

New IPNext10000 Large Capacity Call Manager

High-performance Next Generation Large Capacity Call Manager Solution



AddPac

AddPac Technology

Sales and Marketing

Content

- Product Overview
- Hardware Specification
- System Redundancy Features
- Software Service
- Call Manager Software Component
- IP Telephony Service and Features
- IP Centrex Service
- Network Service and Features
- AddPac User Terminals & Software
- Application Network Diagram
- Web based Smart Multimedia Manager
- Ordering Information



Product Overview

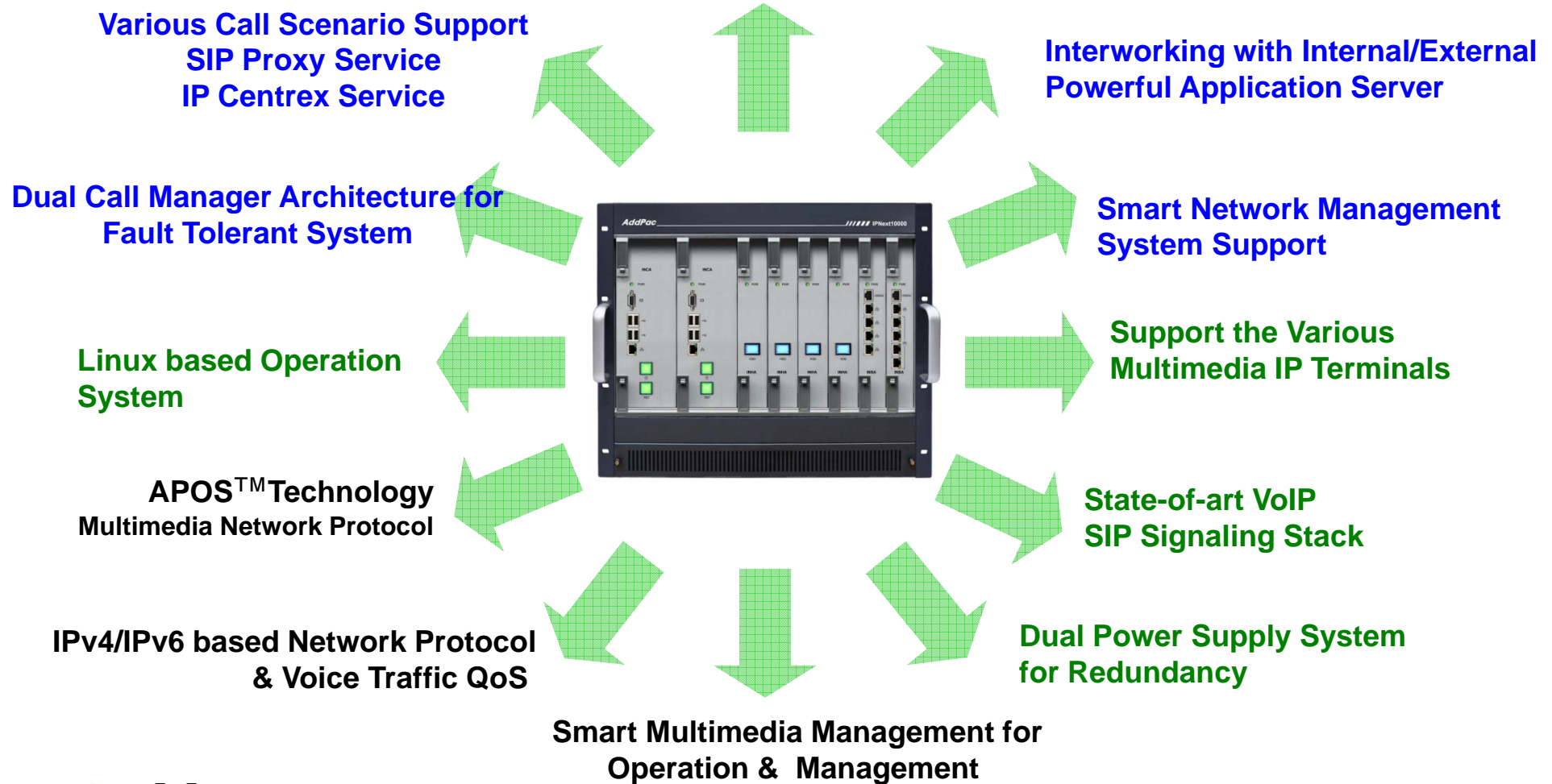
IPNext10000 Large Capacity Call Manager

- SIP Application Server, Proxy, Registrar and Location Server
- Multiple ITSP Trunk with SIP & H.323 Accounts Support
- One(1) System Dual Call Manager Redundancy Architecture
 - One(1) Gigabit Ethernet Interface, Two(2) Hard Disk / IP-PBX
 - Default : Single Call Manager
 - Option A : Dual Call Manager
 - Option B : One(1) Call Manager + One(1) Application Server
- IPv4/IPv6 based Dual Network Protocol Support
- External RTP Proxy Function Support
 - External RTP Proxy Server for Private Address : AP-RS3000
- External Application (IVR, RBT, UMS,etc) Server Support
 - External Application Server : AS10000
- Powerful Management and User Friendly Features
- Fault Tolerant and Scalability Architecture
- High-performance Video, Audio, and Voice Service
- Firmware Upgradeable Architecture
- Linux Operation System
- Smart Multimedia Manager for IP-PBX Management
- Dual Redundancy Power Module

Product Highlights

IPNext10000 Large Capacity Call Manager

IP based Advanced IP-PBX Solution



Hardware Specification

IPNext10000 Large Capacity Call Manager System

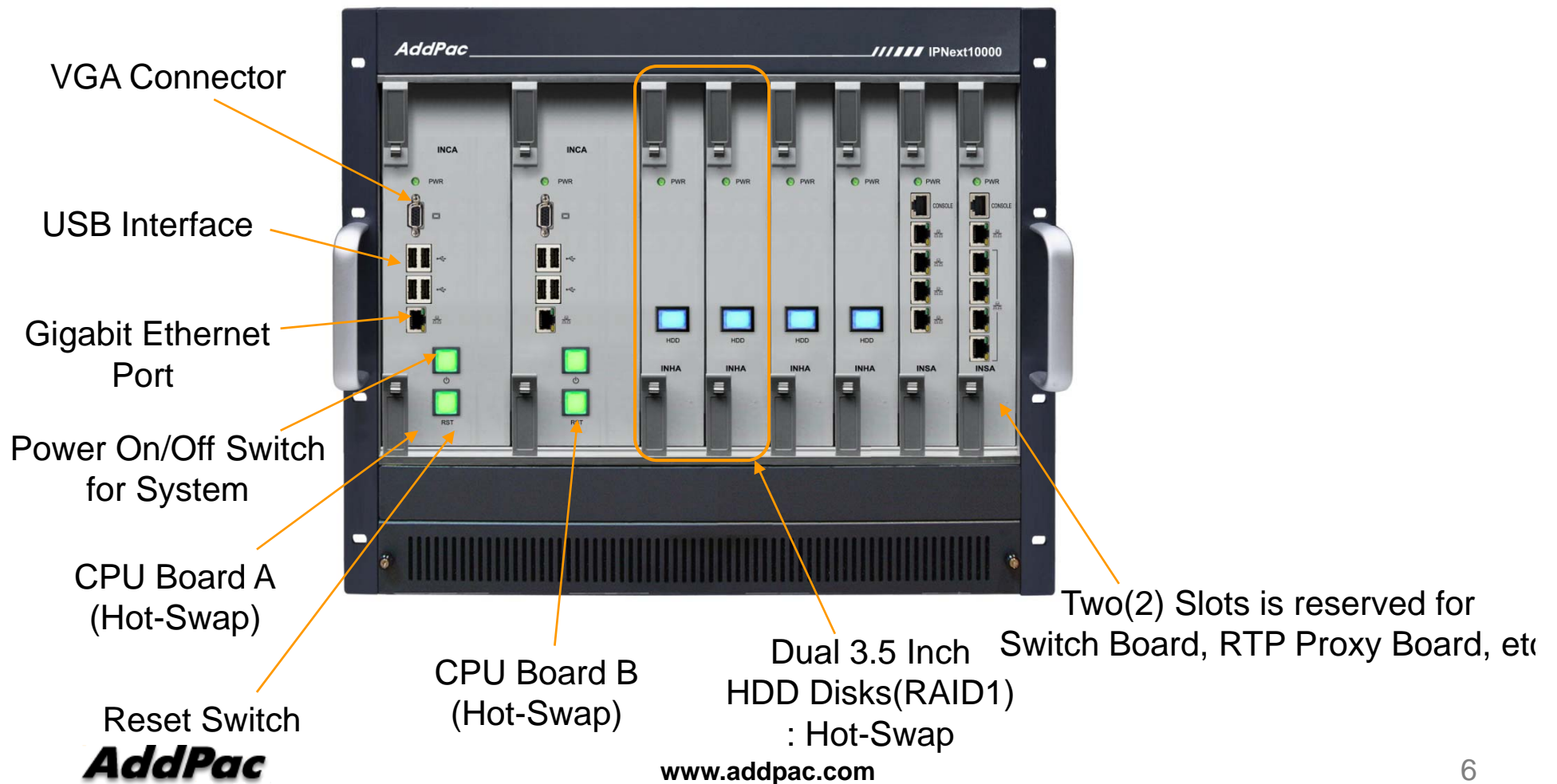
- High-End Computing Power
- Main Chassis
 - Dual Redundancy CPU Boards for System Fault Tolerant
 - One(1) 10/100/1000Mbps Gigabit Ethernet
 - Four(4) USB Interface Port
 - VGA Interface Port for External Video Monitor
 - Two(2) 3.5 Inch Hard Disk Interface Slot (RAID 1)
 - Dual Redundancy Power Supply Module
 - Hot-Swap Features
 - Right most two(2) slots is reserved for Switch Board, Proxy Server, etc

Hardware Specification

IPNext10000 Large Capacity IP-PBX System



IPNext10000 Front Side

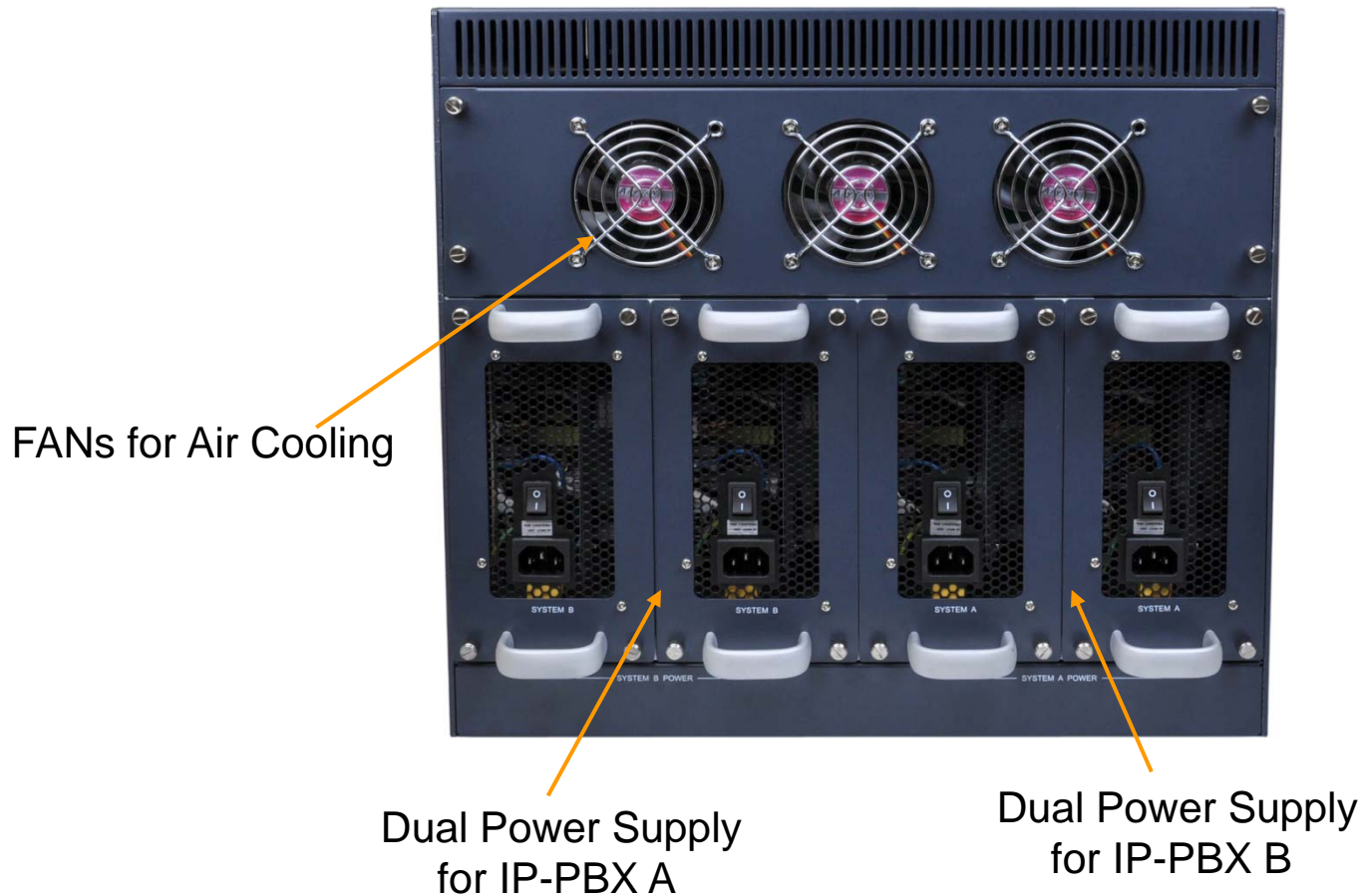


Hardware Specification

IPNext10000 Large Capacity Call Manager

RISC
CPU

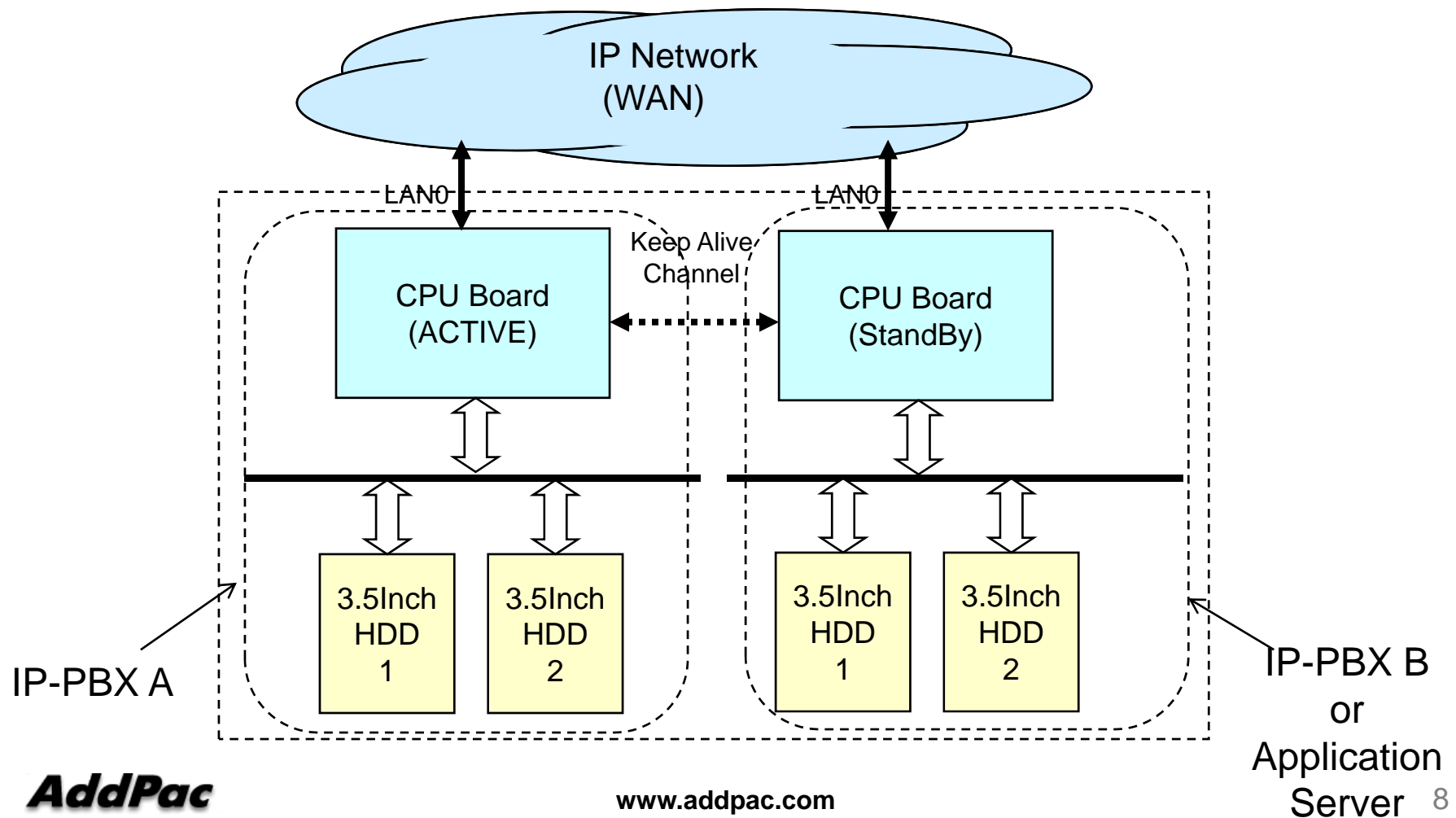
IPNext10000 Back Side



System Redundancy Features

IPNext10000 Large Capacity Call Manager

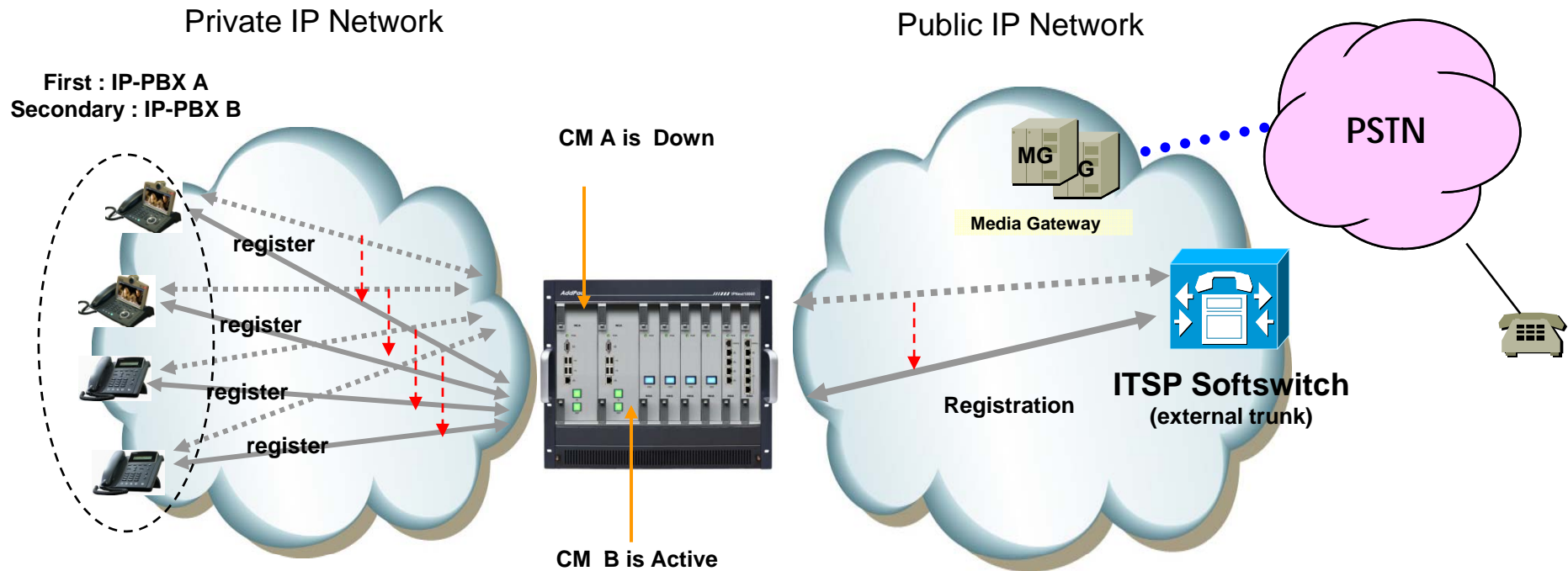
IPNext10000 System Block Diagram



System Redundancy Features

IPNext10000 Large Capacity Call Manager

- Active– Active Duplication Scheme
- Active – Standby Duplication Scheme
- VRRP based Duplication Scheme

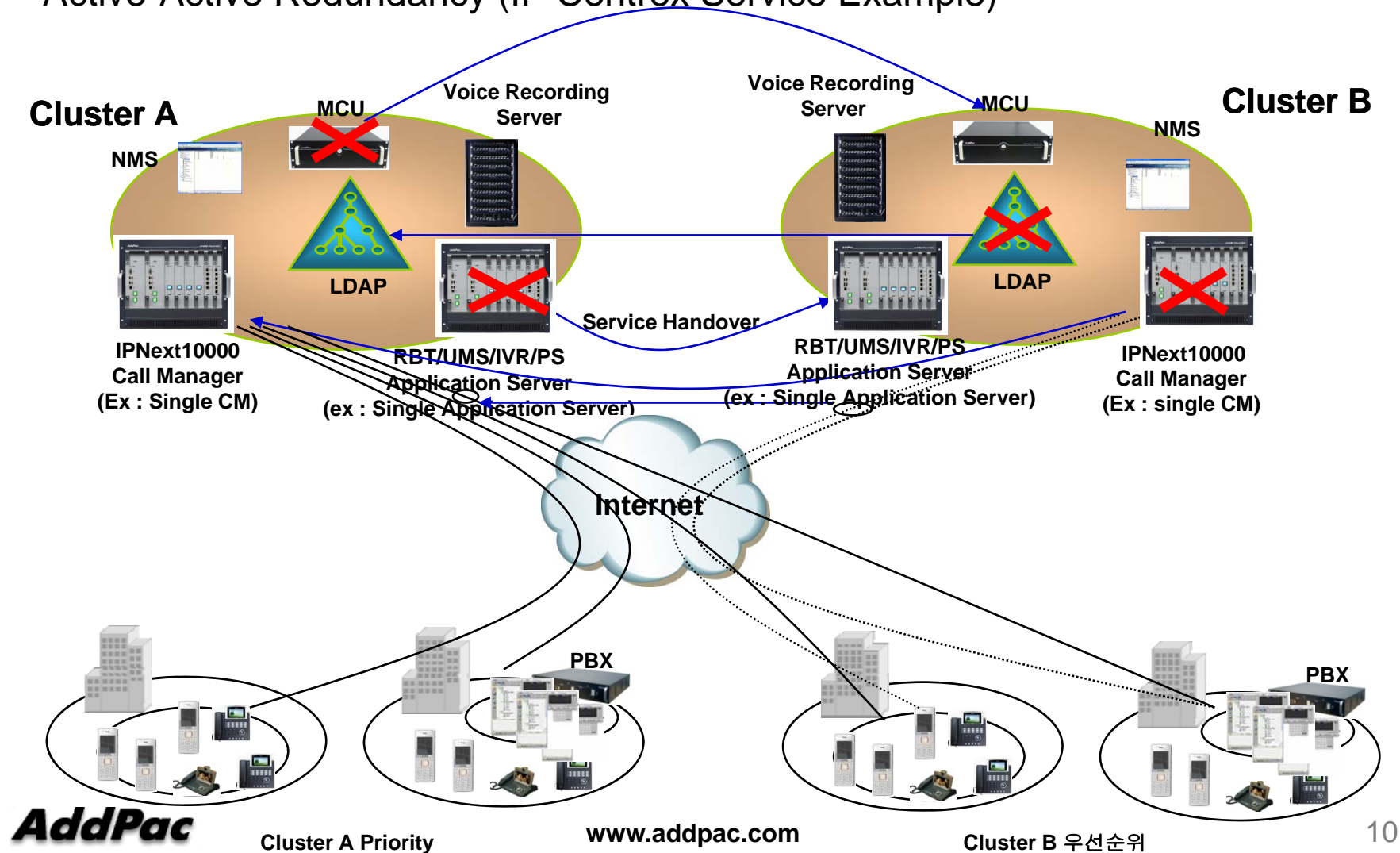


Active – Standby Duplication Scheme (example)

System Redundancy Features

IPNext10000 Large Capacity Call Manager

- Active-Active Redundancy (IP Centrex Service Example)



Software Service

IPNext10000 Large Capacity Call Manager

- **Built-in AddPac APOS Internetworking Software**

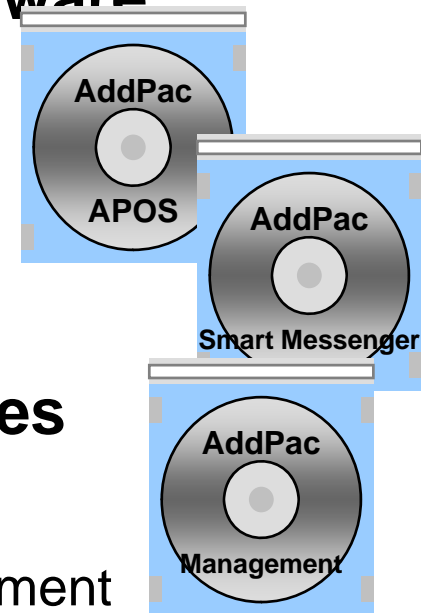
- Scalability, Functionality, and Stability Features
- Advanced Call Manager Features
- QoS Control Features

- **Firmware Upgradeable Architecture**

- **Industry Standard Network Protocol Features**

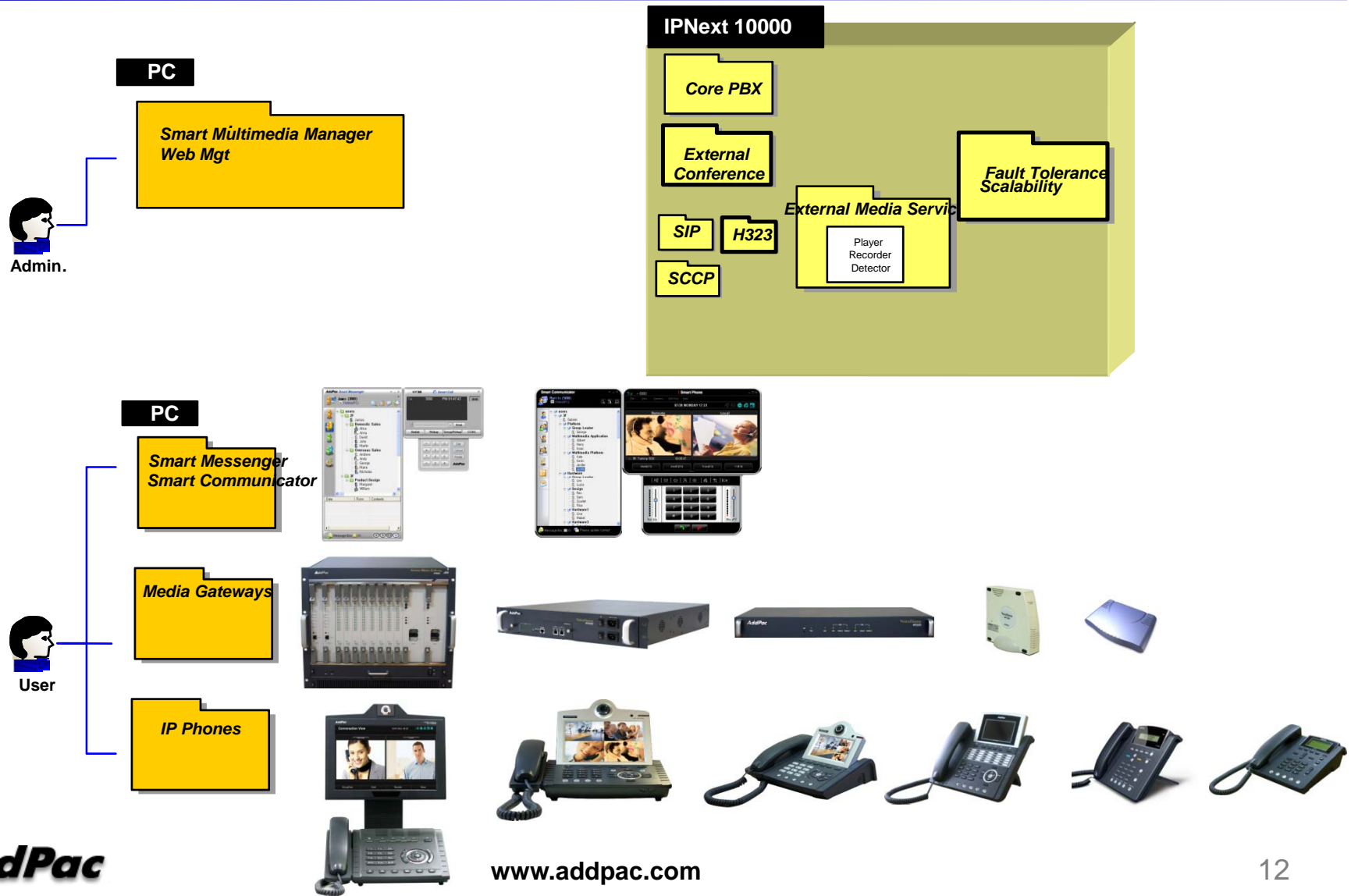
- **Highly User Friendly Management Features**

- Smart Multimedia Manager for Operation & Management



IP-PBX Software Components

IPNext10000 Large Capacity Call Manager



AddPac

www.addpac.com

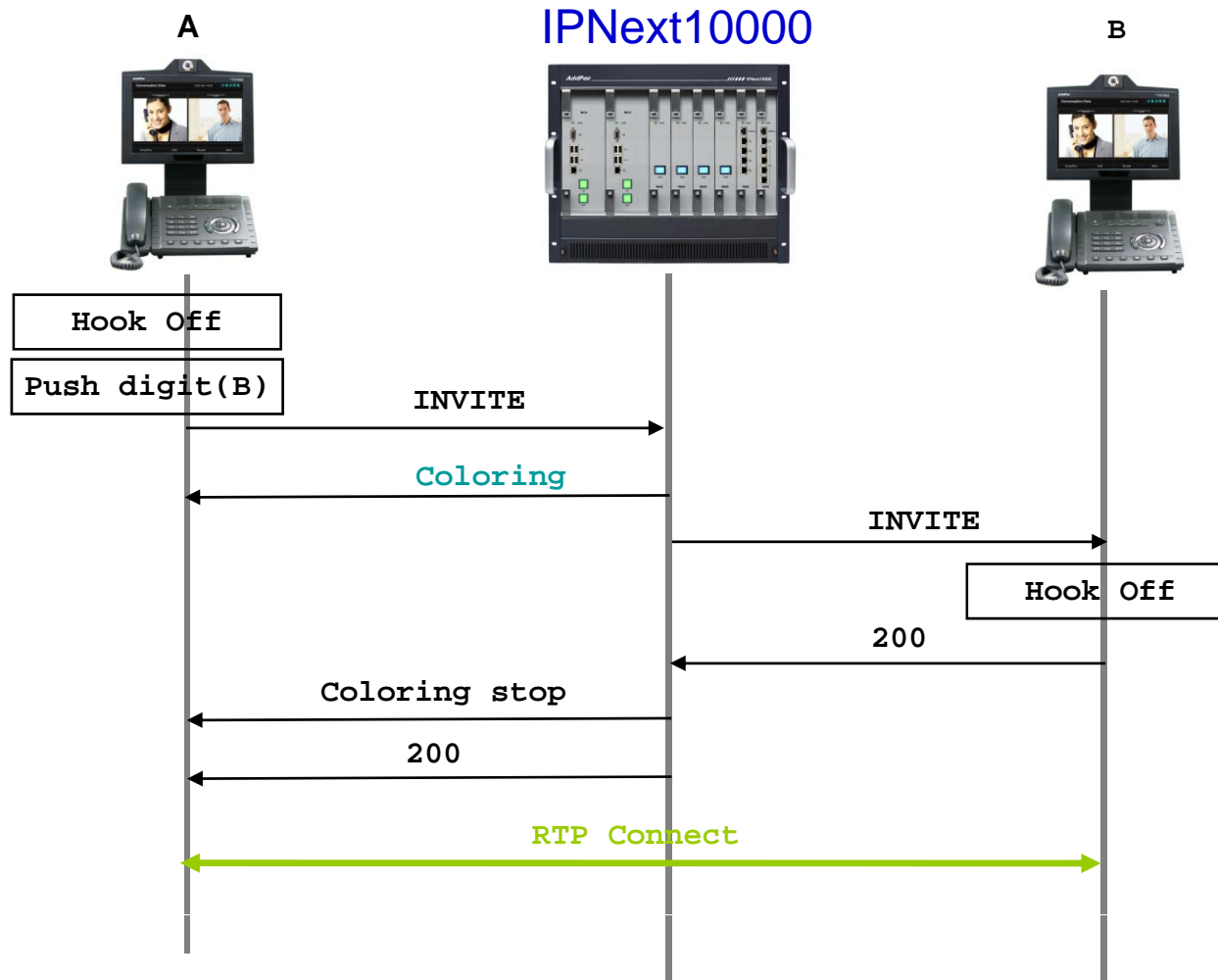
IP Telephony Service and Features

IPNext10000 Large Capacity Call Manager

- Basic Call
- Coloring Service
- Music On Hold
- Blind Transfer
- Call Pickup
- Group Call Pickup
- Consult Call
- Switching Call
- Consult Transfer
- Call Waiting
- Call Park
- Call Pickup Remote
- Hunt Group
- etc

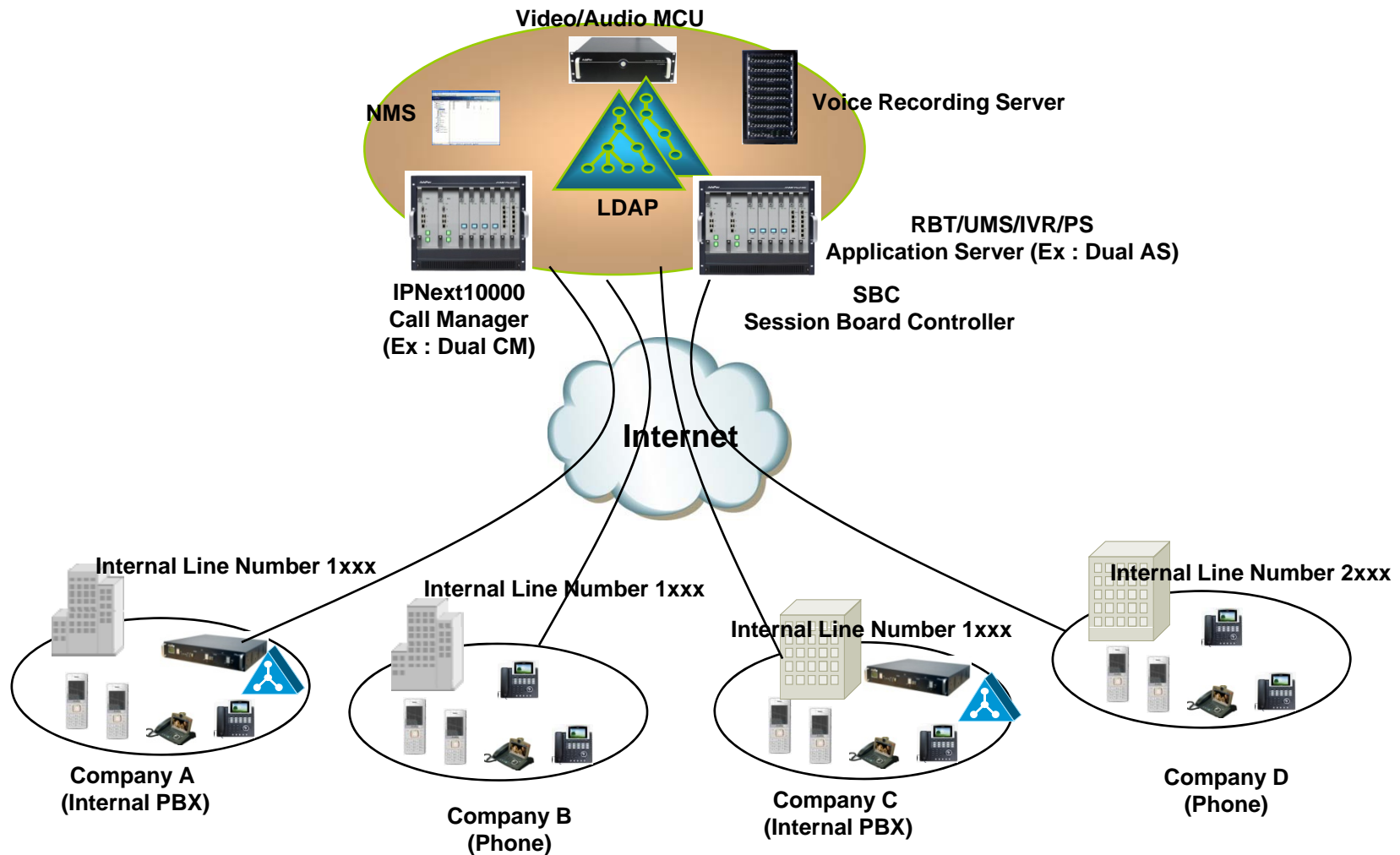
SIP based Basic Call Scenario

IPNext10000 Large Capacity Call Manager



IP Centrex Service

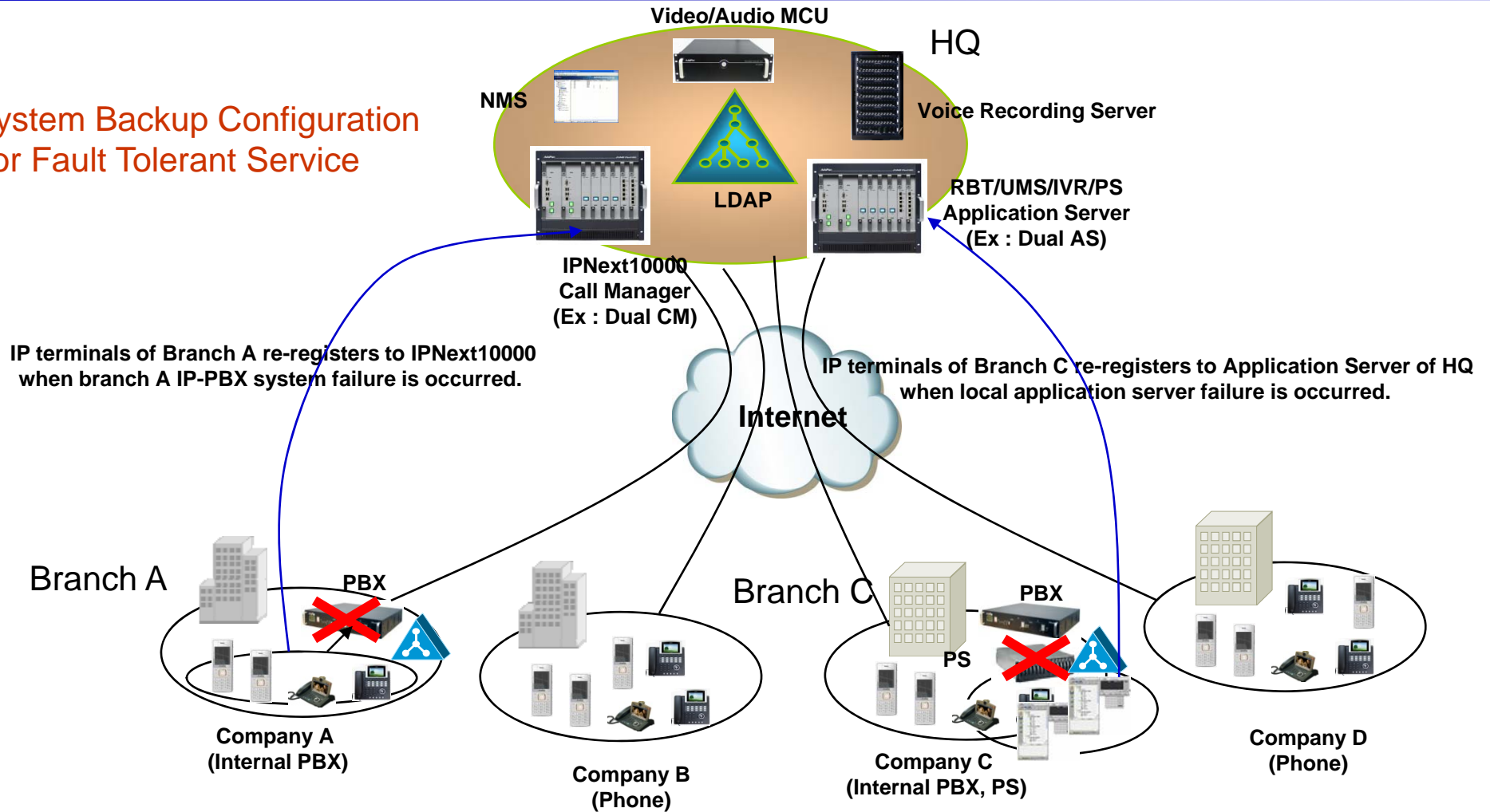
IPNext10000 Large Capacity Call Manager



IP Centrex Service

IPNext10000 Large Capacity Call Manager

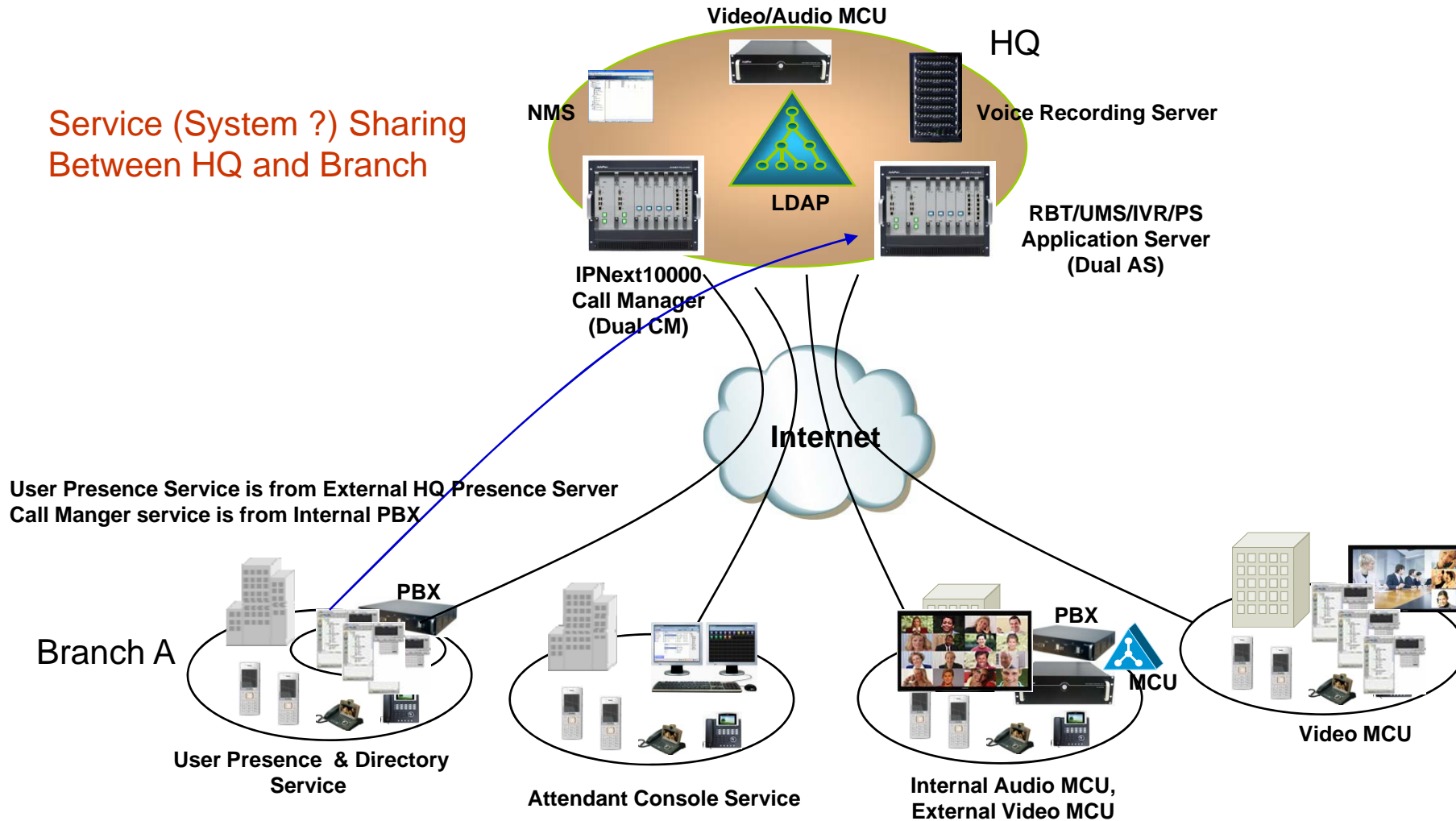
System Backup Configuration for Fault Tolerant Service



IP Centrex Service

IPNext10000 Large Capacity Call Manager

Service (System ?) Sharing
Between HQ and Branch



User Presence Service is from External HQ Presence Server
Call Manger service is from Internal PBX

Network Service and Features

IPNext10000 Large Capacity Call Manager

Basic Network Protocols

- ARP, IP, IPv6, TCP, UDP, ICMP, ICMPv6, SCTP, IGMP, MLD

Routing Protocol

- IPv4 : Static
- IPv6 : Static

Service Protocol

- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog
- IP/IPv6 policy control (QoS)
- VPDN (Virtual Private Dial-up Network : L2TP Server)

Network Service and Features

IPNext10000 Large Capacity Call Manager

IPv4/IPv6 Interworking

- NAT/PAT for IPv4
- IP connect (formerly ip-share) and device cascade for IPv4
- IP/IP, IP/GRE tunneling
- NAT-PT
- 6to4, Autoconfig tunneling

IPv4 Address Configuration

- Fixed (Static)
- DHCP
- PPPoE

IPv6 Address Configuration

- Fixed (Static)
- EUI-64
- Autoconfig (Neighbor Advertisement and Solicitation)

Network Service and Features

IPNext10000 Large Capacity Call Manager

Miscellaneous

- Cisco Style CLI
- Standard & Extended IPv4/IPv6 Access List
- Multi-level User Account Management
- IP accounting
- fsh (Embedded file system shell)
- STUN Client


Network Service and Features

IPNext10000 Large Capacity Call Manager









SNMP MIBs

- MIB-II
- RMON MIBs (Statistics, History, Alarm, Hosts Group)
- RFC2465 Management Information Base for IP Version 6: Textual Conventions and General Group
- RFC2466 Management Information Base for IP Version 6: ICMPv6 Group
- RFC2452 IP Version 6 Management Information Base for the Transmission Control Protocol
- RFC2454 IP Version 6 Management Information Base for the User Datagram Protocol
- AddPac Enterprise MIBs
- etc

IP Phone Comparison Table

	AP-IP300	AP-IP250	AP-IP230	AP-IP160	AP-IP120	AP-IP90
						
LCD Size	4.3 Inch Color LCD	4.3 Inch Color LCD	5 Inch Color LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD
Touch Screen	N/A	Support	Support	N/A	N/A	N/A
Speed-Dial Keys	25 Key with Presence LED	Touch Screen based 25 Keys	Touch Screen based 25 Keys	16 Key with Presence LED	12 Key with Presence LED	N/A
Voice Codec	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
3-Party Conversation	Support	Support	Support	Support	Support	Support
LAN Port	2	2	2	2	2	2
PoE(Optional)	Support	Support	Support	Support	Support	Support
FXO(Optional)	Support	Support	Support	Support	Support	Support




IP Video Phone Comparison Table

	AP-VP500	AP-VP350	AP-VP300N	AP-VP280	AP-VP250	AP-VP230	AP-VP150	AP-VP120
								
LCD Size	12.1 Inch Touch Screen	7Inch Touch Screen	7Inch Touch Screen	7Inch Touch Screen	4.3Inch Touch Screen	5Inch Touch Screen	4.3Inch Touch Screen	4.3Inch
Camera	CCD	CCD	CCD	CMOS	CMOS	CMOS	CCD	CMOS
Video Codec	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Video MCU	N/A	4-Party Video MCU	N/A	N/A	N/A	N/A	N/A	N/A
Voice MCU	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party
LAN Port	2	2	2	2	2	2	2	2
PoE	N/A	N/A	Support	N/A	Support	Support	Support	Support



IP Extend Key Pack for AddPac IP-PBX

IP Extend Key Pack Comparison Table

Model	AP-PT100	AP-PT50	AP-PT20
Service Features			
Key Type	7 inch LCD Touch Screen	Push Button with User Presence Indication LAMP	Push Button with User Presence Indication LAMP
Key Number	Default : 9(row) x 4(column) = 36	60 Key	40 Key
User Presence Indication	Support	LED on, LED off, LED Blink	LED on, LED off, LED Blink
Multiple Cascading	Support	Support	Support
Speaker	Support	Support	Support
LAN Port	2	2	2
PoE(Optional)	Support	Support	Support
Application	IP Phone or Video Phone Extend Key Pack	IP Phone or Video Phone Extend Key Pack	IP Phone or Video Phone Extend Key Pack

IP Wifi Phone

AP-WP100 IP Wifi Phone

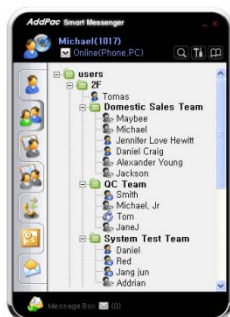
- Wi-Fi IP Phone Solution
- Various Call Scenario Support (IP-PBX)
- State-of-art SIP Signaling
- IEEE802.11b/g up to 54Mbps
- WPA(Wifi Protected Access), 802.11i Security Standard
- Wi-Fi IP Audio Broadcasting Terminal Solution
- External Audio In/Out Port for Headset
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection



Smart Communicator

AP-SMP100 Soft Phone (Video/Voice)

- MS-Window based Soft-Phone
- Smart Multimedia (Video/Voice) Soft-Phone
- IP Real-time Audio/Video Broadcasting Terminal Solution
- Built-In Smart Messenger Function
- Up to 30fps with VGA-Resolution(MPEG-4)
- Video Conference Call Support (AddPac External MCU Inter-working)
- Advanced Voice/Video Traffic QoS
- SIP, H.323* Signaling Support
- Support Various Call Signal via AddPac IP-PBX Inter-working

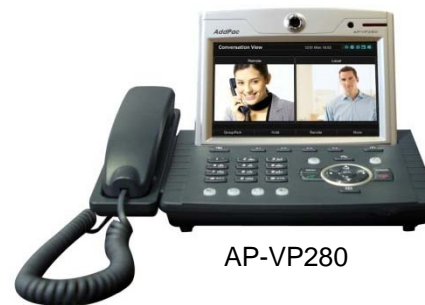


Smart Messenger

- MS-Window based Application
- Support Messenger Service
- Support Various Address Book
- Support User Presence Information
- Support User Search Feature
- Interoperation with Address Book and Smart Phone
- Support Smart Phone Control and Setup
 - Call Control and Forward Setup
- Support Unified Message Box
 - Voice Mail Box
 - Short Message Box



AP-VP300



AP-VP280



Smart Messenger

Smart Window(1/2)

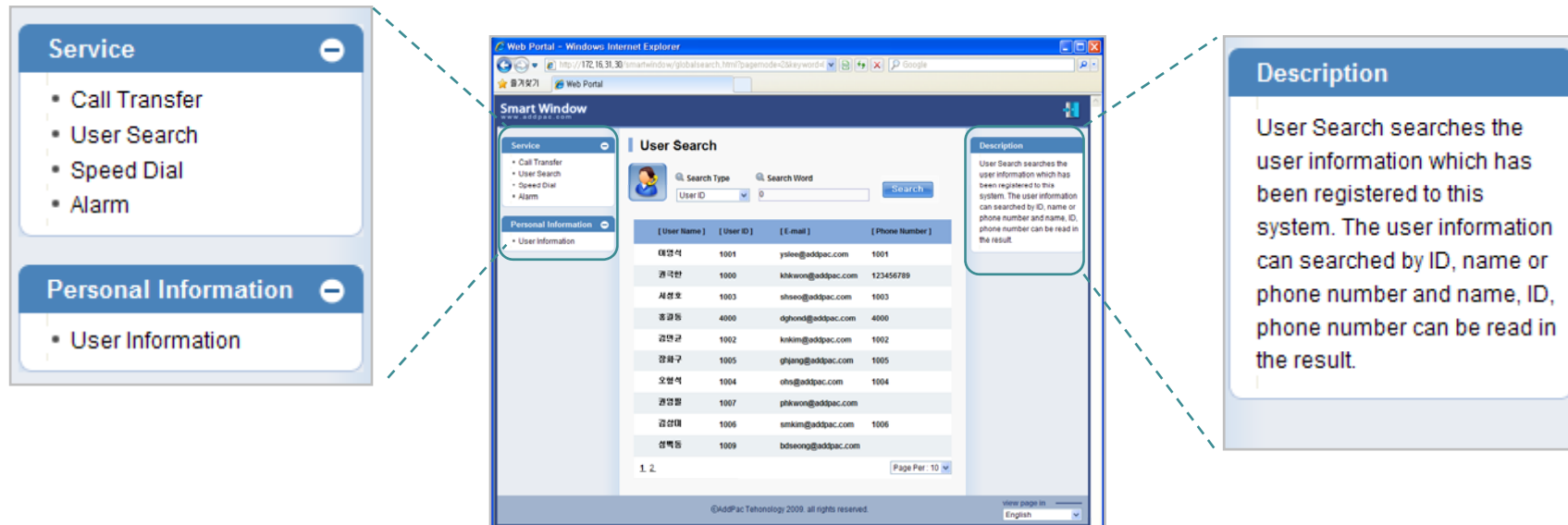
Smart Window is simple web based personal information management program for IP-PBX, Call Manager, etc.

Service Features

- User Information
 - Personal information configuration
- Alarm
 - Alarm event configuration
- User Search
 - User search using search keyword
- Call Forwarding
 - Call forwarding configuration
- Speed Dial
 - Speed dial configuration
- Conference*
 - Configuration of audio, video multiparty conference

Smart Window (2/2)

Main Layout of Smart Window consists of Menu, Contents, Help Message, etc.



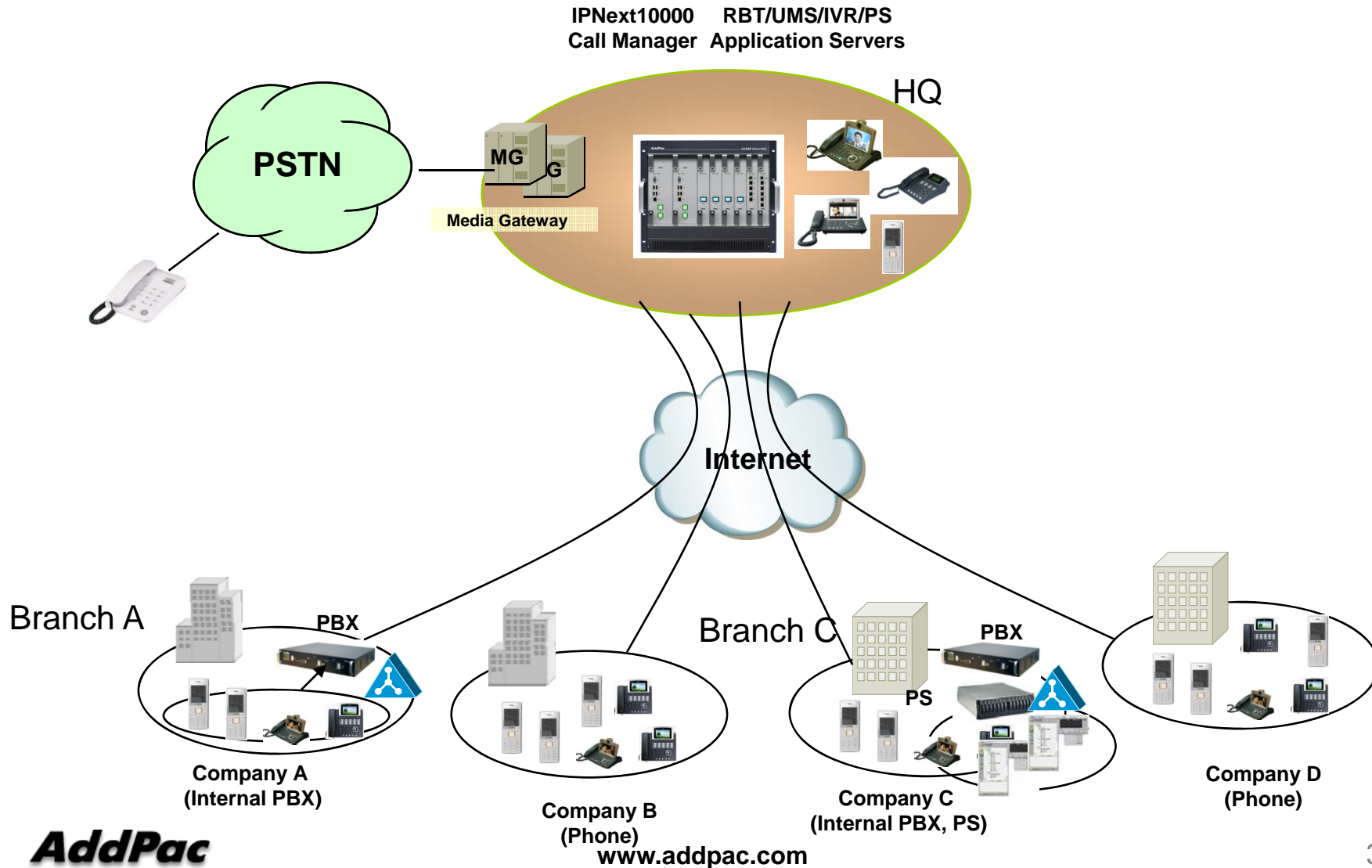
Main Menu

Main Layout

Help Message

Standard IP-PBX Application

IPNext10000 Large Capacity Call Manager



Web Smart Multimedia Manager (WSMM) IP-PBX Series

The screenshot displays the Smart Multimedia Manager (WSMM) web interface. The top navigation bar includes links for Extensions, Trunks, PBX Services, System Admin, and Summary. A 'Getting Started' button is highlighted in green. Below the navigation bar, there are social media links for LinkedIn, Facebook, and YouTube. The main content area features a 'Welcome to AddPac IP-PBX' message for the user 'root', with a last login time of June 08 11:29:56AM (172.16.30.41). An 'Unread Alarm Message' is also visible, indicating a 'login user authentication failed' on 2012-06-01 07:51:12. A 'Quick Menu' section provides shortcuts for adding various system components. The 'Status' section shows a system overview diagram with three main components: User Extensions, System, and Trunks. The System component is further detailed with resource usage statistics: Memory (0%), Storage (7%), Network, Call Manager (0/100), MCU (0/2), Presence (0/100), IVR (0/100), Media (0/100), UMS (0/100), and RtpProxy (0/100). Below these statistics are icons for FXS (1), E1 (0), FXO (1), and GSM (2) ports. The Trunks section shows a list of trunks: Internal Trunk Gateway (0/0), SKN_TG (0/0), and Uacom_IRunk (0/0).

Contents

- Overview
- System Requirement
- WSMM Login
- Extension Management
- Trunks Management
- PBX Services Management
- System Admin Management
- Summary
- User Portal web page



Overview

What`s New in WSMM (Web based Smart Multimedia Manager)

- Simple Menu and Easy Configuration
- Provides Built-In IVR Scenario Editor and Service Configuration
- Provides easy-to-user IP-PBX System API Services and ways to integration with 3rd party systems
- Integrated voice line management such as **FXS, FXO, E1, GSM, 3G**
- **User portal** to configure personal information, call forwarding
- **Diagnostic tool** to analyze SIP Call flow, current status and problems for terminal and trunk

System Requirement

WSMM (Web based Smart Multimedia Manager)

- Windows XP, Vista, Windows 7, Windows Server 2000/2003
- Linux / Unix Platform
- Microsoft Internet Explorer 7.0 / 8.0 / 9.0
- Google Chrome / Mozilla Firefox / Safari / Opera
- Javascript + HTML supported browser (Android, iPhone, iPad,...)

Login

Administration Tool Suite: x
172.16.1.19/main

AddPac
IPNext PBX System

Administrator ID: root
Password: *****

Smart Multimedia Manager
Door Access Control Manager
Time and Attendance Manager

WSMM Login
Execute web browser to enter the IP address of IP-PBX then WSMM accessible login screen will be displayed.

Administrator Authentication
Enter administrator ID and password to complete authentication by clicking login Smart Multimedia Manager.

Help

Smart Multimedia Manager
www.addpac.com

Add an User Extension

Extension * [] 3~8
First Name []
Last Name * []
Voice Mail Password * []
User Password * []
Department []
Title []
Email []
Home Phone []
Mobile Phone []
User ID []
Photo [Select Photo]
Routing Access List [internal]

Advanced Options

Terminal Profile [default]
Security Profile [default]
Use RTP Proxy []
Generate Ring Back Tone at PBX []
Presentation [Default]

Help :: User Extension

- Analog Extension**
- Analog Port**
You should select one of analog FXS port in this PBX. An analog phone or legacy PBX line can be attached to this analog extension. The analog port already assigned to other analog extension will not be shown at the list.
- User Extension**
- Extension**
This is a phone number of this user. For convenience, it is recommended to assign same digits length to user extensions. This user extension is also a user id for login user portal and default user id for SIP registration for registering SIP phone unless setting User ID option.
- First Name / Last Name**
This is user's first name and last name like Michel Jackson.

Copyright © AddPac 1999-2012 All Rights Reserved

HELP
WSMM provides HELP for each functions. Click HELP to display new screen and detail description of setup is clearly explained in homepage.


Related Links

Smart Multimedia Manager
www.addpac.com

Modify the User Extension

Apply Cancel Advanced Options

User Extension

Extension *	1009	3 - 8 digits
First Name	ByoungGoo	
Last Name *	Choi	
Voice Mail Password *	****	4digits and user portal login
User Password *	1111	For SIP registration
Department	root	Search
Title		ex) manager
Email		ex) admin@addpac.com
Home Phone		ex) 123-456-7890
Mobile Phone		ex) 123-456-7890
User ID		SIP registration ID
Photo	 (Maximum File Size: 100KB) Select Photo	
Routing Access List	internal	

Related Links

- WSMM User Portal
- Routing Access Lists
- Terminal Profiles
- Security Profiles
- Pickup Group

Description

A user extension is an IP Phone (SIP / SSCP phone) or a soft phone for end user. It is composed of user profile, phone number and terminal belongs to the user.

Advanced Options

Terminal Profile	default
Security Profile	default
Use RTP Proxy	<input type="checkbox"/>
Back Tone at	<input checked="" type="checkbox"/>
Representation	Default

Copyright © AddPac 1999-2012. All Rights Reserved

Related Links
WSMM setup page provides related link functions. Related links helps easy operation of IP-PBX by providing link.

Diagnostic

Smart Multimedia Manager
www.addpac.com

Terminal Diagnostic 1009 (172.16.18.100)

You can check network connectivity from the PBX to the terminal by **Network Connectivity Test** and also you can check SIP awareness of the terminal by checking response message from the terminal by **SIP Aware Test**.

Step 1.

- 1. **Network Connectivity Test** Successfully pinged 172.16.18.100 which is just provisioned to phone. Reply from 172.16.18.100: time=100ms loss=0% **Succeeded**
- 2. **SIP Aware Test** This phone '172.16.18.100' is successfully responding SIP OPTIONS. **Succeeded**

At this step, you can make a test call on the diagnostic terminal to some destination number. If this terminal has problem on local call, the destination could be a local extension otherwise the destination could be mobile or PSTN number. The call trace shows information whether the call is properly handled or not. This test call can be traced only one administrator at same time and simultaneous test call will not be allowed.

1005 Start Outbound Test

Outbound Call Test Make a test call '1005' Succeeded. **Succeeded**

2012-06-12 20:15:36 deviceId: 70 caller: 1009 callee: 1005 Call Test Start.
----- From 1009 (172.16.18.100:5060) -----

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.17.30:5060;branch=z9hG4bKd84f0b0fa411
From: <sip:dial-service@172.16.17.30>;tag=d84f0b0fa4
To: <sip:1009@172.16.18.100>;tag=dc4fa2c5a4
Call-ID: dca3d74f-519d-a2e8-80c5-0002a4038e2c@172.16.18.100
CSeq: 11 INVITE
Session-Expires: 1800;refresher=uac
User-Agent: AddPac SIP Gateway
Contact: sip:1009@172.16.18.100
Require: timer
Content-Type: application/sdp
Content-Length: 179

v=0
o=1009 1339532254 1339532254 IN IP4 172.16.18.100
s=AddPac Gateway SDP
c=IN IP4 172.16.18.100
t=1339532254 0

/AVP 0
0000/1
(172.16.18.100:5060) -----

172.16.17.30:5060;branch=z9hG4bKd84f0b0fa411
service@172.16.17.30>;tag=d84f0b0fa4
2.16.18.100>;tag=dc4fa2c5a4
f-519d-a2e8-80c5-0002a4038e2c@172.16.18.100

1800;refresher=uac
ic SIP Gateway
9@172.16.18.100

lication/sdp
179

Step 2.

Copyright © AddPac 1999-2012 All Rights Reserved

Diagnostic
It provides to display terminal and trunk status inspection in IP-PBX

Step 1.

- Network Connection Test
- SIP Aware Test

Step 2.

- Outgoing Call Test

Built-in IVR Scenario Editor

Smart Multimedia Manager
www.addpac.com

IVR Scenarios

Apply Save Cancel

IVR Scenario Properties

Name: addpac
Description:

IVR Scenario sequence

- Start
 - Play
 - Menu (AddPac)
 - Multi
 - Check Extension
 - TRUE
 - Play
 - Transfer
 - FALSE
 - Play (Wrong Number)
 - Goto
 - 0
 - Play (Connect)
 - Transfer
 - 1
 - Play (Connect)
 - Transfer (Voice Mail)
 - 2
 - Play (Announcement)
 - Transfer
 - No Match
 - Play (Thank you)
 - Disconnect
 - No Input
 - Play (Please Press Number)
 - Goto

Menu

This action inputs a single digit or multiple digits from user phone and branches to an event handle by matching input digit.

Name * AddPac
File Path hello_full .Open
 Cancelable
If this option is enabled, you can stop the sound by pressing any key.
Initial Timeout 10
Allowable Count 5
Single Digit
Add Single Digit Event of 3
Multi Digit
Add Multi Digit Event with Inter Digit Timeout 1 Sec
and Max Digit Length 4

Description

Using this built in IVR scenario editor, you can create a new IVR scenario or modify it. The created scenario is generated to voice XML file and loaded to interpreter when you apply this scenario. This IVR scenario can be tested by call to IVR extension where this scenario is applied.

Copyright © AddPac 1999-2012. All Rights Reserved

Built-in IVR Scenario Editor
WSMM is embedded with IVR Scenario Editor.
An administrator may create/edit IVR scenario without using special tool
IVR Scenario Sequence

- Start
- Menu / Play / Transfer / Check Extension / Goto / Disconnect

Main

Smart Multimedia Manager
www.addpac.com

Extensons
Trunks
PBX Services
System Admin
Summary

Welcome to AddPac IP-PBX
root
Last Login at June 08 11:29:56AM (172.16.30.41)

Unread Alarm Message
login user authentication failed 2012-06-01 07:51:12

Quick Menu
 > Add an User Extension
 > Add a VoIP Trunk
 > Extensions
 > Add an Analog Extension
 > Add an Outgoing Call Rule
 > Terminals
 > Add a Conference Room
 > Add an Incoming Call Rule

Status

User Extensions
 Registered (2)
 Unregistered (3)
 Unconfigured (0)
 Unused license (95)

System
 Memory Storage: 0%
 Network: 7%
 Call Manager: 0/100
 MCU: 0/2
 Presence: 0/100
 IVR: 0/100
 Media: 0/100
 UMS: 0/100
 RtpProxy: 0/100

Trunks
 Internal Trunk Gateway (0/0)
 SKN_TG (0/0)
 Dacom_Trunk (0/0)

FXS (1) 0 1 2 3
 E1 (0) 0 1
 FXO (1) 4 5 6 7
 GSM (2) 0 1 2 3

Main Menu
Through left "Main Menu", setup IP-PBX policy.

Alarm Message
It displays IP-PBX system errors

Short Cut
A short cut link.

Status
It displays current IP-PBX system major status

Main - Alarm History

Smart Multimedia Manager
www.addpac.com

Start

Welcome to AddPac IP-PBX

root
Last Login at June 08 11:29:56AM (172.16.30.41)

Unread Alarm Message 1
login user authentication 2012-06-01 07:51:12

Quick Menu

- Add a User Extension
- Add a VoIP Trunk
- Extensions
- Add an Analog Extension
- Add an Outgoing Call Rule
- Terminals
- Add a Conference Room
- Add an Incoming Call Rule

Alarm History

Level: All Ack: All Period: 2012-06-01 ~ 2012-06-08 Search Refresh

Level	Messages	DateTime
1 Minor	NTP time sync service started!	2012-06-01 07:54:35
2 Major	ftp service disabled by operator	2012-06-01 07:54:02
3 Minor	network interface fastethernet 0/0 now up	2012-06-01 07:53:35
4 Major	disk upper quota limit exceeded	2012-06-01 07:53:04
	threshold exceeded!	2012-06-01 07:51:55
	authentication failed	2012-06-01 07:51:12

Displaying 1 - 6 of 6

Alarm History
Main page displays alarm message. Click Unread Alarm Message to display alarm history page at the bottom. It also displays IP-PBX system errors.

Main – Quick Menu

The screenshot displays the 'Smart Multimedia Manager' web interface. On the left is a navigation sidebar with options like 'Extensions', 'Trunks', 'PBX Services', 'System Admin', and 'Summary'. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root' and an 'Unread Alarm Message' about a failed login. Below this is a 'Quick Menu' section with several options: 'Add an User Extension', 'Add a VoIP Trunk', 'Extensions', 'Add an Analog Extension', 'Add an Outgoing Call Rule', 'Terminals', 'Add a Conference Room', and 'Add an Incoming Call Rule'. A red dashed box highlights the 'Quick Menu' and the 'Add an User Extension' form below it. A blue arrow points from the 'Add an User Extension' link in the Quick Menu to the form. The form includes fields for Extension, First Name, Last Name, Voice Mail Password, User Password, Department, Title, Email, and Home Phone. A 'Description' box explains that a user extension is an IP Phone or soft phone. A 'Related Links' section lists 'Routing Access Lists', 'Terminal Profiles', 'Security Profiles', and 'Pickup Group'. A yellow callout box at the bottom left explains the Quick Menu's purpose.

Quick Menu
A short cut link for favorite. It provides Extension / Conference Room / Trunk / Call Rule / Terminals short cut link to improve the convenience of user.

Main – Follow Us

The screenshot shows the 'Smart Multimedia Manager' interface. On the left sidebar, there is a 'Follow Us' section with icons for LinkedIn, Facebook, and YouTube. Blue arrows point from these icons to corresponding social media pages overlaid on the main interface. The LinkedIn page shows a post about AddPac Technology Hybrid IP-PBX System. The Facebook page shows the AddPac profile. The YouTube page shows a video titled 'AddPac Hybrid IP-PBX IPNext180 / IPNext187 / IPNext190 (16/24/32 Port)'. A yellow box at the bottom left contains the text: 'Follow Us You may check AddPac product information, solution and etc. through Linked, Facebook, YouTube.'

Main – Status Monitoring

Smart Multimedia Manager
www.addpac.com

Start

Welcome to AddPac IP-PBX

Unread Alarm Message 1
login user authentication failed 2012-06-01 07:51:12

root
Last Login at June 08 11:29:56AM (172.16.30.41)

Quick Menu

- > Add an User Extension
- > Add an Analog Extension
- > Add a Conference Room
- > Add a VoIP Trunk
- > Add an Outgoing Call Rule
- > Add an Incoming Call Rule
- > Extensions
- > Terminals

Status

User Extensions

System

Memory Storage 0%
Network 7%

Call Manager
MCU
Presence
IVR
Media
UMS
RtpProxy

FXS (1) E1 (0)
FXO (1)
GSM (2)

Trunks

Voice Lines

Slot / Port	Type	Status	Pattern	Connection PLAR	Input Gain	Output Gain	Caller ID	Modify
1 1/4	FXO	Idle			0	0	Disabled	
2 1/5	FXO	Idle			0	0	Disabled	
3 1/6	FXO	Idle			0	0	Disabled	
4 1/7	FXO	Idle			0	0	Disabled	
5 2/0	GSM	unreg...			0	0	Disabled	
6 2/1	GSM	unreg...			0	0	Disabled	
7 2/2	GSM	unreg...			0	0	Disabled	
8 2/3	GSM	unreg...			0	0	Disabled	

Slot / Port	Type	Status	Pattern	Connection PLAR	Input Gain	Output Gain	Check Source	Protocol Emul	Modify
1 0/0/0	E1	down			0	0	Master	Network	
2 0/1/0	E1	down			0	0	Master	Network	

Slot / Port	Type	Status	Number	User ID	Password	Input Gain	Output Gain	Caller ID	Modify
1 1/0	FXS	Idle				0	0	Disabled	
2 1/1	FXS	Idle				0	0	Disabled	
3 1/2	FXS	Idle				0	0	Disabled	
4 1/3	FXS	Idle				0	0	Disabled	

Registered (2)
Unregistered (3)
Unconfigured (0)
Unused license (95)

Copyright © AddPac 1999-2012 All Rights Reserved

Status
You may check current IP-PBX major information. It supports Terminal, Trunk Register Status, System Status (Memory, Storage, Network, Service), Voice Module Status (FXS, FXO, E1, GSM) Check and main menu short cut function.

Extension - Extensions

The screenshot displays the 'Smart Multimedia Manager' web interface. The left sidebar contains a navigation menu with 'Extensions' highlighted. The main content area shows a 'Welcome to AddPac IP-PBX' message and an 'Unread Alarm Message' notification. Below this, there is a 'Quick Menu' and a 'Status' section with a 'User Extension' overview. The 'Extensions' table lists various extension types and their details. A detailed view of a 'User Extension' is shown below the table, including an 'Add an Extension' form and descriptive text for different extension types.

	Modify	Delete	Extension Number	Type	Name	Date Created
1			1007	User Extension	Jinsuk Choi	2012-06-08 17:54:53
2			1009	User Extension	ByoungGoo Choi	2012-06-08 17:58:05
3			3000	User Extension	BongYoung Jeong	2012-06-08 17:59:14
4			1008	User Extension	SeongHyun Lee	2012-06-08 18:59:48
5			1010	Analog Extension	JongHwee Kwon	2012-06-08 18:36:34
6				Conference Room	Ad-Hoc Defaults	1999-11-30 08:00:00
7			0001	Voice Mail	vmal_rec	2012-06-08 17:49:53
8			0002	Voice Mail	vmal	2012-06-08 17:49:54
9			0003	Voice Mail	vmal_noauth	2012-06-08 17:49:54

Registered (2)
Unregistered (3)
Unconfigured (0)
Unused license (95)

Add an Extension

Analog Extension
 An analog extension is a kind of user extension who has FXS (Foreign eXchange Station) analog voice line. Normal analog phone is connected at this extension.

Hunt Group
 A hunt group has members of user extensions. Within a hunt group, an available member (user extension) can receive a call to the hunt group extension. A hunt group has one of simultaneous, sequential or random call hunting mode.

Pickup Group
 A pickup group has members of user extensions who can pick up a ringing call within the group. The pickup group extension number is used for picking up a call by other group member.

Conference Room
 A conference room extension is used for making a conference room. In case of dial-out conference, when a privileged user calls to conference room extension, all conference participants receive call to join. In case of meet-me conference, conference participants call to conference extension to join.

IVR Extension
 An IVR (Interactive Voice Response) extension has a role of auto attendant for incoming calls from trunks. If incoming calls from trunk are routed to an IVR extension by incoming call rule, the interactive scenario will be proceed to transfer the call to a proper user extension.

Extension
 Extension setup is possible to operate IP-PBX operation. User Extension / Analog Extension / Hunt Group / Pickup Group / Conference Room / IVR Extension

Extension - Directory

The screenshot displays the 'Smart Multimedia Manager' interface. The left sidebar contains a navigation menu with 'Extensions' selected. The main content area shows the 'Directory' page with a list of users and their extensions. A yellow callout box highlights the 'Directory' section of the sidebar.

Directory
It displays user organization department. Each user may setup department in User Extension. Use directory to use click to call function in user portal web page.

Name	Extension	Notes
BongY Jeong	1101	
SeongHyun Lee	1008	
연구소		
Hardware		
DongHee Jang	1020	
Smart work		
Smart Management		
BongYong Jeong	3000	Hello ~ I am Jeong BongYong
Smart Framework		
BY Jeong	1100	
SangGyun Lee	1005	
HyungSuk Oh	1006	Have a nice day ~
ByoungGoo Choi	1009	

System Status Dashboard:

- Registered: 4
- Unregistered: 4
- Unconfigured: 0
- Unused license: 92
- Memory Storage: 1% (8%)
- Network: [Icon]
- Call Manager: 0/100
- MCU: 0/2
- Presence: 0/100
- IVR: 0/100
- Media: 0/100
- UMS: 0/100
- RtpProxy: 0/100
- Trunks:
 - Internal Trunk Gateway: (0/0)
 - SKN_TG: (0/0)
 - Dacom_Trunk: (0/0)

Extension - Routing Access List

The screenshot shows the Smart Multimedia Manager interface. The left sidebar contains a menu with 'Extensions' highlighted. The main content area displays the 'Routing Access Lists' configuration page. A table lists existing routing access lists, and a form below allows adding a new one. A yellow box at the bottom left contains the following text:

Routing Access List
Apply call rules regarding outgoing call routing for external bound trunk in IP-PBX.

Modify	Delete	Name	Description	Date Created
		internal	internal access control	2012-06-08 17:49:54

Add a Routing Access List

Add Cancel

Routing Access List

Name *

Description

Select Outgoing Call Rules to allow routing. You can adjust routing priority by drag and drop a rule among Allowed Outgoing Call Rules.

Outgoing Call Rules	
Name	

Allowed Outgoing Call Rules	
Name	

Description

You can permit outgoing call routings to specific trunk by adding Outgoing Call Rules.

Related Links

- Outgoing Call Rules

Extension - Terminal Profile

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with the following items: Extensions, Directory, Routing Access Lists, Terminal Profiles, Terminals, Trunks, PBX Services, System Admin, Summary, Getting Started, Follow Us, and social media links for LinkedIn, Facebook, and YouTube. A red dashed box highlights the 'Terminal Profiles' menu item, with a yellow starburst icon and a blue arrow pointing to the 'Terminal Profiles' tab in the main content area.

The main content area shows the 'Terminal Profiles' configuration page. It includes a 'Global Terminal Settings' section with various options:

- Calling Party Presentation: allowed, Restricted
- Language: Korean
- Call Duration Limit: Call Duration Limit, 24 (1~48 Hour)
- Off-net Transfer: Off-net Transfer
- Initial Digit Timeout: 15000 (1000~100000ms)
- First Inter Digit Timeout: 3000 (1000~10000ms)
- Second Inter Digit Timeout: 3000 (1000~10000ms)
- Number of Digit(First Inter Digit Timeout): 4 (1~100)
- Internal Call: default
- External Call: default
- Internal Forwarded Call: default
- External Forwarded Call: default
- Keaplive Timeout: 30 (10~86400sec)

A 'Description' box on the right states: "Below settings are applied whole terminals in this system including trunks. Some VoIP settings can be customized to terminals by Terminal Profile."

At the bottom of the interface, a copyright notice reads: "Copyright © AddPac 1998-2012 All Rights Reserved."

Terminal Profile
Setup SIP/SSCP/Timeout/Ring/VoIP setting in IP-PBX. It supports global setting and terminal profile.

Extension - Terminals

Smart Multimedia Manager

Welcome to AddPac IP-PBX

Unread Alarm Message

No Unread Alarm Message

root
Last Login at June 11 04:38:52AM (172.16.1.50)

Terminals

	Modify	Delete	Diagnose	Extension	Name	User Agent	IP Address	State	MAC Address	Create Time
1				1007	Jinsuk Choi			Unregistered		2012-06-08 17:54:53
2				1008	SeongHyun Lee			Unregistered		2012-06-08 18:59:49
3				1010	JongHwee Kwon	AddPac SIP ...	172.16.17.30	Unregistered		2012-06-08 18:36:35
4				1009	ByoungGoo Choi	AddPac AP-V...	172.16.18.100	Registered	0002.a403.8...	2012-06-08 17:58:06
5				3000	BongYoung Jeong	AddPac SIP ...	172.16.18.101	Registered		2012-06-08 17:59:15

Modify the User Extension

Apply Cancel Advanced Options

Extension * 3000 3 ~ 8 digits

First Name BongYoung

Last Name * Jeong

Voice Mail Password * **** 4digits and user portal login

User Password * 1111 For SIP registration

Department Search

Title ex) manager

Email ex) admin@addpac.com

Home Phone ex) 123-456-7890

Mobile Phone ex) 123-456-7890

User ID SIP registration ID

Photo (Maximum File Size: 100KB)

Description

A user extension is an IP Phone (SIP / SSCP phone) or a soft phone for end user. It is composed of user profile, phone number and terminal belongs to the user.

Related Links

- WSMM User Portal
- Routing Access Lists
- Terminal Profiles
- Security Profiles
- Pickup Group

Terminals
 You may search/setup/change the status of SIP, SSCP, External Terminal status in IP-PBX. Extension, Name, User Agent, IP Address, Register Status, Mac Address, Terminal Create Time

Trunk - Trunks

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains navigation menus for Extensions, Trunks, PBX Services, System Admin, and Summary. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root' and an 'Unread Alarm Message' section. Below this is a 'Quick Menu' and a 'User Extension' section with a status bar showing 2 Registered, 3 Unregistered, 0 Unconfigured, and 95 Unused licenses. The 'Trunks' section is highlighted with a red dashed border and contains a table of existing trunks and an 'Add a Trunk' form.

Modify	Delete	Diagnose	Name	Type	IP Address	State	Description	Date Created
1			Internal Trunk Gateway	VoIP Trunk	127.0.0.1	Registered		2012-06-08 17:...
2			Dacom_Trunk	SIP Proxy Server	172.16.19.201	Unregistered		2012-06-08 18:...
3			SKN_TG	VoIP Trunk	172.16.19.200	Registered		2012-06-08 18:...

Add a Trunk

- VoIP Trunk**
This is a generic VoIP Trunk which can register to this PBX or communicate without registration. The VoIP Trunk could be VoIP gateway which has analog FXS, FXO, E&M line, digital E1, T1 line or mobile GSM line, or IP-PBX or other SIP / H.323 Trunk.
- SIP Proxy Server**
This could be VoIP service provider who operates SIP Proxy Server and provides VoIP service to public telephone network or mobile network or other VoIP network. Also, this could be an IP-PBX who provides SIP server features. This PBX should register to the SIP Proxy Server for receiving incoming calls and sending outgoing calls.
- H.323 Gatekeeper**
This could be VoIP service provider who operates H.323 Gatekeeper and provides VoIP service to public telephone network or mobile network or other VoIP network. Also, this could be an IP-PBX who provides H.323 Gatekeeper features. This PBX should register to the H.323 Gatekeeper for receiving incoming calls and sending outgoing calls.

Description
Using the trunks, user extensions in this PBX can communicate with remote users in public telephone network or mobile network or other VoIP network including branches.

Trunk
A trunk setup for IP-PBX in order to make a call. You may setup VoIP Trunk, SIP Proxy Server, and H.323 Gatekeeper as well as to check the register status in accordance with Trunk types.

Trunk - Outgoing Call Rules

Smart Multimedia Manager
www.addpac.com

Welcome to AddPac IP-PBX

Unread Alarm Message
No Unread Alarm Message

root
Last Login at June 11 04:38:52AM (172.16.1.50)

Outgoing Call Rules

Modify	Delete	Name	Pattern	Trunk	Date Created
		external rule	8T		2012-04-04 09:39:48

Add an Outgoing Call Rule

Name *

Patterns *

Trunks of Outgoing Call *

Called Number Translation

Number Translation

Calling Number Translation

Description
An Outgoing Call Rule controls outgoing call routing to a specific trunk. An outgoing call from user extension can be routed to trunk by selecting an Outgoing Call Rule which has matched pattern with dialed digits of the call. Also, an incoming call from a trunk can be applied to Outgoing Call Rules by an Incoming Call Rule for routing to other trunk.

Related Links
• Trunks

Outgoing Call Rules
A call rule for external call routing. You may apply various options such as Outgoing call rule (Number Translation, Routing Mode, Display Name Presentation, P-Asserted Identity Presentation) for outgoing call rule.

Trunk - Incoming Call Rules

Smart Multimedia Manager
www.addpac.com

Start

Welcome to AddPac IP-PBX

Unread Alarm Message
No Unread Alarm Message

root
Last Login at June 11 04:38:52AM (172.16.1.50)

Quick Menu

Status

User Extension

Getting Started GO

Follow Us

Linked in

facebook f

YouTube

Registered (2)
Unregistered (3)
Unconfigured (0)
Unused license (95)

Smart Multimedia Manager
www.addpac.com

Status Incoming Call Rules

Add an Incoming Call Rule

Add Cancel

Name *

Trunks of Incoming Call *

- Internal Trunk Gateway
- SM_SIP_Provider
- ss
- jschoLgk

The incoming call can be routed to an IVR extension or a single user extension by pattern matching to called party number and calling party number of the call.

Route to an extension by called number + Add Rule

Transfer Rule Modify Delete

Single Extension Routing

Route to an extension by calling number + Add Rule

Transfer Rule Modify Delete

If the called party number of the incoming call contains user extension number, it can be routed to the destination extension using DID (Direct Inward Dialing) rule.

Route to multiple extension by called number(DID) + Add Rule

Route DID Rule Modify Delete

Multiple Extension Routing(DID)

The incoming call from a trunk can be routed to other trunks by applying Outgoing Call Rules.

Called Pattern to delete digits from the front and adding

Trunk Routing to outgoing call rules external rule

Description

The Incoming Call Rule controls incoming call routing from specific trunks by looking up calling party number and called party number of the call. Applying this rule, the incoming calls are routed to IVR extension, user extensions, or other trunks. Using malicious call filter, call might be dropped.

Related Links

- Trunks
- Outgoing Call Rules

Copyright © AddPac 1999-2012. All Rights Reserved.

Incoming Call Rules
A call rule for incoming call through trunk . You may apply various options such as (Number Translation, DID)

PBX Service - Speed Button Profiles

The screenshot displays the Smart Multimedia Manager interface. On the left, a navigation menu highlights 'PBX Services' and 'Speed Button Profiles'. The main area shows the 'Speed Button Profile' configuration page, which includes a table of existing profiles and a form to add a new one. The table contains one entry: 'button profile' created on 2012-04-02 10:43:18. The configuration form has fields for 'Profile Name*', 'Description', and a table for 'Add a Speed Button' with columns for Name, Extension, Type, and a dropdown menu. A 'Description' sidebar on the right explains the function of speed buttons.

Speed Button Profile
 A function for IP/VP-Phone. A newly created speed button list may check in phone. Use idle/Ring/Connect status and touch to call function for each extension.

PBX Service - Announcement and Tones

Smart Multimedia Manager
www.addpac.com

Welcome to AddPac IP-PBX
root
Last Login at June 11 01:51:12PM (127.0.0.1)

Announcement and Tones
Language: Korean | Restore Default Ments | Global Media Setting | Refresh

Modify	ID	Announcement	Description	Custom File	Scheduled
	400110	Connect	연결 중 안내		
	400120	Retry	내선 번호 재 시도 안내		
	400130	No Number	없는 내선 안내		
	400140	Over Count	회수 초과 안내		
	410110	Greeting	인사말		
	410120	Connected to attendant	안내원과 연결		
	410130	Connect 2	연결 중 안내		
	410140	No Number 2	번호 입력 오류		
	410150	Over Time	입력 내용 오류		
	420110	Busy	통화 중 안내		
	420120	No Answer	부재 중 안내		
	420130	System Normal Fail	통화 실패 안내		
	420400	Thank you	미용 감사 안내		
	430110	Press Password	비밀번호 입력 안내		
	430120	Over Count 2	회수 초과 안내		
	430130	Connect 3	연결 중 안내		
	430140	Over Time 2	번호 입력 오류		

Announcement and Tones
Apply | Cancel

Announcement ID: 400110
Description: 연결 중 안내
Language: Korea

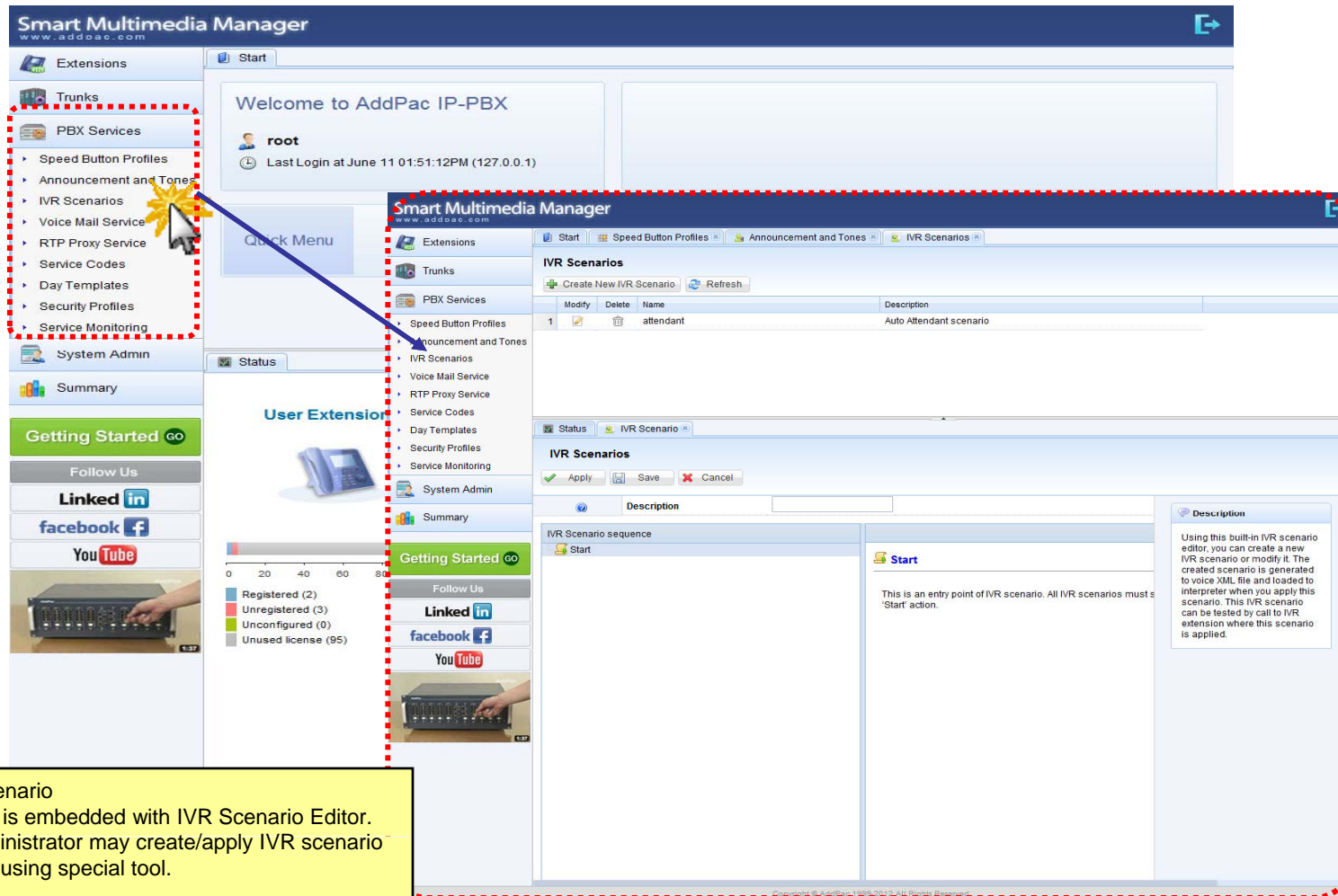
No.	Name	Start date	End Date	Start Time	End T
Create New Schedule					

Description
You can upload new announcements from your PC and each announcement can be assigned to time schedule.

Copyright © AddPac 1999-2012. All Rights Reserved

Announcement and Tones
A setup to manage an announcement (Dial-tone, Consult-tone, Waiting-tone) in IP-PBX service. Announcement may select either Korean/English and administrator may upload Ment File directly.

PBX Service - IVR Scenarios



The screenshot displays the Smart Multimedia Manager (WSMM) interface for managing IVR scenarios. The main window shows a table of IVR scenarios with one entry: 'attendant' with the description 'Auto Attendant scenario'. Below the table, there are buttons for 'Apply', 'Save', and 'Cancel', and a 'Description' field. The interface also includes a sidebar with navigation options like 'Extensions', 'Trunks', and 'PBX Services', and a 'User Extension' status section.

Modify	Delete	Name	Description
		attendant	Auto Attendant scenario

IVR Scenarios

Apply Save Cancel

Description

IVR Scenario sequence

Start

Start

This is an entry point of IVR scenario. All IVR scenarios must start with 'Start' action.

Description

Using this built-in IVR scenario editor, you can create a new IVR scenario or modify it. The created scenario is generated to voice XML file and loaded to interpreter when you apply this scenario. This IVR scenario can be tested by call to IVR extension where this scenario is applied.

Registered (2)
Unregistered (3)
Unconfigured (0)
Unused license (95)

IVR Scenario
WSMM is embedded with IVR Scenario Editor.
An administrator may create/apply IVR scenario
without using special tool.

PBX Service - Voice Mail Services

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories like Extensions, Trunks, PBX Services, System Admin, and Summary. The 'PBX Services' menu is highlighted with a red dashed box, and a yellow starburst icon is placed over the 'Voice Mail Service' option. A blue arrow points from this icon to the 'Voice Mail Service' configuration page. The configuration page includes fields for 'Retrieving Extension by Other Phone', 'Retrieving Extension by Owner Phone', and 'Leave Extension'. It also features an 'Advanced Options' section with settings for 'Audio Message Length', 'Per Extension Hdd Quota', 'Over HDD Quota', 'Use Account Blocking', 'Password Fail Count', 'Enable E-mail Notification', 'Attach File to Email', 'Delete File After Email Notification', 'Enable SMS Notification', and 'SIP Port'. A 'Description' box on the right explains the purpose of the settings. At the bottom of the interface, a copyright notice reads 'Copyright © AddPac 1999-2012. All Rights Reserved.'

Voice Mail Service
Voice Mail Service let you setup Voice Mail Extension, Message Box, Notification, and SMS related setup. Each user may check the received voice-mail, SMS through user portal web page.

PBX Service - RTP Proxy Service

Smart Multimedia Manager
www.addpac.com

Extensions
Trunks
PBX Services

- Speed Button Profiles
- Announcement and Tones
- IVR Scenarios
- Voice Mail Service
- RTP Proxy Service
- Service Codes
- Day Templates
- Security Profiles
- Service Monitoring

System Admin
Summary

Getting Started GO

Follow Us

Linked in
facebook f
YouTube

Welcome to AddPac IP-PBX

root
Last Login at June 11 01:51:12PM (127.0.0.1)

Quick Menu

User Extension

Registered (2)
Unregistered (3)
Unconfigured (0)
Unused license (95)

Smart Multimedia Manager
www.addpac.com

Status RTP Proxy Service

Apply Cancel

RTP Proxy Settings

Idle Timeout 600 (0-7200, default: 600sec)

Packet Loss Event Count 0 (0-65535, default: 0)

*IPv4

Add Network Domain

Network Domain	Minimum	Maximum	DSCP	Modify	Delete
----------------	---------	---------	------	--------	--------

Network Domain

*IPv6

Add Network Domain

Network Domain	Minimum	Maximum	DSCP	Modify	Delete
----------------	---------	---------	------	--------	--------

Description

Manage RTP Proxy Service for NAT traversal. Normally, RTP proxying between private network and public network will be automatically handled by PBX. If you got problem to hear voice from remote side, enable option of RTP proxying in trunk setting or user extension setting.

RTP Proxy Service
RTP Proxy supports smooth call conversation by acting as rtp packet relay for each different network (private/ public) Call. RTP Proxy Service provides various options such as (Port range / DSCP)

PBX Service - Service Codes

The screenshot shows the Smart Multimedia Manager interface. On the left, the 'PBX Services' menu is expanded, with 'Service Codes' highlighted. A blue arrow points from this menu item to the 'Service Codes' configuration page. The configuration page includes a 'General Code' section with fields for 'Call Park', 'Call Pickup', 'Call Forwarding All Register', 'Call Forwarding All Activation', and 'Call Forwarding All Deactivation'. Below this is an 'Advanced Options' section with various call management settings like 'Call Reject(Absence) Activation', 'Call Waiting Activation', and 'CCBS Register'. A 'Description' box on the right explains that a service code is a special digit starting with # or *.

Service Codes
 A function to setup additional service phone number in IP-PBX. It is a service code to use additional service in SIP terminal and start with # or * and may assign maximum of two phone numbers.

PBX Service - Day Templates

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories like Extensions, Trunks, PBX Services, System Admin, and Summary. The 'PBX Services' category is expanded, and 'Day Templates' is highlighted. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root'. Below this, there's a 'Quick Menu' and a 'User Extension' section with a status bar showing 'Registered (2)', 'Unregistered (3)', 'Unconfigured (0)', and 'Unused license (95)'. The 'Day Templates' section is active, showing a table with one entry: 'holiday' created on 2012-03-30 11:24:41. Below the table, there are 'Add' and 'Cancel' buttons, and a form for adding a new template with fields for 'Name' and 'Description'. A tooltip explains that a Day Template is used to specify a period or special day(s) for schedule policy.

Day Templates
Day Template function provides a service in accordance with registered date after registering special date/day as template (date / Day of Week / Weekly)

PBX Service - Security Profiles

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories like Extensions, Trunks, PBX Services, and System Admin. The 'PBX Services' category is expanded, showing options such as Speed Button Profiles, Announcement and Tones, IVR Scenarios, Voice Mail Service, RTP Proxy Service, Service Codes, Day Templates, Security Profiles, and Service Monitoring. A mouse cursor is pointing at the 'Security Profiles' option. The main content area shows the 'Security Profiles' configuration page, which includes a table with columns for Modify, Delete, Name, Description, and Date Created. The table contains one entry: 'default' with description 'default security profile' and date '2012-06-08 19:49:52'. Below the table is the 'Global Security Setting' section, which includes a dropdown menu for 'TLS Cipher Suites' with a list of options: N/A, RC4_40, RC4_128, DES_CBC, 3DES_CBC, AES_128_CBC, AES_256_CBC, SEED_CBC, and ARIA_CBC. A description box on the right explains that these cipher suites are used for secure SIP negotiations.

Security Profiles
IP-PBX supports TLS Cipher Suites.
User may select priority with 3 TLS Suites and may select RC4_40, RC4_128, DES_CBC, 3DES_CBC, AES_128_CBC, AES_256_CBC, SEED_CBC, ARIA_CBC in each suites.

PBX Service - Service Monitoring

Smart Multimedia Manager
www.addpac.com

Extensions
Trunks
PBX Services

- Speed Button Profiles
- Announcement and Tones
- IVR Scenarios
- Voice Mail Service
- RTP Proxy Service
- Service Codes
- Day Templates
- Security Profiles
- Service Monitoring

System Admin
Summary

Getting Started GO

Follow Us
Linked in
facebook f
YouTube

Welcome to AddPac IP-PBX
root
Last Login at June 11 01:51:12PM (127.0.0.1)

Quick Menu

Smart Multimedia Manager
www.addpac.com

Extensions
Trunks
PBX Services

- Speed Button Profiles
- Announcement and Tones
- IVR Scenarios
- Voice Mail Service
- RTP Proxy Service
- Service Codes
- Day Templates
- Security Profiles
- Service Monitoring

System Admin
Summary

Getting Started GO

Follow Us
Linked in
facebook f
YouTube

Service Monitoring
Interval : 10 sec.

Active Calls | Conference

ID	Established Time	Duration	Calling Number	Called Number	Audio Codec	Video Codec	Recording	Drop Call

System Status Dashboard:

- Registered (2)
- Unregistered (3)
- Unconfigured (0)
- Unused license (95)

System Resources:

- Memory Storage: 0%
- Network: 8%
- Call Manager: 0/100
- MCU: 0/2
- Presence: 0/100
- IVR: 0/100
- Media: 0/100
- UMS: 0/100
- RtpProxy: 0/100

Trunks:

- Internal Trunk Gateway (0/0)
- ss (0/0)
- SM_SIP_Provider (0/0)
- JschoL_gk (0/0)

Service Monitoring
It displays Active Call & Conference information in IP-PBX. User may setup monitoring screen renew, interval time setup, and provides active call & conference information.

System Admin - Network Interface

Smart Multimedia Manager
www.addpac.com

Start

Welcome to AddPac IP-PBX

root
Last Login at June 11 01:51:12PM (127.0.0.1)

Quick Menu

System Admin

- Network Interface
- Network Services
- Administrators
- Licenses
- Voice Lines
- Alarm History
- Call History
- Show Command

Summary

Getting Started GO

Follow Us

LinkedIn

facebook

YouTube

User Extension

Registered (2)
Unregistered (3)
Unconfigured (0)
Unused license (95)

Network Interfaces

Apply Cancel

Interface Mode: DHCP Static IP

WAN Interface

IP Address * A.B.C.D

Subnet Mask * A.B.C.D

Default Gateway A.B.C.D

Primary DNS Server

Secondary DNS Server

IPv6 Address XX:XXM

IPv6 Default Gateway XX:XX

LAN Interface

Interface Mode: None Bridge IP Shared NAT Static IP

IP Address A.B.C.D

Subnet Mask A.B.C.D

DHCP Server On Off

DHCP Range ~ A.B.C.D

IPv6 Address XX:XXM

Description

This PBX system can have one or two network interfaces. The WAN interface is a main network interface of this system normally has public IP address for communicating with VoIP providers and Trunk gateways in public domain. The LAN interface normally has private IP address for communicating with IP phones or user terminals in private domain.

Copyright © AddPac, 1998-2012. All Rights Reserved.

Network Interface
IP-PBX Network interface setup.
WAN Interface
- IPv4 / IPv6 Address, DNS, DHCP Client
LAN Interface
- IPv4 / IPv6 Address, DHCP Server

System Admin - Network Services

The screenshot shows the 'Smart Multimedia Manager' interface. The left sidebar contains a 'System Admin' menu with a red dashed box around it, and a yellow starburst icon pointing to the 'Network Services' option. The main content area is titled 'Network Services' and contains a table of configuration options for various protocols. A yellow box at the bottom left contains text describing the network service setup.

Service	Configuration Options
NTP	Time zone: % Unknown command (show clock-http) server enable: <input type="radio"/> On <input checked="" type="radio"/> Off System Datetime: [] - [] - [] : [] : [] Primary NTP Server: [] Secondary NTP Server: [] Interval: [] NTP time resynchronize, in hour (default: 27)
TELNET	Service Enable: <input checked="" type="radio"/> On <input type="radio"/> Off Service Port: 23 (default: 23) Service Enable: <input type="radio"/> On <input checked="" type="radio"/> Off
SNMP	Service Port: [] (default: 161) Community: [] Trap Service IP Address: [] Trap Community: []
HTTP	Service Enable: <input checked="" type="radio"/> On <input type="radio"/> Off Service Port: 80 (default: 80) Authentication: <input type="radio"/> NONE <input checked="" type="radio"/> Basic <input type="radio"/> Digest
FTP	Service Enable: <input checked="" type="radio"/> On <input type="radio"/> Off Control Port: 21 (default: 21) Data Port: 20 (default: 20)
LDAP	Server Port: 389 (default: 389)
SYSLOG	Service Enable: <input type="radio"/> On <input checked="" type="radio"/> Off Service Port: [] (default: 514) Log Life Time: [] (1 ~ 300 Day)

Network Service
IP-PBX network service setup.
User may setup NTP, TELNET, SNMP, HTTP, FTP, LDAP, SYSLOG, Dynamic DNS, CDR, SMTP, DDoS function detail setup.

System Admin - Administrators

The screenshot shows the Smart Multimedia Manager interface. The left sidebar contains a 'System Admin' menu with sub-items: Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, Show Command, and Summary. The main content area displays the 'Administrators' page, which includes a table of existing administrators and a form for creating or editing one.

Modify	Delete	Name	ID	Level	Description
		root	root	Administrator	System Administrator
		administrator	administrator	Administrator	Addpac Administrator

Administrators

General Settings

User name*

Description

ID*

Password*

Level

Application Permission

Door Access Control Manager

Time and Attendance Manager

Description: Add / Delete / Modify an Administrator who configure, operate, and monitor this system.

Registered (2)
Unregistered (3)
Unconfigured (0)
Unused license (95)

Administrator
An administrator creation/change is possible to operate IP-PBX. Level (Administrator / Operator / Monitor) application is possible and may assign additional Application Permission (Door Access Control Manager / Time and Attendance Manager)

System Admin - Licenses

The screenshot shows the 'Smart Multimedia Manager' interface. The 'System Admin' menu is expanded, and the 'Licenses' option is selected. The 'Licenses' page displays a table of licenses and their values.

Service	License	Value	
1	Call Manager	Max Calls	100
2	Call Manager	Max Devices	100
3	Call Manager	Max Subscribers	100
4	MCU	Max Sessions	2
5	MCU	Max Party per Sessions	4
6	Presence	Max Sessions	100
7	IVR	Max Sessions	100
8	IVR	Max Scenarios	100
9	Media	Max Sessions	100
10	UMS	Max Sessions	100
11	UMS	Max Mail-boxes	100
12	RtpProxy	Max Sessions	100

License
To use various service of IP-PBX, License must be created. In accordance with License policy, Max Service is restricted and license upload/download is possible in accordance with policy.

System Admin - Voice Lines

Smart Multimedia Manager
www.addpac.com

Start

Welcome to AddPac IP-PBX

root
Last Login at June 11 01:51:12PM (127.0.0.1)

Click Menu

System Admin

- Network Interfaces
- Network Services
- Administrators
- Licenses
- Voice Lines
- Alarm History
- Call History
- Show Command

Summary

Getting Started GO

Follow Us

Linked in

facebook f

YouTube

User Extension

Registered (2)

Unregistered (3)

Unconfigured (0)

Unused license (95)

Smart Multimedia Manager

www.addpac.com

Status IVR Scenarios Voice Lines

Voice Lines

Apply Cancel

Slot / Port	Type	Status	Pattern	Connection PLAR	Input Gain	Output Gain	Caller ID	Modify
1 1/4	FXO	idle			0	0	Disabled	✓
2 1/5	FXO	idle			0	0	Disabled	✓
3 1/6	FXO	idle			0	0	Disabled	✓
4 1/7	FXO	idle			0	0	Disabled	✓
5 2/0	GSM	unreg...			0	0	Disabled	✓
6 2/1	GSM	unreg...			0	0	Disabled	✓
7 2/2	GSM	unreg...			0	0	Disabled	✓
8 2/3	GSM	unreg...			0	0	Disabled	✓

Analog & Mobile

Trunk

Slot / Port	Type	Status	Pattern	Connection PLAR	Input Gain	Output Gain	Clock Source	Protocol Emulat	Modify
1 0/0/0	E1	down			0	0	Master	Network	✓
2 0/1/0	E1	down			0	0	Master	Network	✓

Digital

Extension Analog

Slot / Port	Type	Status	Number	User ID	Password	Input Gain	Output Gain	Caller ID	Modify
1 1/0	FXS	idle	1100	1100	1111	0	0	Disabled	✓
2 1/1	FXS	idle	1101	1101	1111	0	0	Disabled	✓
3 1/2	FXS	idle				0	0	Disabled	✓
4 1/3	FXS	idle				0	0	Disabled	✓

Description

This is a built-in voice lines such as FXS lines for analog extensions and FXO, E&M, E1, T1, GSM lines for internal trunk gateway. You can add analog extension at extension menu and set internal trunk gateway property at trunk menu. You can set some physical settings at here and detail settings by Smart Web Manager(Internal Voice Line).

Related Links

- Smart Web Manager (Internal Voice Line)
- Analog Extension
- Internal Trunk Gateway

Getting Started GO

Follow Us

Linked in

facebook f

YouTube

Copyright © AddPac 1998-2012. All Rights Reserved

Voice Line
It displays Voice Module information in IP-PBX. Voice modules are including FXS, FXO, E&M, E1, T1, GSM, and 3G. Each module may setup Gain, Caller ID, and Pattern.

System Admin - Alarm History

Smart Multimedia Manager
www.addpac.com

Welcome to AddPac IP-PBX
root
Last Login at June 11 01:51:12PM (127.0.0.1)

System Admin

- Network Interfaces
- Network Services
- Administrators
- Licenses
- Voice Lines
- Alarm History
- Call History
- Show Command

Alarm History

Level: All | Ack: All | Period: 2012-06-05 ~ 2012-06-12 | Search | Refresh

Level	Messages	DateTime
1 Critical	The Call Manager SERVER on node Router , MX250(172.16.17.30) Server is REGISTERED	2012-06-12 19:48:39
2 Major	The Call Manager TERMINAL on node Router , BongY&Jeong(172.16.17.30) Terminal is REGISTERED	2012-06-12 19:27:39
3 Major	The Call Manager TERMINAL on node Router , BY&Jeong(172.16.17.30) Terminal is UNREGISTERED	2012-06-12 19:27:19
4 Major	The Call Manager TERMINAL on node Router , BY&Jeong(172.16.17.30) Terminal is REGISTERED	2012-06-12 19:27:19
5 Major	The Call Manager TERMINAL on node Router , BY&Jeong(172.16.17.30) Terminal is REGISTERED	2012-06-12 19:26:54
6 Major	The Call Manager TERMINAL on node Router , BongYong&Jeong(172.16.18.101) Terminal is REGISTERED	2012-06-12 19:25:16
7 Minor	An Authentication/Connection Success has been identified on network device 172.16.1.50. This message is usually gen...	2012-06-12 19:22:58
8 Major	The Call Manager TERMINAL on node Router , ByoungGoo&Choi(172.16.18.100) Terminal is REGISTERED	2012-06-12 19:21:55
9 Critical	The Call Manager SERVER on node Router , MX250(172.16.17.30) Server is REGISTERED	2012-06-12 19:21:43
10 Critical	The Call Manager SERVER on node Router , MX250(172.16.17.30) Server is REGISTERED	2012-06-12 19:21:43
11 Critical	The Call Manager Service on node Router which was previously OutOfService is now In Service	2012-06-12 19:21:37
12 Critical	The Call Manager SERVER on node Router , MX250(172.16.17.30) Server is REGISTERED	2012-06-12 19:21:37
13 Critical	The Call Manager TRUNK on node Router , Internal Trunk Gateway(127.0.0.1) Trunk is REGISTERED	2012-06-12 19:21:37
14 Critical	The Call Manager SERVER on node Router , MX250(172.16.17.30) Server is REGISTERED	2012-06-12 19:21:37

System Status

User Extensions: Registered (2), Unregistered (3), Unconfigured (0), Unused license (95)

System: Memory Storage (1%), Network (7%), Call Manager (0/100), MCU (0/2), Presence (0/100), IVR (0/100), Media (0/100), UMS (0/100), RtpProxy (0/100)

Trunks: Internal Trunk Gateway (0/0), SKN_TG (0/0), Dacom_Trunk (0/0)

Alarm History
It displays trouble in IP-PBX system. Level (All / Critical / Major / Minor), ACK (All / Acknowledge / Not Acknowledge). User may check various information through period filter.

System Admin - Call History

The screenshot shows the Smart Multimedia Manager interface. The left sidebar contains a 'System Admin' menu with options: Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, Show Command, and Summary. The 'Call History' page is displayed, showing a summary table and a detailed table. A yellow callout box provides the following information:

Call History
It displays device error which was occurred in IP-PBX System. User may check various information through Call Type (Unspecified / Inter-Site Call / PSTN Backup / Service Provider), Period, Number

The 'Call History' page includes a 'Summary' table with the following data:

Summary	Total Call Duration	Total Call Count	ASR	Longest Call Duration	Call Fail Count
472	472	11	81%	90 (sec)	2

Below the summary is a table with columns: Calling Number, Called Number, Established Time, duration (sec), Call State, State Cause, and datetime. The page also features a system status diagram and various resource usage gauges.

System Admin - Show Command

The screenshot displays the Smart Multimedia Manager System Admin interface. The left sidebar contains a navigation menu with 'System Admin' selected, and a sub-menu where 'Show Command' is highlighted. A red dashed box highlights the 'Show Command' option in the sidebar and the corresponding 'Show Command' window in the main content area. A blue arrow points from the sidebar menu to the window. The 'Show Command' window shows a 'Request Command' field with the text 'show call-manager sscp' and a 'Show' button. Below this, the system output is displayed in a code block format. A 'Description' box on the right explains that users can check system status by selecting a category or entering a CLI command.

Request Command: show call-manager sscp

```
SSCP Timer Information.
  retry-counter = 1
  retry-timeout = 5000 (msec)
  keepalive-timeout = 30 (sec)

CM <-> CM_Servers SSCP Information.
  retry-counter = 3
  retry-timeout = 5000 (msec)
  keepalive-timeout = 3 (sec)
  keepalive-retry-timeout = 1000 (msec)
  keepalive-retry-timeout(other server) = 3000 (msec)
  binding accept status = TRUE

SSCP Policy Information.
  cm service status = on
  signaling-port = 8855
  packet-size = 1472

  client-session logging count = 0
  client-session registering count = 0

  event store time = 3(sec)
  event store count = 10
  event total store count = 0

Client Auth Session Information.
  id          ip          timerCount
  -----
  -----

Client Session Information.
  session-id  user-id      ip address      port  status
  -----
  2001        1009        172.16.18.100  5060  in-service
  2002        3000        172.16.18.101  5060  in-service
  -----

Servers Information.
  server-id   binding-id   ip address      port  state
  -----
  10200000    1            172.16.17.30   5101  BIND  ums
  10100000    1            172.16.17.30   5041  BIND  rdt
  10600000    1            172.16.17.30   5021  BIND  tvr
  -----

SessionClientGroup
Group(0) sessionSize(0) :
Group(2) sessionSize(0) :
```

Show Command
User may check the status of IP-PBX System through category and CLI (Command Line Interface)

Ordering Information

- IPNext10000 Next Generation IP-PBX Hardware
 - One(1) System Dual IP-PBX Main Body
- Built-in APOS Internetworking Software for IPNext10000
- Including 1 Year Hardware Warranty
- Product Documents
 - Install and Operation Guide (PDF)
- Pricing
 - AddPac Technology Regional Sales Manager
 - Authorized Sales and Marketing Representatives
 - Please Contact www.addpac.com



Thank you!

AddPac Technology Co., Ltd.
Sales and Marketing

Phone +82.2.568.3848 (KOREA)

FAX +82.2.568.3847 (KOREA)

E-mail sales@addpac.com