

Internet Telephony PBX System



PLANET IPX-1800 series IP PBX system are designed and optimized for the SMB, and SOHO daily communications. The IPX-1800 is the next generation voice communication platform for the small to medium enterprise. Designed as an open, scalable, and highly reliable telephony solution, the IPX-1803 is able to accept 30 extension registrations, and effectively fulfills the demands from SOHO to enterprise. Designed to run on a variety of VoIP applications, the IPX-1803 provides centralized call control, auto-attendant, voice conferencing, PSTN access and IP-based communications. The IPX-1800 series divided into two models: the IPX-1803 integrates up to 4 telephony interfaces, including 3 FXO (Foreign eXchange Office, FXO), 1 FXS (Foreign eXchange Station, FXS); the FXS interface in IPX-1803 provides lifeline functionality; the IPX-1804 integrates 4 FXO telephony interfaces to become a feature-rich PBX system that supports seamless communications between existing PSTN calls, analog, IP phones and SIP-based endpoints.

The IPX-1803 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 IPX-1803, standard SIP phones can be easily integrated in your office; plus the auto-config feature, you may integrate our IP Phone VIP-153T/VIP-153PT, and the ATA (analog telephone adapter) series - VIP-156/VIP-157 to build up the VoIP network deployment in minutes. Allowing distributed IP technology to meet traditional voice services, with proactive management interface, the IPX-1803 in the daily business processes, enterprises can make people more productive, more intelligent tasks, and more customer satisfaction.

KEY FEATURE

SIP features

- Static/Dynamic registration
- Call-based MD5 authentication
- NAT traversal for clients
- Outbound proxy with or without WAN
- SIP trunks for inter-PBX SIP trunking
- Inter-proxy call hand-off
- Auto-config for SIP IP Phones /ATA
- 30 registration/30 voicemail/10 concurrent calls
- Auto NAT Discovery and Traversal
- Built-in STUN Client
- RTP Proxy
- RTP Port Range Designation

Relational Provision

- Logical Partition/Relation on Users and Trunks
- Logical Provision on Outgoing and Incoming Calling Search Scopes
- Rich Dial-Plan Expressiveness thru Route Patterns
- User Privilege Propagation Over Intra-Trunks
- Object-Oriented Provisioning Paradigm

PBX Features

- Support call hold, call waiting, 3-way call conference with feature phones
- Built-in in-line call transfer
- Unconditional, unavailable, busy call forward
- Per-calling-number forward and rejection
- Call Privilege Grouping
- Group-based call pick-up
- Call-parking
- Multi-room meet-me conference
- FXO disconnection tone detection
- In-band/RFC2833/SIP-INFO DTMF Translation
- QoS support
- Lifeline/Emergency call support
- Music on hold
- Outbound 900/0204 blocking
- Blacklist of Number Patterns

Auto Attendant

- Configurable IVR prompts
- Key to reach operator
- Timeout interval and timeout action
- Music on ringing extensions
- Forward to voice mail on no-answer

Voice Mail

- User PIN
- 450 minutes recording time
- MWI notification
- E-mail notification
- Personal reception on unavailability and busy
- Voicemail forwarding
- Voicemail to email

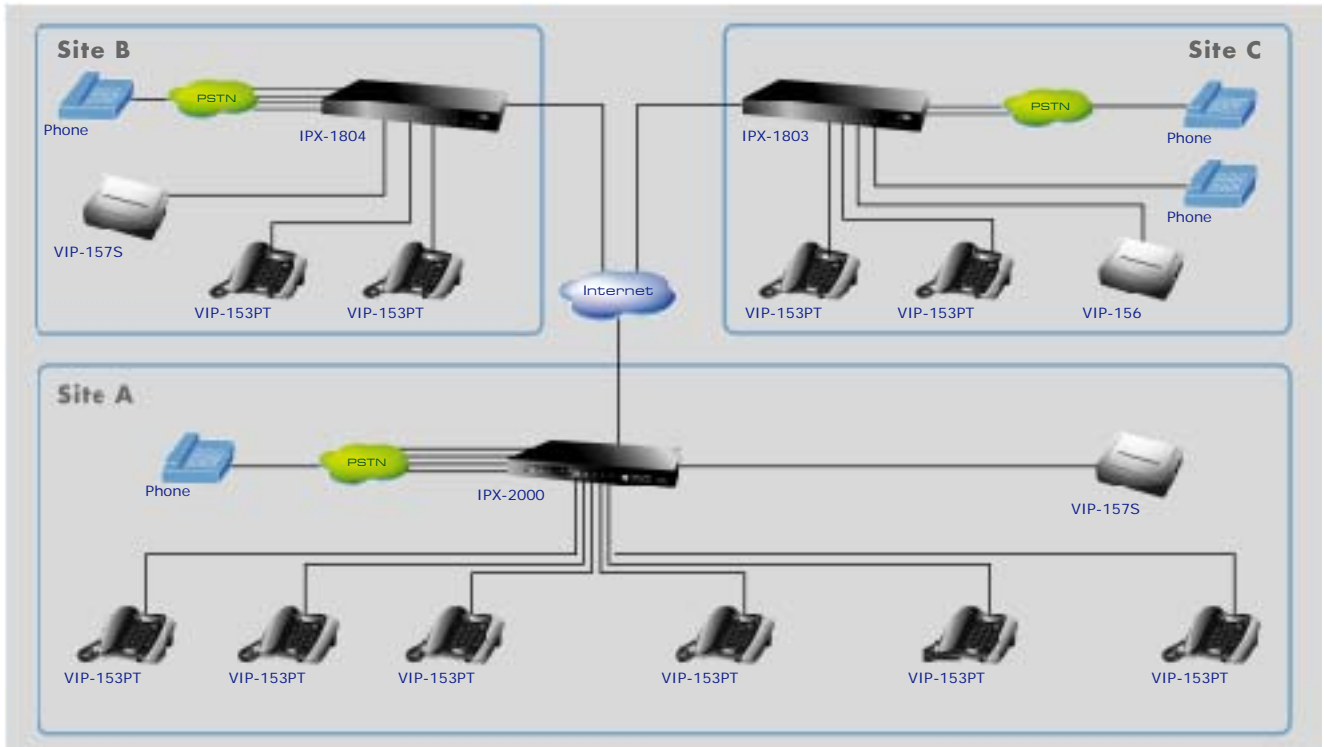
SPECIFICATION

Product **Internet Telephony PBX System**

Model	IPX-1803	IPX-1804
Hardware		
LAN	4 (10Base-T/100Base-TX, Auto-Negotiation)	
WAN	1 (10Base-T/100Base-TX, Auto-Negotiation)	
Telephony ports	3 x FXO, 1 x FXS	4 x FXO
Standards and protocol		
Call control	SIP 2.0 (RFC3261)	
Registration	Max. 30 nodes/SIP IP phones	
Calls	Max. 10 concurrent calls	
Voice CODEC support	G.711, G.726, GSM, G.723.1 (5.3, 6.3kbps), G.729A (8kbps)	
Voice processing	Voice Active Detection,	
PBX features	DTMF detection/generation, G.165/G.168 echo cancellation (ECN) (25 ms.), Comfort noise generation (CNG), Gain Control Support call hold, call waiting, 3-way call conference with feature phones Built-in in-line call transfer Unconditional, unavailable, busy call forward Per-calling-number forward and rejection Group-based call pick-up Call-parking Inter-PBX SIP trunking Multi-room meet-me conference Auto-attendant Voice mail system Call privilege grouping FXO interface for PSTN Inbound/outbound FXO disconnection tone detection FXO hunt group Caller ID detection Echo cancellation In-band/RFC2833/SIP-INFO DTMF translation Music on hold Direct lineOutbound 900/0204 blocking	
Auto Attendant Features	Configurable IVR prompts Key to reach operator Timeout interval and timeout action Music on ringing extensions Forward to voice mail on no-answer	
VoiceMail Features	User PIN Multilingual Multi-folder archive Fast-forward/Rewind/Undelete MWI notification E-mail notification and attachment (Unified messaging) Personal reception on unavailability and busy Voicemail forwarding Voicemail to email Reply call or new call in voicemail menu	
Internet Sharing		
Protocol	TCP/IP, NAT, DHCP, HTTPs, DNS	
Advanced Function	QoS	
Management	HTTP (LAN access). HTTPs (WAN, LAN access)	
Environment		
Operating Temperature	0~50 Degree C, 10~90% Humidity	
Power Requirement	100-240VAC, 50/60Hz	
EMC/EMI	CE, FCC	
Network and Configuration		
Connection type	Static IP, PPPoE, DHCP	
Management	HTTPs (LAN / WAN access), HTTP (LAN access)	

APPLICATIONS

IPX-1803/IPX-1804 Intra Office Voice Communication



ORDERING INFORMATION

IPX-1803	Internet Telephony PBX System (30 user registrations, 3 x FXO, 1 x FXS)
IPX-1804	Internet Telephony PBX System (30 user registrations, 4 x FXO)
Related Products - Auto Provision supports	
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-Port SIP Analog Telephone Adapter
VIP-153T	SIP IP Phone
VIP-153PT	SIP IP Phone with PoE Splitter Built-in
VIP-152T	SIP IP Phone