

VERSO 301 IP PHONE (SIP)

Product Overview

Verso 301 is an all-in-one telephone specially designed for IP PBX and SIP Servers and SMB (small & medium business) environment. High quality voice connection using with IP PBX and SIP Server services.

IP is abbreviation of Internet Protocol. An IP phone is a telephone transporting voice by grouping data package of IP protocol. IP phone can be used on IP Networks from an enterprise-wise LAN to a city-wide WAN.

An IP phone transports voice message over data communication network at extremely low price but of excellent sound quality just like ordinary fixed telephone lines do. With our VERSO 301 Internet phone, you're going to save dramatically on international calls.

With its unique generalized outline and linner line modes. VERSO 301 Internet phone functions much like an ordinary telephone switching between inner and out line, it brings great convenience to the users. When VERSO 301 Internet phone is in generalized inner line mode, it is free of charge to call another VERSO 301 wherever the destination phone is in the world. When VERSO 301 series IP net phone is in generalized out line mode, it can place calls to every ordinary telephones in the world at an extremely low price.

With its stability, sound quality, compatibility and rich functionalities, VERSO 301 leverages your wealth with high quality communications!



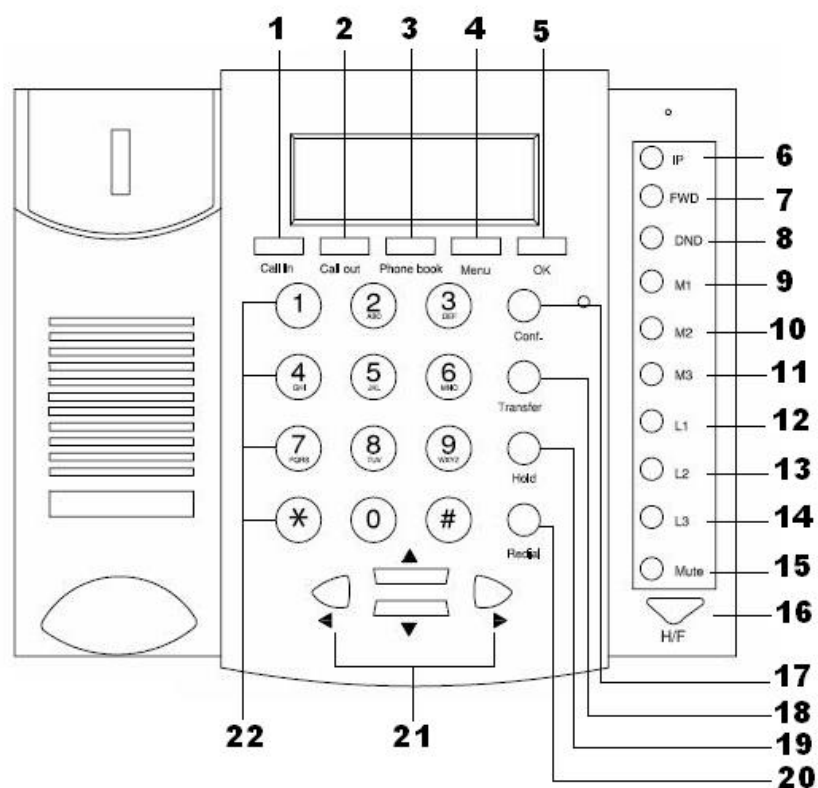
Verso 301 Highlight Features:

- ❖ Support **SIP V1** (RFC2543), **V2** (RFC3261)
- ❖ Provide **1xWAN**, **1xLAN**
- ❖ Perfect sound quality: **VAD**(Voice Activity Detection), **CNG**(Comfortable Noise Generator), **LEC**(Line Echo Canceller), **Packet Loss Compensation**, **Adaptive Jitter Buffer**
- ❖ Support IP/TCP/UDP/RTP/RTCP, ICMP/ARP/RARP/SNTP
- ❖ Support TFTP Client/DHCP Client/ PPPoE Client, Static IP
- ❖ Support Telnet/HTTP Server, DNS Client
- ❖ Support Codec **G.711**, **G.723**, **G.726**, **G.729A** and **G.729B**
- ❖ **Router** function
- ❖ **PoE** (Power Over Ethernet) – **optional**
- ❖ Call Hold, Call Waiting, Caller ID, **3-way Conference**, **DND**
- ❖ Call forward (all forward, busy forward and no answer forward.
- ❖ In-Band DTMF, Out-of Band DTMF, SIP Info
- ❖ Registrar Server (**three SIP Account**)
- ❖ Speaker Phone, Speed dial, Phone book, Flash and Volume Adjustment function
- ❖ HTTP 1.1 basic/digest authentication for Web setup
- ❖ MD5 for SIP authentication (RFC2069/ RFC 2617)
- ❖ **QoS**: ToS field
- ❖ NAT Traversal : **STUN**
- ❖ Web Browser, Telnet, Keypad Configuration
- ❖ TFTP, Telnet, HTTP firmware upgrade



Phone Diagram

1. Call In
2. Call Out
3. Phone Book
4. Menu
5. OK
6. IP
7. FWD
8. DND
9. Memory1
10. Memory2
11. Memory3
12. Line1
13. Line2
14. Line3
15. Mute
16. H/F
17. Conf.
18. Transfer
19. Hold
20. Redial
21. Direction Keys
22. Number keys



Hardware Specifications

Hardware Overview

- Main Chip – CM5000
- Data Memory – 16M
- Program Memory – 64M
- Ethernet Jack – 2 x 10/100M jacks
- AC/DC adapter – Input AC230V, output 9V

Software Overview

- Network Protocol
 - SIP V1 (RFC2543), V2 (RFC3261)
 - IP/TCP/UDP/RTP/RTCP
 - ICMP/ARP/RARP/SNTP
 - TFTP Client/DHCP Client/ PPPoE Client, Static IP
 - Telnet/HTTP Server
 - DNS Client
- Codec
 - G.711
 - G.723
 - G.726
 - G.729A
 - G.729B
- IP Assignment
 - Static IP
 - DHCP
 - PPPoE

- Voice Quality
 - VAD(Voice Activity Detection)
 - CNG(Comfortable Noise Generator)
 - LEC(Line Echo Canceller)
 - Packet Loss Compensation
 - Adaptive Jitter Buffer
- Security
 - HTTP 1.1 basic/digest authentication for Web setup
 - MD5 for SIP authentication (RFC2069/ RFC 2617)
- Configuration
 - Web Browser
 - Telnet
 - Keypad Configuration
- Firmware Upgrade
 - TFTP
 - HTTP

Operating requirements:

- Operation temperature: 0 to 50 °C
- Storage temperature: -10 to 60 °C
- Humidity: 10 to 90% no dew

Electric requirements:

- Voltage: 9V DC
- Power adaptor: AC/DC input 230V, output 9V
- Network interface: 2 X RJ-45 Ethernet connectors

Size:

198 x 176 x 60 mm (L x W x H)