VIP-1260PT

High Definition Color PoE Gigabit IP Phone

Intuitive Design Brings Quality Communication

PLANET VIP-1260PT is a new six-line SIP enterprise Gigabit IP phone with 2.8" LCD display that brings lifelike richness and voice quality to phone calls. The VIP-1260PT is a reliable communication device as it features ease of use, versatility, high-quality design and superb audio performance. Besides, its full duplex speakerphone system with HD voice can definitely make communication between two or more business parties crystal-clear with no noise interference in the background as it supports the G.722 wideband and Opus codec.

Multi-language IP Phone with Color Display

As the VIP-1260PT is a high-performance IP phone, it allows you to make calls from any location easily as long as it is online. Its 2.8-inch (main) LCD monitor comes with multi-language support via screen and web UI. It complies with IPv4/IPv6, IEEE 802.3af/at PoE interface and dual 10/100/1000Mbps Ethernet for flexible deployment and supports superior audio quality delivered by the advanced speaker and microphone system, and the digital signal processor (DSP).

Highlights

- IETE SIP compliant with Transport Layer Security
- Wideband G.722 HD and Opus audio
- 2.8-inch 320 x 240 color LCD monitor
- 6 identities/accounts with 2-line color LED key
- Dual Gigabit and IEEE 802.3af/at PoE compliant
- Multi-language support via the web UI and LCD
- IPv6, VPN, VLAN, QoS, TR069 and auto-provisioning

Advantageous Applications

- Supports SIP 2.0 (RFC3261) over UDP/TCP/TLS
- Inband, SIP info, RFC 2833 DTMF relay
- Soft keys and function keys programmable
- Echo cancellation: Supports G.168, and a maximum filter length of 96ms 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
- 3-way conferencing
- Paging and intercom
- Call park, call pickup and join call
- Redial and click to dial
- Secondary dialing automatically
- Incoming calls, outgoing calls and missing calls (Each supports 100 records.)
- SMS and speed dial
- Phonebook up to 1000 records

Call Control Features

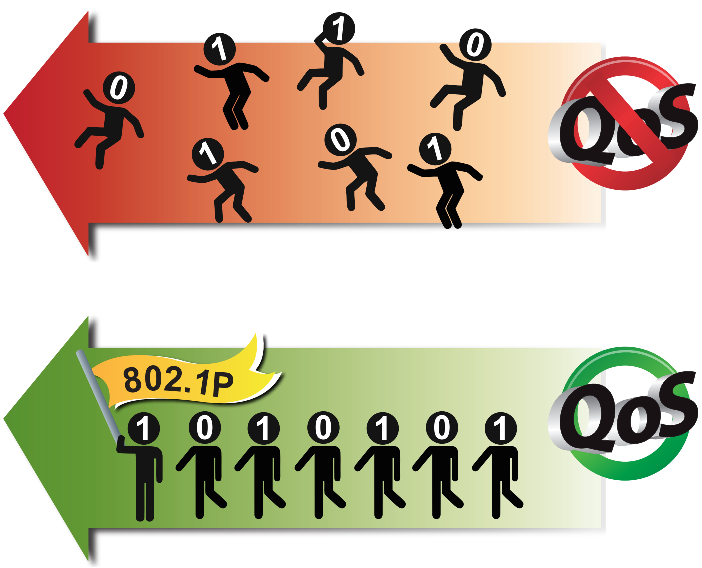
- Flexible dial map, hotline and empty calling no. for rejected service

Compliant with SIP 2.0

SIP phones continue to gain popularity among businesses as the preferred protocol for enhancing communication 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 provided by VoIP infrastructure providers, thus enabling them to provide their customers with 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 secure toll voice quality by utilizing cutting-edge 802.1p QoS (Quality of Service) and IP ToS technology.



- Black list for rejected authenticated calls
- White list and call limit
- Do not disturb (DND)
- Caller ID display
- Dial without registration

Network Features

- IPv4/IPv6
- Static /PPPoE/DHCP client
- 802.1 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

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 (HD speaker and G.722 audio codec) 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 the users.

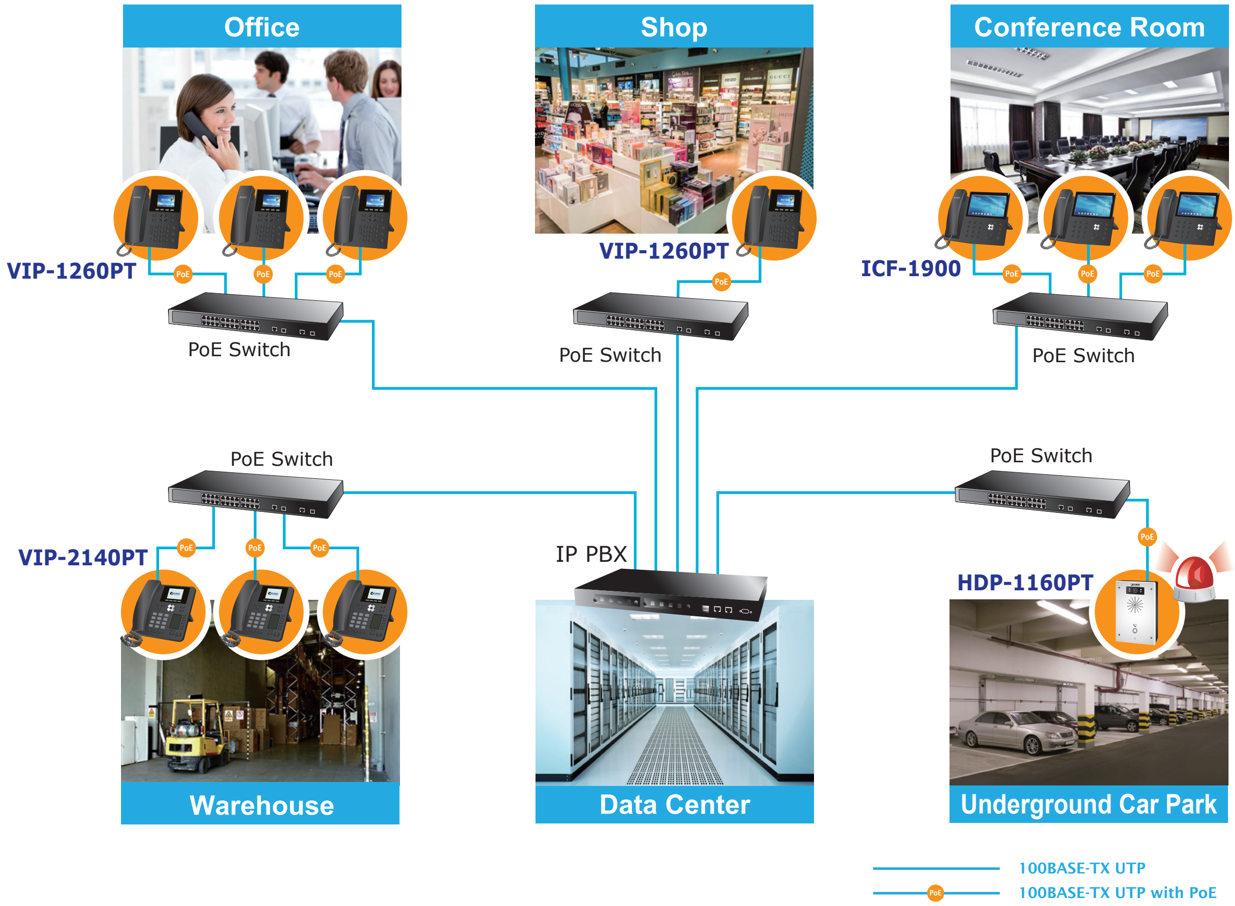
Applications

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

Help 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 Specifications	
Lines (Direct Numbers)	6 SIP Lines
Display	LCD x 1: 2.8 inch (320 x 240) color-screen LCD
Feature Keys	Keypad: 31 keys, including 4 Soft-keys 6 Function keys 5 Navigation keys 12 Standard Phone Digits keys 3 Volume Control keys, Up/Down/Mute (Microphone) 1 Hands-free key
Network Interfaces	RJ45 10/100/1000 Mbps Ethernet jacket x 2: Network x 1 (802.3af PoE Class 2 enabled) PC x 1 (Bridged Network)
Connectors	HD hands-free speaker (0 ~ 7KHz) x 1 HD hands-free microphone (0 ~ 7KHz) x 1 HD handset (RJ9) x 1
Power Requirements	IEEE 802.3af 5V 1000mA (Optional External Power Supply) Power Consumption: Idle – ~ 1.7W, Peak – ~5.7W
Weight	700g
Dimensions (W x D x H)	180 x 210 x 80 mm

Protocols and Standard	
Protocols	SIP2.0 over UDP/TCP/TLS RTP/RTCP/SRTP STUN DHCP IPv6 LLDP PPPoE 802.1x L2TP (basic unencryption) OpenVPN SNTP FTP/TFTP HTTP/HTTPS TR069 AES128 & AES256
Networking	
Networking	Physical: 10/100/1000Mbps Ethernet, dual bridged port for PC bypass IP Model: IPv4/VIPv6/VIPv4&VIPv6 IP Configuration: Static, DHCP, PPPoE Network Access Control: 802.1x VPN: L2TP (basic unencryption), OpenVPN VLAN QoS RTCP-XR (RFC3611), VQ-RTCPXR (RFC6035)
Deployment & Maintenance	Auto-provisioning via FTP/TFTP/HTTP/HTTPS/DHCP/OPT66/SIP PNP/TR069 Web management portal Web-based packet dump Configuration Export, Import Phonebook Import, Export Firmware Upgrade Syslog
Features	
Call Features	Call out, answer, reject Mute, Unmute (microphone) Call Hold, Resume Call Waiting Intercom Caller ID Display Speed Dial Anonymous Call (Hide Caller ID) 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 3-way Conference Hot Line Hot-Desking
Phone Features	Intelligent phonebook (up to 1000 entries in total) Remote phonebook (XML/LDAP, 1000 entries) Call log (100 entries in total, in/out/missed) Black/White List Call Filtering Screen saver Voice Message Waiting Indication (VMWI) Programmable DSS/Soft keys Network Time Synchronization Voice Recording with IP PBX Action URL / Active URI Multi-language support in screen and web UI: English, Chinese (Traditional/Simplified), Japanese, Russian, Italian, Turkish, German, Dutch, Spanish, Hebrew, Polish, French, etc.

Audio Features	HD voice microphone/speaker (handset/hands-free, 0~7KHz frequency response) HAC handset Wideband ADC/DAC 16KHz sampling Narrowband codec: G.711a/u, G.726-32K, G.729AB, iLBC Wideband codec: G.722, Opus Full-duplex acoustic echo canceller (AEC) 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 (RFC2833), SIP info
Environment	
Operating Temperature	0 ~ 45 degrees C
Operating Humidity	10 ~ 95% (non-condensing)
Emission	CE, FCC, RoHS

Ordering Information

VIP-1260PT	High Definition Color PoE Gigabit IP Phone

Related Products

VIP-1120PT	High Definition Color PoE IP Phone (2-line)
VIP-2140PT	High Definition Color PoE IP Phone with Dual Display (4-line)
ICF-1900	High Definition Touch Color Screen Smart Media Android SIP Conference Phone
VGW-420FS	4-Port SIP VoIP Gateway (4 FXS)
VGW-820FS	8-Port SIP VoIP Gateway (8 FXS)
VGW-1620FS	16-Port SIP VoIP Gateway (16 FXS)
VGW-2420FS	24-Port SIP VoIP Gateway (24 FXS)
VGW-3220FS	32-Port SIP VoIP Gateway (32 FXS)
VIP-462DG	802.11g SIP DECT VoIP Router
VIP-156PE	802.3af PoE SIP Analog Telephone Adapter
VIP-157S	2 FXS Analog Telephone Adapter
HDP-1160PT	720p SIP Vandalproof Door Phone with PoE
HDP-5240PT	720p SIP Multi-unit Video Door Phone with RFID and PoE
HDP-5260PT	720p SIP Multi-unit Apartment Vandalproof Door Phone with RFID and PoE
VTS-700P	7-inch SIP Indoor Touch Screen PoE Video Intercom
IPX-330	Internet Telephony PBX System (30 user registrations)
IPX-2100	Internet Telephony PBX System (100 user registrations)
IPX-2200	Internet Telephony PBX System (200 user registrations)
IPX-2500	Internet Telephony PBX System (500 user registrations)
UMG-1000	Desktop Unified Office Gateway

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