AWTEK

Wireless & Communication

VoIP(SIP) phone

AW-322

Voice Features:

- Codec G.711A/u, G.7231 high/low, G.729
- Voice Gain Setting
- RFC2833 DTMF relay
- Support SIP synchronously

SIP Features:

- RFC3261, RFC3262, RFC3666, RFC2543
- Proxy and Register, SIP domain
- Server authentication: none,basic,MD5
- DNS name of SIP server
- SIP signaling port setting
- NAT transverse, STUN
- NAT transverse, SIP Express router
- Public Server/ Private server. Can connect to ISP and Private SIP server at same time
- Dual public server
- SIP INFO for DTMF, interoperate with CISCO SIP device
- Each password for each number
- SIP Call forward/transfer/holding/waiting
- Peer to peer SIP call

Networks Features:

- WAN/LAN port with Router or Bridge Mode
- Basic NAT and NAPT
- NAT ALG
- Under Bridge mode, Access internet by using NAT through PPPoE
- PPPoE for xDSL, automatically keep alive
- DHCP Client on WAN, DHCP server on LAN
- DNS client with 2 servers IP
- DNS relay on LAN
- Auto configuration on LAN for IP and DHCP server
- SNTP,802.1P QoS, Firewall
- Network utilities: ping, trace route, telnet client
- Support 5 SIP phone Number
- **POE Feature (optional)**

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