

Dual Redundancy IP-PBX Solution



AddPac

AddPac Technology

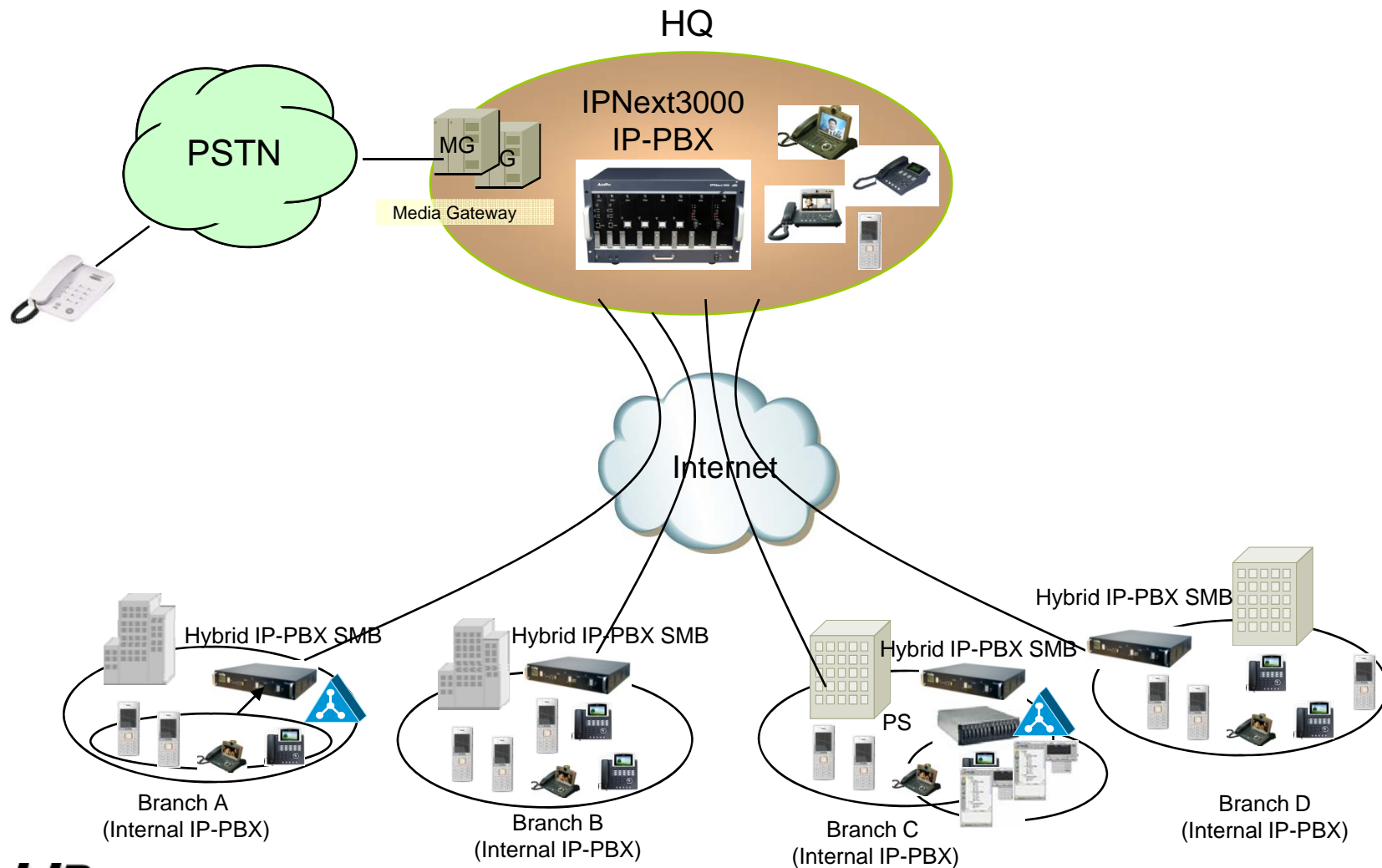
2013, Sales and Marketing

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- IP Phones for AddPac IP-PBX
- IP Extend Key Pack for AddPac IP-PBX
- Wifi IP Phone, Soft Phone, Smart Messenger
- Web based SMM (Smart Multimedia Manager)

IPNext IP-PBX Service Diagram

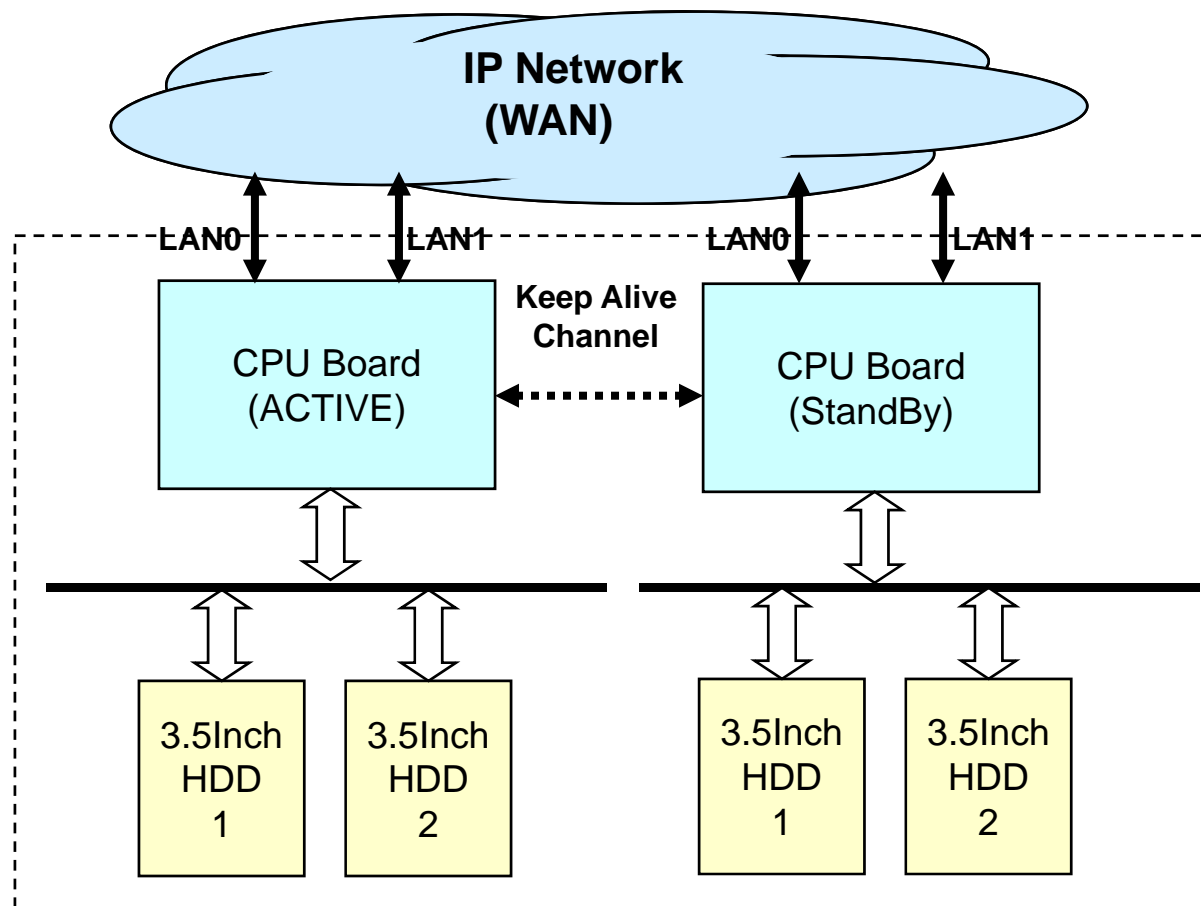


IPNext IP-PBX Comparison Table

Model		IPNext5000	IPNext3000	IPNext2000	IPNext600
Service Features					
Registration User Number		5000	3000	2000	500
Concurrent Call User Number		1000	800	500	100
IPv4/IPv6 Dual Stack Support		Support	Support	Support	Support
VoIP Signaling	Internal	SIP	SIP	SIP	SIP
	External	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support	Support	Support
RTP Proxy Service (IPv6, Private IP)		Support	Support	Support	Support
LAN Port		2	2	2	2
System Duplication		Support(built-in)	Support(built-in)	Support(built-in)	Support(Built-In)

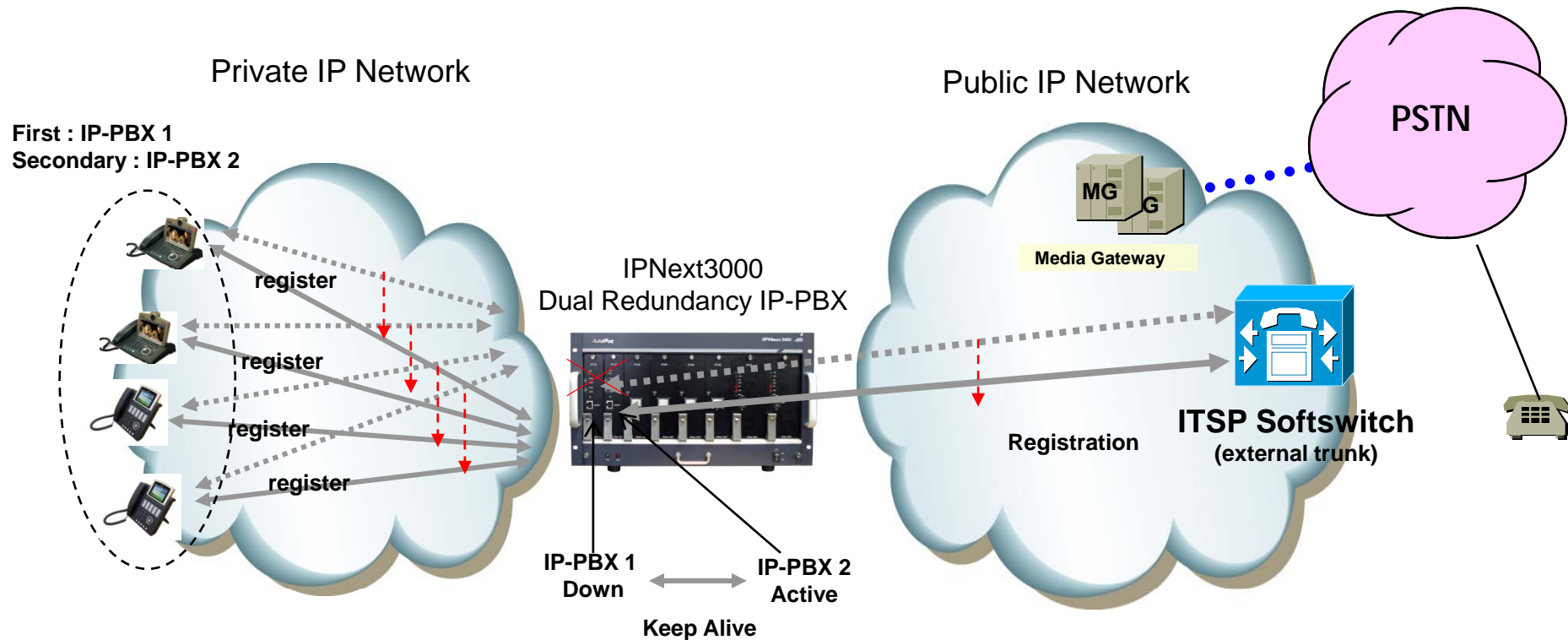
One System Dual Redundancy Architecture

One System Dual Redundancy IP-PBX System Block Diagram




One System Dual Redundancy Architecture

Active-Standby System Redundancy Scheme



ITSP: Internet Telephony Service Provider



IPNext5000 IP-PBX (One System, Dual IP-PBX)

Main Features

IPNext5000 Next Generation IP-PBX

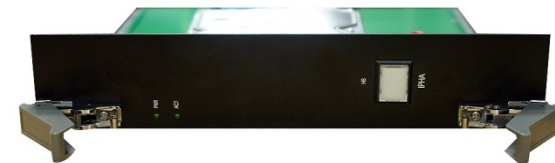
- IP based Advanced Mobile IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Large Office
- System Duplication Support (Dual System Board, Dual Power Supply)
- Powerful Management and User Friendly Features
- Fault Tolerant and Scalability Architecture
- High-performance Video, Audio, and Voice Service
- Firmware Upgradeable Architecture
- IVR Service with Scenario Editor
- Voice Mailing Service
- Presence Service for High-End IP Phone, Video Phone, UC
- RTP Proxy Service for Private IP service
- SIP, H.323 Signaling for Outbound Calls
- Various Call Scenario (Call Pickup, Call Park, Call Transfer, etc)
- Various IP Multimedia Terminal Support

Hardware Specification

IPNext5000 Next Generation IP-PBX

RISC
CPU

- 64bit High-End C Microprocessor Computing Power
- Main Chassis
 - Dual Redundancy CPU Boards for System Fault Tolerant
 - Two(2) 10/100Mbps Gigabit Ethernet
 - One(1) RS-232C Console (RJ45)
 - Two(2) 3.5 Inch Hard Disk Interface Slot (RAID 1)
 - Dual Redundancy Power Supply Module
 - Hot-Swap Features

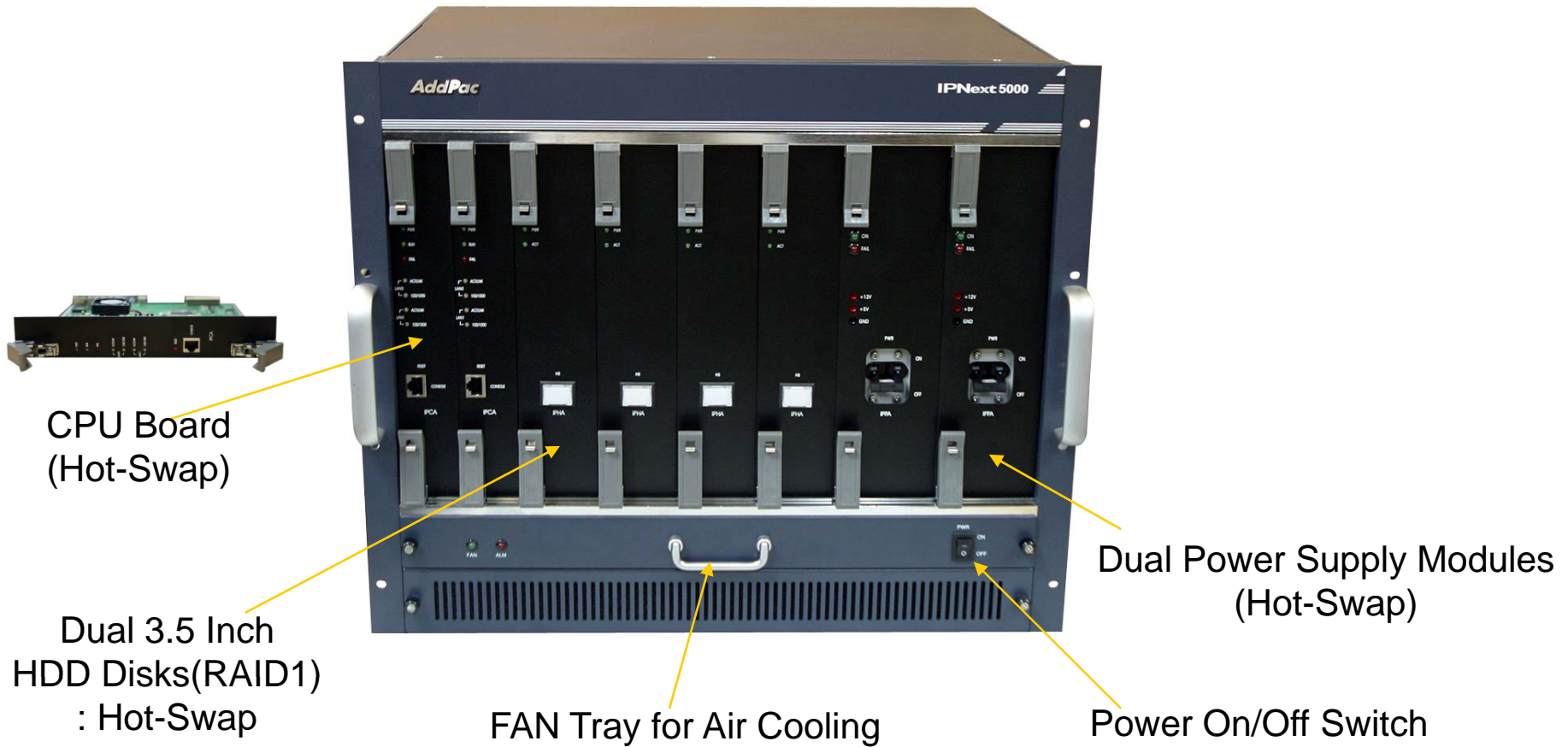


Hardware Specification

IPNext5000 Next Generation IP-PBX



IPNext5000 Front Side



Hardware Specification

IPNext5000 Next Generation IP-PBX

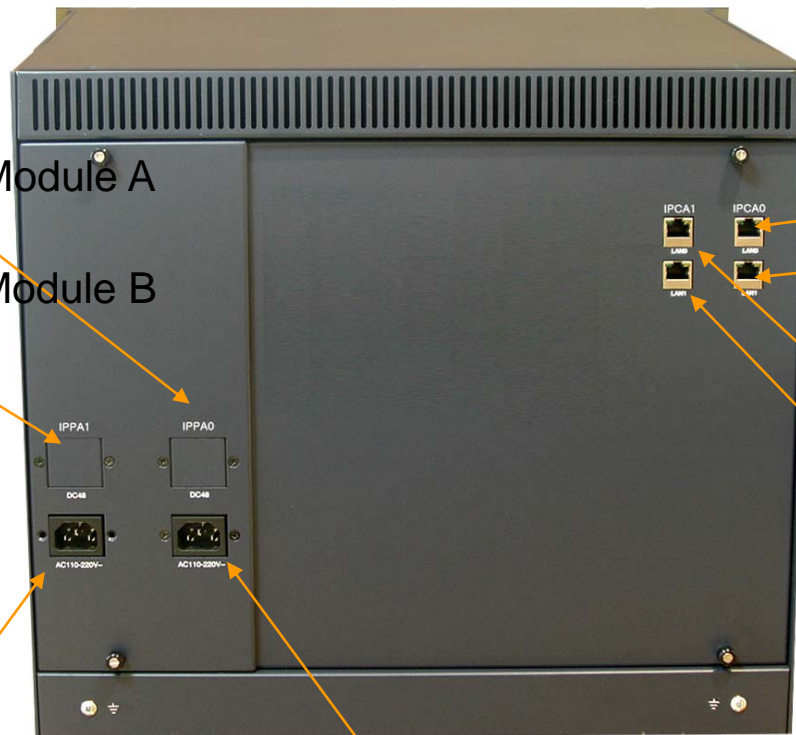
RISC
CPU

DSP

IPNext5000 Back Side

Plug for DC PSU Module A

Plug for DC PSU Module B



LAN0 for Active CPU


LAN1 for Active CPU

LAN0 for Standby
CPU

LAN1 for Standby
CPU

One Plug for Single PSU Modules
(AC PSU Module B)

Plug for AC PSU Module A
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IPNext3000 IP-PBX (One System, Dual IP-PBX)

Main Features

IPNext3000 Next Generation IP-PBX

- IP based Advanced Mobile IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Large Office
- System Duplication Support (Dual System Board, Dual Power Supply)
- Powerful Management and User Friendly Features
- Fault Tolerant and Scalability Architecture
- High-performance Video, Audio, and Voice Service
- Firmware Upgradeable Architecture
- IVR Service with Scenario Editor
- Voice Mailing Service
- Presence Service for High-End IP Phone, Video Phone, UC
- RTP Proxy Service for Private IP service
- SIP, H.323 Signaling for Outbound Calls
- Various Call Scenario (Call Pickup, Call Park, Call Transfer, etc)
- Various IP Multimedia Terminal Support

Hardware Specification

IPNext3000 Next Generation IP-PBX

RISC
CPU

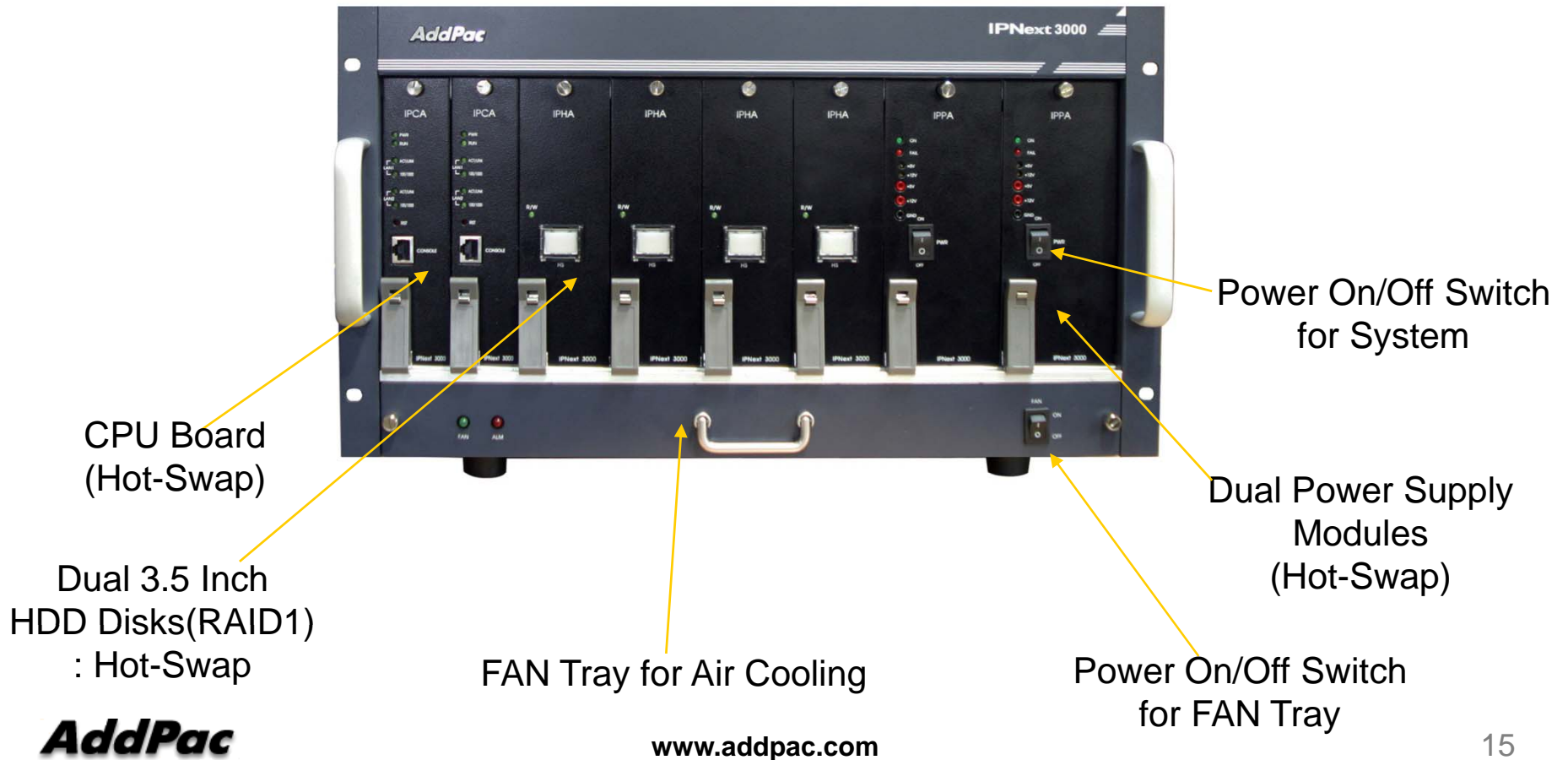
- 64bit High-End Microprocessor Computing Power
- Main Chassis
 - Dual Redundancy CPU Boards for System Fault Tolerant
 - Two(2) 10/100/1000Mbps Gigabit Ethernet
 - One(1) RS-232C Console (RJ45)
 - Two(2) 3.5 Inch Hard Disk Interface Slot (RAID 1)
 - Dual Redundancy Power Supply Module
 - Hot-Swap Features

Hardware Specification

IPNext3000 Next Generation IP-PBX



IPNext3000 Front Side

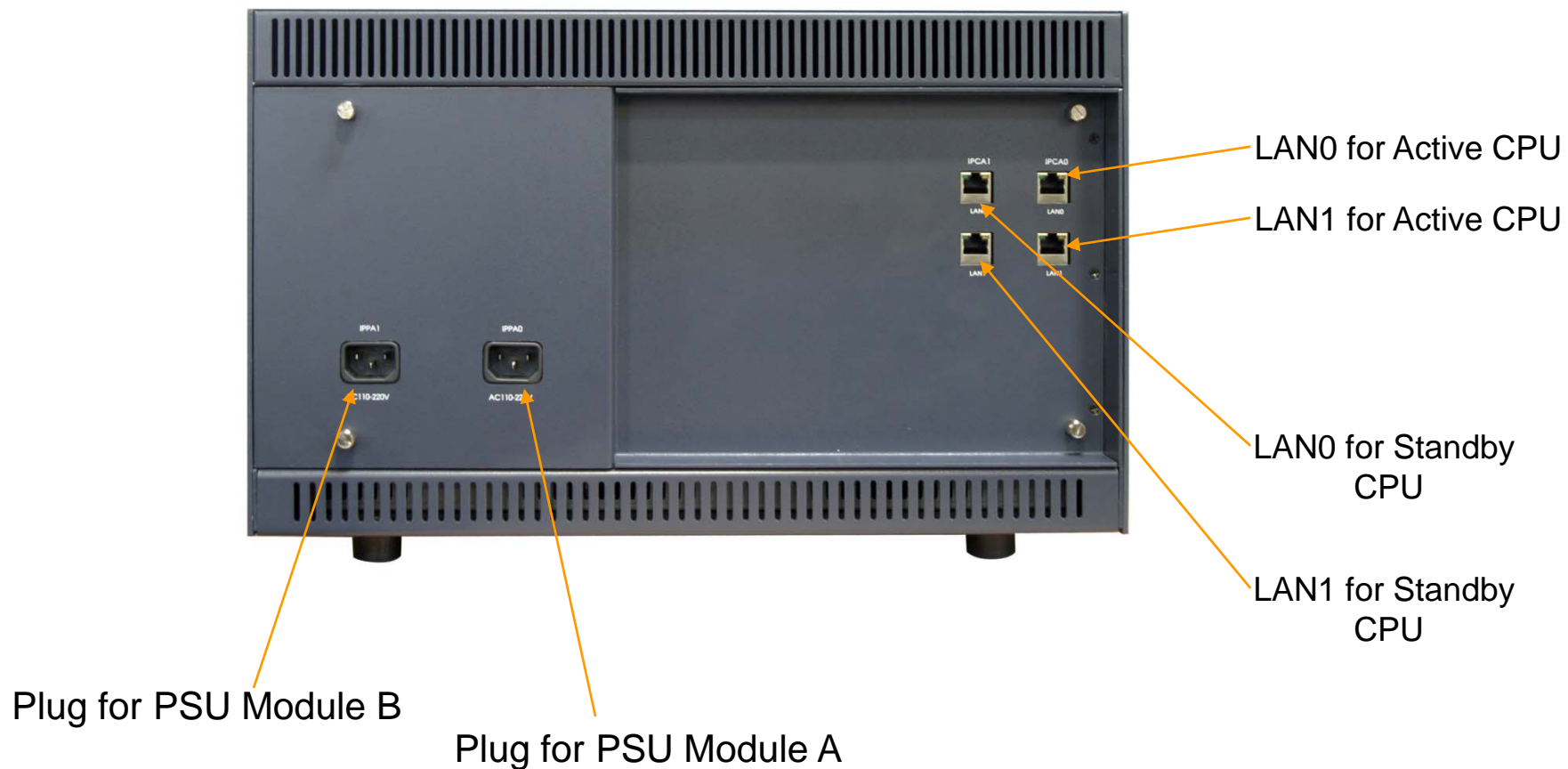


Hardware Specification

IPNext3000 Next Generation IP-PBX



IPNext3000 Back Side





IPNext2000 IP-PBX (One System, Dual IP-PBX)

Main Features

IPNext2000 Next Generation IP-PBX

- IP based Advanced Mobile IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Large Office
- System Duplication Support (Dual System Board, Dual Power Supply)
- Powerful Management and User Friendly Features
- Fault Tolerant and Scalability Architecture
- High-performance Video, Audio, and Voice Service
- Firmware Upgradeable Architecture
- IVR Service with Scenario Editor
- Voice Mailing Service
- Presence Service for High-End IP Phone, Video Phone, UC
- RTP Proxy Service for Private IP service
- SIP, H.323 Signaling for Outbound Calls
- Various Call Scenario (Call Pickup, Call Park, Call Transfer, etc)
- Various IP Multimedia Terminal Support

Hardware Specification

IPNext2000 Next Generation IP-PBX

RISC
CPU

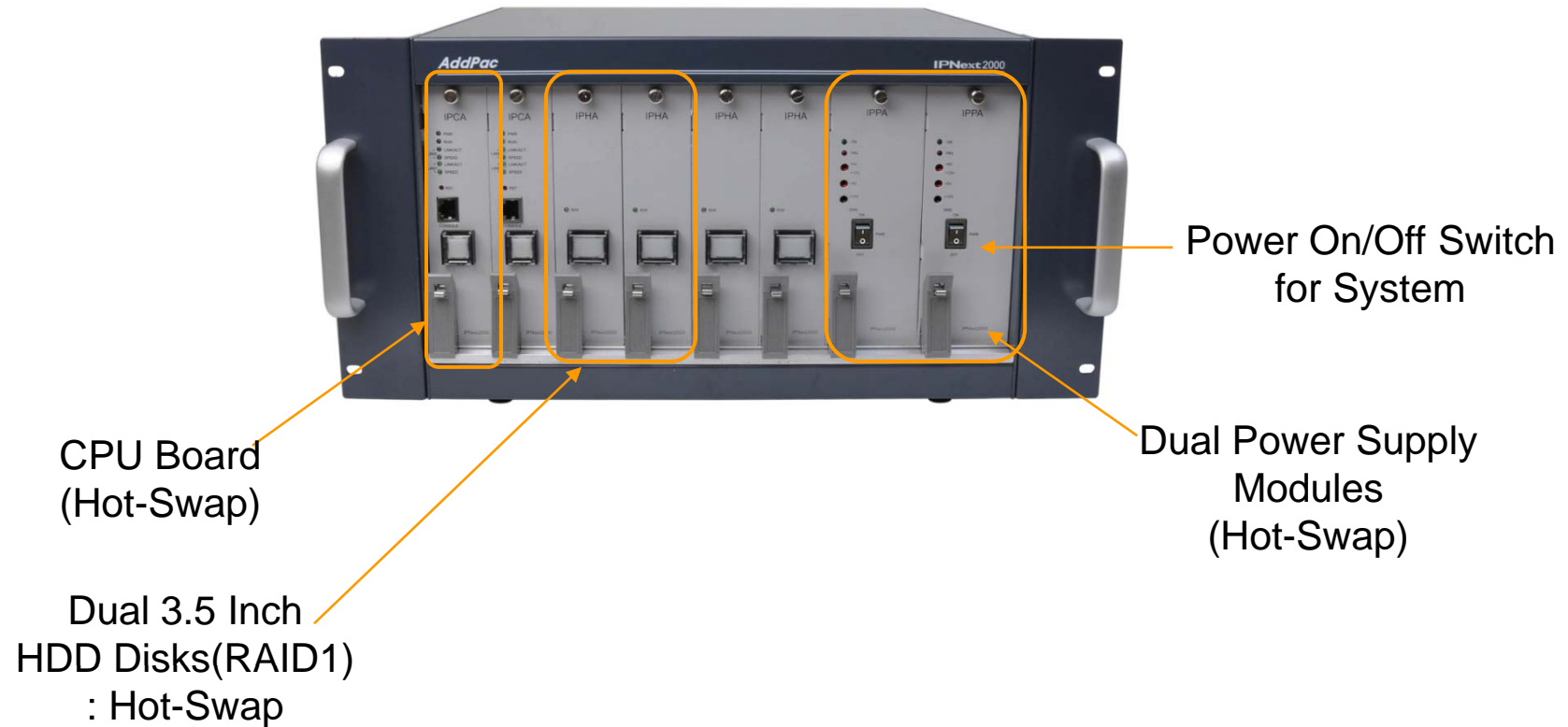
- 64bit High-End Microprocessor Computing Power
- Main Chassis
 - Dual Redundancy CPU Boards for System Fault Tolerant
 - Two(2) 10/100/1000Mbps Gigabit Ethernet
 - One(1) RS-232C Console (RJ45)
 - Two(2) 3.5 Inch Hard Disk Interface Slot (RAID 1)
 - Dual Redundancy Power Supply Module
 - Hot-Swap Features

Hardware Specification

IPNext2000 Next Generation IP-PBX

RISC
CPU

IPNext2000 Front Side

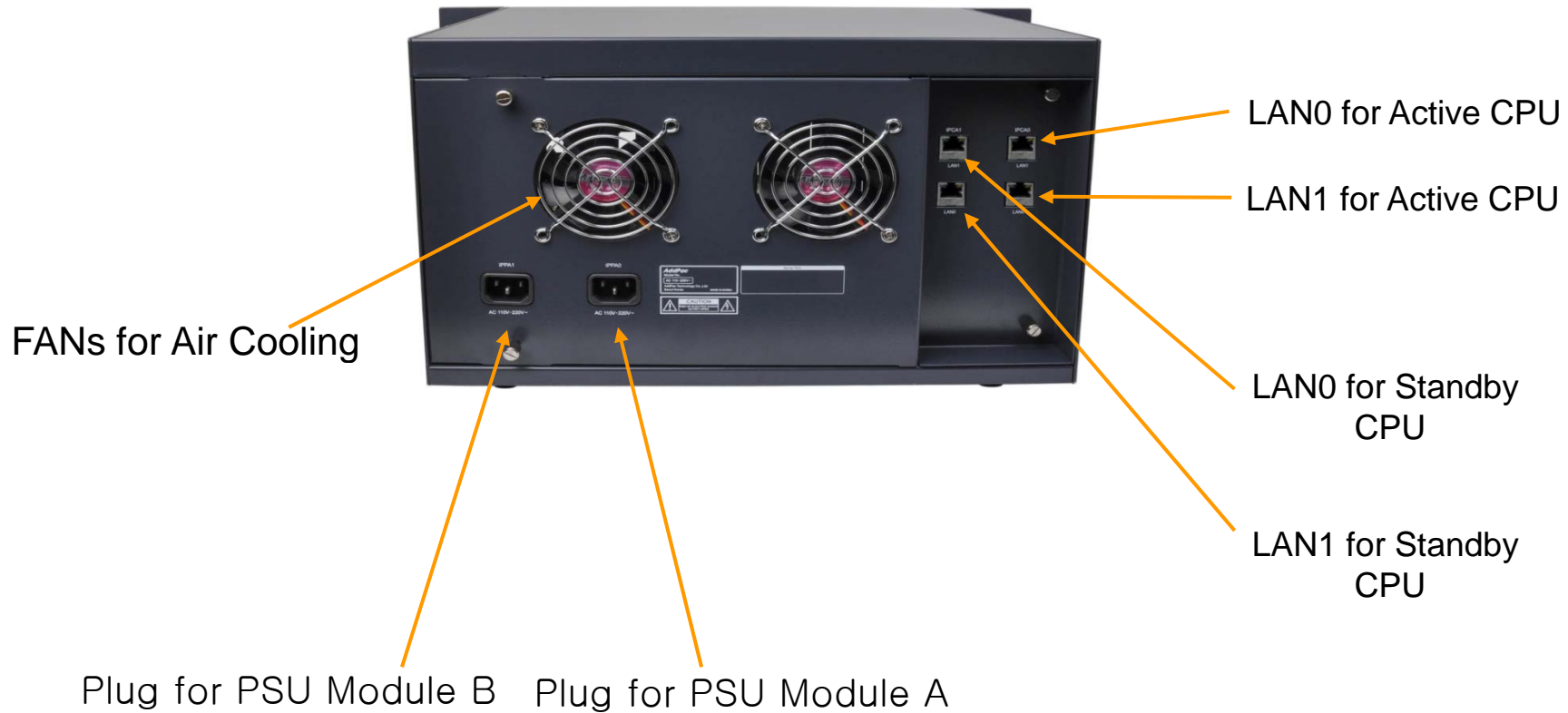


Hardware Specification

IPNext2000 Next Generation IP-PBX

RISC
CPU

IPNext2000 Back Side





IPNext600 IP-PBX (One System, Dual IP-PBX)

Main Features

IPNext600 Next Generation IP-PBX

- SIP Application Server, Proxy, Registrar and Location Server
- Multiple ITSP Trunk with SIP & H.323 Accounts Support
- Dual System Redundancy Architecture
 - Two(2) Fast Ethernet Interface / System
- High Performance RISC Architecture
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- IPv4/IPv6 Dual Stack
- RTP Proxy Function Embedded for Private IP and IPv6 Address Interworking
- User Presence Service Features for Smart Multimedia Messenger and Smart IP Phone
- IVR Scenario Editor, Voice Mail, Media Service (Coloring), Conference
- Firmware Upgradeable Architecture
- Smart Multimedia Manager for IP-PBX Management
- Smart Messenger Service (click to dial) for Unified Communication
- VPMS (VoIP Plug&Play Management System) & Smart NMS for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Dual Redundancy Power Module

Hardware Specification

IPNext600 Next Generation IP-PBX

RISC
CPU

- 64bit High-End Microprocessor Computing Power
- Main Chassis
 - Dual Redundancy CPU Boards for System Fault Tolerant
 - Two(2) 10/100Mbps Gigabit Ethernet
 - One(1) RS-232C Console (RJ45)
 - Dual Redundancy Power Supply Module
 - Hot-Swap Features

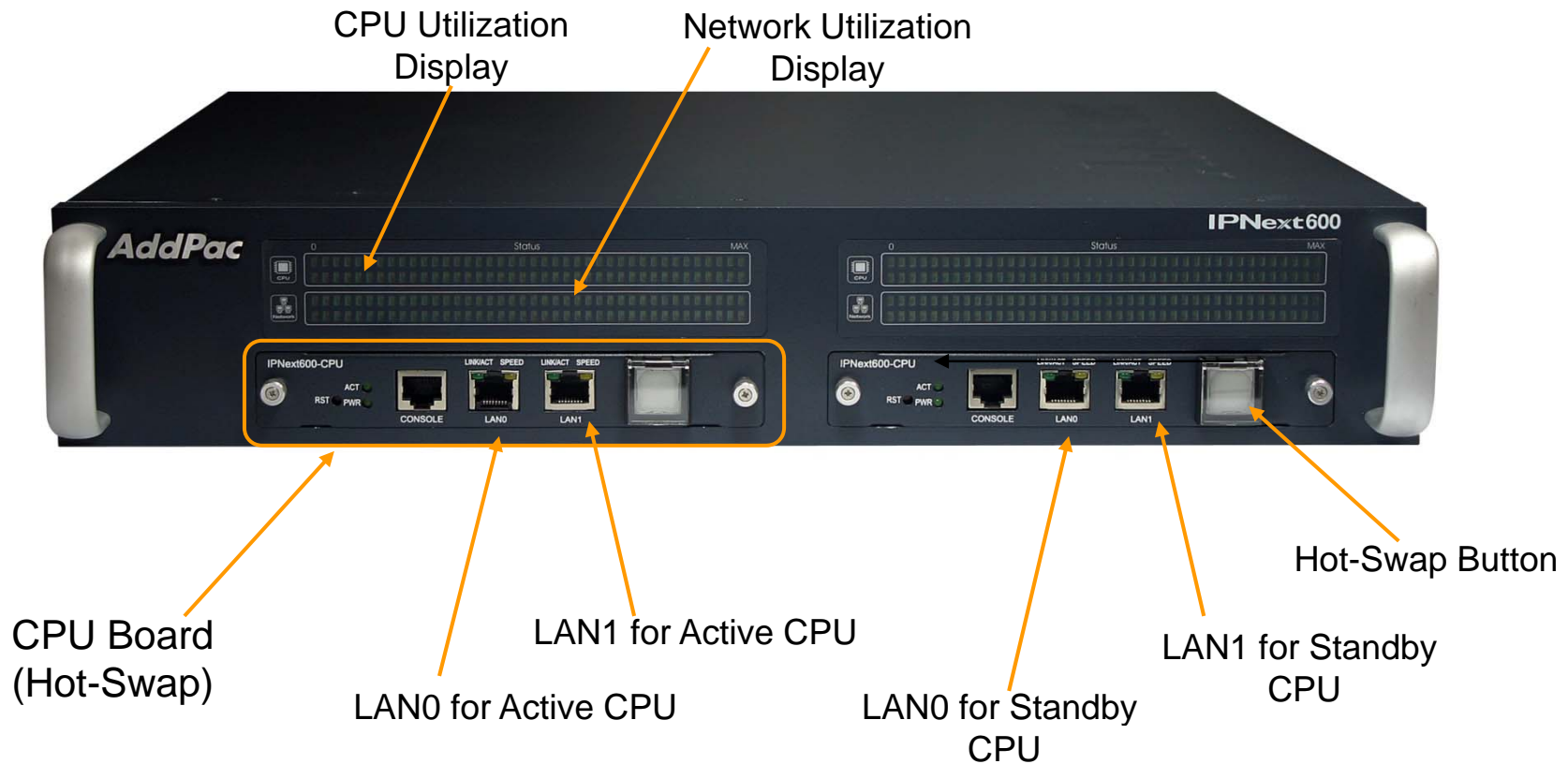


Hardware Specification

IPNext600 Next Generation IP-PBX

RISC
CPU

IPNext600 Front Side



Hardware Specification

IPNext600 Next Generation IP-PBX

RISC
CPU

IPNext600 Back Side

Dual Power Supply
Modules
(Hot-Swap)



PSU Module A

PSU Module B

Power On/Off Switch
for System



AddPac IP Terminals

AddPac IP Terminals

- AP-VP500 Video Phone
- AP-VP350 MCU Video Phone
- AP-VP300 Video Phone
- AP-VP280 Video Phone
- AP-VP250 Video Phone
- AP-VP230 Video Phone
- AP-VP150 Video Phone
- AP-VP120 Video Phone
- AP-IP300 IP Phone
- AP-IP250 IP Phone
- AP-IP230 IP Phone
- AP-IP160 IP Phone
- AP-IP120 IP Phone
- AP-IP90 IP Phone
- AP-WP100 Wi-Fi Phone
- AP-SMP100 Soft Phone



AP-VP500



AP-VP300



AP-VP150



AP-WP100



AP-VP120



AP-IP300



AP-IP230



AP-IP160











AP-IP120



IP Video Phones for AddPac IP-PBX


IP Video Phones for AddPac IP-PBX

	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	Support	Support	Support	Support	Support



IP Phones for AddPac IP-PBX




IP Phones for AddPac IP-PBX

	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 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)

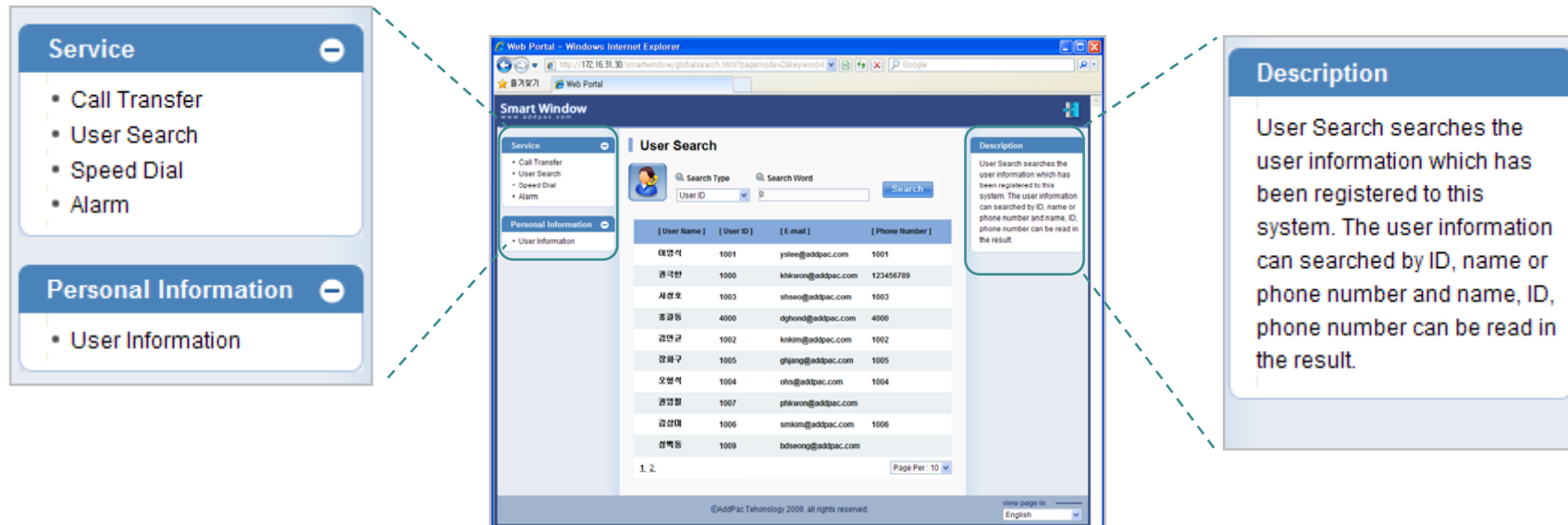
Smart Window is simple web based personal information management program for IP-PBX, Call Manager, etc.

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

Web based Smart Multimedia Manager for IP-PBX



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

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

The screenshot displays the 'Smart Multimedia Manager' (WSMM) interface. The main window is titled 'Add an User Extension' and contains a form with fields for Extension, First Name, Last Name, Voice Mail Password, User Password, Department, Title, Email, Home Phone, Mobile Phone, User ID, Photo, and Routing Access List. Below the form are 'Advanced Options' and 'General Settings' sections. A red dashed box highlights a help icon in the 'User Extension' section, with an arrow pointing to a help window. The help window, titled 'Help :: User Extension', contains the following content:

- 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.

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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

Routing Access List: internal

Advanced Options

Terminal Profile: default

General Settings

Security Profile: default

Use RTP Proxy:

Back Tone at:

Representation: Default

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

Related Links

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

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

Diagnostic

Smart Multimedia Manager
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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**

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
172.16.18.100>;tag=dc4fa2c5a4
f-519d-a2e8-80c5-0002a4038e2c@172.16.18.100

1800;refresher=uac
ic SIP Gateway
@172.16.18.100

lication/sdp
179

Step 2.

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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.

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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

The screenshot shows the Smart Multimedia Manager interface. The top section displays a welcome message for 'root' and an 'Unread Alarm Message' for 'login user authentication' on 2012-06-01 at 07:51:12. A red dashed box highlights this message, with a blue arrow pointing down to the 'Alarm History' page below. The 'Alarm History' page features a table with columns for Level, Messages, and DateTime, listing various system events such as NTP time sync service started, ftp service disabled, and disk upper quota limit exceeded.

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.

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

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 provides a definition of the Quick Menu.

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' web 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 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

Status

User Extensions

System

Memory Storage 0%
Network 7%

Call Manager
MCU
Presence
IVR
Media
UMS
RtpProxy

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	

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 navigation options: Extensions, Trunks, PBX Services, System Admin, and Summary. The main content area shows a 'Welcome to AddPac IP-PBX' message and an 'Unread Alarm Message' notification. Below this is a 'Quick Menu' and a 'Status' section with a 'User Extension' overview and a bar chart showing extension status: Registered (2), Unregistered (3), Unconfigured (0), and Unused license (95).

The 'Extensions' table lists the following data:

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

The 'Add an Extension' section provides definitions for different extension types:

- 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

Smart Multimedia Manager

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

Extensions

Directory

Routing Access Lists

Terminal Profiles

Terminals

Trunks

PBX Services

System Admin

Summary

Getting Started

Follow Us

Linked in

facebook

YouTube

Status

User Extension

Registered (2)

Unregistered (3)

Unconfigured (0)

Unused license (95)

Smart Multimedia Manager

Start Directory Extensions

Directory

Add Modify Delete Refresh

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	

Description

In this directory page, you can add / delete / modify departments of your organization. The users can be added at User Extension page. This directory is used for showing user profile and click to call at user portal web page.

Status

User Extensions

System

Trunks

Memory Storage 1% 8%

Network

Call Manager 0/100

MCU 0/2

Presence 0/100

IVR 0/100

Media 0/100

UMS 0/100

RtpProxy 0/100

FXS (1) 0 1 2 3

E1 (0) 0 1

FXO (1) 4 5 6 7

Internal Trunk Gateway (0/0)

SKN_TG (0/0)

Dacom_Trunk (0/0)

Registered (4)

Unregistered (4)

Unconfigured (0)

Unused license (92)

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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.

Extension - Routing Access List

The screenshot displays the Smart Multimedia Manager interface. The left sidebar contains a navigation menu with 'Extensions' highlighted. The main content area shows the 'Routing Access Lists' configuration page. A table lists existing lists, and a form below allows adding a new one. A yellow callout box provides a definition for a Routing Access List.

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

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

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' section of the main content area.

The main content area is divided into two sections. The top section, titled 'Terminal Profiles', includes a table with the following data:

Modify	Delete	Name	Description	Date Created
1		default		2012-06-08 17:49:40

The bottom section, titled 'Global Terminal Settings', contains various configuration 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
- Keapalive 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."

A yellow box at the bottom left contains the following text:

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 and an 'Unread Alarm Message' notification. Below this, there 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

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.

The screenshot shows the 'Smart Multimedia Manager' interface. The sidebar on the left contains navigation links: Extensions, Trunks, Trunks, Outgoing Call Rules, Incoming Call Rules, PBX Services, System Admin, and Summary. The main content area is titled 'Outgoing Call Rules' and includes a table with one rule: 'external rule' with pattern '8T' and date '2012-04-04 09:39:48'. Below the table is a form to 'Add an Outgoing Call Rule' with fields for Name, Patterns, Trunks of Outgoing Call, Called Number Translation, and Calling Number Translation. A description on the right explains that an Outgoing Call Rule controls routing to a specific trunk.

Trunk - Incoming Call Rules

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

Smart Multimedia Manager
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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
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Status Incoming Call Rules

Add an Incoming Call Rule

Add Cancel

Name *

Trunks of Incoming Call *

Internal Trunk Gateway
 SM_SIP_Provider
 ss
 jschoI_gk

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

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PBX Service - Speed Button Profiles

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.

The screenshot shows the 'Smart Multimedia Manager' interface. The left sidebar contains a navigation menu with 'PBX Services' expanded to show 'Speed Button Profiles'. The main content area displays the 'Speed Button Profile' configuration page, which includes a table of existing profiles and a form for adding a new one. The table shows one profile named 'button profile' created on 2012-04-02. The configuration form includes 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 that speed buttons can be applied to IP phones and that their maximum number is determined by the phone's OSD or physical buttons.

PBX Service - Announcement and Tones

The screenshot shows the 'Smart Multimedia Manager' web interface. The left sidebar menu has 'PBX Services' highlighted with a red dashed box, and a yellow starburst icon points to the 'Announcement and Tones' option. The main content area displays the 'Announcement and Tones' configuration page, which includes a table of announcements and a detailed view for a selected announcement.

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	최수 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	최수 초과 안내		
	430130	Connect 3	연결 중 안내		
	430140	Over Time 2	번호 입력 오류		

The detailed view for announcement ID 400110 shows the following information:

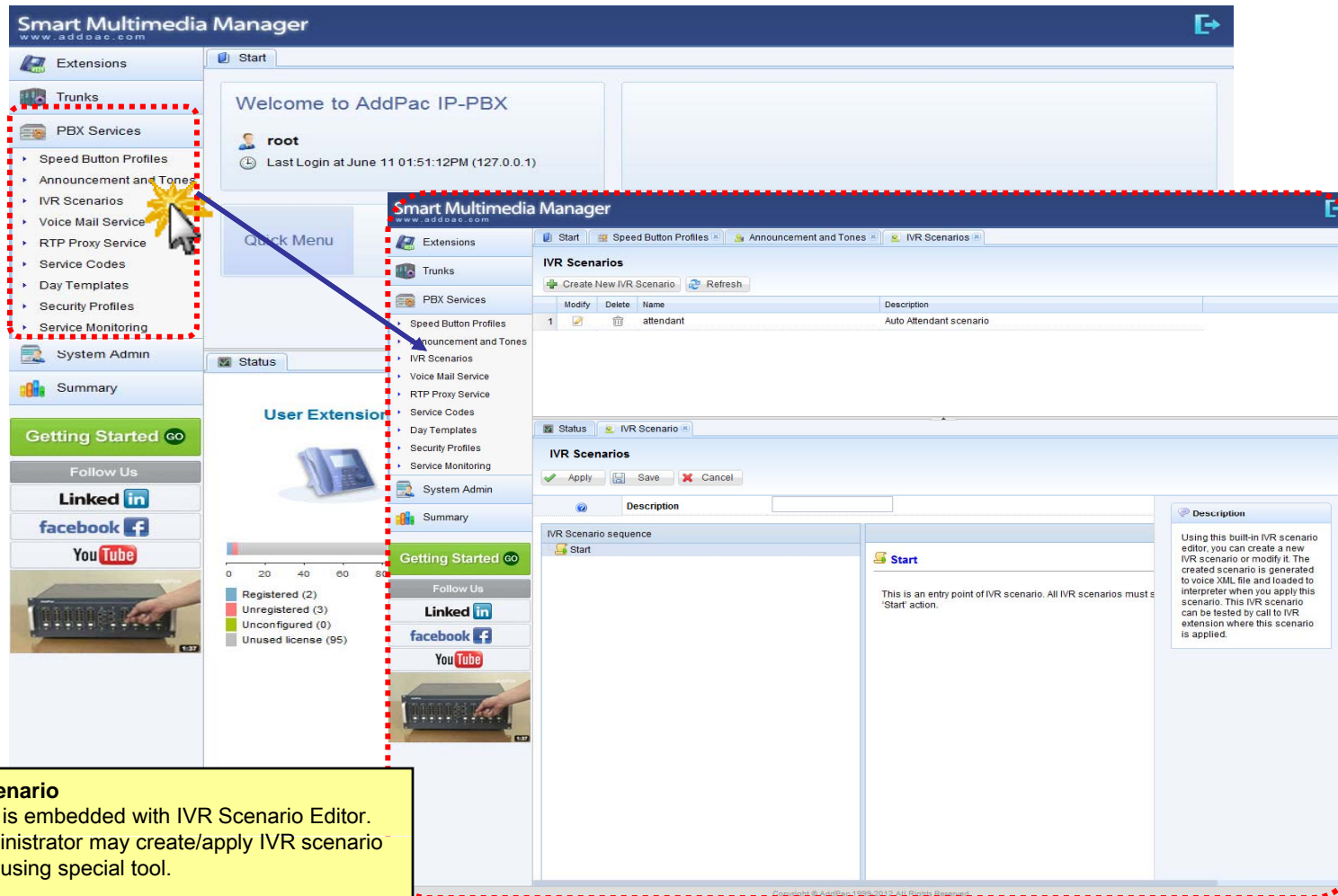
- Announcement ID:** 400110
- Description:** 연결 중 안내
- Language:** Korea
- File List:**

File name	File type	Media type	Version	Upload
400110_kr.audio.ulaw.wav	package	audio	8.50	2012
- Schedule Settings:**

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

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. The left sidebar contains a navigation menu with 'PBX Services' expanded to show 'IVR Scenarios'. A red dashed box highlights this menu item, with a yellow starburst and a blue arrow pointing to the 'IVR Scenarios' sub-section of the main interface. The main interface shows a 'Welcome to AddPac IP-PBX' message for user 'root'. Below this is a 'Quick Menu' and a 'User Extension' status section. The 'IVR Scenarios' section is active, showing a table with one scenario: 'attendant' (Auto Attendant scenario). Below the table are buttons for 'Apply', 'Save', and 'Cancel'. The 'Description' field is empty. The 'IVR Scenario sequence' section shows a 'Start' action. A 'Description' box on the right explains that this is a built-in IVR scenario editor used to create or modify scenarios, which are generated to XML files and loaded to be interpreted by the system.

Modify	Delete	Name	Description
		attendant	Auto Attendant scenario

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 expanded, and 'Voice Mail Service' is highlighted. The main content area shows the 'Voice Mail Service' configuration page, which includes fields for 'Retrieving Extension by Other Phone', 'Retrieving Extension by Owner Phone', and 'Leave Extension'. There are also sections for 'Advanced Options' (Audio Message Length, Per Extension Hdd Quota, etc.), 'Message Box Settings', 'Notification Settings', and 'SMS Settings'. A 'Description' box on the right explains the purpose of the settings.

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

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
----------------	---------	---------	------	--------	--------

*IPv6
Add Network Domain

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

Network Domain

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.

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

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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

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.

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 service codes are special digits used to activate services.

PBX Service - Day Templates

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories: Extensions, Trunks, PBX Services, System Admin, and Summary. The 'PBX Services' category is expanded, showing sub-items: Speed Button Profiles, Announcement and Tones, IVR Scenarios, Voice Mail Service, RTP Proxy Service, Service Codes, Day Templates, Security Profiles, and Service Monitoring. A red dashed box highlights the 'Day Templates' sub-item, with a blue arrow pointing to the main content area. The main content area shows the 'Day Templates' configuration page. At the top, there is a 'Welcome to AddPac IP-PBX' message for user 'root'. Below this is a 'Quick Menu' and a 'User Extension' status section. The 'Day Templates' section includes a table with columns: Modify, Delete, Template Name, Description, and Date Created. A table entry is visible with '1' in the Modify column, a trash icon in the Delete column, 'holiday' in the Template Name column, and '2012-03-30 11:24:41' in the Date Created column. Below the table, there are 'Add' and 'Cancel' buttons. The 'General Settings' section has input fields for 'Name' and 'Description'. A 'Description' tooltip is visible on the right, stating: 'Specify period or a special day(s) to apply in 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 shows the Smart Multimedia Manager interface. The left sidebar contains a navigation menu with 'PBX Services' expanded to show 'Security Profiles'. The main content area displays the 'Security Profiles' configuration page, which includes a table of existing profiles and a 'Global Security Setting' section. The 'Global Security Setting' section has a dropdown menu for 'TLS Cipher Suites' with the following options: N/A, RC4_40, RC4_128, DES_CBC, 3DES_CBC, AES_128_CBC, AES_256_CBC, SEED_CBC, and ARIA_CBC. A yellow callout box provides additional information about these cipher suites.

Modify	Delete	Name	Description	Date Created
		default	default security profile	2012-06-08 19:49:52

Global Security Setting

Apply Cancel

TLS Cipher Suites

1. N/A
2. RC4_40
3. RC4_128

DES_CBC
3DES_CBC
AES_128_CBC
AES_256_CBC
SEED_CBC
ARIA_CBC

Description

In case of SIP, below cipher suites are used to negotiate with terminal for secure TLS. The cipher suites can have preferences as below three suites.

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

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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

Start 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

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

Smart Multimedia Manager
www.addpac.com

System Admin

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

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User Extension

Registered (2)

Unregistered (3)

Unconfigured (0)

Unused license (95)

Network Services

Apply Cancel

NTP	Time zone	% Unknown command (show clock-http)
	server enable	<input type="radio"/> On <input checked="" type="radio"/> Off
	System Datetime	- : : <input type="button" value="Apply"/>
	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
	Service Port	(default:161)
SNMP	Community	
	Trap Service IP Address	
	Trap Community	
	Service Enable	<input checked="" type="radio"/> On <input type="radio"/> Off
HTTP	Service Port	80 (default:80)
	Authentication	<input type="radio"/> NONE <input checked="" type="radio"/> Basic <input type="radio"/> Digest
	Service Enable	<input checked="" type="radio"/> On <input type="radio"/> Off
FTP	Control Port	21 (default:21)
	Data Port	20 (default:20)
LDAP	Server Port	389 (default:389)
	Service Enable	<input type="radio"/> On <input checked="" type="radio"/> Off
SYSLOG	Service Port	(default:514)
	Log Life Time	(1 ~ 300 Day)

Description

You can change properties of system network services such as TELNET, SNMP, HTTP, FTP, LDAP, SYSLOG, and so on.

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 item, which is highlighted with a red dashed box. A blue arrow points from this menu item to the 'Administrators' page. The 'Administrators' page displays a table of administrators and a form for creating or editing an administrator.

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

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 left sidebar contains a navigation menu with 'System Admin' selected. The main content area displays the 'Licenses' page, which includes a table of license settings and a description box.

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

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User Extension

Registered (2)

Unregistered (3)

Unconfigured (0)

Unused license (95)

Smart Multimedia Manager

Extensions Trunks PBX Services System Admin

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

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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

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

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	✓

Extension Analog

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

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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

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

Status

User Extension

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

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Smart Multimedia Manager
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Start Security Profiles Alarm History

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

Status

User Extensions

Registered (4)
Unregistered (4)
Unconfigured (0)
Unused license (92)

Getting Started

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System

Memory 1%
Storage 7%
Network

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

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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

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

System Admin - Show Command

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
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- Voice Lines
- Alarm History
- Call History
- Show Command
- Summary

Show Command

Categories

- System
- VoIP
- Call Manager
- Command Line Interface

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)



Thank you!

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