Empowering DBS & VoIP
Communication Solution Provider

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VoIP Gateway, 1 Port - Model : ATS1000

ATS-1000 is a VoIP Gateway for Home & SOHO users. It's powered by a SoC, 16-bit DSP, 32-bit RISC CPU with 125MHz clock rate. Supporting T2Free Super SIP server and RFC3261, RFC2543 SIP protocol. It is compatible with different SIP proxy and VoIP devices in the market. Full features and configuration access is by telephone key pad.

Features

- Voice communication gateway which is based on wide band and independent of PC
- WAN 10/100Base T RJ-45 for LAN
- LAN 10/100Base T RJ-45 for PC
- PSTN 1 RJ11 for Lifeline
- Voice Features: SIP v1 (RFC2543), v2(RFC3261),
- Echo cancellation: Support G. 165, 168 and Voice Gain Setting, Jitter Buffer, CNG (Comfortable noise generator)
- Support dual GK, call forward and Peer to Peer,
- Support NAT transverse
- Support SIP domain & SIP authentication , DNS name of server, Peer to Peer
- DTMF: SIP info, In-band DTMF, Out-of-band DTMF
- Support TFTP/PPPoE Client, DHCP Server/Client,Telnet/HTTP Server
- Support DHCP Client on WAN
- Support DHCP Server on LAN

Specifications

TCP/IP

- Supports SIP v1 (RFC2543), v2(RFC3261),TCP/IP/UDP, RTP/RTCP, IP/ICMP, ARP/RARP. TFTP/PPPoE Client, DHCP Server/Client,Telnet/HTTP Server.
- NAT/DHCP Server, DNS Client

NAT Traversal

- STUN, Outbound Proxy

IP Assignment

- Static IP, DHCP, PPPoE

Security :

- HTTP1.1 basic/digest authentication for Web. Setup
- MD5 for authentication (RFC2069/RFC2617)

Call Function

- Call Hold, Call Transfer, Call Forward, Call Mute, Caller ID Tone
- Ring Tone, Ring Back Tone, Dial Tone, Busy Tone ,Programming Tone

Codec

- G.711 64k bit/s(PCM) G.723.1 6.3k/5.3k bits/s
- G.729A 8kbit/s(CS-ACELP) G729B adds VAD & CNG to G.729.
- G.726 16k/24k/32k/40k

Voice Quality

- VAD (Voice activity detection), CNG(Comfortable noise generator) ,LEC(Line echo canceller) AEC, G.165, G.168. Adaptive Jitter Buffer, Packet loss Compensation.

Configuration

- Web Browser, IVR/Keypad

Firmware Upgrade

- HTTP, TFTP

Dial Methods

- Direct IP Call without SIP Proxy, Dial number via SIP server

Call Function :

- Call Hold, Waiting, Forward , Call ID , 3-way conference.

Phone Function

- Volume Adjustment, Speed dial key, Phone book, Flash

DTMF Function

- In-band DTMF, Out-of-band DTMF, SIP Info

Hardware and certification

- Port: RJ45 x 2 , RJ11 x1(FXS x1)
- Certification: CE, FCC approval