

Wireless Analog Telephone Adapter



Combining the cutting edge of Internet telephony and ATA manufacturing experience, PLANET now introduces the latest member of PLANET Wireless ATA family: the VIP-161SW.

To bring the most satisfaction to customers, the VIP-161SW not only provides the high quality of voice communications and wired Internet sharing capabilities but also offers Access Point (AP) function for daily wireless communication. With advanced router and VoIP DSP processor technology, the VIP-161SW is able to make calls via SIP proxy voice communications plus the IP sharing and the QoS mechanism.

The VIP-161SW is the ideal choice for Voice over IP communication and integrates Internet sharing for the daily tasks. To give most flexibility to users, the Wireless ATA provides direct analog interface for fax machine and analog telephones. Users can not only make the daily VoIP communication but also enjoy the convenience brought by FoIP communications.

With the VIP-161SW, home users and companies are able to save the cost of installation and extend their previous investments in telephones, conferences and speakerphones. The VIP-161SW equipped with two telephony interfaces, so users may register to different SIP proxy servers and establish up to 2 concurrent VoIP calls for more flexibility in the voice communications. The VIP-161SW can be the bridge between traditional analog telephones and IP network with an extremely affordable investment.

The VIP-161SW includes two Ethernet interface for Internet (PPPoE, DHCP or Fixed IP) or office LAN connection. The dual Ethernet design brings the greatest convenience when deploying VoIP network. With a built-in IEEE 802.11b/g wireless AP/CPE, the Wi-Fi ATA offers wireless connectivity via 54Mbps data transmissions.

KEY FEATURE

Product Features

- IEEE 802.11b/g compliant
- Multi-mode: AP, AP-Client Mode
- Smart QoS mechanism to ensure the voice quality
- Auto-config feature for flexible, ease-of use system integration
- NAT Router, Static Routing, Virtual Server, DMZ
- Smart QoS mechanism to ensure the voice quality
- IP ToS (IP Precedence) / DiffServ

VoIP Features

- SIP 2.0 (RFC3261) compliant
- Up to 2 concurrent VoIP calls
- Voice codec support: G.711, G.729 AB, G.723, G.276
- T.38 FAX transmission over IP network (G.711 Fax pass-through)
- In-band and out-of-band DTMF Relay (RFC 2833)
- Three-way conference calls
- Call Waiting / Forward / Transfer / Hold / Resume / Screen
- Caller ID Detection/Generation: DTMF, Bellcore, ETSI, NTT
- Voice processing: VAD, CNG, Dynamic Jitter Buffer, G.168~2000 echo cancellation

