Hybrid IP-PBX System Solution

IPNext 190

IPNext190 Hybrid IP-PBX System

IPNext190 is a next-generation hybrid IP-PBX system and a new IP telephony solution for all IP environments. IPNext190 inter-works with various IP terminals of AddPac (e.g. AP-VP300, AP-VP280, AP-VP150, AP-VP120 IP video phone, and AP-IP300, AP-IP230, AP-IP160, AP-IP120 IP Phone, etc) and analog phones (low cost compared with IP terminals, also reused) to provide multimedia IP telephony services as well as the traditional voice-based PBX features. This product is based on the advanced embedded RISC that enables firmware upgrade, and can be equipped with PSTN VoIP interfaces (FXS, FXO, etc) depending on module options. IPNext190 is suitable for small and medium size companies, and inter-works with the VoIP and video products of AddPac Technology to provide a variety of IP application services appropriate for your network. Also, IPNext190 supports H.323 VoIP inter-working as well as SIP protocol for outbound call.
The front panel of IPNext190 hybrid IP-PBX system supports LED displays of device status and has two(2) 10/100Mbps Fast Ethernet Ports, the RS-232C console port for Command Line Interface (CLI), Power Inlet and Power ON/OFF Switch. Its rear panel supports various VoIP interfaces such as FXS, FXO, Digital E1/T1 depending on module options. IPNext190 can support maximum 32 port VoIP Interface Ports (Four(4) slots x 8Port VoIP Module). IPNext190 supports media gateway features as well as IP-PBX features.
The call scenarios supported by IPNext190 provide SIP-based basic calls, color ring services, blind transfer, call pickup, group call pickup, consult calls, switching calls, consult transfer, call waiting, call waiting notification, call park, call pickup remote, and hunt group. This product is designed to provide application services that require much memory such as voice mails using internal memory. Also, RTP Proxy Service for private IP address, intelligent IVR service, Media Service such as music on hold, announcement, music ring back tone, Unified Messaging Service (voice mail) are additional IP keyphone service features beside basic SIP call manger service.
AddPac Technology provides various product families appropriate for your network environment. To meet your needs, AddPac supports VoIP and media gateways, audio/video terminals, audio/video MCU, IP audio/video broadcast, CCTV VMS solution, audio/video recording solutions, and traffic controller QoS device solutions. In the future all IP-based multimedia telephony environment, various audio/video resources should be shared on an IP network; thus, the integration of the entire solution and that of solutions for each area are very important. AddPac IP-PBX is designed considering the integrated multimedia solution, and can meet your various needs.

Product Highlights

- New 'All-In-One' hybrid IP-PBX system Concept with Video, Voice, Audio, Data Integrated for Complete Multimedia IP Telephony System
- VoIP Interface for PTSN service :
  Maximum 32 Port (4 VoIP Module Slots)
  - 8 Port FXS Module
  - 8 Port FXO Module
  - 4 Port FXS & 4 Port FXO Module
  - Digital E1/T1 Module
- Upgradeable System Architecture based on Programmable RISC & DSP
- Two(2) 10/100Mbps Fast Ethernet Interface
- RS-232C Console Interface(RJ45) for CLI (Command Line Interface)
- Supports Various AddPac IP end-point terminal : Video Phones, IP Phones
- RTP Proxy Service Features for Private IP and IPv6 Address
- Internal : SIP signaling
  Outbound Call : SIP and H.323 Signaling
- Various IP Telephony Call Scenario
  (Call Transfer, Call Park , Call Pickup etc)
- Intelligent IVR service via Scenario Editor
- Unified Messaging Service (Voice Mail, etc)
- User Presence Service
- Media Service (RingBack, Music on Hold, Announcement)
- Voice H/W MCU Module (8Ch): Option
- Remote Firmware Upgrade Through FTP & TFTP Protocol
- Web based Smart Multimedia Manager
- Essential Scalability Features such as DHCP Server & Relay, NAT/PAT, IEEE Transparent Bridging, and Debugging/Diagnostics, etc.

IPNext 190 Application

- IP Telephony System Integration for Small Office, Medium Size Enterprise Network
- IP based Call Center System

IPNext190

Hybrid IP-PBX System Solution

Hardware Specification
- RISC PowerPC Microprocessor
- Memory
  - Boot memory 512Kbyte Flash Memory
  - Main memory 128Mbyte SDRAM
  - Flash Memory 2Gbyte
- Ethernet Interface
  - 10/100Mbps 2-Port Ethernet (RJ-45)
- Console Port
  - RS-232C 1-Port Console (RJ-45)
- PSTN VoIP Interface : Maximum 32Port
  - AP-N1-FXS8 : 8-Port FXS (8 x RJ11)
  - AP-N1-FXO8 : 8-Port FXO (8 x RJ11)
  - AP-N1-FXS4O4 : 4-Port FXS & 4-Port FXO (8 x RJ11)
  - AP-N1-E1/T1 : 1-Port Digital E1/T1 (1 x RJ45)
- Voice MCU Module : Option
  - Internal H/W Voice MCU Module : G.711, G.729. G.723.1 (4 Party, 2 Session, Max 8 Channel)

Power & Operational environment
- Power Requirement : VAC 110~220V, 50/60Hz, 5V 15A, 12V 8A
- Operation Temperature 0℃~+50℃
- Storage Temperature -40℃~+85℃
- Humidity 5%~95%

Dimensions
- Dimension (H x W x D) : 56mm x 440mm x 313mm -19" Rack Mountable Chassis
- Weight : 5 Kg

Voice over IP Service
- H.323, SIP VoIP Signaling Protocol
- VAD, DTMF, CNG, G.168
- Voice Codec : G.723.1, G.729, G.726. G.711,
- Various VoIP Features
- Interoperable with Diverse VoIP Gateways
- Interoperable with Diverse VoIP Gatekeeper, SIP Proxy Server

IP Telephony Service
- Internal Call : SIP signaling
- Outbound Call : SIP and H.323 Signaling
- Speed Dial, Phone Book
- Basic Call Scenario, Coloring Service Music on Hold, Blind Transfer, Call Pickup
- Consult Call, Switching Call, Consult Transfer
- Call Waiting, Call Waiting Notify
- Call Park, Call Pickup Remote, Hunt Group
- Web based Smart Multimedia Manager Program
- MS-Window based Smart Messenger Support

RTP Proxy Service
- Private IP Address Support
- IPv6 Address Support
- Public and Private IP Address Inter-working

IVR Service
- IVR Scenario Editor Support
- IVR Scenario Simulation
- IVR Scenario Run-Time Debugging

User Presence Service
- Real-time User Presence Service
- Interworking with Smart Messenger and Presence IP Terminal

Media Service
- Ring Back Tone, Announcement, Music on Hold
- Scheduled based Media Service

Unified Messaging Service
- IVR Scenario for Voice Mail Recording and Retrieval
- Voice Mail Notification via E-Mail
- Quota Management for each user

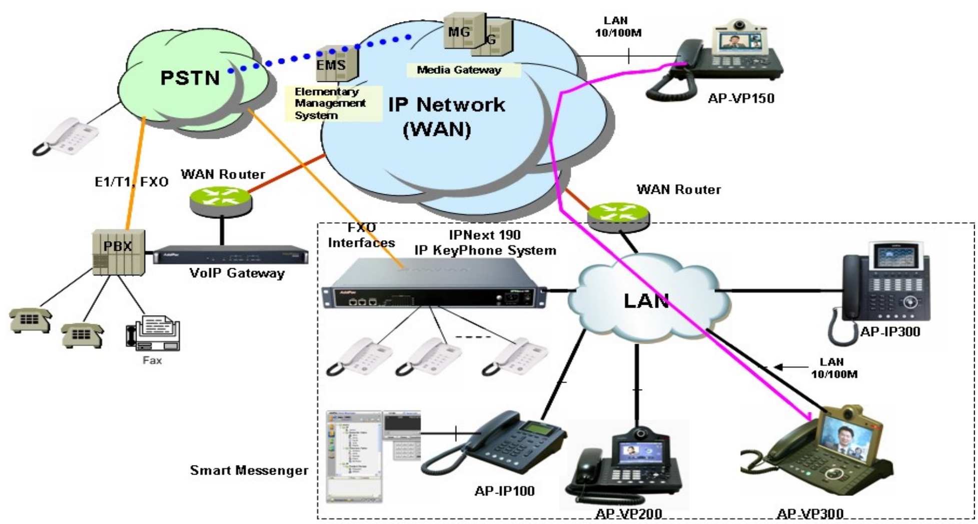
Network Management
- Standard SNMP Agent ((MIB v2) Support
- Console, Telnet, Web Based Management
- Remote Download via FTP/TFTP

Traffic QoS Control
- Traffic QoS Control Feature for Services
- Voice, Data, Video Prioritizing Control
- Various QoS Algorithm Support

Security Feature
- IP Packet Filtering / Access List
- Access Control and Data Protections
- Enable/Disable for Specific Protocols
- Multi-level User Account Management
- Auto-disconnect for Telnet/Console Sessions
- PPP User Authentication Support
  - Password Authentication Protocol (PAP)
  - Challenge Handshake Authentication Protocol (CHAP)

Other Feature
- DHCP Server & Relay Functions
- DDNS , NTP (Network Time Protocol)

Network Diagram



Ordering Information
- IPNext 190 IP Keyphone System
  - Two(2) Fast Ethernet Port
  - One(1) RS-232C Console Port
  - RISC CPU
  - 2GB Flash, 128 MB SDRAM
  - Four(4) VoIP Module Slots
  - APOS v8.xx Manual
- CAB-LAN Ethernet Cable
- CAB-CON RS-232C Console Cable

PSTN VOIP Interface Module Option :
- AP-N1-FXS8
- AP-N1-FXO8
- AP-N1-FXS4O4
- AP-N1-E1/T1

AddPac

www.addpac.com

AddPac Technology Co., Ltd.

2,5F, Kyung-An Bldg., 769-12, Yeoksam-Dong Kangnam-Gu, Seoul, 135-080, Korea

Tel: (02)568-3848, Fax: (02)568-3847, e-mail: sales@addpac.com