KE2001A

QUICK INSTALL GUIDE

Table of Contents

1 Check for Required Items

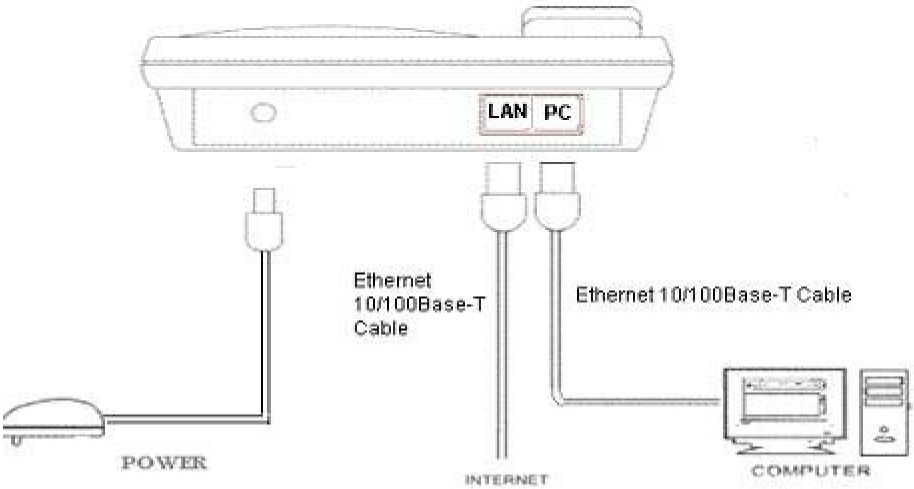
Please check to be sure that you have all of the following components:

• Broad internet access • One phone cord • One power adapter

• One IP phone • One handset • One Ethernet (RJ45) cable

2 Installation Steps

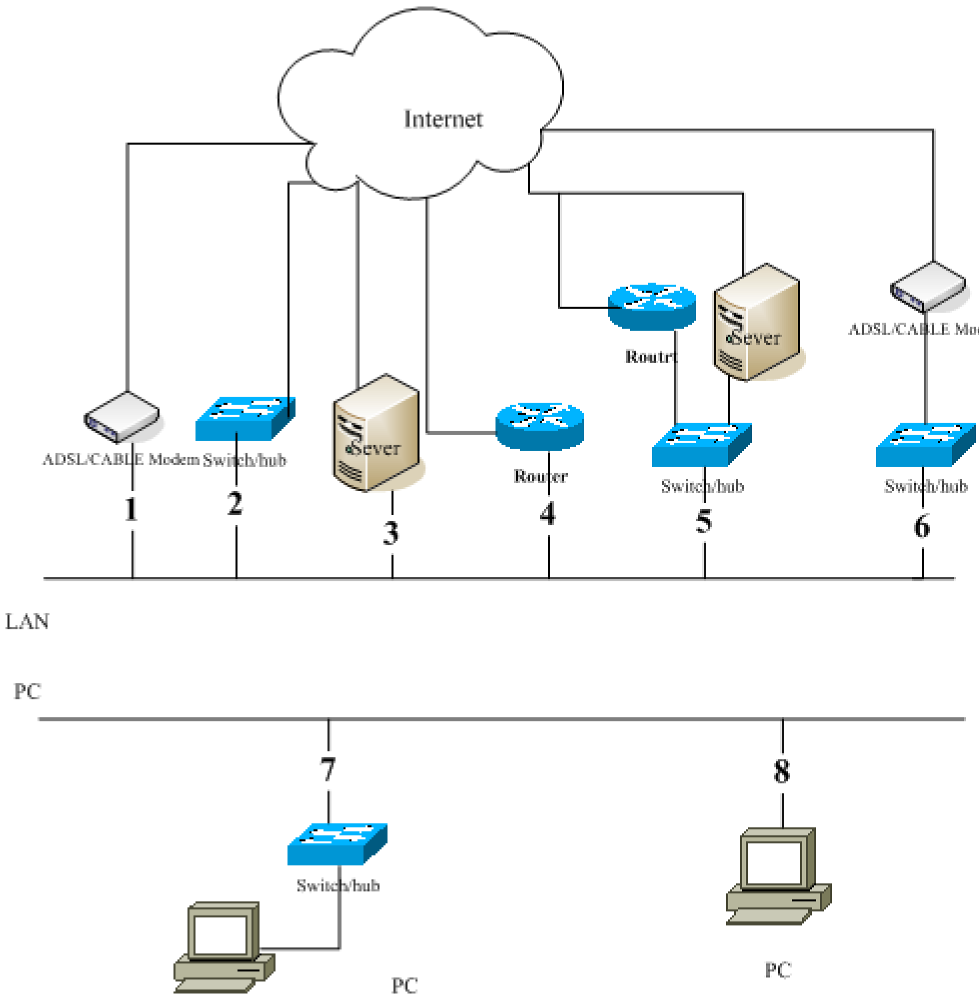
2.1 Installation View



Please install your KE2001A as the connection chart above.

! Note: After the IP phone powers up, the PC port has the same function as LAN port. The PC connection will not operate until the IP phone is fully operational.

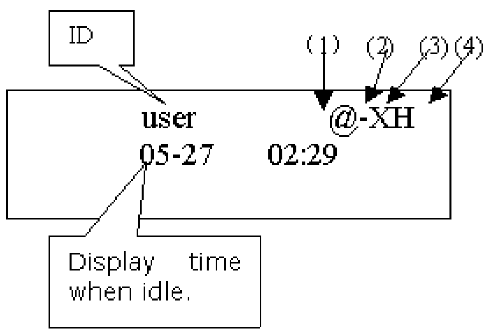
2.2 Connection Chart to Determine Cable Types



To determine the type of cable that you will need to connect to your terminal, look at the above diagram and determine the number of your configuration. The lines marked 2, 4, 5,6,7,8 are Straight-through Ethernet cable, the line marked 3 is Crossover Ethernet cable. If the ADSL/CABLE Modem has the auto-reversal function (it usually has the function), the line marked 1 is Straight-through Ethernet cable, and otherwise it is Crossover Ethernet cable.

Note: when the port LAN connects with line1,2,6,the equipments connected with port PC will not get IP address(i.e., they can't connect with Internet)

3 LCD Display



There are 4 status indicators, they are positioned in the upper right corner of the display. In the above picture they are labeled 1 through 4.

1) Connection Status: will display one of: **!,** －＞ **or** @

!= login failure

－＞= in process of Logging In

@ = successfully Logged In

2)Connection Method: will display one of: **－, A or S**

－: NAT mode not in use. If your service provider supports NAT, you do not need to use NAT mode.

A: using stun mode

S: using static mode

3)PC/LAN Port Status: will display one of: **H, F or X**

X: No PC connected

H: 10M bandwidth connected

F: 100M bandwidth connected

More detailed information about installation KE2001A Terminal, please refer to the user manual.

4 Ready to Use

When the boot process is complete, some prompt characters will be displayed on LCD. If the first character displays '**!**'or no display, it means that you didn't registered successfully. You may need to make settings on your phone.

4.1 Changing the Setting via Your Phone

4.1.1 KE2001A Input Keys

<table>
<tr><th>Key</th><th>Characters</th></tr>
<tr><td>1</td><td>1 → ! # $ % ^ & * ( ) _
+ | ” < > ? - = ; ’ / [ ]
{ } `￥1</td></tr>
<tr><td>2</td><td>a b c A B C 2</td></tr>
<tr><td>3</td><td>d e f D E F 3</td></tr>
<tr><td>4</td><td>g h i G H I 4</td></tr>
<tr><td>5</td><td>j k l J K L 5</td></tr>
<tr><td>6</td><td>m n o M N O 6</td></tr>
<tr><td>7</td><td>p q r s P Q R S 7</td></tr>
<tr><td>8</td><td>t u v T U V 8</td></tr>
<tr><td>9</td><td>w x y z W X Y Z 9</td></tr>
</table>

Key	Characters
0	. , @ 0
*****	. :

! NOTE:

1. Press “ALPHA” key to change input mode between digits and alphanumeric. Default: alphanumeric
2. The ‘*’ key can be used to input ’.’
3. If you need to enter the same character consecutively, press the pound key (#) to confirm the last entry, and then press the number key again to enter the next character.

   For example, if you are entering “Cab”:

   - Press the 2 key 7 times
   - Press the 2 key twice
   - Press the 2 key 3 times

4.1.2 Network settings

First, you should set network type according to your actual network status. DHCP is default status**.** In DHCP status, your LAN automatically assigns all the required network parameters to the phone. Press the SET/OK key,Use “ ▲/▼ ”key to select **Network_Type**. Press” SET/OK”, and then use” ▲/▼ ” to select DHCP, Static IP or PPPoE. Press ESC twice, you will see the system activate the settings automatically.

- ***Static IP Settings***

If your network is Static IP, then you should set static IP address. Press“ SET/OK” key, use“ ▲/▼ ” to select Static IP Settings. Press “SET/OK” , then set IP, gateway and subnet mask. After set complete,press “ESC” key to save automatically.

- ***PPPoE Settings***

If your network is PPPoE, you should make PPPoE settings. Press “ SET/OK” key, use “ ▲/▼ ” to select PPPoE. Press “SET/OK” key, then enter the PPPoE user name and password. After set complete,press “ESC” key to save automatically.

Important: If your service provider makes Presetings such as user name, password, IP phone number, SIP server address, you do not need to do the following operations.

4.1.3 Account settings

Press“SET/OK”key, use“ ▲/▼ ”key to select Account Settings. In account settings, you should select **Server Register on** first, then set server address, user name, phone number, password, domain .etc.

Please refer to **5.2**

4.1.4 Saving Settings

After doing an operation, press the “ESC” key twice and the display should show **Activate**

Settings…, it will takes several seconds to save these settings.

4.2 Changing the Settings via Web Browser

Press the **ESC** key on your phone to check IP address.

Open an Internet Explorer browser on a PC that is connected to the same network as the Terminal, then type the IP address into the address bar and press Enter

USERNAME：user

PASSWORD：voip

Once in the Settings screen, you can verify or enter your network and VoIP related settings, click on **Submit** to save any changes.

5 Work Normally

After the phone reboots and properly programmed, the LCD of the phone will display @. In addition, when the phone is off-hook, you will now hear a dial tone and can make calls.

6 Configuration Parameters

6.1 Network configuration parameters

Parameter	**Description**
Network Type	The terminal can have 3 ways to get network parameters: DHCP, Static IP or PPPoE, please select one. If you do not know, please ask your network administrator or service provider.
DHCP	If your terminal is configured to use DHCP, The required network parameters such as IP, subnet mask will automatically be configured as soon as it is connected to the LAN and powered up.
Static IP	Select this item to authorize users set IP address, subnet mask and gateway IP address of the terminal manually.
PPPoE	Select this item to enable PPPoE protocol which is designed for ADSL and Cable Modem users. With this system, ADSL ISP automatically assigns all the required IP parameters to any device connected to it when the device log on.
PPP User Name/ Password	With PPPoE selected, please enter the user name and password here.
IP Address /Subnet Mask/ Gateway	With Static IP selected, please enter IP address, subnet mask, gateway IP of the terminal here. Note that this address should match the IP address assigned to you by network administrator or your ISP.

Parameter	**Description**
First/second DNS IP	This field defines the primary or secondary DNS (Domain Name Server) address. With Static IP selected, plese enter DNS IP of the terminal here.
NTP Server	Enter NTP (Time Source) Server address
NTP Time Zone	Enter time zone where you live. The time zone value is according to GMT. Please confirm your time zone first and then input from +12 to -12
NAT_TRAVESRAL	nat traversal: When the IP phone with private IP address need communicate with other IP phones in a different LAN or on Internet, please select an item NAT_TRAVERAL Radio Button. none: Select this item when the sip proxy server and IP phone in the same LAN, or the SIP server supports the IP phone working behind the LAN. static: When the system does not support IP phone working behind the LAN, please select this item to manually type public IP address of the NAT device. With this item selected, “NAT External IP” field will be activated. stun: Select this item with Stun server used according to requirement of system. STUN presents a working solution for most NATs that are not symmetric, e.g., most of the household routers have non-symmetric NAT and in this case, it is OK to use STUN. However, STUN does NOT work with symmetric NAT and if your routers have built-in symmetric NAT, do not use STUN. If your configuration is STUN-friendly, please configure your phone to use it. With this item selected, STUN URL field is activated. Please ask your service provider if you do not know which one to select.
STUN URL	When NAT Traversal is set to Stun, please put the URl of the stun server into STUN URL, in the format as “domain name/IP address : service port”.
NAT External IP	With the **static** selected in NAT_TRAVERSAL, please type the public IP of NAT server.

6.2 VoIP Configuration Parameters

! Note: Please check with your ISP for the protocol settings. Changing the settings without proper guidance may result in the the terminal not functioning.

Parameter	**Description**
Preferred Audio Codec	You can select the type of audio codec: default, G.723_only, G.711u_only, G.711a_only or G.729 only
DTMF Relay Type	You can select the type of DTMF relaying: off, 2833, sip_info or inband.

Parameter	**Description**
	These options determine whether user can dial extension telephone. If you select off, you will can not dial extensions. Please consult with your ISP for the correct choice.
Select VAD	Enable/disable VAD (Voice Activation Detection)
SIP Server	SIP server address, Please ask your service provider if you do not know what to fill in.
Register Port	The local SIP port used by terminal to send register packets. The default port number is 5060 or 5063. Please ask your service provider if you do not know what to fill in.
Using Outbound Proxy Server	Choose whether to use proxy server.” on” means using this function, while “off” means not
Outbound Proxy URL	Enter the URL of Outbound Proxy server
Server Register	Choose whether to let the phone send register packet. Usually, you must choose “on”
Display Name	Enter the name which will display on you phone’s LCD
Domain Name	It’s the SIP realm. Please ask your service provider if you do not know what to fill in.
User Name	The user name used to login SIP server for authentication.
Password	The password used to login SIP server for authentication.
Phone Number	The number of this terminal, usually is allocated by system.
Expire Time	Send registration request every expire time. Default:60 s
Using Call Waiting	Choose whether to use Call Waiting function. “on” means using this function, while “off” means not
Use # to Dial Out	Choose whether to use # as a quick dial key. “on” means using this function, while “off” means not
Dial Out In X Sec	Enter the second, after you dialing the destination number, your call will be dialed out automatically after X seconds.
Timer on/off	Whether IP phone display current call duration when getting through a

Parameter	Description
	call.
Balance on/off	Choose whether to enable balance display function.
Dialtone Type	Choose the dial tone standard from the dropdown menu.
Ringing Type	Choose the ringing standard from the dropdown menu.

6.3 Advanced Configuration Parameters

Parameter	Description
Edit Speed Dial	The Edit Speed Dial section allows you to program the speed dial numbers for the M1-M8 keys on the IP phone. Simply enter the number to be dialed in the field beside the Key you wish to assign it to. (Remember to follow the Dialing rules set by your service provider)
Edit AbbrDial	The Abbreviated Dialing Plan is a table of 10 entries that will allow you to program frequently dialed phone numbers or “prefixes”. When using this feature, you can enter specific number patterns, which when dialed will actually dial out different or addition digits. Once programmed there are no special buttons to press to initiate this feature.
Software Upgrade	To upgrade from the remote upgrade server: You can choose **Update Program** to upgrade the function menu, key function, configuration page and kernel program of your phone; choose **Update Bootload** to upgrade the booting program for phone starting. To upgrade from a local PC: You can select the file from your PC and then click ‘Upgrade’..
Handset In	Set the input volume of handset
Handsfree In	Set the input volume of speaker

7 Other Settings

Parameter	Description
Edit Phone Book	Here you can edit the contact’s name, phone number
Change Password	Enter the password you want it changed to, and then re-enter for confirmation. Press Submit to save changes

8 FAQ

Problem	Possible Cause	Solution
Poor Voice Quality	Your internet connection bandwidth is less than 30Kbps	Your internet connection may not be performing correctly. Please contact your Internet Service Provider or Network administrator.
	Unstable network connection: Network delay > 500ms or network trembling > 100ms or network packet loss > 10%	
Phone continues to display '!' as the first character of prompt character string	Phone's connection to the internet is obstructed by a firewall.	Please contact the service provider or network administrator for advice on how to configure your firewall or proxy server.
	Your network has a proxy server which is blocking the phone's connection to the internet.	
	Your phone is configured with incorrect account settings.	Please contact the service provider or network administrator for advice on how to reconfigure your account details.
	Your phone is configured with an incorrect server address.	Please contact the service provider or network administrator to confirm your phone has the correct server details.
Failure call	If the LCD display NO Proxy and you can hear busy tone, you may not set the relative parameters about proxy server when you use NAT	Please set the relative parameters about proxy server by your IP phone or web.

Problem	Possible Cause	Solution
	If you only hear busy tone when you make a call, the causes may be as following: A. Insufficient balance B. The called party in busy line C. The called party off line D. The call can't make through due to some reasons.	Find the correct reasons or consult with your service provider or network administrator.

9 Performance and Features

9.1 Characteristics

- Program memory—2048 KB Flash memory
- Ethernet interface—two RJ45 connector compatible with IEEE 802.3 10/100M Base-T
- Keypad—besides the standard keys 0-9,*,＃, there are another 12 function keys for operation and setting of phone
- Status indicator—LCD to indicate working status
- Hand free function—full duplex speakerphone, dial-tone, speaker and loudspeaker volume can be digitally adjusted independently
- Power adapter— input 100~120V (USA, Japan etc.) or 220~240V (China, Europe etc.) or 100~240V,47-63Hz
- Log on Soft-switch
- Automatic find Soft-switch
- Support DHCP: Automatically obtain local IP, subnet mask, router IP
- Support PPPoE
- Voice Active Detection
- Comfortable Noise Generation
- 16ms Echo Cancellation
- Dynamic Buffer Management—minimize effect to voice quality caused by audio delay jitter

9.2 Standards and Protocols

- IEEE 802.3 10/100M Base-T
- G.711A, G.711μ, G.723.1 5.3K/6.3 Kbps and G.729 audio codec
- SIP
- TCP/IP: Internet Transport and Control Protocol

- RTP: Real-time Transport Protocol
- RTCP: Real-time Control Protocol
- G.723.1、G.729 VAD/CNG economical bandwidth
- G.165 16ms Echo cancel
- DHCP: Dynamic Host Configuration Protocol
- DNS: Domain Name Server
- NTP: Network Time Protocol
- HTTP: Hyper Text Transfer protocol

9.3 Specification

- 208 × 197 × 68mm (L × W × H)

9.4 Electronic Characteristics

- Voltage: 12V DC, 500mA
- Power adapter: AC input 100~120V (USA, Japan etc.) or 220~240V (China, Europe etc.) or 100~240V,47-63Hz
- Network Interface: IEEE 802.3 10/100 Base-T
- EMC: FCC Part15 CLASS B /CE

9.5 Operational Environment

- Operational Temperature: -10 to 40° C
- Storage Temperature: -40° to 55° C
- Humidity: 10% to 90% no dew

9.6 Recommend Network Condition

- Delay: Less than 400ms
- Jitter: Less than 100ms
- Package Loss Rate: Less than 10%
- Required Bandwidth: Minimums 30Kbps

Safety Warning: Please do not place this product under fire and high temperature. Avoid heavy impact, and do not leave the product in rainy or highly humid environments!