VIP-1140PT

High Definition Color PoE IP Phone

Intuitive, Aesthetic Design and Quality Communication

PLANET's VIP-1140PT is a 4-line SIP VoIP phone set featuring color display, user-friendly, aesthetic user interface design and HD voices.

It is low profile when sitting on desktop but can perfectly provide great assistance to efficiently get communications done with ease of use, quality voice and the capability of up to 6-way conferencing.

For the support of wideband audio, in addition to the basic G.722 codec, the phone set also supports IETF's Opus codec which has very low latency and significantly reduces the possibility of voice distortion or discontinuity.

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

The VIP-1140PT allows you to make digital phone calls utilizing existing broadband networks in homes and offices without installing new analog connections (e.g. copper wires). Its 2.4-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

- 2.4-inch WVA color LCD monitor
- 3-line key with color LED
- 4 SIP identities
- Wideband G.722 HD audio
- Support G.723.1 and G.726 for efficient low-bandwidth and high-quality wideband audio
- Opus codec support
- Power over Ethernet (PoE) compliant
- QoS, TR069, auto-provisioning and multi-language

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) & Comfort Noise Generation (CNG)
- Full duplex hands-free speakerphone
- Hands-free headset ringing choice
- Voice codec setting for each SIP line

SIP Applications

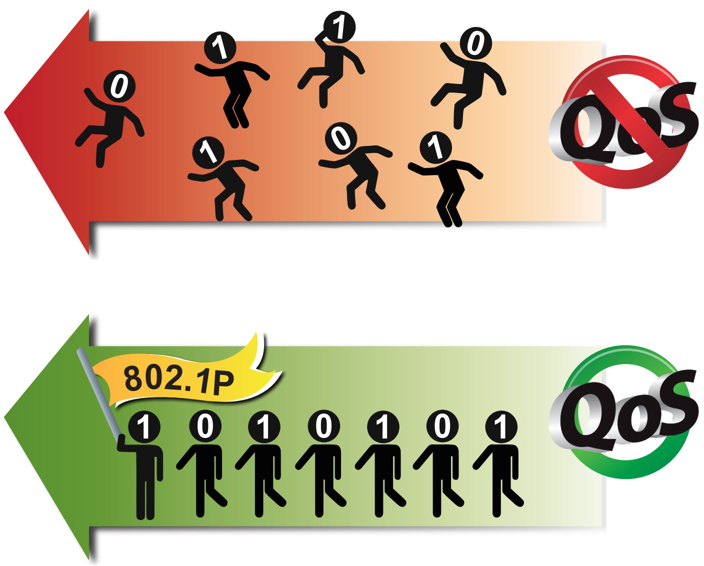
- Call forward and transfer (blind/attended)
- Call holding and waiting
- 6-way conferencing
- Paging and intercom
- Call park, call pickup and join call
- Redial and click to dial
- Automatic secondary dialing
- Incoming calls, outgoing calls and missing calls (Each supports 600 records)
- SMS and speed 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-1140PT supports Session Initiation Protocol 2.0 (RFC 3261) for easy integration with general voice over IP system. The VIP-1140PT 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-1140PT 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
- 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

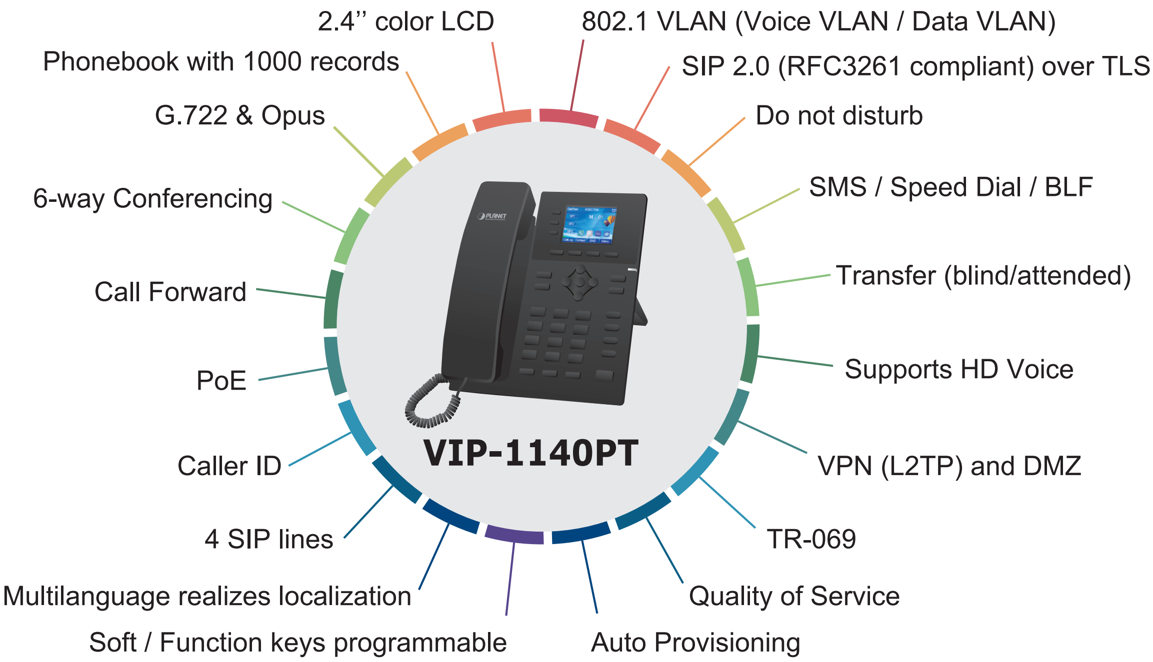
High-quality G.722 HD and Opus Audio Codes

The VIP-1140PT 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-1140PT keeps bringing the most premium sound for users.

Applications

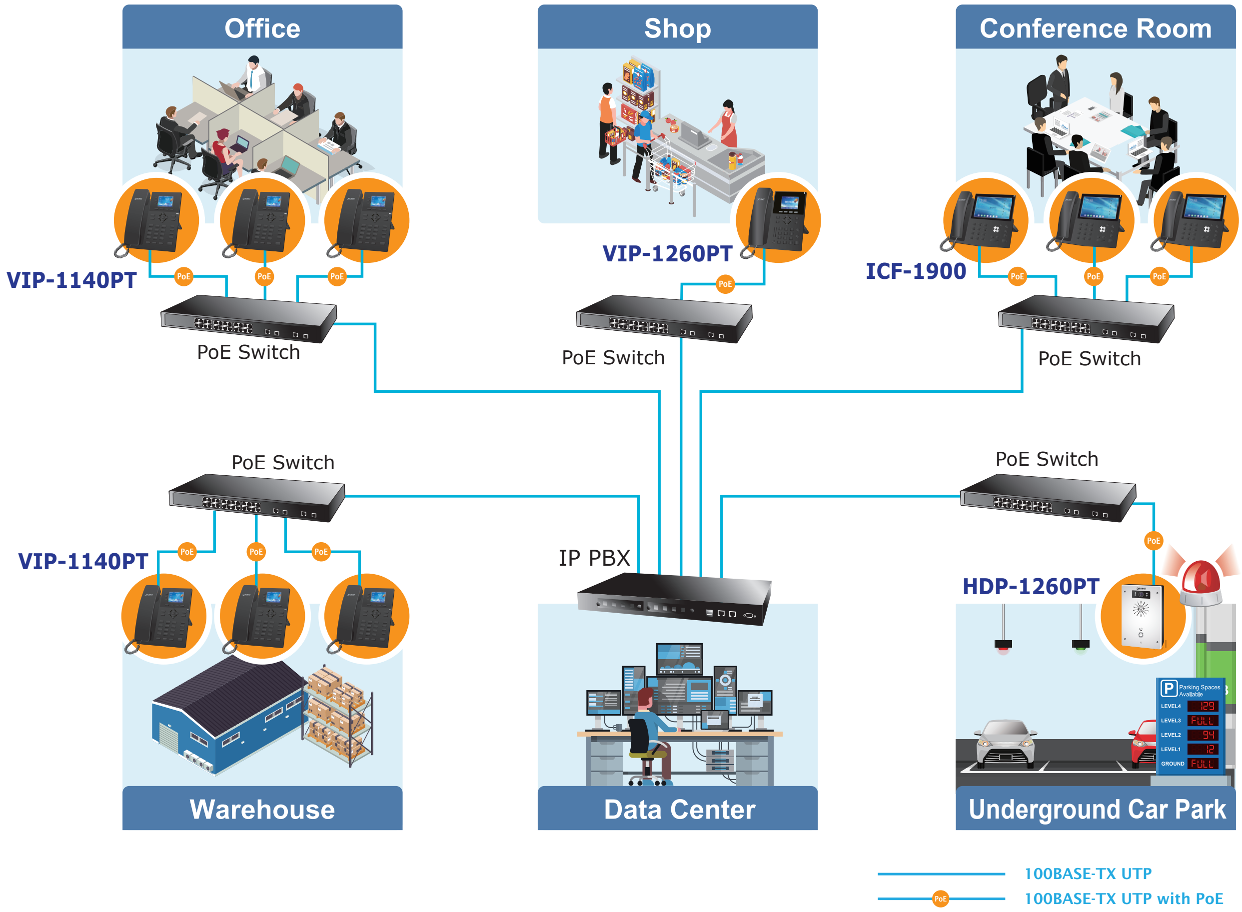
Enhanced, Full-Featured SIP IP Phone

The VIP-1140PT 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-1140PT

Help your teams move faster with exceptional audio quality and built-in flexibility. The VIP-1140PT 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-1140PT
Hardware	
Lines (Direct Numbers)	4 SIP Lines
Physical Interfaces	LCD x 1: 2.4-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
Connectors	RJ9 phone jacket x 2: ● Handset x 1 ● Headphone x 1 RJ45 10/100BASE-TX Ethernet jacket x 2: ● Network x 1 (802.3af PoE Class 1 enabled) ● PC x 1 (Bridged Network)

Power Requirements	IEEE 802.3af Power over Ethernet 5V 600mA
Weight	665g
Dimensions (W x D x H)	210 x 168 x 60 mm
Protocols and Standard	
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
Request for Comments (RFCs)	354, 1321, 1350, 1769, 1889, 1890, 2131, 2132, 2616, 2617, 2661, 2833, 2976, 3261, 3262, 3263, 3264, 3265, 3268, 3311, 3489, 3711, 4346, 4566, 5630, 5865
Networking	
Networking	● Physical: 10/100Mbps Ethernet, dual bridged port for PC bypass ● IP Configuration: Static, DHCP, PPPoE ● Network Access Control: 802.1x ● VPN: L2TP (basic unencryption), OpenVPN ● VLAN ● QoS
Deployment & Maintenance	● Auto-provisioning via FTP/TFTP/HTTP/HTTPS/DHCP/OPT66/SIP PNP/TR069 ● Web management portal ● Web-based packet dump ● Configuration backup/restore ● 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 500 entries in total) ● Remote phonebook (XML/LDAP) ● Intelligent contact number matching/filtering ● Call log (300 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
Environment	
Operating Temperature	0 ~ 45 degrees C
Operating Humidity	10 ~ 65% (non-condensing)
Emission	CE, FCC, RoHS

Ordering Information

VIP-1140PT	High Definition Color PoE IP Phone (4-Line)

Related Products

VIP-1120PT	High Definition PoE IP Phone (2-Line)
VIP-1260PT	High Definition Color PoE Gigabit IP Phone (6-Line)
HDP-1260PT	1080p SIP Vandalproof Door Phone with RFID and 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
ICF-1900	High Definition Touch Color Screen Smart Media Android SIP Conference Phone
IPX-330	Internet Telephony PBX System (30 user registrations)
IPX-1800N	Internet Telephony PBX System with ISDN Support
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)
VGW-400 series	4-port SIP VoIP Gateway
VGW-800 series	8-port SIP VoIP Gateway
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)
GSW-2620VHP	24-port 10/100/1000T 802.3at PoE + 2-port Gigabit SFP Ethernet Switch with LCD PoE Monitor (300W)
FGSW-2622VHP	24-port 10/100TX 802.3at PoE + 2-port Gigabit TP/SFP Combo Ethernet Switch with LCD PoE Monitor (300W)

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