TELETRONICS INTERNATIONAL INC. ®

EZL⬤⬤P ®
VoIP Phone

EZLoop® LAN VoIP Phone is a full-featured IP-based telephone set for office telephony via Ethe base communication. Over the office LAN connection, it provides IP-PBX solution such as station-station call, IP call and local PSTN/PBX extension call via PSTN Gateway. The traditional PBX functionality is provided with H.450 features together with Gatekeeper or IP/PBX Call Manager. 10/100BaseT embedded switch/hub RJ-45 ports allow connect to office LAN and PC on your tab

Item#21-103

It is easily interface with ADSL/Cable Modem that is provided by ITSP, ISP or carrier company to provide VoIP services to residential and SOHO application. An integrated Analog Phone features provides IP call or PSTN call selection. Or to be a Plain Old Telephone set (POTs) when external power is failure.

It provides programmable keys and feature buttons, an internal high-quality speakerphone with microphone mute, HOLD function, FORWARD and TRANSFER feature buttons. The EZLoop® LAN VoIP Phone also provides a dot matrix of two lines 24 characters LCD display each. The display provides features such as date and time, calling party name, calling party number, and digits dialed. An LED associated with each feature and line buttons provides feature and line status.

TELNET and LCD front panel provide local and remote configuration, Management and software download. By using this feature, ITSP/ISP provides centralized control, management and software upgraded remotely. Several examples are listed below: IP/Gatekeeper addressing setting, dial plan, voice coder, Gatekeeper and Peer to Peer mode selection, QoS and call progress tone setting.

Features

- ISP/ITSP (Internet Telephony Service Provider)
- Multi-nation enterprise communication
- SOHO Telephony
- IP-PBX with office telephony services
- Phone to PC, Phone to Phone

Benefits

- Easy access to Internet phone call
- Cost Saving - Telephone call from VPN or public Internet
- Follows the existing telephone call dial plan
- Easy interface to ADSL/Cable Modem or Leased line equipment
- Built-In two 10/100BaseT Switch/Hub
- Built-In analog line telephone function
- Provided IP call or PSTN call selection
- (Or) to be a Plain old Telephone set when external power is failure

Interface

- Two 10/100BaseT Ethernet port with Switch/Hub inside
- One RJ-11 PSTN analog line interface
- Input Voltage 9Vdc (External AC input, DC output adaptor)

Features

- Support ITU-T H.323 V2/V3/V4 protocol
- PPPoE, when PPPoE re-connect, LAN VoIP Phone will reboot, and on LCD will display related messages
- Behind NAT router or IP sharing device
- Support two LAN phones behind the same NAT
- If the NAT supports multi public IP, the LAN phones that behind the NAT can be found by its IP address.
- Support both Fixed IP and DHCP
- Support H.235 security function
- Support Fast Start and H.245 Tunneling
- Provide H.450 services (Hold, Transfer, Forward)
- Up to ten E.164 phone number
- LED configuration password protection
- Automatically Gatekeeper Discovery
- Alternate Gatekeeper selection
- Provide Peer-to-Peer Mode (Non Gatekeeper needed) selection
- 2 x 24 dot matrix LCD display
- Speaker phone and Handset operation
- Ring tone, Speaker and Handset volume adjustable
- 10 sets last number redial
- LCD Display: Time, Date, Caller ID, Call Duration
- 12 Number Keypads: 0 to 9, # and *
- 4 LCD operation keys : OK, C, LEFT and RIGHT
- 10 function keys for Memory dial or Multi-Line function
- 8 one touch function keys : Speaker, Redial, Mute, Hold, Transfer,
- Call Forward, PSTN, Message
- 5 LED display : PSTN, Message, Hold, Mute and Speaker
- Enter IP phone and Gatekeeper IP address from Keypads
- Configure Dial path selection
- MS-NetMeeting v3.0 compatible
- Support QOS by setting TOS (Type Of Service) parameters of VoIP packet
- Support 802, 1P1Q, VLAN
- Phone book via web interface can set 20 sets
- Support DDNS server (www.dyndns.org or www.3322.org)

Audio Performance

- G.711 A/u-Law, G.723.1, G.729A, G.729
- VAD, CNG
- G.168/G.165 compliant echo cancellation
- Programmable Dynamic Jitter Buffer
- Bad Frame Interpolation
- Completed voice band signaling support
- Provide both Inband and H.245 Outband DTMF generation/detection
- Gain/Attenuation Settings
- Provide Progress Tone: Dial tone, busy tone, call-holding tone and ring-back tone
- PSTN port is working when power is failure

Management Features

- TFTP/FTP download
- LCD Front Panel
- TELNET

Environmental Features

- Humidity: 10 to 95 %, Non-condensing
- Operational Temperature : 0 to +50 ºC
- Storage Temperature : -10 to +70 ºC

LED Indicator for system status

- SPEAKER : Speaker Phone working status
- MESSAGE : Message indication
- LINK/ACT : 10/100BaseT status
- HOLD : IP call put on hold indication
- MUTE : IP call is in Mute status
- PSTN : Selection between IP call and analog line call

Chassis

- Dimension: 215mm(W) x 71mm(H) x 198mm(D)
- Weight (unit): 834 g