VIP-1260PT

High Definition Color PoE Gigabit IP Phone

Intuitive, Aesthetic Design and Quality Communication

PLANET VIP-1260PT is an enterprise Gigabit IP phone with six lines and a 2.8" color LCD display. This device provides lifelike richness and high-quality voice to phone calls.

With ease of use, versatility, high-quality design, and superb audio performance, the VIP-1260PT is a reliable communication device. Additionally, its full duplex speakerphone system with HD voice supports the G.722 wideband and Opus codec, making communication between two or more parties crystal-clear, without any background noise interference.

Multi-language VoIP Telephony with Wide-Viewing-Angle Color Display

The VIP-1260PT allows you to make digital phone calls utilizing the existing broadband networks in homes and offices without installing new analog connections (e.g. copper wires). Its 2.8-inch wide-viewing-angle color display with a resolution of 320 x 240 pixels offers a clear depiction of caller's information as well as supports up to 19 languages. Compliance with IEEE 802.3af PoE standards makes deployment convenient and flexible. In addition, the advanced speaker/microphone and the dedicated Digital Signal Processor assures superior audio quality.

Highlights

- 2.8-inch WVA color LCD monitor
- 3-line key with color LED
- 6 SIP identities
- Dual Gigabit and IEEE 802.3af/at PoE compliant
- Wideband G.722 HD Opus audio
- Support G.723.1 and G.726 for efficient low-bandwidth and high-quality wideband audio
- Multi-language support via the web UI and LCD
- IPv6, VPN, VLAN, QoS, TR069 and auto-provisioning

Advantageous Applications

- Supports SIP 2.0 (RFC 3261)
- Inband, SIP info, RFC 2833 DTMF relay
- Soft keys and function keys programmable
- Echo cancellation: Supports G.168, and 96ms max. filter length in hands-free mode
- Supports voice gain setting, Voice Activity Detection (VAD) and Comfort Noise Generation (CNG)
- Full duplex hands-free speakerphone
- Hands-free headset ringing choice
- Voice codec setting for each SIP line

SIP Applications

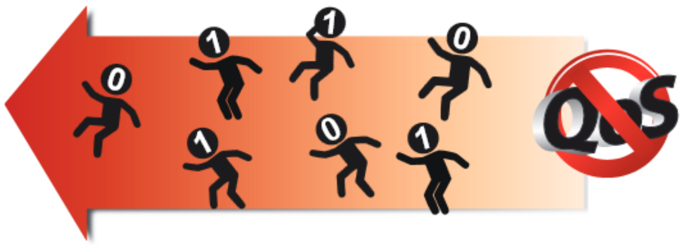
- Call forward and transfer (blind/attended)
- Call holding and waiting
- 6-way conferencing
- Paging and intercom
- Call park, call pickup and join call
- Redial and click to dial
- Automatic secondary dialing
- Incoming calls, outgoing calls and missing calls (Each supports 600 records)
- SMS and speed dial
- Phonebook up to 1000 records

Compliant with SIP 2.0

SIP phones continue to gain popularity among businesses as the preferred protocol for enhancing communication experience across IP networks. The VIP-1260PT supports Session Initiation Protocol 2.0 (RFC 3261) for easy integration with general voice over IP system. The VIP-1260PT is able to broadly interoperate with equipment supplied by VoIP infrastructure providers, thus enabling them to offer their customers better voice over IP services.

Affordable for All Businesses

The VIP-1260PT is definitely affordable for all business establishments who want flexible deployment options and expansion. It can effortlessly deliver secured toll voice quality by utilizing cutting-edge 802.1p QoS (Quality of Service) and IP ToS technology.



Call Control Features

- Flexible dial map, hotline and empty calling no. for rejected service
- Black list for rejected authenticated calls
- White list and call limit
- Do not disturb (DND)
- Caller ID display
- Dial without registration

Network Features

- IPv4 and IPv6
- Static IP/PPPoE/DHCP client
- 802.1Q VLAN (voice VLAN/data VLAN)
- VPN (L2TP) and openVPN
- Quality of Service

Maintenance and Management

- Integrated web server provides web-based administration and configuration
- Telephone keypad configuration via display menu/navigation
- Automated provisioning and upgrade via HTTPS, HTTP, TFTP
- User authentication for configuration pages
- SNTP time synchronization
- TR069
- PLANET Easy DDNS and PLANET DDNS
- PLANET Smart Discovery Utility for deployment management

High-quality G.722 HD and Opus Audio Codes

The VIP-1260PT delivers with Harman Kardon speaker, wideband G.722 HD and Opus audio codec whose both hardware and software HD functions are the next generation of voice quality for telephony audio, making the quality of voice better than that (toll quality) of the standard digital telephony, and come close to that of a room conversation. HD voice is transmitted in the audio frequency range of 50Hz to 7KHz or higher over telephone lines, resulting in higher quality voice and clearer communication. The VIP-1260PT keeps bringing the most premium sound for users.

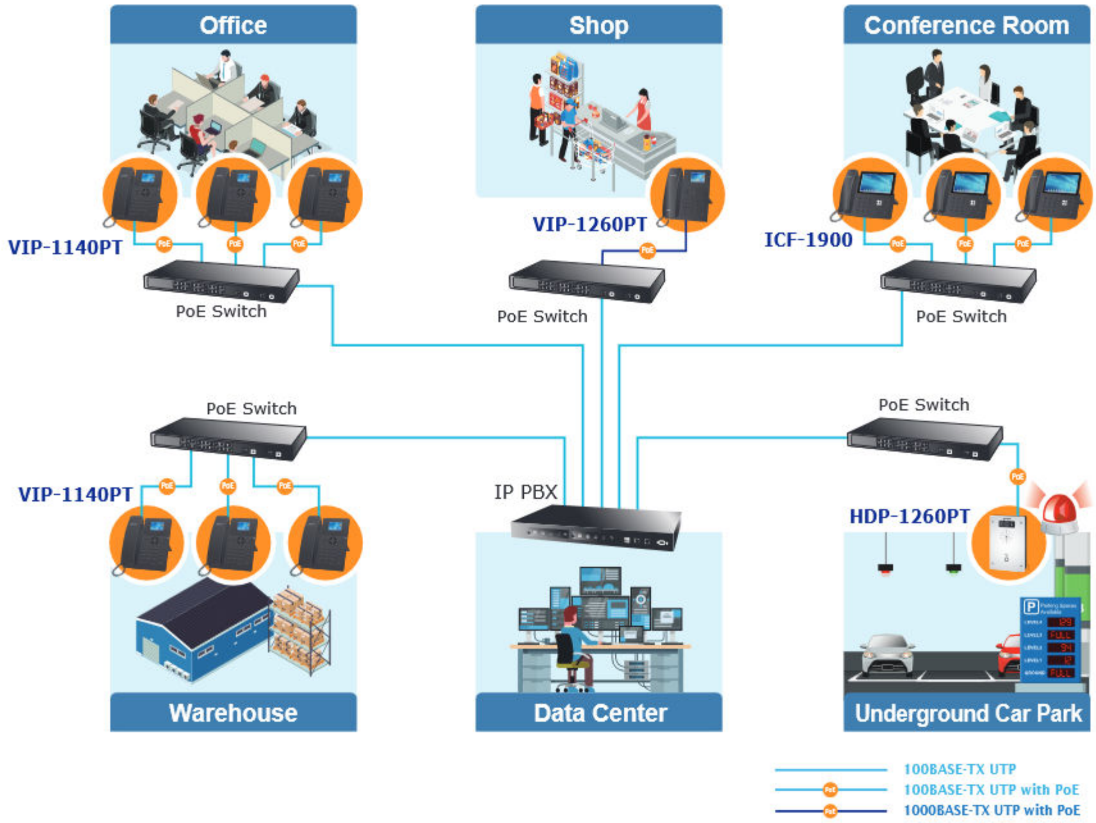
Applications

Enhanced, Full-Featured SIP IP Phone

The VIP-1260PT is optimized for executive use for administrative assistants and those working with bandwidth-intensive application on collocated PCs. Four programmable extension keys could be configured as IP PBX features like BLF, MWI, DND, Call Forward, Call Park and many others.

Enterprise IP Telephony Deployment of VIP-1260PT

The VIP-1260PT helps your teams move faster with exceptional audio quality and built-in flexibility. The VIP-1260PT is much easier to install and configure than the traditional phone system. Its low-cost and high-definition voice quality give you value for money. Based on standard SIP 2.0, it is compatible with all the standard SIP-based servers.



Specifications

Product	VIP-1260PT
Hardware	
Lines (Direct Numbers)	6 SIP Lines
Physical Interfaces	LCD x 1: 2.8-inch (320 x 240) WVA color LCD Keypad: 36 keys, including ■ 3 line keys with tri-color LED ■ 4 soft-keys ■ 5 navigation keys ■ 8 function keys ■ 12 standard phone digit keys ■ 3 volume control keys -- Up, Down, Mute (microphone) ■ 1 hands-free key HD hands-free speaker (0 ~ 7KHz) x 1 HD hands-free microphone (0 ~ 7KHz) x 1 HD handset (RJ9) x 1
Network Interfaces	RJ45 Ethernet jacket x 2: ■ Network x 1 (Gigabit, 802.3af PoE Class 1 enabled) ■ PC x 1 (Gigabit, Bridged Network)
Connectors	RJ9 phone jacket x 2: ■ Handset x 1 ■ Headphone x 1
Power Requirements	IEEE 802.3af Power over Ethernet 5V 600mA
Weight	669g
Dimensions (W x D x H)	210 x 168 x 60 mm

Protocols	
Protocols	■ SIP2.0 over UDP/TCP/TLS ■ RTP/RTCP/SRTP ■ STUN ■ DHCP ■ PPPoE ■ 802.1x ■ L2TP (basic unencryption) ■ OpenVPN ■ SNTP ■ FTP/TFTP ■ HTTP/HTTPS ■ TR069
Networking	
Networking	■ Physical: 10/100/1000Mbps Ethernet, bridged port for PC bypass ■ IP Configuration: Static, DHCP, PPPoE ■ VLAN ■ QoS ■ RTCP-XR (RFC3611), VQ-RTCPXR (RFC6035)
Security	■ Network Access Control: 802.1x ■ VPN: L2TP (basic unencryption), OpenVPN
Deployment & Maintenance	■ Auto-provisioning via FTP/TFTP/HTTP/HTTPS/DHCP/OPT66/SIP PNP/TR069 ■ Web management portal ■ Web-based packet dump ■ Configuration Import/Export ■ Auto/Manual online software upgrade ■ Syslog
Features	
Call Features	■ Call out, answer, reject ■ Mute/unmute (microphone) ■ Call hold, resume ■ Call waiting ■ Intercom ■ Caller ID display ■ Call forwarding (always/busy/no answer) ■ Call transfer (attended/unattended) ■ Call parking, pick-up (depending on server) ■ Redial ■ Do-not-disturb (per line/per phone) ■ Auto-answering (per line) ■ Voice message (on server) ■ Local 6-way conferencing ■ Hot line
Phone Features	■ Phone accessibility control by user PIN ■ Intelligent phonebook (up to 1000 entries in total) ■ Remote phonebook (XML/LDAP) ■ Intelligent contact number matching/filtering ■ Call log (600 entries in total, in/out/missed) ■ Voice message waiting indication (VMWI) ■ Network time synchronization ■ Multi-language support in screen and web UI: English, Chinese (Traditional/Simplified), Russian, Italian, German, French, Hebrew, Spanish, Catalan, Euskara, Galego, Turkish, Slovenian, Czech, Dutch, Korean, Ukrainian, Portuguese
Audio Features	■ HD voice microphone/speaker (handset/hands-free, 0~7KHz frequency response) ■ Wideband ADC/DAC 16KHz sampling ■ Narrowband codec: G.711a/u, G.723.1, G.726-32K, G.729A/B. iLBC ■ Wideband codec: Opus, G.722 ■ Full-duplex acoustic echo canceller (AEC) – 96ms tail-length in hands-free mode ■ Voice activity detection (VAD), comfort noise generation (CNG), background noise estimation (BNE), noise reduction (NR) ■ Packet loss concealment (PLC) ■ Dynamic adaptive jitter buffer up to 300ms ■ DTMF: In-band, out-of-band – DTMF-relay (RFC 2833), SIP info