

VoIP SIP ATA (Analog Telephony Adapter) Series



- VA01** VoIP SIP ATA with 1 FXS port
- VA02** VoIP SIP ATA with 2 FXS ports
- VA11** VoIP SIP ATA with 1 FXO, 1 FXS ports

❖ Call Free for long distance or international calls

Features

- Compliant with IEEE 802.3, 802.3u 10/100M Ethernet standard
- Supports SIP 3261 (Radvision SIP stack), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, and DHCP
- Interoperable with various third party SIP end user devices
- Supports popular voice-coders including G.729, and G.711 (A-law and μ -law)
- Supports standard voice features such as Caller ID, Consultation Hold, Call Waiting, Blind Call Transfer, Attended Call Transfer, Call Forwarding, Mid-Call DTMF (only using RTP) and 3-Way Conferencing
- Supports Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), and Line Echo Cancellation
- Supports DIGEST authentication (MD5 only)
- Provides easy configuration through a web interface and limited configuration through the handset dial-pad
- Voice over data prioritization ensure high-quality telephone service
- Supports Local PSTN call transfer to VoIP call or VoIP call transfer to local PSTN call
- Selective PSTN / VoIP voice communications
- PSTN Backup support through Hardware Relay—applicable when power is down or when WAN interface is down (Optional)
- STUN support for NAT traversal

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Specifications

Standard <ul style="list-style-type: none"> • IEEE 802.3 / 802.3u 10/100M Ethernet Standard • SIP v1(RFC2543), v2(RFC 3261) • Voice codec : G.711(A-law, μ-law), G.729A/B
Physical Interface <ul style="list-style-type: none"> • 10/100Mbps Ethernet, RJ45 • RJ11 Port for Phone • RJ11 Port for PSTN (Optional) • Power DC 9V, 1A
Voice Feature <ul style="list-style-type: none"> • Voice codec <ul style="list-style-type: none"> G.711 : 64k bit/s (PCM) G.723 : 6.3k / 5.3k bit/s G.726 : 16k / 24k / 32k / 40k bit/s (ADPCM) G.729A : 8k bit/s (CS-ACELP) G.729B : adds AD & CNG to G.729 • Supports silence Suppression, VAD(Voice activity detection), CNG (Comfortable noise generator) and LEC (Line echo canceller) • Supports Packet Loss Compensation
Call Function <ul style="list-style-type: none"> • Call ID • Call Waiting Caller ID • Consultation Hold • Blink Transfer • Attended Transfer • 3-Way Conferencing • Call Waiting • Call Forwarding • Call Return • IP Dialing • Speed Dialing
Network Function <ul style="list-style-type: none"> • PPPoE • DHCP Client • Friendly web-based configuration
PSTN Backup (Optional) <ul style="list-style-type: none"> • Supports PSTN Backup
Management Feature <ul style="list-style-type: none"> • Web Browser • Telnet • Voice configuration • HTTP

Security <ul style="list-style-type: none"> • HTTP 1.1 basic/digest authentication for Web setup • MD5 for SIP authentication (RFC2069/RFC2617)
Led indicator <ul style="list-style-type: none"> • Phone, Line, Ethernet and power indicator
Temperature <ul style="list-style-type: none"> • 32~122°F (0 ~50°C)
Humidity <ul style="list-style-type: none"> • Max. 95% (Non-Condensing)
Regulatory Approvals <ul style="list-style-type: none"> • FCC, CE

Application:

