

# KE2100 Datasheet

## Introduction:

KE2100 is a SIP based VoIP terminal, it uses your existing investment in the Internet to reduce your long distance costs. The KE2100 connect directly to your Local Area Network or Cable Modem to use the internet to carry your voice calls to any telephone or other VoIP terminal. It is designed to be easy to use, easy to upgrade and offers you a wealth of features and services in an economical fashion.



## KE2100 Specification

<b>Standards</b>	IEEE 802.3
<b>Device Management</b>	Web-Based - Internet Explorer v6 or later;
<b>Function</b>	Router/Bridge
<b><u>VoIP Functions</u></b>	<b><u>VoIP Specifications</u></b>
<b>Call Control Protocol</b>	SIP
<b>Compliance</b>	G.711A, G.711 $\mu$ , G.723.1, G.726 G.729 audio codec and
<b>Voice Compression</b>	GSM
	Echo Cancellation: G.165
<b>Ethernet Ports</b>	LAN: 10/100Base-T Ethernet Port (MDI-II)
	PC: 10/100Base-TX Fast Ethernet Ports (MDI-X)
	IEEE 802.3 10/100 BASE-T Ethernet compliance
	TCP/IP, UDP, ARP, ICMP, TFTP, HTTP
<b>Network Protocols</b>	DHCP: Dynamic Host Configuration Protocol client
	PPP over Ethernet Client
<b>Support DDNS</b>	
<b>Support VLAN</b>	
<b>Support Tos Diff-Serv</b>	Signal and RTP
<b>Support Syslog</b>	Can send the server the request log type
<b>Support DMZ</b>	
<b>Support PPTP</b>	
<b>Support Virtual Server</b>	
<b>DTMF</b>	Inbound/SIP INFO/RFC2833
<b>Network Management</b>	Telnet function
	Manage functions through an intuitive web-based graphical user interface

	HTTP: Hyper Text Transfer Protocol provides firmware upgrade
<b>Security</b>	Administration password through Web MD5
<b><u>Technical Parameters</u></b>	
<b>LCD</b>	2*16 Character LCD
<b>Number of Ports</b>	Two LAN port :( 10/100 Base-T Ethernet) RJ45 ports to connect to your Internet access modem or Local Area Network
<b>Power Input</b>	External Power Supply DC 5 V, 1A
<b>Storage Temperature</b>	-10° to 65° C
<b>Humidity</b>	10% to 90% no dew
<b>Operating Temperature</b>	0 to 50° C
<b>Dimensions</b>	
<b>Certification</b>	FCC Part15 CLASS B /CE Delay: Less than 400ms
<b>Recommend usage net condition</b>	Jitter: Less than 100ms Package lost percent: Less than 10% Bandwidth: More than 28.8Kbps
<b><u>Characters</u></b>	
<b>Program Memory</b>	2048KB Flash memory
<b>Keypad</b>	Standard keys 0-9,*,#, plus 23 other function keys for operation and setting of phone
<b>Hand Free Function</b>	Half duplex speakerphone, dial-tone, speaker and loudspeaker volume can be digitally adjusted independently
<b>Power Adapter</b>	input 100~120V (USA, Japan etc.) or 220~240V (China, Europe etc.) or 100~240V,47-63Hz
<b>Log on Soft-switch</b>	Automatic searching for soft-switch platform
<b>DHCP</b>	Support DHCP
<b>PPPoE</b>	Support PPPoE
<b>Voice Quality Guarantee</b>	Voice Active detection Comfort noise generation 16ms echo cancellation Dynamic buffer management—minimize effect to voice quality caused by audio delay jitter
<b><u>Functions</u></b>	
<b>Call History</b>	Stores the most recent: 64 Missed, 64 Received, and 64 Dialed Calls.
<b>Phonebook</b>	Programmable for up to 140 telephone numbers
<b>Speed Dialing</b>	8 speed dial buttons
<b>Volume Adjust</b>	Adjust the volume of handset, speaker and ring
<b>Call duration display</b>	Show real time call duration
<b>Redial</b>	Supports Last Number Redial

<b>Dialing Plan</b>	Dialing Plan support
<b>Status Indicator</b>	LCD/Web displays register status
<b>Speaker</b>	Hands free
<b>Mute</b>	Mute
<b>Support Local update</b>	Support local HTTP update
<b>Support remote configuration update</b>	Support automatically remote HTTP/FTP/TFTP configuration update
<b>Hot line</b>	The terminal will auto dial out the number you preset when you pick up it.
<b>Wake up alarm</b>	The terminal will ring at the time you preset.
<b><u>Advanced Features</u></b>	
<b>Support Caller Display</b>	LCD displays incoming call information: number or name
<b>Support Balance Display</b>	LCD displays the balance of user account
<b>Call hold</b>	During a call press "HOLD" to put a part on hold and press "hold" again to retrieve the party.
<b>Call transfer</b>	When an incoming call is answered, press XFER key, can make a call transfer. When a new call comes in while you are on a call, you will receive "Call waiting tone". To receive this call, press the "HOLD" button on the device.
<b>Call waiting</b>	
<b>MWI</b>	Support MWI information form softswitch
<b>Do Not Disturb</b>	The incoming calls will be reject, and will be add to MISSED CALLS
<b>3-Way calling</b>	Support 3-way calling
<b>Multiple lines</b>	Support 3 different accounts register to 3 different servers. Includes offline call forwarding and online call forwarding.
<b>Call forwarding</b>	There are 3 types of Online forwarding: No Answer, Busy, and Always.