VIP-5060PT

Professional HD PoE IP Phone (6-Line)

Cost-effective, High-performance PoE VoIP Phone

To build high-performance VoIP communications at a low cost, PLANET has launched a new member of its IP Phone family, the VIP-5060PT enterprise-class 6-Line PoE IP Phone. It complies with IEEE 802.3af PoE interface for flexible deployment. The VIP-5060PT makes it simple for the enterprise featuring voice and data system or expanding voice system to new locations. It helps the company to save money on long distance calls; for example, the remote workers can dial in through a Unified VoIP Communication System just like an extension call but no long distance call charge would occur. The VIP-5060PT also allows call to be transferred to anyone at any location within the voice system, which enables the enterprise to communicate more effectively and is helpful to streamline business processes.

Highlights

- Dual 10/100/1000 Gigabit Ethernet
- Supports SIP 2.0 (RFC3261)
- Supports six SIP voice lines
- IEEE 802.3af Power over Ethernet compliant
- Supports multiple road calls waiting in line
- Supports HD voice
- Supports SRTP (Secure Real-time Transport Protocol) and Busy Lamp Field (BLF)
- Supports 5 extension consoles; max. 130 definable keys

Advanced Features

- SIP supports SIP domain, SIP authentication (none, basic, MD5), DNS name of server, Peer to Peer / IP call
- Inband, SIP info, RFC 2833 DTMF Relay
- 9 kinds of ring types and 3 user-defined music rings
- Large dot matrix LCD display and soft keys make user easier to use
- Soft keys and function keys programmable
- Multilanguage realizes localization
- Echo cancellation: Supports G.168, and hands-free can support 96ms
- Full duplex hands-free speaker phone
- Hands-free headset ringing choice
- Supports Voice Gain Setting, VAD, CNG
- Voice codec setting for each SIP line

SIP Applications

- Call forward / Transfer (blind/attended)
- Call Holding / Waiting
- 3-way conference
- Paging and Intercom
- Call park / Call pickup / Join call
- Redial and click to dial
- Secondary dialing automatically
- Incoming calls / outgoing calls / missing calls (Each supports 100 records)
- SMS and Speed Dial
- Phonebook up to 500 records
- XML phonebook / browser

High Quality HD VoIP Voice

The VIP-5060PT delivers HD voice (High-Definition Voice) which is 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 even close to that of a room conversation. HD voice is transmitted in the audio frequency range of 50 Hz to 7 kHz or higher over telephone lines, resulting in higher quality voice and clearer communication.

Standard Compliance

The VIP-5060PT supports Session Initiation Protocol 2.0 (RFC 3261) for easy integration with general voice over IP system. The VIP-5060PT is able to broadly interoperate with equipment provided by VoIP infrastructure providers, thus enabling them to provide their customers with better multi-media exchange services.

Compliant with standard SIP RFC 3261

Enhanced, Full-Featured Business IP Phone

The VIP-5060PT is a full-featured enhanced business IP Phone that addresses the communication needs of the enterprises. It provides 6 voice lines and dual 10/100/1000Mbps Ethernet. Furthermore, the VIP-5060PT delivers user-friendly design containing a 128 x 64 LCD with white backlight, 4 Line keys and 4 soft keys. It supports 5 extension consoles with each consisting of 26 keys.

The VIP-5060PT supports all kinds of SIP based phone features including Call Waiting, Auto Answer, Music on Hold, Caller ID and Call Waiting ID, 3-way Conferencing, Call Hold, Call Forwarding, Black List, DTMF Relay, In-Band, Out-of-Band (RFC 2833) and SIP INFO, among others. Besides office use, the VIP-5060PT is also the ideal solution for VoIP service offered by Internet Telephony Service Provider (ITSP).

Call Control Features

- Flexible dial map / Hotline / Empty calling no.
- Reject service / Black list for reject authenticated call
- White list / Limit call
- Do not disturb (DND)
- Caller ID / CLIR (reject the anonymous call) / CLIP (make a call with anonymous)
- Dial without register

Network Features

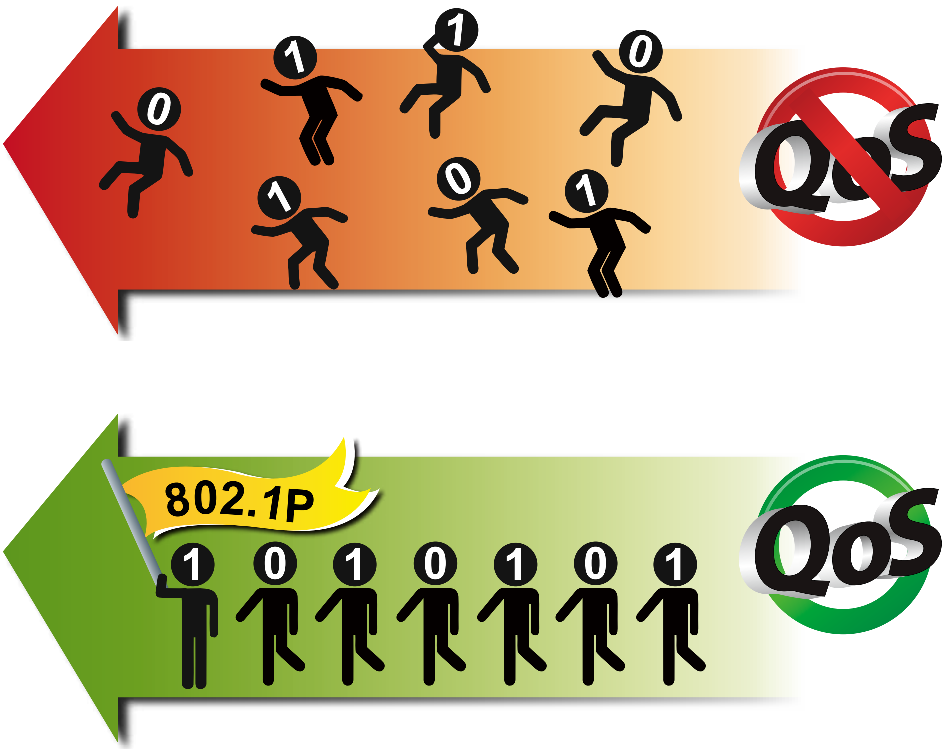
- Route and Bridge modes
- PPPoE / DHCP client on WAN
- 802.1 VLAN (voice VLAN / data VLAN)
- VPN (L2TP) and DMZ
- Main DNS and secondary DNS server
- DNS Relay, SNTP Client, Firewall, openVPN

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
- Local and Remote Syslog (RFC 3164)
- SNTP Time Synchronization
- TR069

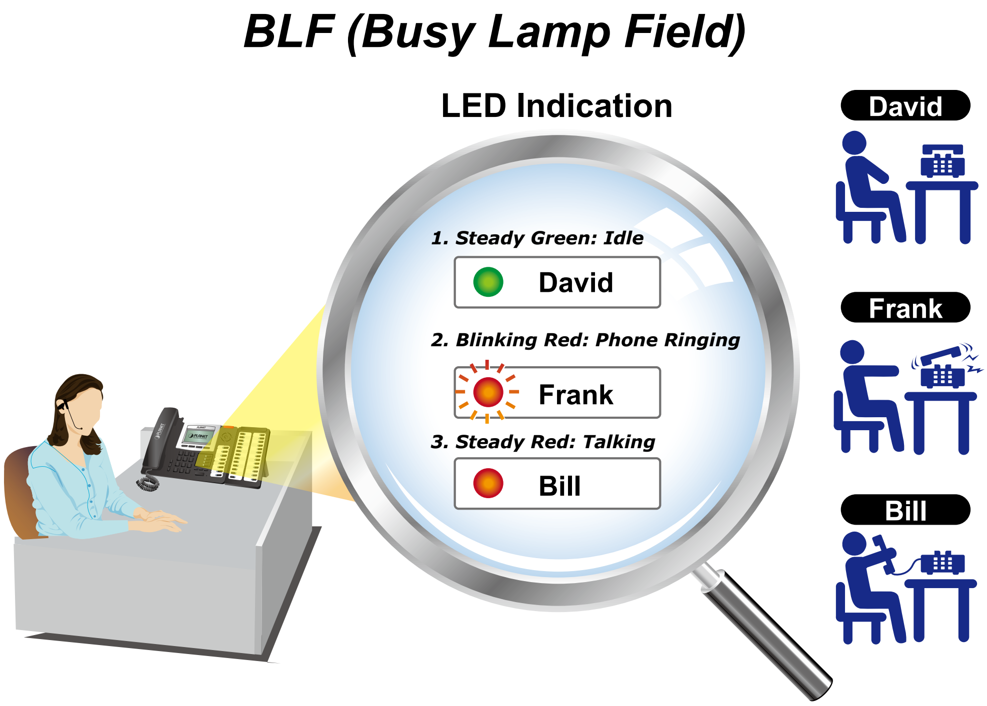
Secure, High-Quality VoIP Communication

The VIP-5060PT can effortlessly deliver secured toll voice quality by utilizing cutting-edge 802.1p QoS (Quality of Service), 802.1Q VLAN tagging, and IP TOS (Type of Service) technology. Using voice and data VLAN can easily separate the data and voice, thus maintaining the best quality.



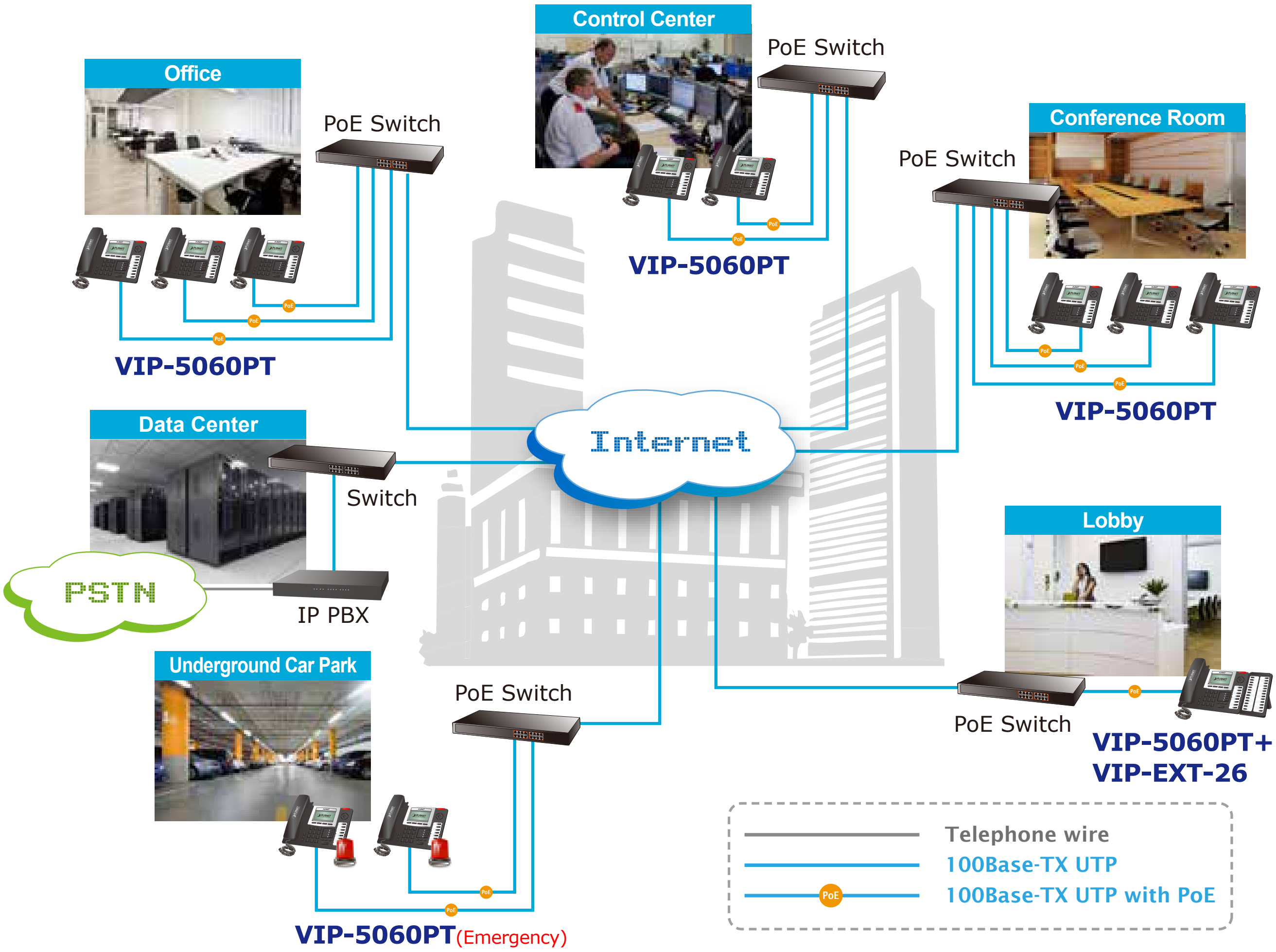
Professional Application

The VIP-5060PT supports Busy Lamp Field (BLF) function that, via the lights on the phone, enables users to easily identify the status of other phones which are connected to the same IP PBX, such as busy, idle, ringing, etc. The connected IP PBX must also support BLF feature. The BLF function is helpful for a receptionist on the front desk to route all incoming calls smoothly.



Applications

Professional IP Telephony System Deployment with the VIP-5060PT



Specifications

Model	VIP-5060PT
Hardware	
Lines (Direct Numbers)	6-Line enterprise-class IP phone
Display	80 x 43 mm / 128 x 64 pixel LCD with blue backlight
Feature Keys	4 line keys 8 DSS keys 4 Soft Keys 12 dialing buttons (0~9, *, #) 12 fixed function buttons
Network Interfaces	2 x 10/100/1000Base-T RJ-45 Auto Negotiation, Auto MDI Network-port with 802.3af PoE support
Protocols and Standard	
Data Networking	MAC Address (IEEE 802.3) IPv4 (RFC 791) Address Resolution Protocol (ARP) DNS: A record (RFC 1706), SRV record (RFC 2782) Dynamic Host Configuration Protocol (DHCP) client (RFC 2131) Internet Control Message Protocol (ICMP) (RFC 792) TCP (RFC 793) User Datagram Protocol UDP (RFC 768) Real Time Protocol RTP (RFC 1889, 1890) Real Time Control Protocol (RTCP) (RFC 1889) Differentiated Services (DiffServ) (RFC 2475) Type of service (ToS) (RFC 791, 1349) VLAN tagging 802.1p Layer 2 quality of service (QoS) Simple Network Time Protocol (SNTP) (RFC 2030) Backward compatible with RFC 2543 Session Timer (RFC 4028) SDP (RFC 2327) NAPTR for SIP URI Lookup (RFC 2915)
Voice Gateway	SIP version 2 (RFC 3261, 3262, 3263, 3264) SIP supported in STUN (RFC 3489) Message Waiting Indicator (RFC 3842) Voice algorithms: - G.711 (A-law and μ-law) - G.723.1 high/low - G.729a/b - G.722.1 (HD Voice) - G.726 Dual-Tone Multi-Frequency (DTMF), In-Band and Out-of-Band (RFC 2833) (SIP INFO) Voice Activity Detection (VAD) with Silence Suppression Adaptive Jitter Buffer Management Comfort Noise Generation Echo Cancellation Message
Features	
Advanced Features	SIP 2.0 (RFC3261) IEEE 802.3af Power over Ethernet (PoE) compliant Multiple road call waiting in line Supports HD voice Supports SRTP and BLF SIP supports SIP domain, SIP authentication (none, basic, MD5), DNS name of server, Peer to Peer / IP call Inband, SIP info, RFC 2833 DTMF Relay 9 kinds of ring types and 3 user-defined music rings Large dot matrix LCD display and soft keys make user easier to use Supports 5 extension consoles with each consisting of 26 keys Soft keys programmable; function keys programmable Multilanguage realizes localization Echo cancellation: Support G.168, and Hands-free can support 96ms Full duplex hands-free speaker phone Hands-free headset ringing choice Supports Voice Gain Setting, VAD, CNG Voice codec setting for each SIP line

	Call forward / Transfer (blind/attended) Call Holding / Waiting 3-way conference Paging and Intercom Call park / Call pickup / Join call Redial and click to dial Secondary dialing automatically Incoming calls / outgoing calls / missing calls (Each supports 100 records) SMS and Speed Dial Phonebook for 500 records XML phonebook/browser
Call Control Features	Flexible dial map / Hotline / Empty calling no. Reject service / Black list for reject authenticated call White list / Limit call Do not disturb (DND) Caller ID / CLIR (reject the anonymous call) / CLIP (make a call with anonymous) Dial without register
Network Features	Route and Bridge modes PPPoE / DHCP client on WAN 802.1 VLAN (voice VLAN / data VLAN) VPN (L2TP) and DMZ Main DNS and secondary DNS server DNS Relay, SNTP Client, Firewall, openVPN
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 Local and Remote Syslog (RFC 3164) SNTP Time Synchronization TR069
Environments	
Power Requirements	5V DC, 1A IEEE 802.3af Power over Ethernet
Operating Temperature	0 ~ 40 degrees C
Operating Humidity	10 ~ 65% (non-condensing)
Weight	990 g
Dimensions (W x D x H)	290 x 260 x 60 mm
Emission	CE, FCC, RoHS
Connectors	Two 10/100/1000 BASE-T RJ-45 Ethernet ports Handset: RJ-9 connector Headset: RJ-9 connector RJ-11 Ext. connector DC power jack Built-in speakerphone and microphone

Ordering Information

VIP-5060PT	Professional HD PoE IP Phone (6-Line)

Accessories

VIP-EXT-26	Expansion Module for VIP-2020PT/VIP-5050PT

Related Products

VIP-2020PT	Enterprise HD PoE IP Phone (2-Line)
VIP-362WT	802.11n Wireless Desktop IP Phone
VIP-256PT	802.3af PoE SIP IP Phone
VIP-156	SIP Analog Telephone Adapter
VIP-156PE	802.3af PoE SIP Analog Telephone Adapter
VIP-157	1 FXS / 1 FXO SIP Analog Telephone Adapter
VIP-157S	2 FXS Analog Telephone Adapter
ICF-1700	Touch Screen Internet Multimedia Phone
IPX-330	Internet Telephony PBX System (30 user registrations)
IPX-2100	Internet Telephony PBX System (100 user registrations)
UMG-1000	Desktop Unified Office Gateway
UMG-2200	Unified Office Gateway (8-Port FXO)
VIP-281 series	2-Port FXS H.323 / SIP / GSM VoIP Gateway
VIP-480 series	4-Port FXS H.323 / SIP VoIP Gateway
VIP-880 series	8-Port FXS H.323 / SIP VoIP Gateway
VIP-1680 Series	16-Port FXS H.323 / SIP VoIP Gateway
VIP-2480 Series	24-Port FXS H.323 / SIP VoIP Gateway
VGW-402	4-Port SIP VoIP Gateway (2*FXS + 2*FXO)
VGW-400FS	4-Port SIP VoIP Gateway (4*FXS)
VGW-400FO	4-Port SIP VoIP Gateway (4FXO)

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