer is:

A calls and talks to B.
A asks B to transfer the call to C.
B presses the “HOLD” key.
B hears a dial tone.
B dials the number of C.
C is ringing.
C answers.
B talks to C.
B hangs up or presses the “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 the third party is not dialed in 8 seconds, the IP Phone will go in 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 the Phone Book button at the right bottom corner, users are able to restore, in total 50 phone numbers with this function. When users press the Phone Book button, the two options shown in LCD display are Phone Book and Speed Dial.

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

To set up the speed dial function, users should select Speed Dial and M1-M10 will be displayed in the menu. When selecting each slot, two options are shown; Current Info and Change Setting. Current Info displays the phone number for a slot. By selecting Change Setting, users are able to change the person in the phone book list for that individual slot.

After the speed dial items M1-M10 are set, users can make speed dial calls by directly pressing the M1-M10 key when the phone is off-hook.

6 Attachment

6.1 Hot Key Definitions

Hot keys are a sequence of keys pressed when the IP Phone is in an 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
#*112*	Upgrade software of VP3302
#*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 SIP Interoperability List

The followings are the lists of some of the devices that have been tested to be interoperable with VP3302 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
-中興通訊： 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