

# 2 FXS SIP Analog Telephone Adapter



**KEY FEATURES** 

### **PRODUCT FEATURES**

- Feature-rich telephone service over home Internet / Intranet connection
- Up to 2 concurrent VoIP calls
- Cost-effective, easy-to-use Analog Telephone Adapter solution
- Web-based utility and machine configuration
- Remote administrator authentication
- Voice prompt for machine configurations

#### **VOIP FEATURES**

**APPLICATIONS** 

• SIP 2.0 (RFC3261) compliant

ATA-150S, the latest PLANET Analog Telephone Adapter solution, enables home users and companies to convert existing standard telephones to IP-based networks in less installation cost. The ATA-150S is equipped with two telephony interfaces, thus users may register to different SIP proxy servers, IP PBX and establish up to 2 concurrent VoIP calls for more flexibility in the voice communications.

The ATA-150S can be the bridge between the traditional analog telephones to IP network with an extremely affordable investment and easy installation. The service providers and enterprises are able to offer users enhanced telephony communication services via the existing broadband connection to the Internet or corporation network.

 Voice codec: G.711(A-law /µ-law), G.729 AB, G.723 (6.3 Kbps / 5.3Kbps)

> 100Base-TX UTP 100Base-TX UTP w PoE Telephone wire

- FoIP : T.38 FAX Relay, G.711 Fax pass-through
- QoS : IP TOS (IP Precedence) / DiffServ
- Call Waiting / Hold / Resume / Transfer / Forward
- 3-Way Conference / Caller ID Generation
- VAD / CNG / Dynamic Jitter Buffer

Phone

• SNMP v1/v2, TR-069 and Auto Provision



Phone

FA>

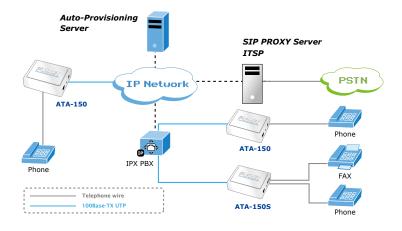
VIP-254P

VIP-550PT

VIP-351PT



# **ATA-150S**



Product	2 FXS SIP Analog Telephone Adapter
Model	ATA-150S
Hardware	
LAN	1 x 10/100Mbps RJ-45 port
FXS	2 x RJ-11 connection
Protocols and Standard	
Standard	SIP 2.0 (RFC3261), STUN (RFC 3489), UPnP, MD5 for SIP authentication (RFC 2069 / RFC 2617)
Voice codec	G.711, G.723, G.729
	Voice activity detection (VAD)
	Comfort noise generation (CNG)
	G.168: Line echo canceller (LEC)
	Jitter Buffer
Voice Standard	DTMF Detection and Generation
	In-Band and Out-of-Band (RFC 2833), (SIP INFO)
	QoS : IP TOS (IP Precedence) / DiffServ
	FAX support : T.38 FAX Relay, G.711 Fax pass-through
	Call Waiting
	Call Hold / Resume
	Call Transfer: Blind Transfer / Attended Transfer
Telephony Features	Call Forward: On Busy Forward / No Condition forward / No Answer Forward
	Call Screen: Incoming Call Screen (Reject or Forward Incoming Call) / Outgoing Call Screen (Blocking
	Outgoing Call)
	3-Way Conference
Protocols	TCP/IP, UDP, DHCP, RTP, HTTP, ICMP, ARP, DNS, TFTP, PPP, PPPoE
	Web-based Graphical User Interface
	Remote management over the IP Network
Configuration &	Web-based firmware upgrade
Management	Backup and Restore Configuration file
	SNMP v1/v2
	TR-069
Network and Configuration	in the second
Access Mode	Static IP, DHCP, PPPoE
Management	Web, Auto-provision
Dimension (W x D x H)	94 x 72 x 30 mm
Operating Environment	0~40 Degree C, 10~95% humidity
Devues Descriptions and	12V DC
Power Requirement	

## ORDERING INFORMATION

ATA-1505

SIP Analog Telephone Adapter (1 x LAN, 2 x RJ-11)

# **Data Sheet**

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