PLANET
Networking & Communication

IPX-2000

Internet Telephony PBX system

PLANET IPX-2000 is the next generation voice communication platform for the small to medium enterprises. Designed as an open, scalable and highly reliable telephony solution, the IPX-2000 is able to accept 200 extension registrations, and effectively scales from under 100 users to as many as 200 in a standard rack-mountable unit. The PLANET IPX-2000 is also designed to operate on a variety of VoIP applications; it provides centralized call control, auto-attendant, voice conferencing, and PSTN access, digital and IP-based communications.

The IPX-2000 integrates up to 8 calls via the IPX-FXO (Foreign eXchange Office, FXO) module to become a feature-rich PBX system that supports seamless communications between existing PSTN calls, analog, IP phones and SIP-based endpoints.

Moreover, the IPX-2000 also integrates telephony call processing, call control, voice mail, and a widely PBX application programming interface into a highly scalable architecture designed to support both traditional circuit-based and the Internet telephony service within a distributed enterprise communications network.

With the IPX-2000, standard SIP phones can be easily integrated in your office, plus the auto-config feature, you may integrate our IP phone VIP-254 series, and the analog telephone adapter - VIP-156/157 series to build up the VoIP network deployment in minutes.

Allowing distributed IP technology to meet traditional voice services, with the IPX-2000 proactive management interface in the daily business process, it brings enterprises higher employee productivity and customer satisfaction.

Key Features

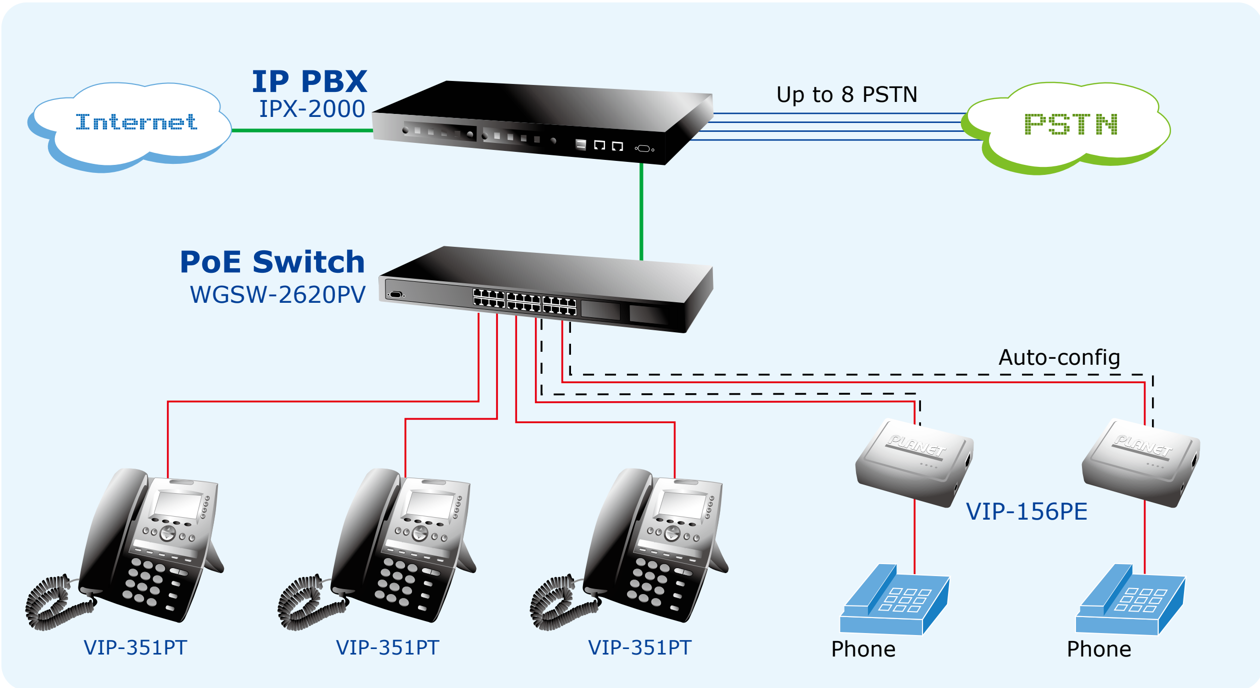
SYSTEM HIGHLIGHT

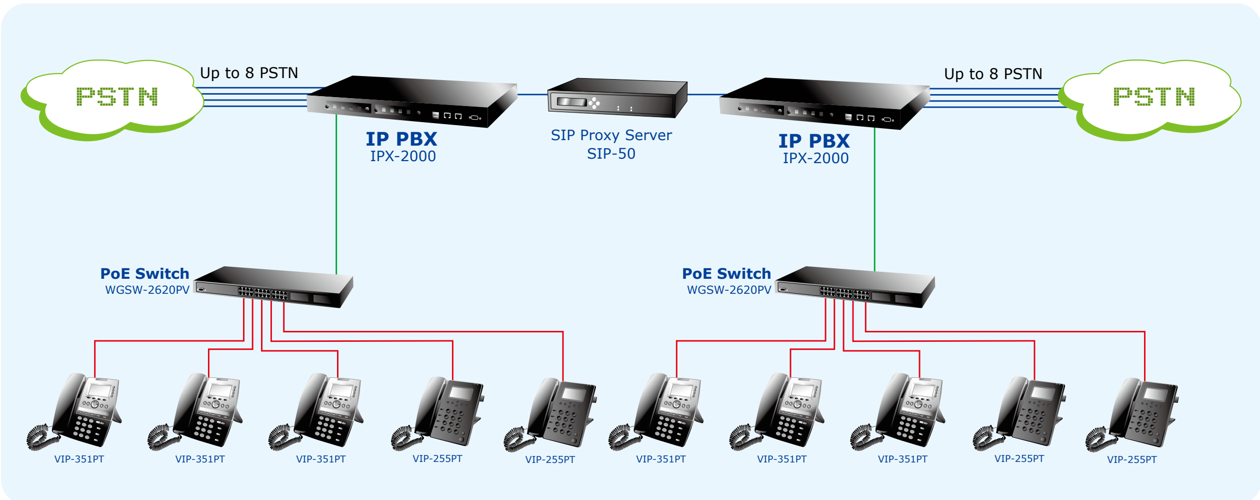
- 200 max users/extensions with voicemail account
- 50 max concurrent sessions
- Highly integrated, embedded system for stability
- Immediate VoIP connections
- Analog interface (FXO/FXS) support
- Flexible dialing plans
- On-ramp/off-ramp calls for VoIP and PSTN
- Remote management
- Wizard for easy configuration
- Command Line Interface (CLI) for quick-batch configuration
- Seamless integration with legacy PBX
- Auto provision and auto firmware upgrade for feature IP phones

FEATURE HIGHLIGHT

- SIP registrar and SIP proxy
- Multi-language voice prompts for international business
- Customizable 3-layer IVR
- ACD (Automatic Call Distribution) to form basic Call Centers
- Stackable design for scalability and investment protection
- UMS (Unified Messaging System) support
- Meet-me conference for virtual meeting room
- System status and system log
- Call Detail Record (CDR)
- Built-in STUN client
- Function-rich voicemail System
- CAC (Call Admission Control)
- Call keep alive

Applications





Specification

Product	Internet Telephony PBX system
Model	IPX-2000
Hardware	
Interface	One RJ-45 10/100 base-TX WAN Ethernet port One RJ-45 10/100 base-TX LAN Ethernet port Two USB 2.0 ports One RS-232 serial port Two expandable PCI interface slots
Standards and protocol	
Registration	Max. 200 nodes / SIP IP phones
Calls	Max. 50 concurrent calls
Call control	RFC 3261, RFC 3311, RFC 3515, RFC 3265, RFC 3892, RFC 3361, RFC 3842, RFC 3389, RFC 3489, RFC 3428, RFC 2327, RFC 2833, RFC 2976, RFC 3263, RFC 3264, RFC 3362, RFC 4612
SIP Registrar	Static/Dynamic registration, Configurable Expiry Time, MD5 authentication Handle loose RFC-compliant SIP devices, Resilient message retry mechanism Cache client registrations
SIP Proxy	Stateful proxy server, NAT traversal for clients, Inter-proxy call hand-off Outbound Proxy behind NAT Device
PBX System	
Call features	Codec G.711 (μ/A-law), G.723.1 (6.3k/5.3k bit/s), G.729A, and G.726 (16k/24k/32k/40k bit/s) supported Transcoding channel 0~16, subject to add-on card In-band/RFC2833/SIP-INFO DTMF translation Two expandable slots for telephony interfaces 50 SIP trunks for ITSP account or private trunking shared by extensions 200 DID SIP trunks to extensions Support gateway trunk mode per SIP trunk Enable/Disable NAT Traversal per SIP trunk Call admission control of call count or bandwidth per SIP trunk Long call audit Support SIP OPTIONS keep alive NAT session keep alive Configurable RFC 2833 payload type per SIP trunk FXS/FXO analog trunking FXO disconnection tone detection FXO disconnection tone parameter setting FXS hot line FXS warm line Caller ID detection Trunk hunting Digits manipulation during hunting Life-line priority call Support SIP Call Hold, Call Waiting Support SIP phone 3-way conference Support Blind/ Attended Transfer In-line Call Transfer Unconditional, Unavailable, Busy Call forward Call Back on Busy between extensions Per calling number forward and rejection Blacklist of number patterns 32 call pick-up groups Call Park and Retrieve Recording on demand with PLANET IP phones Remote extension registration via Internet Direct line to extension (DID to Extension) Direct line by called number (DID by Number) Direct line by privilege (DID by Privilege)

	Echo Cancellation (G.168) Flexible numbering plan Call privilege grouping Configurable Music on Hold Memo Call for extension Schedule-based Broadcast Support T.38 FAX over IP Support T.30, T.38 FAX pass through ENUM resolution
NAT	Auto NAT discovery and traversal Built-in STUN client RTP proxy RTP port range designation
IVR	50 configurations of 3-layer IVR Worktime/Holiday setting for different IVR Configurable greeting prompts Music on Ringing extensions Forward to Voice Mail on No-answer Support 3 languages in IVR tree Hot key to operator
Voice	User Authentication by PIN Multilingual, 3 languages Multi-folder Archive Fast-forward /Rewind /Undelete MWI notification VMWI notification E-mail notification and attachment (Unified messaging) Personal greeting on unavailability and busy Record personal greeting through phone Voicemail Forwarding Reply call or new call after logged in Voicemail menu Built-in 40GB hard disk drive for Voicemail Support USB 2.0 interface for Voicemail, CDR, and system configuration backup Support NFS remote backup for Voicemail, CDR, and system configuration
Meet-me conference	24 conference rooms with configurable number and PIN Up to 24 parties among all conference rooms Lock/Mute/Join/Drop control for administrator Music on First Dial-in Party Hot key to leave the conference Hot key for administrator to manage the conference
Automatic Call Distribution	32 queues with 32 agents among all queues. 32 inbound call among all queues Configurable waiting length for individual queue Support five distribution policies including round robin, ring all, least recent, fewest call, and random Configurable waiting time for each queue Allow agent remotely log-in Agent can participate multiple queues Agent phones also allow extension calls
Stackable	Support LAN stacking up to 4 units in the same model Automatic intra-trunking creation among stacking units Automatic configuration publishing from Master to Slaves Automatic load balancing in hosting feature phones3
Internet Sharing	
System management	Web-based configuration with session control User and administrator configuration mode Automatic expiring the idle sessions Support firmware upgrade through the Internet Configuration Wizard for mass extensions and users creation Step-by-Step Wizard for adding users, extensions and trunks Built in online help in wizard Command Line Interface (CLI) for configuration System event Syslog

Network management	DHCP/PPPoE/Static IP on WAN Support MAC Clone on WAN Static LAN routing Firewall on predefined services Virtual Server for client device NAT for outbound traffic from LAN WAN QoS queuing mechanism for VoIP and data traffic Support TOS setting DNS forwarder and dynamic DNS SNMPv2 with standard MIB format Adaptive WAN bandwidth and DSP channel saving
Environment	
Operating Temperature	Operating temperature 0~50% Storage temperature -10~70% Humidity (RH) 10~80% non-condensing
Power Requirement	100~240V AC, 50~60 Hz
EMC/EMI	CE, FCC

Ordering Information

IPX-2000	Internet Telephony PBX system (200 user registrations, 2 slots)
IPX-FXO	4-Port FXO module for IPX-2000
IPX-FSL	4-Port Life-Line module for IPX-2000 (3FXO+1FXS)

Available Modules

VIP-156	SIP Analog Telephone Adapter (2 x RJ-45, 1 x RJ-11)
VIP-156PE	SIP Analog Telephone Adapter with 802.3af splitter built-in (2 x RJ-45, 1 x RJ-11)
VIP-157S	2-Port SIP Analog Telephone Adapter (2 x RJ-45, 2x RJ-11)
VIP-254T	SIP IP Phone
VIP-255PT	SIP PoE IP Phone
VIP-351PT	4-Line Enterprise PoE IP Phone

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