IPNext

[System Installation Guide]

Version 1.2



AddPac Technology Co., Ltd. www.addpac.com



IPNext

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Version 1.00	2008, 8, 25	Initial Released	AddPac Technology
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Chapter 1. IPNext Introduction

Outline

IPNext is a next-generation SOHO IP-PBX system for interworking with PSTN interface and various IP terminals of AddPac (AP-VP300 IP video telephone, AP-IP300 IP Phone, etc) to provide multimedia IP telephony services as well as traditional IP telephony. This product is built based on high performance embedded RISC and suitable for the small and medium size companies. IPNext interworks well with the various VoIP/video products to provide IP application services

Function

1) Next Generation SOHO IP-PBX System

The front system panel of AddPac IPNext NGN (Next Generation Network) SOHO IP-PBX is built with device status LED displays. It has two 10/100 Mbps fast Ethernet ports, the RS-232C console port for Command Line Interface (CLI). Also, it supports 2-ports FXO analog interface. FXO VoIP interface performs media gateway function as a way of PSTN interfacing. The call features supported by IPNext including; SIP-based basic calls, ring tone, music on hold, blind transfer, call pickup, group call pickup, consult transfer, consult call, call waiting, call waiting notification, call park, call pickup remote, and hunt group. In addition, it is designed to support SIP, H.323 concurrently. A high capacity Flash Memory is installed to increase the stability while providing many application services such as Voice Mail.

2) RTP Proxy Feature

Enterprise network environment (made up of both SIP server and IP phone) requires several IP addresses so that either IP version6 or private IP address in the NAT environment requires for enterprise networks due to the public IP address depletion. The RTP proxy feature is required for reliable multimedia communications between End-to-End terminals in private address IP address. The RTP proxy server feature is used for communication between a private IP terminal and a public IP terminal in between edge terminals such as IP phones, communication between a private IP terminal and a public IP terminal and a public IP terminal in the NAT environment, communication between private IP terminals, provide audio/video broadcasting in private and public IP environments, and enable audio/video conference calls in private and public IP environments. The RTP proxy



feature can be operated regardless of VoIP signaling protocols such as H.323, SIP, MGCP and supports both IP Version4/IP Version6 dual address systems.

3) User Presence Feature for Unified Communication

As User Presence feature provides user presence indication information in Smart Messenger Program (MS Window based PC platform environment) or in IP terminal (next generation IP telephony solution). User presence feature operates based on IP-PBX system and AddPac protocol SSCP (Smart Service Control Protocol) in between IP terminal (or in Smart Messenger). User presence function displays user on line status, user away and user busy and performs broadcasting regard of user information to all terminals (terminal with presence capability) from the collected information in each End-Point terminals.

4) Intelligent IVR

One of the most important features of SOHO IP-PBX solution is; ARS and IVR. IVR requires different IVR features for each different field including public office, company and Call Center. To accommodate the needs of user, AddPac IP-PBX solution provides the IVR scenario Editor; thus, user can map out in accordance with call scenario and provides IVR support feature tools to manage IP Key Phone System.

5) Unified Messaging Service

AddPac IPNext IP PBX system supports the network based media service. It is a total solution for interworking with VoIP gateway, IP video phone, IP phone to perform an announcement, ring tone, music on hold. You may change the ring back tone and music on hold file in accordance with the schedule. Video codec supports the MPEG-4, multimedia ring back tone and multimedia CID (MCID).

6) Unified Messaging Service for Voice Mail

IPNext SOHO IP-PBX system network messaging feature supports voicemail by interworking with VoIP gateway, IP video phone and IP phone in next generation multimedia telephony solution. It supports the SIP VoIP signaling, voice mail, multimedia message, IVR scenario for retrieving. User may setup the IVR voicemail environment by using VXML based IVR scenario editor program. It supports the memory quota setup function for storing voicemail and supports voicemail notification through email. You may play the voicemail through PSTN, VoIP, email and AddPac messenger program.

7) Firmware Upgradeable Structure

IPNext's high performance RISC CPU is a programmable structure so that it is capable of improving functions, altering, and adding new features constantly. When additional features or



alterations are needed, just directly download from the homepage or setting an auto upgrade option. You may use the latest features without much effort. Also, firmware and version upgrade such as video phone, IP phone, smart messenger program for AddPac IP keyphone system solution can be independently downloaded from the homepage and may be upgraded by downloading from the keyphone. For example, PC software Smart Messenger Program (act as terminal) is designed to upgrade by checking version. When IP Keyphone system administrator manages the terminal software version to upgrade, you may be able to upgrade by bringing new firmware into IP Keyphone system through version check while booting or when the IP Phone power is on. IP Keyphone systems and terminals are designed to manage software version management system in top-down structure for the efficient software version management..

8) Reliable IP-PBX Solution

IPNext is an integrated network device that supports routing services, NAT/PAT, DHCP Server/Relay, and Quality of Service (QoS). In order to adapt into the variety of network environments such as xDSL, cable network, FTTH, Metro Ethernet, Metro ATM, dedicated lines, flexible IP environment, an advanced QoS (Quality of Service), security feature should be supported along with abundant network services. In this sense, IPNext supports two 10/100 Mbps Fast Ethernet interfaces. Based on this, IPNext supports advanced LAN-to-LAN routing and bridge services as well as various network and security services such as NAT/PAT. AddPac IPNext IP-PBX is a reliable solution built by using excellent technologies.

9) Audio/Video Privacy Protection

IPNext IP-PBX provides the network security of 'Standard & Extended IP Access List' for equipment access control and service security as well. It supports the security features enhanced by VPN, Secure RTP function as well as the data security and voice/video secure call functions for privacy protection.

10) AddPac IP PBX Total Solution

AddPac technology is not just a vendor of SOHO IP-PBX BOX for the customer satisfaction, but it provides various products for appropriate network environment for VoIP, Media Gateway, audio/video terminals, audio/video MCU, IP audio/video broadcast, VoD solutions, network DVR solution, audio/video recording solutions, and traffic controller QoS device solutions. In all IP based multimedia telephony environment, various audio/video resources should be shared on an IP Network; thus, the integration of solutions for each area and entire solutions are very important. AddPac IP telephony solution is designed to consider the integrated multimedia solution so that it can satisfy the various needs of customer.

AddPac's various VoIP Gateway series and multimedia network devices have been fully



recognized in terms of its performance and stability throughout the world. IPNext will provide full satisfactions for customers who seek for a next generation IP telephony system with our accumulated experience in the enterprise market and service provider market.

Main Characteristics

High Performance RISC Microprocessor Structure Two(2) FXO VoIP Interface for PSTN interconnection(Option) Network Interface: 10/100Mbps Fast Ethernet InterfaceX2 User Interface: 1Port RS-232C Serial Console Interface User Terminals : AP-VP500, AP-VP300, AP-VP250, AP-VP150, AP-VP120, AP-IP300, AP-IP150, AP-IP100, Smart Messenger

AP-VP500	AP-VP500	AP-VP500 Video Phone High Performance & Powerful Communication Method
AP-VP300	AP-VP300	AP-VP300 Video Phone High Performance & Powerful Communication Method
AP-VP250	AP-VP250	AP-VP250 Video Phone High Performance & Powerful Communication Method
AP-VP150	AP-VP150	AP-VP150 Video Phone High Performance & Powerful Communication Method



AP-VP120	AP-VP120	AP-VP120 Video Phone High Performance & Powerful Communication Method
AP-IP300	AP-1P300	AP-IP300 IP Phone High Performance & Powerful Communication Solution
AP-IP150	AP-IP150	AP-IP150 IP Phone High Performance & Powerful Communication Solution
AP-IP100	AP-IP100	AP-IP100 IP Phone High Performance & Powerful Communication Solution

Call Manager Feature: Call Scenario, Coloring Service, Music on Hold SIP, H.323 Signaling Support Support Scenario Editor/Supporting Tool for IVR Service Voice Mail, UMS (Unified Messaging Service), Announcement, RingBack Tone, Music

on Hold.

Media Gateway function for PSTN interface (PSTN) Fault tolerant and Scalability

Service	Detail	Required Devices for Connection
IP Keyphone	Call Scenario, Call Transfer, Call	· AddPac AP-VP300 Video Phone
	Forwarding, Coloring Service.,	· AddPac AP-VP150 Video Phone
	Music on Hold.	· AddPac AP-IP300 High-End IP Phone
	Three Way Conference.	· AddPac AP-IP150 IP Phone
	Voice Mail, IVR	· AddPac AP-IP100 IP Phone

Table 1-1 IPNext



		· AddPac Smart Messenger
PSTN Interface	FXO VoIP Gateway Function	·Embedded 2-Port FXO PSTN Interface
(Option)	For PSTN	(Option)



Configuration



Picture 1-1 IPNext Network Configuration



Hardware Specification

Category	Specification	
Microprocessor	CPU	High Performance RISC Integrated Host Processor
Memory	Main Memory	128Mbyte SDRAM
	Flash Memory	512Mbyte
	Boot Memory	512Kbyte Flash Memory
	LAN0 Port	One(1) 10/100Mbps Fast Ethernet
Fixed Network	LAN1 Port	One(1) 10/100Mbps Fast Ethernet
Interface	Console Port	One(1) RS-232C Interface for CLI
	USB Port	USB 1.1 Host Mode Interface
DCTN Interface	FXO Port	None (Model A)
PSTN Interface		2-Port FXO Voice Interface (2 x RJ11) (Model B)
USB Module	USB Port USB 1.1 Host Mode Interface	
Power Requirement	Power VAC 110~220 VAC, 50/60Hz, 5V 3A	
Operating	$0^{\circ}C \approx 50^{\circ}C (32^{\circ} \approx 122^{\circ}E)$	
Temperature		
Storage	-40°C ~ +85°C (-40° ~ +185°F)	
Temperatures		
Relative Humidity	5% ~ 95%	
Dimensions	38 x 182 x 182 (mm	, H x W x D)
Weight	0.46Kg	

Table 1-2 IPNext Hardware Specification



Software Function

	Table 1-5 IF Next Software Function
Category	Specification
	Trunk Hunting by Preference or Sequential
	Calling Hunting by Preference, Simultaneous, Random
	Calling Hunting by Chained Hunting Group
	Partition for Address Grading
	Call Class for Call Access Control
	Number Translation Rule for Inbound/Outbound Call
	Centrex with Prefix Support
	Multiple Shared Devices with One Number
Number & Call	Multiple Numbers on One Device
Routing	Individual Call Park within Park Number Pool
	Group Call Park within a Group or Other Group
	Call Pickup of Ringing Call of Same Group or Other Group
	Call Pickup of Parked Call
	Call Transfer- Blind, Consult
	Call Forwarding – Unconditional, Busy, No Answer, Voice Mail
	Call Waiting
	Call Swapping
	Call Hold
Advanced	Multiple Call Handling with Call Status and Calling Line Number and Name
Features with	Plug and Play with Auto Discovery Function
AddPac IP phone,	Cottion Man Download and Control
Video Phone, etc	
	Voice Mail List View
	Parked Call List View
Telephony and	Call Forwarding Setting
Sorvico	Recent Call List View
Service	Calling Number and Name Identification
æ Features	Individual Call Park within a Group or Other Group by Softkey
i catules	Group Call Park within a Group or Other Group by Softkey
	Call Pickup of Ringing Call of Same Group or Other Group by Softkey

Table 1-3 IPNext Software Function



Call Pickup of Parked Call by Softkey Call Transfer - Blind, Consult by Softkey Call Waiting Indication Call Swapping by Softkey Conference Control Conference Control IP-PBX Signaling Protocols Protocols IVR Content Role for Registering to ITSP SIP Server - H.323 Gatekeeper Client Role for Registering to ITSP H.323 Gatekeeper Server Default Auto Attendant Support UR Function E Auto Attendant Voice Response E Auto Attendant Support Voice Mail with IVR Voice Mail Conference Conference Conference Conference Conference Conference Dial-Out Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Multiple Concurrents Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Herachical Organization Auto Discovery of IP Phones & IP Video Phones Music & Auto Status of Phones Distinctive Ring by Calling User Auto Config Supparle IP-PBX Miscellaneous Function Personal Directory Personal Personal Directory Personal Directory Personal Directory Personal Personal Directory Personal Pers									
Call Transfer - Blind, Consult by Softkey Call Waiting Indication Call Swapping by Softkey Conference Control Call Swapping by Softkey Conference Control SIP Application Server, Proxy, Registrar and Location Server(RFC3261) Multiple TTSP Trunk with SIP & H.323 Account Support - IP UA Client Role for Registering to ITSP SIP Server - H.323 Gatekeeper Client Role for Registering to ITSP H.323 Gatekeeper Server Default Auto Attendant Support UR Cinteractive Voice Response) & Auto Attendant Support Voice Mail With IVR Voice Mail Voice Mail Support Voice Mail With IVR Conference Gate: Support Voice Mail With IVR Conference Gate: Support Voice Mail With IVR Conference Gate: Support Ad-Hoc Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Music on Hold Replaceable Announcements Dialing Music/Tone Service IDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Musice Auto Config & Uploady Contense Personal Directory Personal Personal Directory Personal Directory Personal Directory Personal Personal Personal Directory Personal Persona		Call Pickup of Parked Call by Softkey							
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(Interactive Voice Response) & Auto AttendantUR FunctionUpload/Download Scenario by Smart IVR Scenario& Auto AttendantIVR FunctionEditor& Support Recordable IVR PromptsSupport Recordable IVR PromptsVoice MailSupport Voice Mail with IVRVoice MailAccess from Remote Site via Trunk SupportVoice MailVoice Mail Notification SupportG.711 u-law, G.711 a-law, G.726 Internal 3-Party Audio ConferenceSupportAd-Hoc ConferenceSupportAd-Hoc ConferenceMusic & AnnouncementMusic on HoldMusic & AnnouncementReplaceable AnnouncementsDialing Music/Tone ServiceDalling Music/Tone ServiceIP-PBX User & Device ManagementLDAP(Lightweight Directory Access Protocol) SupportIP-PBX Miscellaneous FunctionOptional ServiceIP-PBX Miscellaneous FunctionDistinctive Ring by Calling UserAuto Config & UpgradeIntercomPersonal DirectoryPersonal Directory	IVR		Provides with GUI-based Smart IVR Scenario Editor						
Voice Response) & Auto Attendant IVR Function Editor Supports Multiple Concurrent Scenarios Support Recordable IVR Prompts Voice Mail Support Voice Mail with IVR Access from Remote Site via Trunk Support Access from Remote Site via Trunk Support Voice Mail Access from Remote Site via Trunk Support Voice Mail Notification Support G.711 u-law, G.711 a-law, G.726 Internal 3-Party Audio Conference Support Ad-Hoc Conference Music Ad-Hoc Conference Meet-me Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Announcement Music on Hold Replaceable Announcements Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Miscellaneous Distinctive Ring by Calling User Auto Config & Upgrade Intercom	(Interactive		Upload/Download Scenario by Smart IVR Scenario						
& Auto Attendant Supports Multiple Concurrent Scenarios Support Recordable IVR Prompts Support Voice Mail with IVR Access from Remote Site via Trunk Support Voice Mail Access from Remote Site via Trunk Support Voice Mail Notification Support G.711 u-law, G.711 a-law, G.726 Internal 3-Party Audio Conference Support Ad-Hoc Conference Dial-Out Conference Meet-me Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Announcement Music on Hold Replaceable Announcements Dailing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Miscellaneous Distinctive Ring by Calling User Auto Config & Upgrade Intercom Personal Directory Personal Directory	Voice Response)	IVR Function	Editor						
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Voice Mail Access from Remote Site via Trunk Support Voice Mail Notification Support Conference Support Ad-Hoc Conference Ad-Hoc Conference Dial-Out Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Announcementa Replaceable Announcements Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Miscellaneous Function Personal Directory		Support Voice Mail	with IVR						
Voice Mail Notification Support G.711 u-law, G.711 a-law, G.726 Internal 3-Party Audio Conference Support Ad-Hoc Conference Dial-Out Conference Meet-me Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Announcement Music on Hold Replaceable Announcements Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Miscellaneous Function Personal Directory	Voice Mail	Access from Remote Site via Trunk Support							
G.711 u-law, G.711 a-law, G.726 Internal 3-Party Audio Conference Support Ad-Hoc Conference Dial-Out Conference Meet-me Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Announcement Music on Hold Replaceable Announcements Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Distinctive Ring by Calling User Auto Config & Upgrade Intercom Personal Directory		Voice Mail Notification Support							
Support Ad-Hoc Conference Dial-Out Conference Meet-me Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Music on Hold Replaceable Announcements Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Musicellaneous Function Intercom Personal Directory		G.711 u-law, G.711 a-law, G.726 Internal 3-Party Audio Conference							
Conference Ad-Hoc Conference Dial-Out Conference Meet-me Conference Meet-me Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Conference Chair and Participants Management Music & Announcement Music on Hold Replaceable Announcements Dialing Music/Tone Service Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Distinctive Ring by Calling User Auto Config & Upgrade Intercom Function Personal Directory		Support							
ConferenceDial-Out ConferenceMeet-me ConferenceMultiple External MCU Support (Video, Audio, etc) : AP-MC1000Conference Chair and Participants ManagementMusic & AnnouncementMusic on HoldReplaceable AnnouncementsDialing Music/Tone ServiceLDAP(Lightweight Directory Access Protocol) Support- Support Hierarchical OrganizationAuto Discovery of IP Phones & IP Video PhonesMonitoring Status of PhonesIP-PBXMiscellaneousFunctionIntercomPersonal Directory		Ad-Hoc Conference							
Meet-me Conference Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Music on Hold Replaceable Announcements Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Distinctive Ring by Calling User Auto Config & Upgrade Intercom Personal Directory	Conference	Dial-Out Conference							
Multiple External MCU Support (Video, Audio, etc) : AP-MC1000 Conference Chair and Participants Management Music & Announcement Replaceable Announcements Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Distinctive Ring by Calling User Auto Config & Upgrade Intercom Personal Directory		Meet-me Conference							
Conference Chair and Participants Management Music & Music on Hold Announcement Replaceable Announcements Dialing Music/Tone Service Dialing Music/Tone Service IP-PBX User & LDAP(Lightweight Directory Access Protocol) Support - Support Hierarchical Organization - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones IP-PBX Distinctive Ring by Calling User Auto Config & Upgrade Intercom Personal Directory Personal Directory		Multiple External MCU Support (Video, Audio, etc) : AP-MC1000							
Music & Announcement Music on Hold Replaceable Announcements Replaceable Announcements Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support IP-PBX User & Device Management LDAP(Lightweight Directory Access Protocol) Support Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Distinctive Ring by Calling User Auto Config & Upgrade Intercom Personal Directory		Conference Chair and Participants Management							
Music & AnnouncementReplaceable AnnouncementsDialing Music/Tone ServiceDialing Music/Tone ServiceIP-PBX User & Device ManagementAuto Discovery of IP Phones & IP Video PhonesMonitoring Status of PhonesIP-PBX Miscellaneous FunctionIP-PBX Personal Directory		Music on Hold							
Announcement Dialing Music/Tone Service Dialing Music/Tone Service LDAP(Lightweight Directory Access Protocol) Support IP-PBX User & Device Management - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Distinctive Ring by Calling User Auto Config & Upgrade Intercom Personal Directory	Music &	Replaceable Announcements							
IP-PBX User & LDAP(Lightweight Directory Access Protocol) Support Device - Support Hierarchical Organization Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones IP-PBX IP-PBX Miscellaneous Function Personal Directory	Announcement	Dialing Music/Tone Service							
IP-PBX User & - Support Hierarchical Organization Device Auto Discovery of IP Phones & IP Video Phones Management Monitoring Status of Phones IP-PBX Distinctive Ring by Calling User Miscellaneous Auto Config & Upgrade Intercom Personal Directory		LDAP(Lightweight Directory Access Protocol) Support							
Device Management Auto Discovery of IP Phones & IP Video Phones Monitoring Status of Phones Monitoring Status of Phones IP-PBX Distinctive Ring by Calling User Miscellaneous Function Auto Config & Upgrade Intercom Personal Directory	IP-PBX User &	- Support Hierarchical Organization							
IP-PBX Monitoring Status of Phones IP-PBX Distinctive Ring by Calling User Miscellaneous Auto Config & Upgrade Function Intercom Personal Directory	Device	Auto Discovery of IP Phones & IP Video Phones							
IP-PBX Distinctive Ring by Calling User Miscellaneous Auto Config & Upgrade Function Intercom Personal Directory	manayement	Monitoring Status	of Phones						
IP-PBX Auto Config & Upgrade Miscellaneous Intercom Function Personal Directory		Distinctive Ring by	Calling User						
Miscellaneous Intercom Function Personal Directory	IP-PBX	Auto Config & Upgi	rade						
Personal Directory	Miscellaneous	Intercom							
	гипсиоп	Personal Directory							



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	Downloadable Ring						
	IPv4/IPv6 Dual Stack						
Basic Routing		Telnet, FTP, TFTP, SSH, SNMP, Syslog support					
	Management Routing	Packet filtering (Access-list)					
		Static					
	Standard SNMP Agent (N	1IB v2) Support					
Network	Traffic Queuing						
Management	Remote Management using Console, Rlogin, Telnet						
	Web based Management	s using HTTP Server Interface					
	Standard & Extended IP	Access List					
	Access Control and Data	Protections					
	Enable/Disable for Speci	fic Protocols					
Security	Multi-Level User Account	Management					
Functions	Auto-disconnect for Telnet/Console Sessions						
	PPP User Authentication Supports						
	\rightarrow Password Authentication Protocol(PAP)						
	ightarrow Challenge Handshake Authentication Protocol (CHAP)						
	System Performance Analysis for Process, CPU, Connection I/F						
	Configuration Backup & Restore for APOS Managements						
Operation	Debugging, System Audi	ting, and Diagnostics Support					
م Management	System Booting and Auto	o-rebooting with Watchdog Feature					
	System Managements with Data Logging						
	IP Traffic Statistics with	Accounting					
	DHCP Server & Relay Functions						
	Network Address Translation (NAT) Function						
	Port Address Translation (PAT) Function						
Other Scalability	Transparent Bridging (IEEE Standard) Function						
Features	\rightarrow Spanning Tree Bridgir	ng Protocol Support					
	\rightarrow Remote Bridging Supp	port					
	\rightarrow Concurrent Routing and Bridging Support						
	Cisco Style Command Line Interface(CLI)						
	Network time Protocol(N	TP) Support					



Input/Output Configuration

Front



Picture 1-2 IPNext Front

Table 1-4 IPNext Front

No.	Table	Explanation
(1)	POWER	Display if the power is provided properly through Power LED.(Red)
(2)	LAN 0	LED display LAN 0 Link status (Green)
(3)	LAN 1	LED display LAN 1 Link status (Green)
(4)	FXO 0	LED display for FXO voice port status (Blue)
(5)	FXS 1	LED display for FX0 voice port status (Blue)



Rear



Picture 1-3 IPNext Rear

Table 1-5 IPNext Rear

No.	Table	Explanation				
(1)	ON 5V3A	A switch to supply/shut off the system (DC 5V 3A power adapter				
_		connecting terminal and system power.				
(2)	LAN 0	10/100Mbps fast Ethernet interface for WAN connection (RJ45)				
(3)	LAN 1	10/100Mbps fast Ethernet interface (RJ45) for LAN connection				
(4)	CONSOLE	Console interface for network management (RJ45).				
(5)	USB	USB terminal that comply with standard 1.1. Maximum				
_		transmission rate is 12Mbps and user may connect USB memory.				
(6)	PSTN(FXO) 0	1 port PSTN (FXO) voice interface input/output section. (Option)				
_	(Option)					
(7)	PSTN(FXO) 1	1 port PSTN (FXO) voice interface input/output section. (Option)				
	(Option)					



Chapter 2. IPNext Installation

Installation Requirement

Requirement

Following details are recommendations for product safety.

- After IPNext installation, use it in clean environment.
- When opening IPNext cover, work in safe place..
- Do not wear a loose shirt. Don't let your tie or scarf slip down. Roll up your sleeves.

Electric Safety Recommendation

IPNext may face with two electrical issues. One is safety concern in power supply and the other is device damage from electrostatic.

• Electrical Safety

- ✓ Make sure to work in a location where you can shut off the system immediately when accidents occur.
- ✓ Shut off the power while installing device or taking off the cover.
- ✓ Do not work alone in dangerous environment.
- \checkmark Do not assume that the power is off. Be sure to check the power.
- ✓ Be cautious when working in humid area or without ground connection

• Prevent Electrostatic

- ✓ IPNext Chip-Set is very sophisticated components. If you mishandle it, it would cause some damages.
- ✓ Be sure to wear electrostatic prevention waist strap if you have one.
- ✓ If you do not have waist tap, be sure to hold the device metal sash. It will prevent electrostatic.



General Installation Requirement

IPNext is usable anywhere. For the maximum performance, we recommend places as below.

- Maintain level and adequately ventilated.
- Please attach the product in a safe place.
- Do not put other objects on the device.
- Avoid direct sun light and install in cool location.
- Keep a safe distance from fire, flammable liquid, and magnetic material

Preparation before Device Installation

When installing IPNext, user must consider EMI (EIA standard) and distance restriction. Following explains Ethernet cable, console cable and preparation.

Necessary equipments and cables are not included in the box unless you order them separately. To install IPNext, please prepare the following devices and equipments.

Standard Screw Driver Set

- Cable to connect LAN and console port
- RJ-45 to RJ-45 cable for LAN port
- RS-232C console cable that has RJ-45 connector (included in the box)

• Ethernet Port

IPNext has two RJ45 type Ethernet ports at the back and LED display for port status. Use standard cable and connector when accessing in LAN network. Please refer to appendix cable specification for Ethernet cable PIN specification.

Console Port



At the back of IPNext, it has RJ-45 type RS-232C Female DCE connector interface. Use may initialize the setup, monitoring, and debugging through this port. Make sure to use cable and connector. Please refer to appendix cable specification for RS-232C console cable PIN specification.

After Sales Service

Please contact via 02)568-3848 or FAX 02)568-3847 for technical problems. Normal business hours: 9AM ~6PM For company location and further information, visit our website http://www.addpac.com



Remove Product Packaging and Contents Check

Make sure to check the package damage before unpacking. Please check the following contents.

No	Name	Contents	Qt.
1	IPNext		1
2	LAN Cable (RJ45 to RJ45 Option)		1
2	Console Cable (RJ45 to DB9 Option)		1
4	Power Cable and Adapter		1

 Table 2-1 IPNext Product Package

Please contact AddPac Technology if you find any damaged items. (Tel: (02)568-3848)



Installation

Async Serial Interface Connection

Connect RJ-45 connector of RS-232C serial console cable to console port. The opposite serial connector connects to serial port such as IPNext control PC.



Picture 2-1 IPNext Async Serial Interface Connection



Ethernet Interface Connection

For internet connection WAN interface, connect with WAN device (router or ADSL/cable modem) LAN interface and RJ45 standard UTP cable. Cross-Over may be used when connecting to the router or modem directly. Use direct-through cable when connecting to HUB.

Connect LAN0/LAN1 fast Ethernet interface by using RJ-45 connector.



Picture 2-2 IPNext LAN0/LAN1 Interface Connection



PSTN (FXO) Interface Connection (Option)

Connect PSTN (FXO) port with external PSTN and RJ11 standard cable.

PSTN (FXO) port is interface which provides RJ-11 connector to interwork with telephone network and standard PBX.



Picture 2-3 IPNext PSTN Interface Connection



Booting Process and Working Status

It is booting process of IPNext.

- IPNext checks the CPU, memory, and interface.
- Boot Loader will be executed and find appropriate IPNext s/w image file. IPNext is designed to upload s/w in default configuration.
- If s/w image file cannot be found in flash memory, boot loader will wait in boot mode until it downloads the proper IPNext s/w. (Use TFTP or FTP protocol to download proper Next50 s/w)
- After IPNext loading, IPNext will be operated in accordance with saved information. IPNext will run with initial value if there is no saved setup information. For normal network operation, administrator must setup pertinent details.

After IPNext installation and interface connection, power must be supplied. Make sure to connect power cable with IPNext. Do not connect IPNext after providing power cable. Also, use 110V power cable if the power supply is 110V. IPNext automatically recognizes both 110V and 220V so using proper power cable and additional operation is not necessary.

After normal booting, following message will be shown.

System Bootstrap, Version 1.2 Decompressing the image: #############################[OK] ? System Boot Loader, Version 5.1.3

Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

[DUAL-BOOT] Start application (0xbc000000)...



System Bootstrap, Version 1.2

Decompressing the image:

AddPac IP-PBX Series (IPNEXT_G2) 32BIT RISC Processor With 125MHz Clock 128 Mbytes System Memory. 512 Kbytes System Boot Flash Memory 32 Mbytes System Flash Memory

1 RS232 Serial Console Interface

IPNEXT_G2 System software Revision 8.47.024 Released at Thu Nov 6 19:26:52 2008 Program is 5057752 bytes, checksum is 0x28ce9ed4

UTC Time is Mon Dec 1 12:00:10 2008 Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

Allocating system mbuffer counter: 2560 Loading file system(ver2.2), flash-base: 0xffff0000 ram-base: 0x9912836c Ethernet port initialization complete Ethernet port initialization complete System utilization reference (51/20/20/22) USB OHCI Host Controller Driver v5.3 [USB] HUB 2 ports detected Initializing USB Mass Storage driver... USB Mass Storage support registered. v0.0.1 (2005/09/09):USB HID Mouse driver v0.0.1 (2005/09/09):USB HID Keyboard driver Loading module: rt2570usb - v2.0.7 (2006/3/7) [USB] Start ROOT HUB timer Start Target Debug Server



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Attach FastEthernet Interface at Slot 0, Port 0-1, <0-0>/<0-1> [USB] Start HUB event processing FastEthernet0/0: link is up 100 Mbps (full duplex) FastEthernet0/1: link is up 100 Mbps (full duplex) Interface FastEthernet0/0, changed state to UP Interface FastEthernet0/1, changed state to UP Hardware Type ID = 0 Hardware Revision ID = 0x0 Slot (0) Module type : FXO

Start SendMail Server can't open configuration file [flash:/flash/apos.cfg] Start File Transfer Protocol Server (listen tcp/21) HTTP: document_root : /hd/smartclient Start RtpCallScanTimer for group default

Press RETURN to get started.

RTA Module Ready CPU internal DSP SRAM OK Audio DSP S/W download ... OK AudioConference Module Ready

Add default voipPeer(1000)

VoipGateway::Init1 - No IP address on the VoIP Interface CM CREATE DOMAIN CM CREATE DOMAIN

Welcome, APOS(tm) Kernel Version 8.47.024. Copyright (c) 1999-2008 AddPac Technology Co., Ltd.

Login:



Chapter 3. IPNext Console Command

Outline

Use Console Terminal by Using HyperTerminal

Terminal Emulator Application must be setup when using PC as console terminal. Use hyper terminal application when using MS-Windows type.



Picture 3-1 MS-Windows Terminal Emulator HyperTerminal



Execute hyper terminal and decide connection name. User may choose connection name freely. We setup as AddPac.

New Connection	n - HyperTerminal							
File Edit View C	all Transfer Help							
		onnection Descri New Conne Enter a name and c Name: AddPac Icon:	otion ction hoose an ict	on for the o	connecti	?] or: Cancel	×	
Disconnected	Auto detect	Auto detect	SCROLL	CAPS	NUM	Capture	Print echo	1.

Picture 3-2 Enter HyperTerminal Connection Name



Setup interface that is connected with console cable.

Console cable usually connects with RS-232C 9Pin Serial Port. Choose proper port in accordance with user environment. We have connected COM1.

AddPac - HyperTermit	nal Insfer Help								
		Connect To AddPare Enter details for Country/region Arga code: Phone number: Cognect using:	the phone r	number that	t you wa	? X			
Disconnected	Auto detect	Auto detect	SCROLL	CAPS	NUM	Capture	Print echo	1	_ //.

Picture 3-3 Setup Value When Connecting Console Cable to Serial Port



Decide each setup value in interface register information as below. We have used COM1 as a standard.

AddPac - HyperTerminal File Edit View Call Transfer	Help		×
D 2 88 98 1			
	COM1 Properties Port Settings	?×	•
	Bits per second: 9600 Data bits: 8	x	
	Parity: None Stop bits: 1	▼ ▼	
	Flow control: None	Tore Defaults	
	OKCancel	Apply	_
Disconnected Auto	detect Auto detect SCROLL CAPS NUM	M Capture Print echo	

Picture 3-4 COM1 Port Setup Example

After setup, press enter to see booting message on hyper terminal screen



APOS Command Usage

NOTE All AddPac technology devices are embedded with APOS (AddPac Operating System). Thus CLI (Command Line Interface) environment are all identical.

All command in IPNext can connect to console or telnet terminal (VT-100 terminal). Command provides to view system restriction items, user mode to provide access function, look at the system status. Administrator mode to use system debuggin function and change the setup environment or setup a new environment.

Following characteristics are IPNext related command input.

- It automatically recognizes without typing all command. For example, to run "show" command, type "sh" or "sho". It will automatically recognizes as "show"
- It provides on-line help function. When typing system command incorrectly, it shows possible items for command and command usage screen.
- More function provides to display unable information on screen.
- It provides Help and "?" function to display all possible command and explanation.
- It provides "History" function. User may use Prompt number without retyping command.
- System command structures are divided into 3 modes. Each mode has different command. Command for each mode is as below.


General User Mode Command

General user mode command is a function for all logged-in users.

General user prompt is displayed as "IP-PBX>".

Command	Explanation	
enable	Change to administrator mode	
exit	Move from current prompt to lower prompt	
help	APOS Help output	
quit	Same as exit	
show	Command for system working status and setup status	
terminal	Decide number of lines to print out from terminal at a time.	
who	Display access users through vty (command)	
whoami	Display how user accesses (command)	

Table 3-1 General User Mode Command

Administrator Mode Command

Administrator mode command is command that logged in administrator can use. To access in system setup mode, it must be under logged in status. It shows more information according to options even if it is same as command such as "show".

In administrator mode, user may use all command that is used in regular user mode.

Administrator mode prompt is displayed as "IP-PBX#".

Command	Explanation	
clear	Command to initialize Interface Counter, Statistic	
clock	Setup current year, date, time	
configure	Enter as setup mode	
сору	Copy running config as startup config	
debug	Debug command for system	
disable	Enter as user mode	
disconnect	Command to close VTY() Connection	

Table 3-2 Administrator Mode Command



dnsquery	Command that is used in DNS query test
dnsrv	Command that is used in DNS SRV record
end	Enter as administrator mode
erase	Delete config file
exit	Move to previous mode
fsh	Enter as file shell
ftp	Connect ftp client
help	Display APOS help
no	Command to delete current setup
nsupdate	Command to transmit upgrade information to name server
ntpdate	Command to receive time from ntp server
ping	Network connectivity confirmation tool
ping6	Network connectivity confirmation tool (IPV6)
quit	Same as exit
reboot	System rebooting command
show	System working status/setup status command
telnet	telnet access command
terminal	Setup number of lines from terminal
tftp	Command to transmit the file to tftp server
traceroute	IPv4 routing route check command
traceroute6	IPv6 routing route check command
undebug	Command to deactivate the debugging function
who	Command to display all logged in users by vty
whoami	Command to display how it is accessed currently
write	Command to save operating configuration



System Setup

Log-in Account

IPNext let users to make several user accounts. Each account has access restriction as well as APOS command setup in accordance with setup level. User account access restrictions are as below. (Include "root").

- User account "root" delete is impossible
- Only "root" is able to check in an entire account information
- If the user level is same as admin. User who is not a "**root**" can only read their own account information.

```
NOTE IP-PBX products root account default password is setup as "router".
```

Step	Command	Explanation
1	Welcome, APOS(tm) Kernel Version 8.47.024.	Log-in by root account.
	Copyright (c) 1999-2008 AddPac Technology Co., Ltd.	
	Login:	
	Login: root	
2	Password: <password></password>	Enter default password "router". Enter
	IP-PBX> enable	enable in IP-PBX>mode to enter
	IP-PBX#	APOS command setup mode.

Table 3-3 Log-In by Root Account



Step	Command	Explanation
1	IP-PBX# show username	Check user account
	username root password router ;	information by using APOS
	password is clean text type administrator access	command (do not display
	username abc password abc ;	upper account which is
	password is clean text type administrator access	higher than accessed
	username abcd password abcd ;	account)
	password is clean text type operator access	
	username abcde password abcde ;	
	password is clean text type user access	

Table 3-4 User Account Information Check

Step	Command	Explanation
1	IP-PBX#	Enter as APOS command input
	IP-PBX# configure terminal	mode
	IP-PBX(config)#	
2	IP-PBX(config)# username ?	Setup whether to authenticate the
	WORD User name	password when Logging in
	<pre>IP-PBX(config)# username addpac ?</pre>	nopassword : noncertified mode
	nopassword No password is required for the	password: certified mode
	user to log in	
	password Specify the password for the	
	user	
3	IP-PBX(config)# username ?	Setup whether to print out
	WORD User name	password
	<pre>IP-PBX(config)#username addpac password ?</pre>	0: Enter password for encrypted
	0 Specifies an UNENCRYPTED password will	password (password display)
	follow	7: Enter unencrypted password
	7 Specifies a HIDDEN password will	(password not displayed)
	follow	
	WORD The UNENCRYPTED (cleartext) user	WORD: Enter unencrypted
	password	password (password display)

Table 3-5 New User Account Registration



4	IP-PBX(config)# username ?				Setup administrator level	
	WORD User name					
	IP-PBX(confi	g)# us e	ername a	addpac	password	
	router ?					
	administra	Syst	em administ	rator		Administrator: administrator level
	tor					
	operator	Syst	em operator	and mon:	itor	Operator: operator level
	user	Syst	em end-user			User: user level
5	IP-PBX(confi	g)#	username	addpac	password	
	router opera	tor ?				
	<cr></cr>					
	IP-PBX(confi	g)#	username	addpac	password	User ID is addpac, password is
	router opera	tor				router, create level as administrator
	IP-PBX(confi	g)#				level.

Table 3-6 Log-in Under Regular User Account

Step	Command	Explanation
1	IP-PBX# exit	Register as new registered user
	IP-PBX> exit	account from the existing setup.
	Welcome, APOS(tm) Kernel Version 8.41.008.	
	Copyright (c) 1999-2006 AddPac Technology Co., Ltd.	
	Login:	
2	Login:	Open up a new user account by
	Password: <pre>password></pre>	entering registered password
	IP-PBX> enable IP-PBX#	"addpac".
3	IP-PBX# show username	When reading user account
	username addpac password router ;	information, it doesn't show the
	password is clean text type operator access	upper level account information
	username abcde password abcde ;	than the accessed account
	password is clean text type user access	information.

Table 3-7 User Delete

Step	Command	Explanation
1	IP-PBX# configure terminal	User Delete Command
	IP-PBX(config)# no username <user-name></user-name>	



Password setup

User only has an authority to do the "show command" after accessing to console. To have powerful authority, user must enter as enable mode. If the regular user enters as "enable mode", they have all the rights to change the system setup. Thus, be sure to setup the password only for the administrator.

Table 3-8 password Setup

Step	Command	Explanation
1	IP-PBX#	Enter as APOS command input
	IP-PBX# configure terminal	mode.
	IP-PBX(config)#	
2	<pre>IP-PBX(config)# enable password {password}</pre>	Password setup

Host Name Setup

Host Name shows prompt name in CLI environment. It informs to user what types of devices are being accessed through telnet and console. It is convenient when using device location or model name.

Following shows an example to setup device host name as IPNext.

	Table 3-9 hostname Setup		
Step	Command	Explanation	
1	IP-PBX#	Enter as APOS command input	
	IP-PBX# configure terminal	mode	
	IP-PBX(config)#		
2	<pre>IP-PBX(config) # hostname IPNext</pre>	Setup host name as IPNext	



Clock Setup

Clock setup shows the time of system. System time can be checked as "show clock" command in CLI environment as well as check running time up to now. Be sure to set your clock same as current time.

Following shows how to setup current time

Step	Command	Explanation
when	IP-PBX#	Enter as APOS command input
	IP-PBX# configure terminal	mode.
	IP-PBX(config)#	
2	IP-PBX(config) # clock time 2008 12 01 14 30 00	Setup present time as "2008, 12. 01
		14hr 30min 00sec"

Table 3-10 Clock Setup

Interface Setup

IP Address must be setup to communicate with other network devices. Interface speed and duplex is automatically adjusted with other devices (optimum value). However, it occurs when duplex does not fit. In this case, manual setup is required for speed and duplex.

Following example shows how to setup IP address "172.20.101.100" in interface fast Ethernet 0/0 and setup link speed 100M and full duplex manually.

Step	Command	Explanation			
1	IP-PBX#	Enter as APOS command input			
	IP-PBX# configure terminal	mode.			
	IP-PBX(config)#				
2	<pre>IP-PBX(config)# interface fastethernet 0/0</pre>	Enter as interface setup mode.			
3	IP-PBX (config-if) # ip address 172.20.101.100 Setup IP address "172.20.10				
	255.255.0.0	MASK '255.255.0.0"			
4	IP-PBX (config-if)# no speed auto	Setup 100/full duplex manually if			

Table 3-11 Fast Ethernet Interface Setup



5	<pre>IP-PBX (config-if) # speed 100</pre>	auto negotiation is not working
6	<pre>IP-PBX (config-if) # duplex full</pre>	properly. (Default: auto)

Default Router Setup

Setup where to send if it was accessed by unspecified network. Be sure to setup with device IP.

Following shows how to setup default router as "172.20.1.1"

Step	Command		Explanation			
1	IP-PBX#	Enter	as	APOS	command	input
	IP-PBX# configure terminal	mode.				
	IP-PBX(config)#					
2	IP-PBX(config)# ip route 0 0 172.20.1.1	Setup		Default	router	as
		"172.2	0.1.	1"		

Table 3-12 Default route Setup



System Service Setup

HTTP Server Setup

HTTP Server is activated. Make sure to setup when it is not activated. It is setup for PBX HTTP CLI execution, call manger status check in SMM, and to use auto upgrade server.

		•		
Step	Command	Explanation		
1	IP-PBX#	Enter as APOS command input mode		
	IP-PBX# configure terminal			
	IP-PBX(config)#			
2	IP-PBX(config)# http server	HTTP server activation. (default)		
3	<pre>IP-PBX(config)# no http authentication</pre>	(Optional) Setup to access in server without		
		HTTP access authentication. W		
		recommend using http authentication.		
4	IP-PBX(config)# http document-root /hd	Setup default directory for accessing web		
		based SMM. (IPNext v8.47.024. afte		
		version default: /hd)		

Table 3-13 HTTP Setup



Telnet Server Setup

Use telnet to manage the device which is located far. Telnet is already activated in system. So user does not require using command unless it is activated separately. Use TCP23 through telnet port. It is possible to modify in accordance with environment.

Step	Command	Explanation			
1	IP-PBX#	Enter as APOS command input mode.			
	IP-PBX# configure terminal				
	IP-PBX(config)#				
2	IP-PBX(config)# telnet server	telnet service activation			
3	IP-PBX(config)# telnet port 23	(optional) setup when modifying default			
		telnet port 23			

Table 3-14 Telnet Setup

FTP Server Setup

FTP is activated and if it is deactivated, make sure to setup. If user wishes compose ftp server, do ftp allow anonymous command.

Following example shows how to activate ftp server and allow anonymous account access.

Step	Command	Explanation
1	IP-PBX#	Enter as APOS command input mode.
	IP-PBX# configure terminal	
	IP-PBX(config)#	
2	IP-PBX(config)# ftp server	FTP server activation
3	<pre>IP-PBX(config)# username root password</pre>	FTP user account "root/router" creation
	router	
4	IP-PBX(config)# ftp port 21	(optional) setup when modifying default ftp
		port 21
5	IP-PBX(config)# ftp allow anonymous	(optional) permit anonymous account.

Table 3-15 FTP Setup



SNMP Setup

Use SNMP to monitor the remote device. Activate snmp server and assign snmp community value.

Following shows how to setup SNMP community value as "public".

Table 3-16 SNMP Setup

Step	Command	Explanation
1	IP-PBX#	Enter as APOS command input mode.
	IP-PBX# configure terminal	
	IP-PBX(config)#	
2	IP-PBX(config)# ip snmp server	logging activation
3	<pre>IP-PBX(config)# snmp community public</pre>	logging user command
	ro 0.0.0.0	



Network Setup Check

Setup Status Check

Following information shows IPNext default-config status.

```
Welcome, APOS(tm) Kernel Version 8.47.024.
Copyright (c) 1999-2008 AddPac Technology Co., Ltd.
IP-PBX# show run
Building configuration...
Current configuration:
!
version 8.47.024
ļ
hostname IP-PBX
!
username root password router administrator
!
interface Loopback0
 ip address 127.0.0.1 255.0.0.0
!
interface FastEthernet0/0
 no ip address
 speed auto
 no qos-control
!
interface FastEthernet0/1
 no ip address
 speed auto
 no qos-control
ļ
!
ftp server
http server
http document-root /hd
```





```
!
!
! VoIP configuration.
!
! Voice service voip configuration.
!
voice service voip
 fax protocol t38 redundancy 0
 fax rate 9600
 h323 call start fast
 h323 call tunnel enable
!
! Voice port configuration.
!
! FXO
voice-port 0/0
 no caller-id enable
I
! FXO
voice-port 0/1
 no caller-id enable
!
!
! Pots peer configuration.
!
! Voip peer configuration.
!
dial-peer voice 1000 voip
 destination-pattern T
 session target sip-server
 session protocol sip
 no vad
 dtmf-relay dual-mode
 huntstop
!
! Gateway configuration.
!
gateway
```



```
h323-id
 signalling-port 1721
 no ignore-msg-from-other-gk
SIP UA configuration.
!
sip-ua
 signaling-port 5070
 rport enable
!
! MGCP configuration.
!
mgcp
 codec g711ulaw
 vad
!
Tones
!
call-manager sip
 reg-expire-value-fixed enable
!
call-manager sscp store-event-time 3
call-manager sscp store-event-count 10
!
call-manager logger disable
call-manager logger level info
!
call-manager h323
 signalling-port 1720
!
! Network Domain interface configuration.
!
call-manager interface ip FastEthernet0/0 domain public
call-manager interface ip FastEthernet0/1 domain private
!
rtp-proxy
!
line console
I
```



line vty

!

character-set encoding usa ascii

mount mem 8192 /apcm

! mount mem 1024 /tmp

!

```
ldap
```

data-dir /hd/ldap suffix "dc=addpac,dc=com" rootdn "cn=Manager,dc=addpac,dc=com" rootpw secret include /hd/ldap/schema/core.schema include /hd/ldap/schema/cosine.schema include /hd/ldap/schema/inetorgperson.schema include /hd/ldap/schema/addpac.schema include /hd/ldap/schema/apcm.schema include /hd/ldap/schema/apglobal.schema include /hd/ldap/schema/apmessage.schema include /hd/ldap/schema/apms.schema include /hd/ldap/schema/apmd.schema include /hd/ldap/schema/apums.schema ! presence server configuration. ļ I presence service disable sscp store-event-time 1 sscp store-event-count 1 logger disable

logger level info

!

! media server configuration.

!

media

rbt disable

!

Idapclient

ldap disable



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!

end

IP-PBX#



IP, Default Route Setup Confirmation

Following example shows network status for communication. It checks whether the network is properly connected by IPNext ping test.

IP-PBX# configure terminal IP-PBX(config)# interface Fastethernet 0/0 IP-PBX(config-if)# ip address 172.17.201.115 255.255.0.0 IP-PBX(config-if)# exit IP-PBX(config)# ip route 0.0.0.0 0.0.0.0 172.17.1.1 IP-PBX(config)# end IP-PBX# IP-PBX# IP-PBX# write Proceed with write? [confirm] Building configuration... [OK] Configuration saved to flash:/apos.cfg IP-PBX# IP-PBX# ping 172.17.1.1 PING 172.17.1.1 (172.17.1.1): 56 data bytes 64 bytes from 172.17.1.1: icmp_seq=0 ttl=255 time=0 ms 64 bytes from 172.17.1.1: icmp_seq=1 ttl=255 time=0 ms 64 bytes from 172.17.1.1: icmp_seq=2 ttl=255 time=0 ms 64 bytes from 172.17.1.1: icmp_seq=3 ttl=255 time=0 ms 64 bytes from 172.17.1.1: icmp_seq=4 ttl=255 time=0 ms --- 172.17.1.1 ping statistics ---5 packets transmitted, 5 packets received, 0% packet loss'

Table 3-18 IP, Default Route Setup



round-trip min/avg/max = 0/0/0 ms

IP-PBX#

LDAP Setup

Ldap (Lightweight Directory Access Protocol) is software protocol to find file or device such as organization, individual, internet, and intranet. It is a part of directory service standard X.500 in network.

IPNext uses LDAP to setup/save the necessary IP-PBX operation data. Following example shows LDAP setup.

Step	Command	Explanation		
1	IP-PBX#Enter as APOS command input mode.			
	IP-PBX# configure terminal			
	IP-PBX(config)#			
2	IP-PBX(config)# ldap	Enter as Idap setup mode		
3	IP-PBX(config-ldap)# slapd	Idap activation		

Table 3-19 LDAP Setup

Table 3-20 LDAP Notification Server Setup

Step	Command	Explanation
1	IP-PBX#	Enter as APOS command input mode.
	IP-PBX# configure terminal	
	IP-PBX(config)#	
2	IP-PBX(config)# ldap	Enter as Idap setup mode
3	<pre>Router(config-ldap)# notification 5389</pre>	Idap notification server setup and port setup
		(Default TCP Listen Port 5389)

Table 3-21 LDAP Client Setup

Step	Command Explanation			
1	IP-PBX#	Enter as APOS command input mode.		
	IP-PBX# configure terminal			
	IP-PBX(config)#			
2	IP-PBX(config)# ldapclient	Enter as Idapclient setup mode		
3	IP-PBX(config-ldapclient)# name	Setup device nameto register in Ldap		



	IPNext_test	
4	IP-PBX(config-ldapclient)# host	Ldap server IP and Port setup.
	127.0.0.1 389	Default IP 127.0.0.1 Default Port 389
5	IP-PBX(config-ldapclient)# ldap enable	Ldap Client service activation

LDAP Setup Confirmation

* Check LDAP setup status by show run

Table 3-22 LDAP Execution and Confirmation

```
IP-PBX#
IP-PBX# show run
Building configuration...
Current configuration:
!
hostname IP-PBX
!
username root password router administrator
l
interface Loopback0
 ip address 127.0.0.1 255.0.0.0
L
interface FastEthernet0/0
 ip address 172.17.201.79 255.255.0.0
 speed auto
!
interface FastEthernet0/1
 speed auto
!
-----omit-----
!
! APOS File System
L
mount hdd 0 /hd
```



mount mem 8192 /apcm ! mount mem 512 /tmp ! share global workgroup WORKGROUP encrypt-passwords L character-set encoding usa ascii mount mem 8192 /apcm ! mount mem 1024 /tmp ! Idap data-dir /hd/ldap suffix "dc=addpac,dc=com" rootdn "cn=Manager,dc=addpac,dc=com" rootpw secret include /hd/ldap/schema/core.schema include /hd/ldap/schema/cosine.schema include /hd/ldap/schema/inetorgperson.schema include /hd/ldap/schema/addpac.schema include /hd/ldap/schema/apcm.schema include /hd/ldap/schema/apglobal.schema include /hd/ldap/schema/apmessage.schema include /hd/ldap/schema/apms.schema include /hd/ldap/schema/apmd.schema include /hd/ldap/schema/apums.schema => When "slapd" and "notification" configuration does not exist Execute slapd and notification server

!

End

* LDAP Activation and LDAP Notification Server Execution



IP-PBX #

IP-PBX # con t

IP-PBX (config)#

IP-PBX (config)# Idap

IP-PBX (config-ldap)#

IP-PBX (config-ldap)# slapd

LDAP daemon is now active. => LDAP active Status Check

IP-PBX(config-ldap)# notification 5389

IP-PBX(config-ldap)#

Start Ldap Notification Server (listen tcp/5389) => LDAP Notification Server Status Check

IP-PBX #

IP-PBX #

IP-PBX # show run

----- skip ------

```
!!
```

Idap data-dir /hd/ldap suffix "dc=addpac,dc=com" rootdn "cn=Manager,dc=addpac,dc=com" rootpw secret include /hd/ldap/schema/core.schema include /hd/ldap/schema/cosine.schema include /hd/ldap/schema/inetorgperson.schema include /hd/ldap/schema/addpac.schema include /hd/ldap/schema/apcm.schema include /hd/ldap/schema/apglobal.schema include /hd/ldap/schema/apmessage.schema include /hd/ldap/schema/apms.schema include /hd/ldap/schema/apmd.schema include /hd/ldap/schema/apums.schema slapd => LDAP Working Status Check notification 5389 => LDAP Notification Server Status Check **IP-PBX**# **IP-PBX**#



* Items to be confirmed when LDAP is not properly working

IP-PBX # IP-PBX # fsh fsh:/> fsh:/> II total(bytes) 33552 drwxrwxrwx 1 root apos 752 Sep 13 9:10 apcm/ d------ 1 root apos 0 Sep 13 9:09 flash/

drwxrwxrwx 1 root apos 32768 Sep 13 9:09 hd/ drwxrwxrwx 1 root apos 32 Sep 13 11:36 tmp/ fsh:/> fsh:/> cd hd => Check the existence of Idap data in hd. (It's normal if below lists are all included and Execute Ldap Recovery process if one is missing) Ldap Recovery is Smart Multimedia Manager Manual Prefer) fsh:/hd> fsh:/hd> II d------ 1 root apos 2048 Jan 01 12:04 addpac/ d------ 1 root apos 2048 Jan 01 12:05 cdr/ d------ 1 root apos 2048 Jan 01 12:05 en/ d------ 1 root apos 2048 Jan 01 12:06 ko/ d-----1 root apos 2048 Jan 01 12:08 ldap/ drw-rw-rw-1 root apos 2048 Aug 21 09:01 lost+found/ d------1 root apos 2048 Jan 01 12:08 mbox/ d------ 1 root apos 2048 Jan 01 12:08 music/ d------ 1 root apos 2048 Jan 01 12:08 scenario/ d------ 1 root apos 2048 Jan 01 12:08 smartclient/ d------ 1 root apos 2048 Jan 01 12:09 storage/ d------1 root apos 2048 Jan 01 12:12 tone/ d------ 1 root apos 2048 Jan 01 12:12 voice/ fsh:/hd> exit



IP-PBX # IP-PBX

* LDAP Client Setup and Execution

IP-PBX# conf t

IP-PBX(config)# Idapclient

IP-PBX(config-Idapclient)# name IPNext_Idap

IP-PBX(config-Idapclient)# host 127.0.0.1 389

IP-PBX(config-Idapclient)# Idap enable

IP-PBX(config-Idapclient)#

[LDAP_CLIENT] Auto-Registration Complete.device id(56) => LDAP Client Working Status Check



APOS Upgrade

AddPac Technology IPNext permits to access for APOS image file transmitting by using FTP. Also, relevant protocol let user to do service on/off independently.

Upload/download network Configuration is as below.



Picture 3-5 APOS Image File Upgrade by Using FTP

FTP Service Activation

To upload/download the APOS, please activate FTP service in IPNext.

Table 3-23 FTP Server Execution

IP-PBX# configure terminal

IP-PBX(config)#

IP-PBX(config)# ftp server

IP-PBX(config)# Start File Transfer Protocol Server (listen tcp/21)

IP-PBX(config)#



APOS Upload

Use ftp in DOS screen to upload APOS of IPNext. Press window start button and type "cmd" on "run screen" to open DOS screen. Move to image file folder and access to server.

D:\>dir 2008-08-14 15:21p <DIR> 2006-08-14 15:21p <DIR> 2008-08-14 15:21p 5,057,752 IPNext_g2_v8_47_024.bin D:\> D:\> ftp 172.17.113.50 => access to server Connected to 172.17.113.50. 220 IP-PBX FTP server (Version 8.47.010) ready. User (172.17.113.50:(none)): root 331 Password required for root. Password: 230 User root logged in. ftp> binary =>Change to binary mode 200 Type set to I. ftp> hash =>Setup Hash mark input Hash mark printing On ftp: (2048 bytes/hash mark). ftp> put IPNext_g2_v8_47_024.bin => IPNext_g2_v8_47_024.bin file uploading 200 PORT command successful. 150 Opening BINARY mode data connection for 'IPNext_g2_v8_47_024.bin'. ----- 중략 -----226 Transfer complete. ftp: 5057752 bytes sent in 1.92Seconds 2720.69Kbytes/sec. ftp> quit 221 Goodbye. => Even if goodbye message shows up, do not reboot IP-PBX until the "system software" is updated message shows up in console. D:\>

Table 3-24 APOS Upgrade (DOS Screen)

Table 3-25 APOS Upgrade (Console Screen)

IP-PBX#



5057752(0x4fca68) bytes are received and version is "8.47.024" => upgrade processing IP-PBX# IP-PBX#

The "system software" is updated. => upgrade complete



Chapter 4. IPNext Initialization

To initialize (call management file, database schema and basic file) when HDD/flash memory data is damaged due to unexpected errors. Following example shows how to initialize the IPNext data.

* Following procedure uses when Smart Directory Server is damaged or to upgrade. User needs to take extra cautious.

System Setup

Interface Setup

Following example shows how to setup IP address "172.17.111.20" in interface fast Ethernet 0/0.

Step	Command	l	•	Expla	nation		
1	IP-PBX#		Enter as	APOS	command	input	
	<pre>IP-PBX# configure terminal</pre>	IP-PBX# configure terminal		mode			
	IP-PBX(config)#						
2	<pre>IP-PBX(config)# interface fa</pre>	stethernet 0/0	Enter as i	nterface	setup mode	•	
3	IP-PBX(config)# ip addr	ess 172.17.111.20	Setup	IP	address	as	
	255.255.0.0		"172.17.1 [,]	11.20" in	interface		

Table 4-1 Interface Setup



Default Route Setup

This is an example of setup default router as "172.17.1.1".

Table 4-2 Default Route Setup

Step	Command	Explanation
1	IP-PBX#	Enter as APOS command input mode.
	IP-PBX# configure terminal	
	IP-PBX(config)#	
2	IP-PBX(config)# ip route 0.0.0	0 0.0.0.0 Setup default route as "172.17.1.1"
	172.17.1.1	



System Service Setup

HTTP Server Setup

This is an example of setup HTTP Sever activation and Http document-root Directory. (Default: http server, http document-root /hd)

Step	Command	Explanation	
1	IP-PBX#	Enter as APOS command input mode	
	IP-PBX# configure terminal		
	IP-PBX(config)#		
2	IP-PBX(config)# http server	HTTP server activation. (default)	
3	<pre>IP-PBX(config)# no http authentication</pre>	(optional) Setup to access in server without	
		HTTP access authentication. We	
		recommend to use http authentication	
4	<pre>IP-PBX(config)# http document-root /hd</pre>	Default directory setup for Web Based SMM	
		access. (IPNext v8.47.024. after version	
		default: /hd)	

Table 4-3 HTTP Server Setup



FTP Server Setup

This is an example of FTP server activation. (Default: ftp server, ftp port 21)

Table 4-4 FTP Server Setup

Step	Command	Explanation
1	IP-PBX#	Enter as APOS command input mode.
	IP-PBX# configure terminal	
	IP-PBX(config)#	
2	IP-PBX(config)# ftp server	FTP server activation. (default)



File System Initialization (Optional)

File System Initialization (Optional)

This is a setup example to initialize the file system in IPNext. Following setup will initialize saved data/file system in IPNext. User must backup the data/file by using SMM/SMT. (Please refer to Smart Multimedia Manager and IPNext System Maintenance Tool manual for backup method by using SMM and SMT)

······			
Step	Command	Explanation	
1	IP-PBX#	Enter as APOS command input mode.	
	IP-PBX# configure terminal		
	IP-PBX(config)#		
2	<pre>IP-PBX(config)# hdd nand format</pre>	Initialize all file system in IPNext.	
		It may take a while to initialize the file	
		system.	

Table 4-5 File System Initialization Setup

Call-Manager Initialization

System Maintenance Tool Execution and Initialization Process

It performs system maintenance tool process and initializes process. (Please refer to IPNext System Maintenance Tool manual Chapter. 2 System Recovery (system initialize part). System Maintenance Tool installation and initialize process method.



LdapClient Setup

After call manager initialize process and device rebooting is completed through system maintenance tool, setup Ldapclient for call manager service.

	Table 4-0 Luap Client Setup	
Step	Command	Explanation
1	IP-PBX#	Enter as APOS command input
	IP-PBX# configure terminal	mode.
	IP-PBX(config)#	
2	IP-PBX(config)# ldapclient	Enter as Idapclient setup mode
3	IP-PBX(config-ldapclient)# name IPNext_test	Device name setup to register in
		Ldap
4	<pre>IP-PBX(config-ldapclient)# host 127.0.0.1 389</pre>	Ldap server IP and Port setup.
		Default IP (127.0.0.1) Default Port
		(389)
5	<pre>IP-PBX(config-ldapclient)# ldap enable</pre>	Ldap Client service activation

Table 4-6 Ldap Client Setup



Web Based SMM Access

1) Run internet explorer screen. Enter IPNext IP address then press enter.



Picture 4-1 Web Based SMM Access Picture1



2) Move to LDAP input setup screen by clicking click icon.



Picture 4-2 Web Based SMM Access Screen 2



 Enter LDAP access ID, Password, Port information in IPNext. Click on Login icon. Default ID: root, Default Password: router, Default Port: 389)



Picture 4-3 Web Based SMM Access Screen 3



4) Initial main screen will be displayed after access is completed.



Picture 4-4 Web Based SMM Access Screen 4


Chapter 5. Appendix

Console Port Signal and Pin Out

Following appendix explains the specification of cable pins in IPNext.

- Console port signal and pin out (RJ-45 to DB9)
- Pin out of UTP cable (RJ-45 to RJ-45)

Console Port Signal and Pin Out

To connect router console port and terminal emulator software PC, user must use built-in RJ-45 to DB9 (Female DTE connector) cable.

Table 5-1 Console Port Pin Out					
Console Port (DTE)	RJ-45 DB-9		Console Device (PC)		
Signal	RJ-45 Pin	DB-9 Pin	Signal		
RTS	1	8	CTS		
DTR	2	6	DSR		
TxD	3	2	RxD		
GND	4	5	GND		
GND	5	5	GND		
RxD	6	3	TxD		
DSR	7	4	DTR		
CTS	8	7	RTS		



UTP Cable (RJ-45 to RJ-45) Pin Out

Use RJ-45 to RJ-45 Ethernet cable to connect router and other devices (HUB). RJ-45 connector pin order is shown in Picture 4-1.



Picture 5-1 100Base-TX RJ-45 Connector

RJ-45	Signal	Direction	RJ-45 Pin
1	Tx +	\rightarrow	1
2	Tx -	\rightarrow	2
3	Rx +	\leftarrow	3
4	-	-	4
5	-	-	5
6	Rx -	\leftarrow	6
7	-	-	7
8	-	-	8

Table 5-2 Series Ethernet Cable Signal and Pin out

- 1. This specification is cable specification for series cable in between router and hub.
- 2. Use cross cable to directly connect router to PC or router to router.



Abbreviation and Terminology

Abbreviation/Term	Definition and Explanation		
ADSL	Stands for Asymmetric Digital Subscriber Line. If you use ADSL, the central office		
	will be connected to each home directly in a 1:1 method. In a down-link where data		
	is transferred downward from the central office to the users, high-speed data		
	communication of at least 1.5 Mb can be made. On the contrary, in an up-link		
	from the users to the central office, communications are made very slowly. Thus,		
	this service is called an asymmetrical service not a symmetrical service.		
API	Stands for Application Programming Interface. API is a function call legend		
	standard that defines service interfaces.		
APOS	Stands for AddPac Internetworking Operation System. This is an operating system		
	that supports the network products developed by AddPac Technology.		
Authentication	Operation of verifying the identification of a person or a process. This is a security		
	feature.		
BNC Connector	IEEE 802.3 10Base-2 coaxial cable is standard connector to connect MAU(Media		
	Access Unit)		
Boot Loader	This is a chip installed into a printed circuit board used to send executable boot		
	commands to a network device.		
Bps	Stands Bits per second. Typically called bps. Refer to bit rate.		
Cable Modem	This device converts analog signals to digital signals in order to enable the Internet		
	through a cable network. Since telephone networks are made of copper wires and		
	cable networks are made of coaxial and optical cables, the bandwidth of cable		
	networks are much wider than that of telephone networks. However, the		
	modulation/demodulation technology, which converts digital to analog and vice		
	versa, is required for cable networks when data is transferred.		
Call Center	Call Center is a central place where calls from customers and other people are		
	processed systematically. Computer automation is implemented in Call Center to		
	some degree. Typically, Call Center processes many calls simultaneously,		
	categorizes calls, connects the calls to personnel, and records calling logs		
	automatically. Call Center is typically used for mail order catalog firms,		
	telemarketing firms, customer centers for PC products, and large enterprises that		
	sell products or provide services.		
Caller ID	Caller ID is phone service which sends caller's phone number to call receiver.		
	However, digital reader must be attached to the phone.		
Category 5 cabling	One of the five-level UTP cable connection methods specified by the EIA/TIA-586		
	standard. Category 5 cabling enables data to be transferred at a rate of up to		



	100Mbps.
Checksum	This is a method for checking the integrity of transferred data. Checksum is an
	integer calculated from the octet sequence obtained by a series of operations. This
	value is calculated by the recipient again for verification.
Coaxial cable	This coaxial cable is made of an external cylinder-type conductor that wraps an
	internal wire conductor. Examples of the coaxial cables used for LAN include 50 $\!\Omega$
	cables used for digital signal processing and 75Ω cables used for high-speed
	digital signal processing.
CODEC	Abbreviation of Coder-Decoder 1. Convert analog signal to digital bit stream by
	using purse code modulation, and convert digital signal to analog signal again.
	2. DSP software algorithm to compress/decompress voice signal or audio signal
	such as Voice over IP, Voice over Frame Relay, Voice over ATM.
Console	DTE interface (It is a path to enter host)
CoS	Stands for Class of Service. CoS refers to the standard method that enables a
	higher-level protocol to make a lower-level protocol process messages. For the
	SNA lower-level area routing, CoS is used to determine the optional path for lower
	level area nodes to set a given session. CoS consists of a virtual path number and
	a transmission priority field. Also called ToS
Decryption	Decryption means restoring data to the original non-encrypted state by applying
	the encryption algorithm to the encrypted data in reverse.
DHCP	Stands for Dynamic Host Configuration Protocol. DHCP has a mechanism that
	reassigns an IP address dynamically in order for the host to recycle unnecessary
	IP addresses.
DNS	Stands for Domain Name Server. This is a server system used for the Internet to
	convert the name of a network node name to an address.
DS-3	Stands for Digital Signal level 3. This is a frame processing standard used to
	transmit digital signals at a rate of T3 (44.736Mbps).
DSP	Stands for Digital Signal Processor. This is a dedicated processor that processes
	only digital signals. DSP is used as a sub-processor for voice processing in NEXT.
DTMF	Stands for Dual Tone MultiFrequency. Two voice-band tones are simultaneously
	used for dialing (just like touch tones).
E&M	Stands for either recEive and transmit or Ear and Mouth. Typically, this is a
	trunking device used for switch-to-switch or switch-to-network two-way
	communications. The analog E&M interface of Cisco is a RJ-48 connector for PBX
	trunk lines. E&M is available for E1/T1 digital interfaces.
E1	This is a wide area digital transmission technique used mainly in Europe. E1
	enables data transfer at a rate of 2,048Mbps. E1 can be lent by regular service
	providers for a private use.
Encryption	To apply specific algorithm for those who do not have right to access into data.
Fast Ethernet	Baseband LAN standard initiated by Xerox Corporation and co-developed by



	Xerox, Intel, and DEC. CSMA/CD is used for Ethernet networks, which operate
	through a variety of cables at a rate of 10Mbps. Ethernet is similar to the IEEE
	802.3 standards. Refer to 10Base-2, 10Base5, 10Base-F, 10Base-T, 10Broad-36,
	Fast Ethernet and IEEE 802.3.
FAX	Abbreviation of Facsimile. FAX refers to the transmission of scanned texts or
	images to a printer or an output device connected to another phone number by
	using a telephone line. Once the original document is read by a facsimile, the
	facsimile treats the document as a fixed graphic image, and converts it to bitmap.
	In this digital form, data is transferred in the form of an electrical signal through a
	phone system. The receiving facsimile restores the data to a encoded image, and
	prints it on a sheet of paper.
FTP	Stands for File Transfer Protocol. FTP, which is an application protocol, is part of
	the TCP/IP protocol stack used for file transfer between network nodes. FTP is
	defined in RFC 959.
FXO	Stands for Foreign Exchange Office. The FXO interface is connected to the
	switching center of Public Switched Telephone Network (PSTN), and is provided
	by a regular phone. The FXO interface of Cisco is a station interface of the
	switching center or PBX on PSTN, and is a RJ-11 connector for analog connection
	devices.
FXS	Stands for Foreign Exchange Station. The FXS interface is directly connected to a
170	standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS
1.0	standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX.
G.711	standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX. This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under
G.711	standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX. This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under G.711 in an appropriate format that enables digital voice transmission over either
G.711	standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX. This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under G.711 in an appropriate format that enables digital voice transmission over either PSTN or PBX. G.711 is specified under the ITU-T standard of G-series
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G.711 G.723.1	standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX. This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under G.711 in an appropriate format that enables digital voice transmission over either PSTN or PBX. G.711 is specified under the ITU-T standard of G-series recommendation. This is one of the H.324 standards, and specifies a compression technique that enables voice or audio signal elements to be compressed at a very low bit transmission rate. This CODEC is related to the bit transmission rates of 5.3Kpbs and 6.3Kpbs. The high bit transmission rate is based on the MLMLQ technology, and provides high quality sounds. The low bit transmission rate is based on CELP,
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G.723.1 G.726	 standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX. This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under G.711 in an appropriate format that enables digital voice transmission over either PSTN or PBX. G.711 is specified under the ITU-T standard of G-series recommendation. This is one of the H.324 standards, and specifies a compression technique that enables voice or audio signal elements to be compressed at a very low bit transmission rate. This CODEC is related to the bit transmission rates of 5.3Kpbs and 6.3Kpbs. The high bit transmission rate is based on the MLMLQ technology, and provides high quality sounds. The low bit transmission rate is based on CELP, and ensures high flexibility for system designers. This standard is specified under the G-series ITU-T standard.
G.723.1 G.726	 standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX. This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under G.711 in an appropriate format that enables digital voice transmission over either PSTN or PBX. G.711 is specified under the ITU-T standard of G-series recommendation. This is one of the H.324 standards, and specifies a compression technique that enables voice or audio signal elements to be compressed at a very low bit transmission rate. This CODEC is related to the bit transmission rates of 5.3Kpbs and 6.3Kpbs. The high bit transmission rate is based on the MLMLQ technology, and provides high quality sounds. The low bit transmission rate is based on CELP, and ensures high flexibility for system designers. This standard is specified under the G-series ITU-T standard. This standard specifies ADPCM coding performed at a rate of 40Kbps, 32Kbps, 24Kbps, or 16Kbps. If the PBX network is configured to support ADPCM, you can
G.711 G.723.1 G.726	 standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX. This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under G.711 in an appropriate format that enables digital voice transmission over either PSTN or PBX. G.711 is specified under the ITU-T standard of G-series recommendation. This is one of the H.324 standards, and specifies a compression technique that enables voice or audio signal elements to be compressed at a very low bit transmission rate. This CODEC is related to the bit transmission rates of 5.3Kpbs and 6.3Kpbs. The high bit transmission rate is based on the MLMLQ technology, and provides high quality sounds. The low bit transmission rate is based on CELP, and ensures high flexibility for system designers. This standard is specified under the G-series ITU-T standard. This standard specifies ADPCM coding performed at a rate of 40Kbps, 32Kbps, 24Kbps, or 16Kbps. If the PBX network is configured to support ADPCM, you can exchange ADPCM encoding voice with packet voice networks, PSTN, or PBX
G.711 G.723.1 G.726	 standard phone, and provides a ring-back tone, voltage, and a dial tone. The FXS interface of Cisco is a RJ-11 connector for basic telephone service devices, keyset, and PBX. This specifies the PCM voice coding technique of 64Kbps. Voice is encoded under G.711 in an appropriate format that enables digital voice transmission over either PSTN or PBX. G.711 is specified under the ITU-T standard of G-series recommendation. This is one of the H.324 standards, and specifies a compression technique that enables voice or audio signal elements to be compressed at a very low bit transmission rate. This CODEC is related to the bit transmission rates of 5.3Kpbs and 6.3Kpbs. The high bit transmission rate is based on the MLMLQ technology, and provides high quality sounds. The low bit transmission rate is based on CELP, and ensures high flexibility for system designers. This standard is specified under the G-series ITU-T standard. This standard specifies ADPCM coding performed at a rate of 40Kbps, 32Kbps, 24Kbps, or 16Kbps. If the PBX network is configured to support ADPCM, you can exchange ADPCM encoding voice with packet voice networks, PSTN, or PBX networks. This standard is specified under the ITU-T standard of G-series



G.728	This standard specifies variations that ensure low delay of CELP voice
	compression performed at 16Kbps. The CELP voice coding should be converted to
	a public telephony format for transmission over either PSTN or PSTN. This
	standard is specified under the ITU-T standard of G-series recommendation, and
	defines the CELP compression that encodes G.729 voice to a stream of 8Kbps.
	G.728 has two variations (G.729 and G.729 Annex A), and the variations are
	different in terms of calculation complexity. The two variations have voice quality
	similar to ADPCM of 32Kbps. G.728 is specified under the ITU-T standard of G
	series recommendation.
Gatekeeper	This is the component of the H.323 video conference system that analyzes a caller
	ID, controls access authorization, and manages the subnet bandwidth. A
	gatekeeper is H.323 entity that provides the features that enable address
	conversion and LAN access control to the H.323 terminal and gateway on LAN.
	Gatekeepers can provide other services such as bandwidth control and search for
	a gateway to the H.323 terminal and gateway. This device manages a device
	registry on a multimedia network. The devices are registered with the gatekeeper,
H.225	ITU standard for H.225.0 session setup and packet process application. H.225.0
	actually regulated various protocols such as RAS, Q.931 usage, RTP usage.
H.245	ITU standard for H.245 endpoint control.
H.323	This standard is an extension of the ITU-T standard H.320 that enables voice
	conferences over LAN or another packet switching network as well as video
	transmission over the Internet.
HBD3	A type of line code that is used in E1 line.
HDLC	Stands for High-Level Data Link Control. HDLC is a transmission protocol used in
	the data link layer, which is the second layer of the 7-layer OSI model. HDLC is
	used in the X.25 packet switching network. Data consists of frames in HDLC, and
	frames are transmitted through a network. The destination verifies if the frames
	have been successfully transmitted. The HDLC protocol includes data for
	controlling data flow and troubleshooting errors in a data frame.
Hookflash	This is short on-hook duration of a device such as phones during a call.
	Hookflash means that a phone attempts to make a dial tone recall through PBX.
	This is usually used to perform call transfer.
HTTP	An abbreviation of Hypertext Transfer Protocol. It is a protocol to send text file or
	graphic file.
IPSec	Stands for Internet Protocol Security protocol. IPSec is a still developing standard
	for the security of networks or the packet processing layer of network
	communications. In the previous security techniques, security has been included in
	the application layers of a communication model. IPSec is particularly useful for the
	implementation of remote user access through dial-up access to Virtual Private



	Networks (VPN) and regular private networks. The main advantage of IPSec is that
	security can be ensured without replacing an individual user PC with a new one.
	Cisco takes the initiative of suggesting IPSec as the standard, and has embedded
	support to this feature into its network router.
IPv6	IPv6 is the latest IP, and has been embedded into part of IP support into many
	products including the operating systems of PC. IPv6 is called IP Next Generation
	(IPng), that is the next-generation IP. IPv6 is the formal IETF standard. IPv6 is
	designed as an evolutional version of the currently used IP version 4. Network
	hosts or intermediate nodes that adopt either IPv4 or IPv6 can process any
	packets formulized by either IPv4 or IPv6; thus, the users and service provides can
	upgrade their IP to IPv6 individually without collaboration.
ISP	Stands for Internet Service Provider. ISP refers to service providers that provide
	Internet access services, Web site construction and Web hosting services to
	individuals or enterprises. ISP has devices and communication lines required for
	Internet access, and large ISPs have their own high-speed dedicated lines in order
	to provide services that have better quality and are less dependent on telephone
	network service providers to their customers. The large nationwide ISPs of the U.S.
	are AT&T WorldNet, IBM Global Network, MCI, Netcom, UUNet, and PSINet.
	Those of Korea are INet, Channeli, Netsgo, and Netian. The users access the
	Internet through online service providers. The main online service providers of the
	U.S. are America Online and Compuserve, and those of Korea are Chollian, Unitel,
	and Hitel.
ITU-T	Stands for International Telecommunication Union Telecommunication
	Standardization Sector. This is an international organization that develops global
	standards on communication technologies. ITU-T performs the previous tasks of
	CCITT.
IVR	Stands for Interactive Voice Response. IVR refers to a system that provides data in
	the form of recorded messages through phone lines as a response to user input in
	the form of human voice or mainly DTMF signal processing. Examples are banks
	that allow you to check balance by using a phone or automated stock quotations
	system.
LAN	Stands for Local Area Network. This is a low-error, high-speed data network that
	covers relatively small geographical areas of up to several thousand meters. LAN
	inter-connects workstations, peripherals, terminals, and other devices in a building
	or a geographically limited area. The LAN standard specifies a cable connection
	and signal processing method in the physical layer and data link layer of the OSI
	model. Reference: MAN, WAN.
Link	This is a network communication channel configured with lines or a transmission
	path between the transmitter and receiver and related devices. A link mainly refers



	to WAN connections, and is sometimes called a line or a transmission link.
Loopback test	This test is performed as follows: Transmit a signal or return it to the transmitter at
	a location on the communication path. This loopback test is usually performed to
	test the availability of network interfaces.
MAC Address	Stands for Media Access Control Address. This is a standard data link layer
	address required for any and all ports and devices connected to LAN. Other
	devices on a network use this address to locate a specific port within the network
	and to create or update a routing table and data structure. A MAC address is 6
	bytes long, and is managed by IEEE. A MAC address is called as a hardware
	address, a MAC-layer address, or a physical address. Compare to: Network
	Address
MAN	Stands for Metropolitan-Area Network. This network covers the entire area of a
	large city. The operation area of MAN is geographically larger than that of LAN;
	however, is smaller than that of WAN. Compare to: LAN, WAN.
MGCP	MGCP, which is also known as H.248 or Megaco, is a standard protocol required
	to operate signals required during a multimedia conference or to manage sessions.
	This protocol defines a method of communications between the media gateway
	that converts the data format required for a circuit switching network to the one
	required for a packet switching network and the media gateway control device.
	MGCP may be used to set up, manage, and complete calls among multiple
	endpoints. Megaco and H.248 are the improved version of MGCP.
NAT	Stands for Network Address Translation. NAT is a mechanism for reducing the
	need for globally unique IP addresses. NAT allows you to access the Internet as an
	organization whose address is not globally unique converts the address to an
	address space where the address can be globally routed. NAT is also called
	Network Address Translator.
NTP	Stands for Network Time Protocol. NTP, which is built based on TCP, sets a local
	time accurately based on a wireless clock and an atomic clock on the Internet. NTP
	can synchronize a distributed clock in the unit of milliseconds for a long time.
PABX	Stands for Private Automatic Branch eXchange. PABX is a switch for phones used
	at enterprises. PABX is used in Europe, while PBX is used in the U.S.
Packet	A packet is a group of logical data that contains user data and a header where
	control data is contained. A packet mainly refers to the unit of network layer data.
РВХ	Stands for Private Branch eXchange. PBX, which is located in a subscriber
	building, is a digital or analog phone switchboard used to connect private networks
DINO	to public phone networks.
PING	Stands for Packet INternet Groper. ICMP echo-processes a response between
	messages. PING is used for an IP network to test the accessibility of network
	devices.



Point to Point Connection	One of the two basic connection types. In ATM, the point to point connection may		
	be either a one-way connection or a two-way connection between two ATM end		
	systems.		
Pont to Multipoint	One of the two basic connection types. In ATM, the point to multipoint connection		
Connection	is a one-way connection method that enables a transmitting end-system (root		
	node) to be connected to multiple receiving end-systems (riff). Compare to: Point to		
	Point Connection		
POTS	An abbreviation of Plain Old Telephone Service. Reference Item: PSTN.		
PPP	Stands for Point-to-Point Protocol. This protocol is the advanced version of SLIP		
	that enables a router-to-router connection or a host-to-network connection through		
	synchronous or asynchronous lines. SLIP is designed to be used on an IP, while		
	PPP is used along with network layer protocols such as IP, IPX, and ARA. PPP		
	has a bulletin board security mechanism such as CHAP and PAP. PPP has two		
	subprotocols, LCP and NCP. Reference: CHAP, LCP, NCP, PAP, and SLIP		
Protocol Stack	This is a collection of communication protocols that inter-work with one another		
	and that process communications in part or all of the seven layers of the OSI		
	reference model. All protocol stacks are not related to each layer of the OSI model,		
	and one protocol of a stack can process multiple layers at one time. TCP/IP is a		
	typical protocol stack.		
PSTN	An abbreviation of Public Switched Telephone Network. A general term for various		
	telephony network services. It is also known as POTS.		
PVC	Stands for either Permanent Virtual Circuit or Permanent Virtual Connection. PVC		
	is a virtual circuit installed permanently. PVC allows you to reduce a bandwidth for		
	setting up or releasing a circuit when a specific virtual circuit must always exist. As		
	an ATM term, PVC is called Permanent Virtual Connection.		
Q.931 Signaling	This is an ITU standard that specifies ISDN signal processing methods. The		
	H.225.0 standard uses a variation of Q.931 to set up or disconnect the session of		
	Н.323.		
QoS	Stands for Quality of Service. QoS is the criterion of measuring the performance		
	(e.g. transmission quality and service availability) of a transmission system.		
RAM	An abbreviation of Random-Access Memory. It is a memory which microprocessor		
	Can read or write.		
GAN	an appreviation or Registration, Admission, and Status protocol. This protocol finds datekeeper and use H.323 for communication		
RISC	An abbreviation of Reduced Instruction Set Computing.		
IP-PBX	This is a network laver device that determines the optional route to which network		
	traffic is delivered by using one or more metrics. A router forwards packets from a		
	network to another network based on the network layer information. A router is		
	sometimes called a gateway. (A gateway in this meaning is getting older.)		
	sometimes called a gateway. (A gateway in this meaning is getting order.)		



	Compare		
	to: Gateway; Reference: Relay		
RS-232	Physical layer interface. It is known as EIA/TIA-232.		
RTCP	An abbreviation of RTP Control Protocol. It monitors QoS (IPv6 RTP connection) and delivers the processing session related information. Reference Item: RTP (Real-Time Transport Protocol)		
RTP	1. Stands for Routing Table Protocol. This VINES routing protocol based on RIP		
	distributes network topology data, and helps the VINES server that searches for		
	adjoining clients, servers, and routers. A delay time is used as a routing metric.		
	Reference: SRTP		
	2. Stands for Rapid Transport Protocol. RTP provides facing and error recovery		
	services to the APPN data when the data passes the APPN network. RTP allows		
	you to check error recovery and flow control synthetically. RTP does not recover		
	but prevents traffic congestion.		
	3. Stands for Real-Time Transport Protocol. This is one of the IPv6 protocols. RTP		
	is designed to enable the synthetic network transmission feature in the application		
	that transfers real-time data such as audio, video, and simulation data through		
	multicast or unicast network services. RTP enables the real-time application to		
	identify a payload type, specify a sequence number, perform time-stamping, and to		
	monitor a transmission procedure.		
SIP	Stands for Session Initiation Protocol. SIP is an application layer control protocol		
	based on very simple texts, and allows more than one user to make, correct, or		
	complete a session. Examples of sessions include remote conferences, phones,		
	meetings, event notifications, and instant messaging on the Internet. SIP is		
	independent to lower-level packet protocols (e.g. TCP, UDP, ATM, and X.25).		
SNMP	Stands for Simple Network Management Protocol. This is a network management		
	protocol almost dedicated to TCP/IP networks. SNMP monitors and controls		
	network devices, and manages setup, collection of statistical data, operation		
	performance, and security features. Reference: SGMP and SNMP2		
SSCP	An abbreviation of Smart Service Control Protocol. It is a protocol between		
	AddPac exclusive terminal and server. It is used for powerful system and service		
 T1	Control by Improving SIP		
11	speed of 1.5444Mbps through phone switching network Comparison Item: E1.		
	Reference Item: AMI, B8ZS, DS-1.		
TCP/IP	An abbreviation of Transmission Control Protocol/Internet Protocol. It is a general		
	name for protocol suit to support worldwide internetwork establishment. TCP and		
	IP are well known protocols. Reference Item: IP, TCAP.		
Telco	An abbreviation of Telephone Company. Telco indicates a company that provides		
	priorie service to the customers; it usually means independent inner city phone		
	providers such as beli operating company. Sometimes, it means a company that		



	provides long distance phone provider.				
Telnet	A standard terminal emulation protocol that is included in TCP/IP protocol stack.				
	Telnet is used for remote terminal connection. Telnet is defined in RFC 854.				
VDSL	An abbreviation of Very-high-data-rate Digital Subscriber Line. VDSL provides				
	13Mbps~52Mbps downstream and 1.5Mbps~2.3Mbps through single twisted fair				
	cooper line. A range of VDSL is restricted in between 1,000ft and 4,500ft. Compare				
	Item: ADSL, HDSL, SDSL.				
VoIP	An abbreviation of Voice over IP. It is capable of delivering same function,				
	reliability, and voice quality such as POTS. Voice traffic (ex. Call/fax) can be				
	delivered by using Voice over IP function. DSP breaks down the voice signal into				
	frames, and these frames are saved in voice packet.				
VPN	An abbreviation of Virtual Private Network. Because of traffic encrypt, TCP/IP				
	network can be moved safely.				
WAN	An abbreviation of Wide-Area Network. It is a data communication network to				
	provide service to the users in wide area and use digital transmission service that				
	is provided by communication operator. (EX. Frame relay, SMDS, and X.25 are				
	examples of WAN) Compare Items: LAN, MAN.				



Warranty

Name	IPNext (Serial No.:))			
Date	20	• •	. ^	-	20	•	(1 Year)
	Address						
User	Company						TEL
	Name						
	Address						
Seller	Company						TEL
	Name						

Product Warranty Regulation

- If the product breakdown under the normal operation, we will repair the product for free of charge.
- Our company provides the repair, exchange without extra charge. Any removed parts will belong to our company.
- This paper never guarantees the breakdown due to natural disaster, catastrophe, transportation, modification and etc.
- An extra service charge will be incurred if the service is not included in this warranty. This warranty only valid in Korea.
- Addpac is not responsible for a claim for damages from the third party.
- A product repair, exchange and refund follow the consumer protection board.



AddPac

AddPac Technology

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