

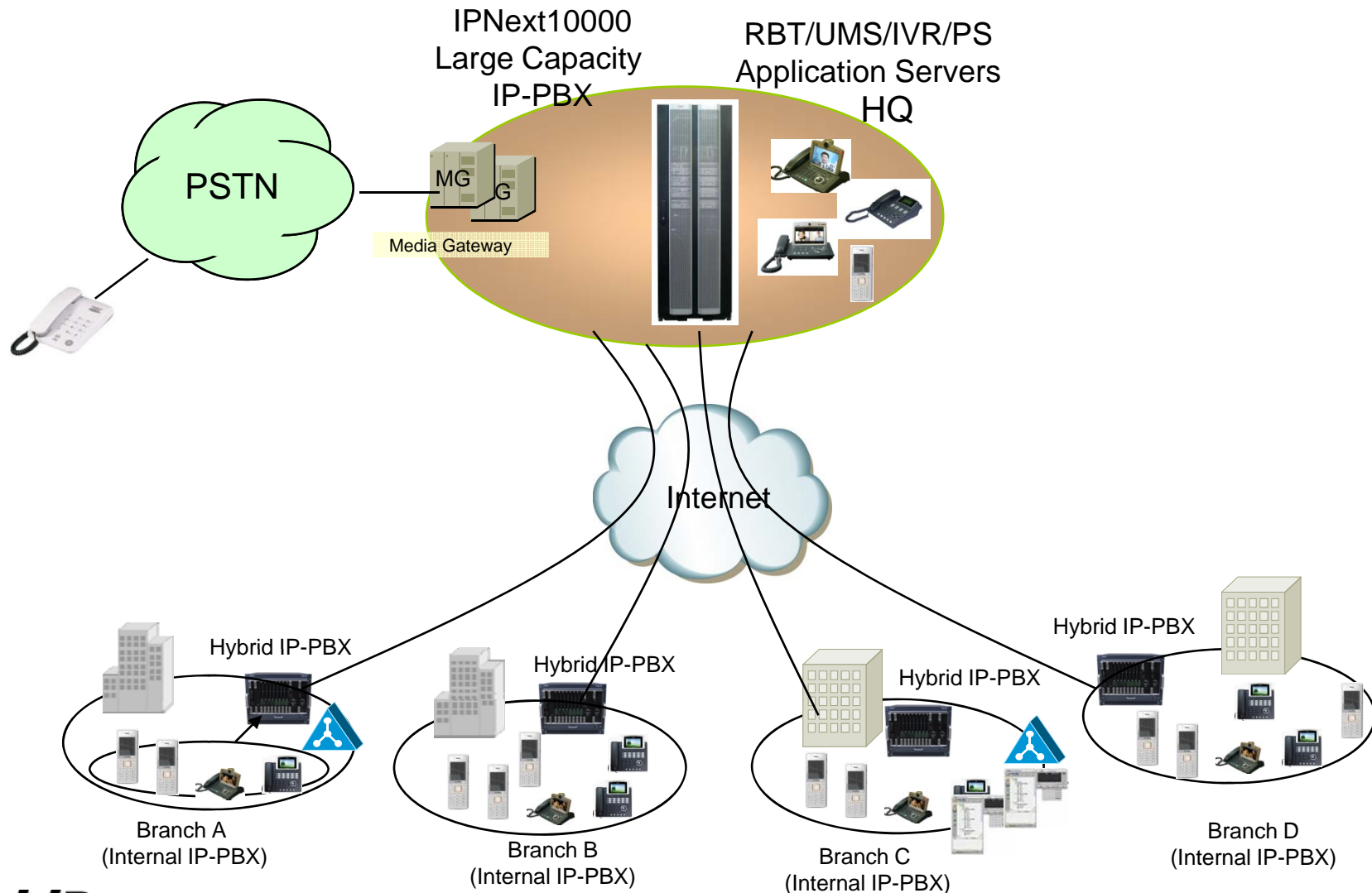
# Large Capacity Hybrid IP-PBX Solution





# Contents

- Large Capacity Hybrid IP-PBX Service Diagram
- Large Capacity Hybrid IP-PBX Comparison Table
- Large Capacity Hybrid IP-PBX Series
  - IPNext350 Hybrid IP-PBX
  - IPNext320 Hybrid IP-PBX
- AddPac IP Terminals, Soft Phone, Smart Messenger
- IP Video Phones for IPNext IP-PBX
- IP Phones for IPNext IP-PBX
- WSMM (Web based Smart Multimedia Manager) for IPNext IP-PBX

# Large Capacity Hybrid IP-PBX Service Diagram



# Large Capacity Hybrid IP-PBX Comparison Table

Model		IPNext350	IPNext320
			
Service Features			
Registration User Number		500	350
Concurrent Call User Number		150	100
IPv4/IPv6 Dual Stack Support		Support	Support
VoIP Signaling	Internal	SIP	SIP
	External	H.323/SIP	H.323/SIP
Powerful IVR, UMS, Media Service, User Presence Service		Support	Support
RTP Proxy Service(IPv6, Private IP)		Support	Support
LAN Port (Gigabit)		2	2
VoIP Module Slots for PSTN		8 Slots x (32-Port Module) = 256 Ports	4 Slots x (32-Port Module) = 128 Port
VoIP Interface		FXS, FXO, etc	FXS, FXO, etc



# IPNext350 Hybrid IP-PBX

# Main Features

## IPNext350 Hybrid IP-PBX System

- IP based Advanced Hybrid IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Medium Office
- PSTN Interface (FXO, FXS, etc) Support
- Up to 256 Port Analog Channel Support
- 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, 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 Terminal Support

# Hardware Specification

IPNext350 Hybrid IP-PBX System

RISC  
CPU

High-end  
DSP

- RISC Microprocessor Computing Power
- Main Chassis
  - Network Interface
    - Two(2) 10/100/1000Mbps Gigabit Ethernet
    - One(1) RS-232C Console (RJ45)
  - Eight(8) VoIP Module Slots for FXS, FXO, etc
  - Hot-Swap VoIP Module
  - Module Type Dual Power Supply for Redundancy

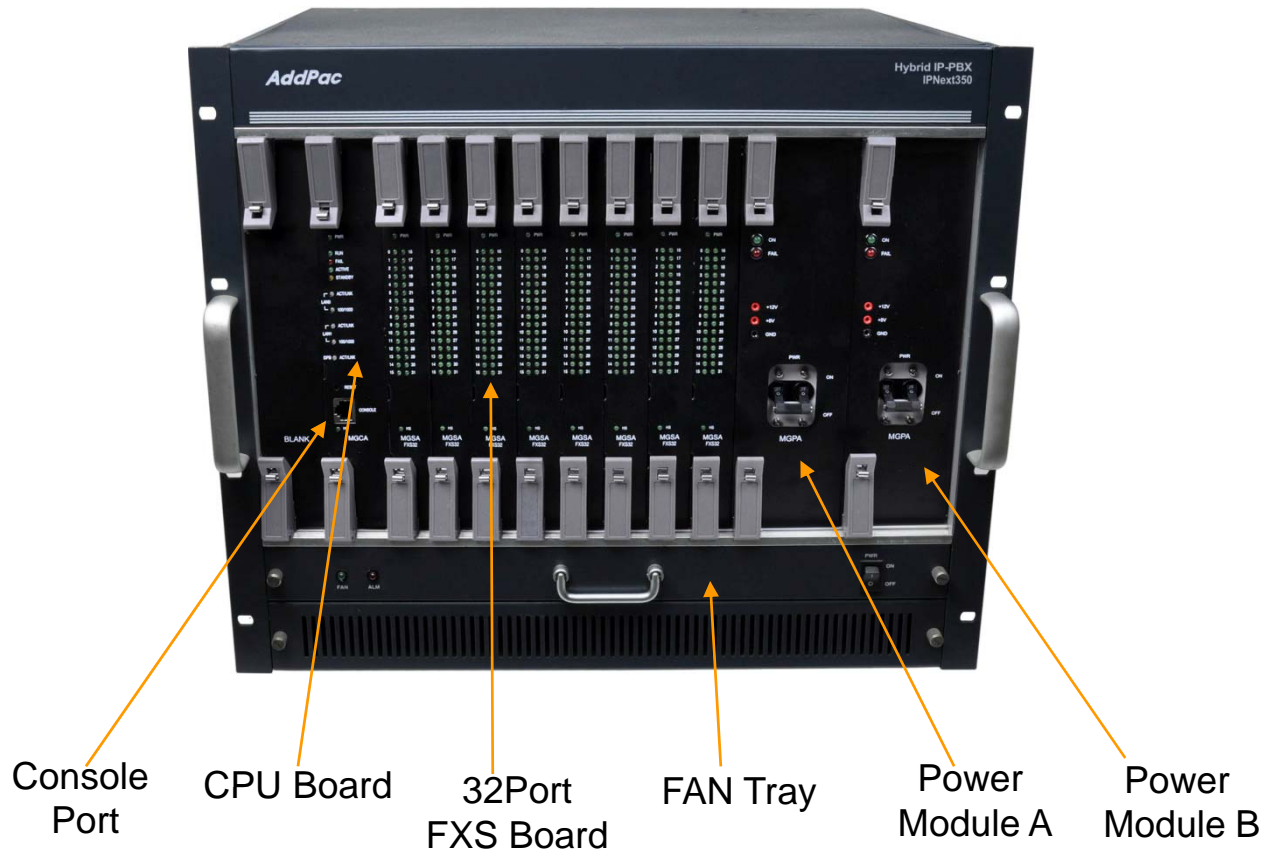
# Hardware Specification

IPNext350 Hybrid IP-PBX System

RISC  
CPU

DSP

Front Side





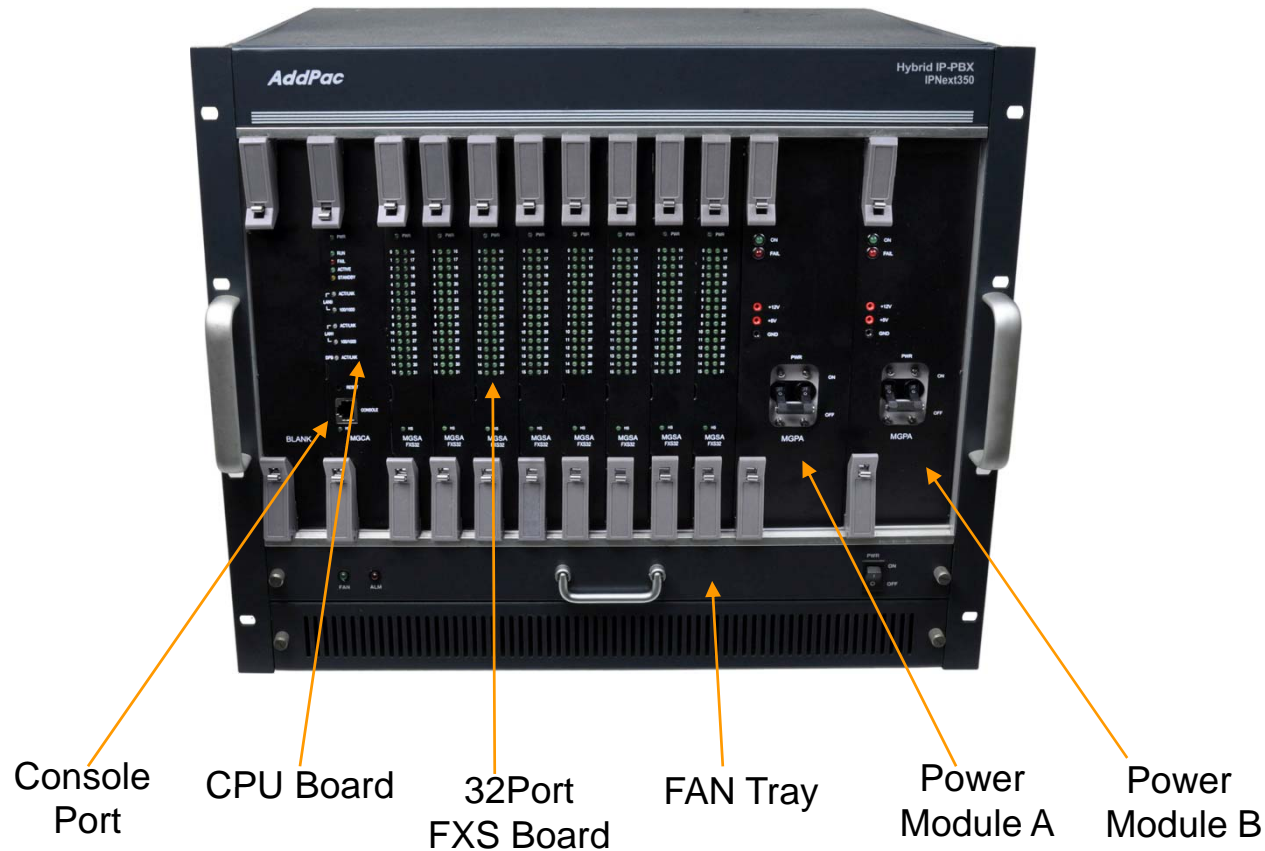
# Hardware Specification

IPNext350 Hybrid IP-PBX System

RISC  
CPU

DSP

## Front Side (MGSA-32FXS-N1 Module)



# Hardware Specification

IPNext350 Hybrid IP-PBX System

RISC  
CPU

DSP

## Back Side



AC Power  
Input

32-Port Analog Interface  
Champ Connector

Gigabit Ethernet  
Interface

# Hardware Specification

IPNext350 Hybrid IP-PBX System

RISC  
CPU

DSP

## MGCA

Media Gateway Control Board Assembly  
: CPU Board



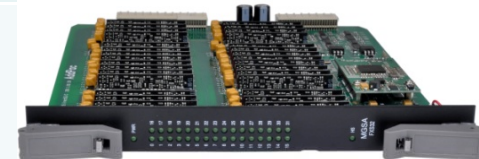
## MGSA-32FXS

Media Gateway Subscriber Board Assembly  
: 32-Port FXS for Long Distance



## MGSA-32FXS-N1

Media Gateway Subscriber Board Assembly  
: 32-Port FXS for Small & Medium Distance



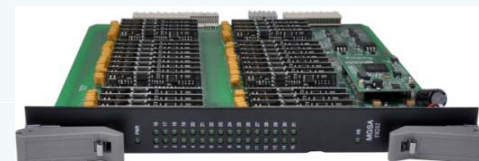
## MGSA-32FXS-N2

Media Gateway Subscriber Board Assembly  
: 32-Port FXS for Small & Medium Distance



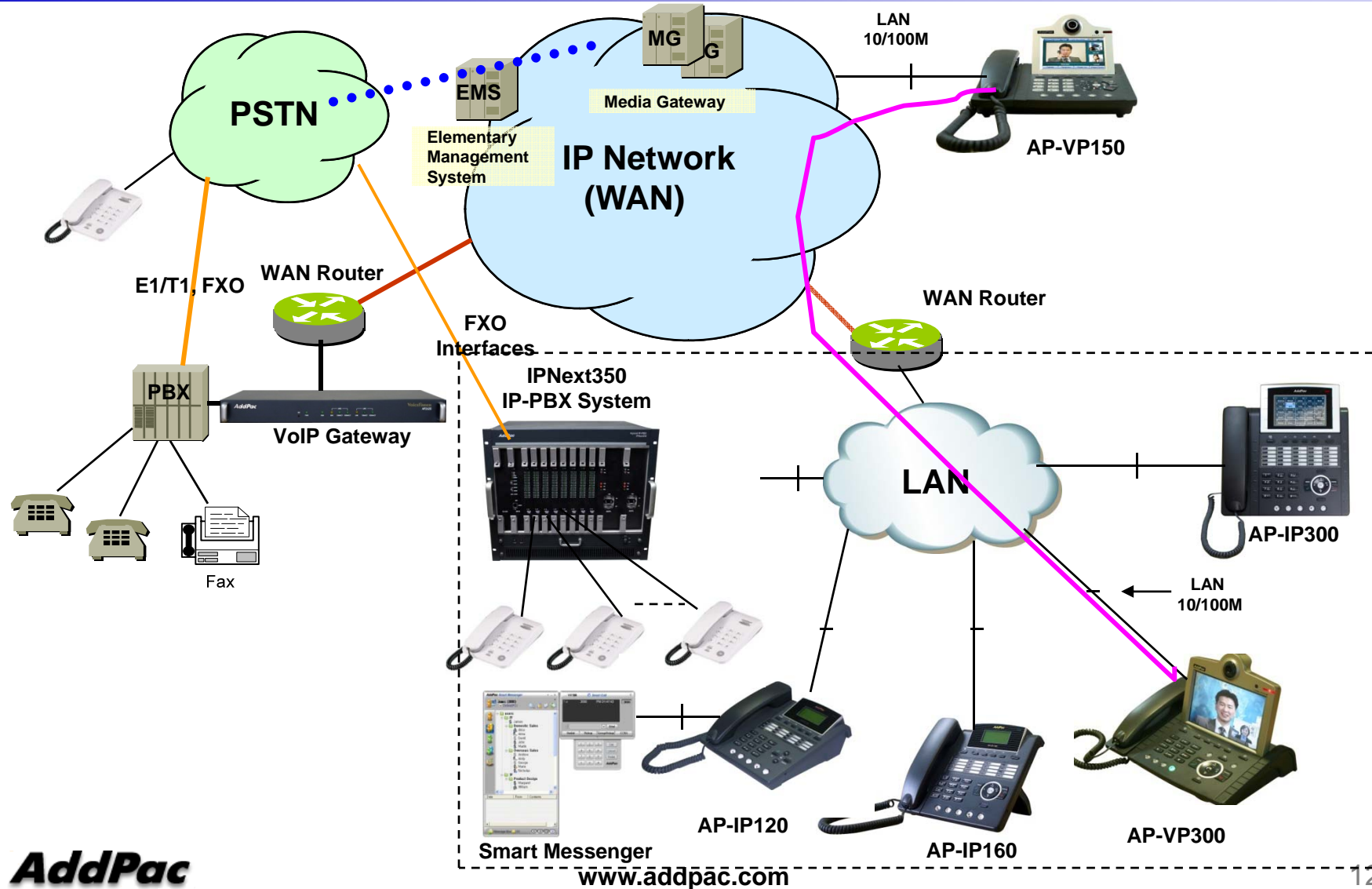
## MGSA-32FXO

Media Gateway Subscriber Board Assembly  
: 32-Port FXO



# Standard IP-PBX Application

IPNext350 Hybrid IP-PBX System





# IPNext320 Hybrid IP-PBX

# Main Features

## IPNext320 Hybrid IP-PBX System

- IP based Advanced Hybrid IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Medium Office
- PSTN Interface (FXO, FXS, etc) Support
- Up to 128 Port Analog Channel Support
- 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, 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 Terminal Support

# Hardware Specification

IPNext320 Hybrid IP-PBX System

RISC  
CPU

High-end  
DSP

- RISC Microprocessor Computing Power
- Main Chassis
  - Network Interface
    - Two(2) 10/100/1000Mbps Gigabit Ethernet
    - One(1) RS-232C Console (RJ45)
  - Four(4) VoIP Module Slots for FXS, FXO, etc
  - Hot-Swap VoIP Module
  - Module Type Dual Power Supply for Redundancy

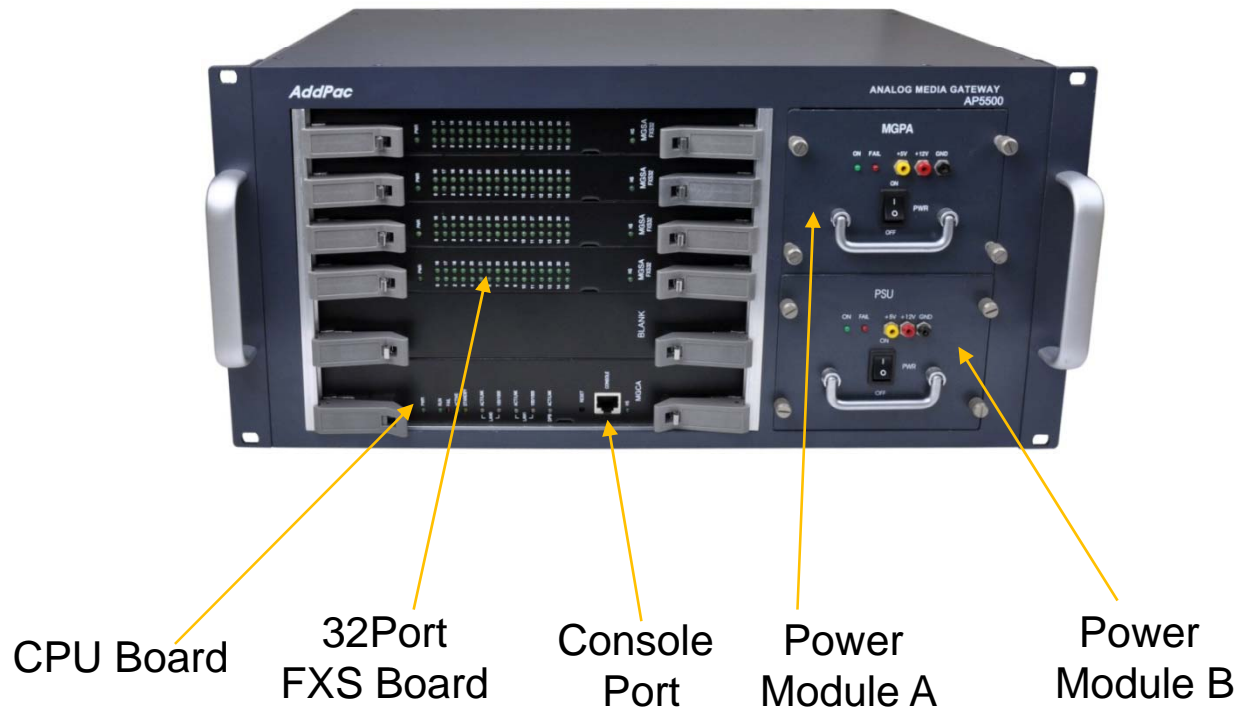
# Hardware Specification

IPNext320 Hybrid IP-PBX System

RISC  
CPU

DSP

Front Side





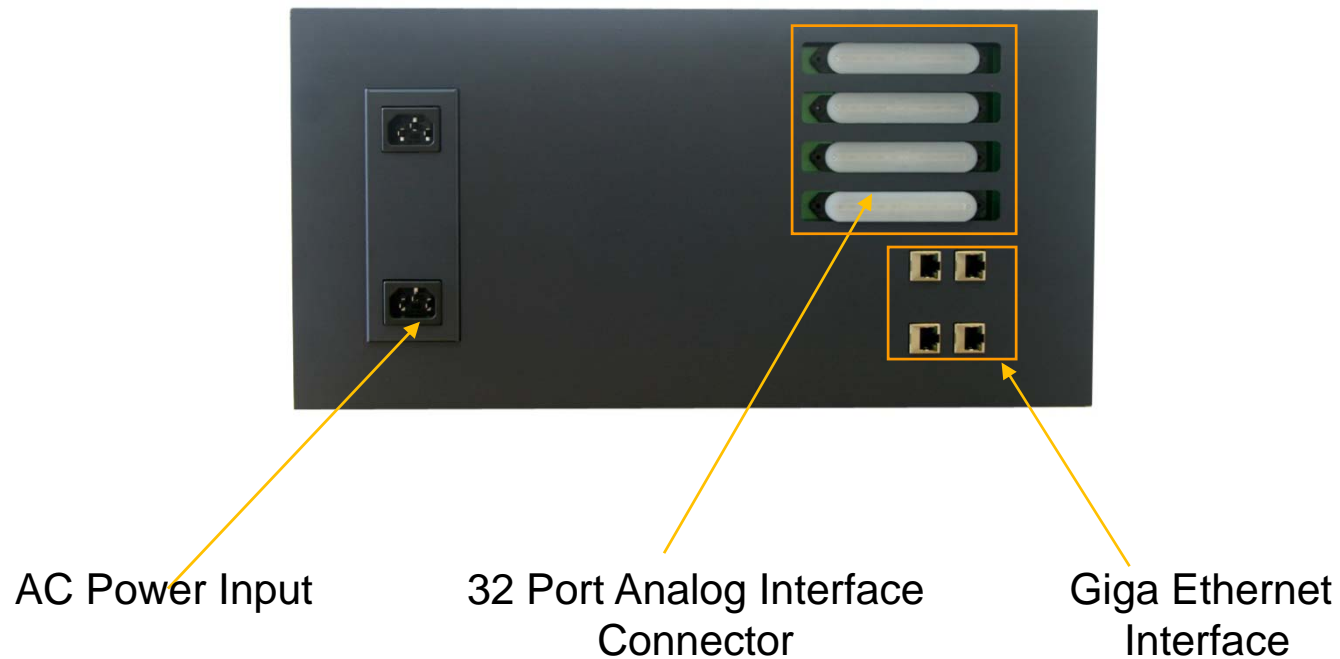
# Hardware Specification

IPNext320 Hybrid IP-PBX System

RISC  
CPU

DSP

## Back Side



# Hardware Specification

IPNext320 Hybrid IP-PBX System

RISC  
CPU

DSP

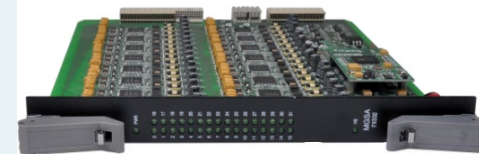
## MGCA

Media Gateway Control Board Assembly  
: CPU Board



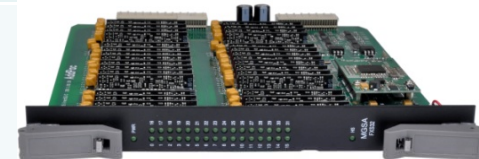
## MGSA-32FXS

Media Gateway Subscriber Board Assembly  
: 32-Port FXS for Long Distance



## MGSA-32FXS-N1

Media Gateway Subscriber Board Assembly  
: 32-Port FXS for Small & Medium Distance



## MGSA-32FXS-N2

Media Gateway Subscriber Board Assembly  
: 32-Port FXS for Small & Medium Distance



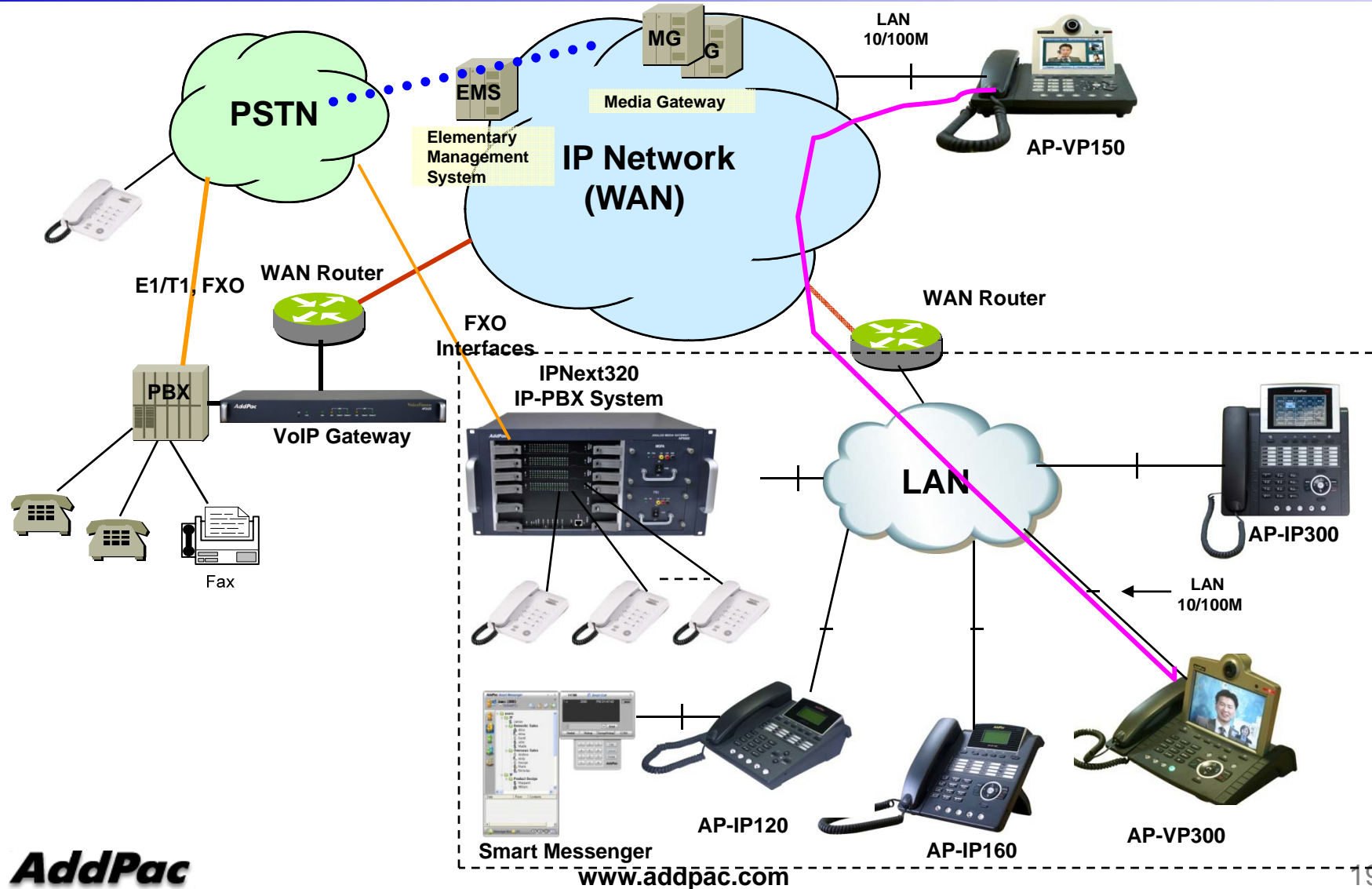
## MGSA-32FXO

Media Gateway Subscriber Board Assembly  
: 32-Port FXO



# Standard IP-PBX Application

IPNext320 Hybrid IP-PBX System



**AddPac**

[www.addpac.com](http://www.addpac.com)



# 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

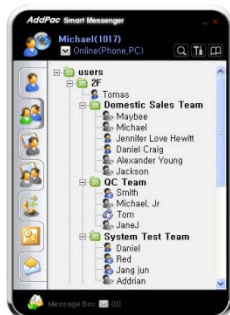


AP-IP120

# 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



Smart Messenger



AP-VP280



IPNext350 IP-PBX

[www.addpac.com](http://www.addpac.com)



# Smart Window(1/2)

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

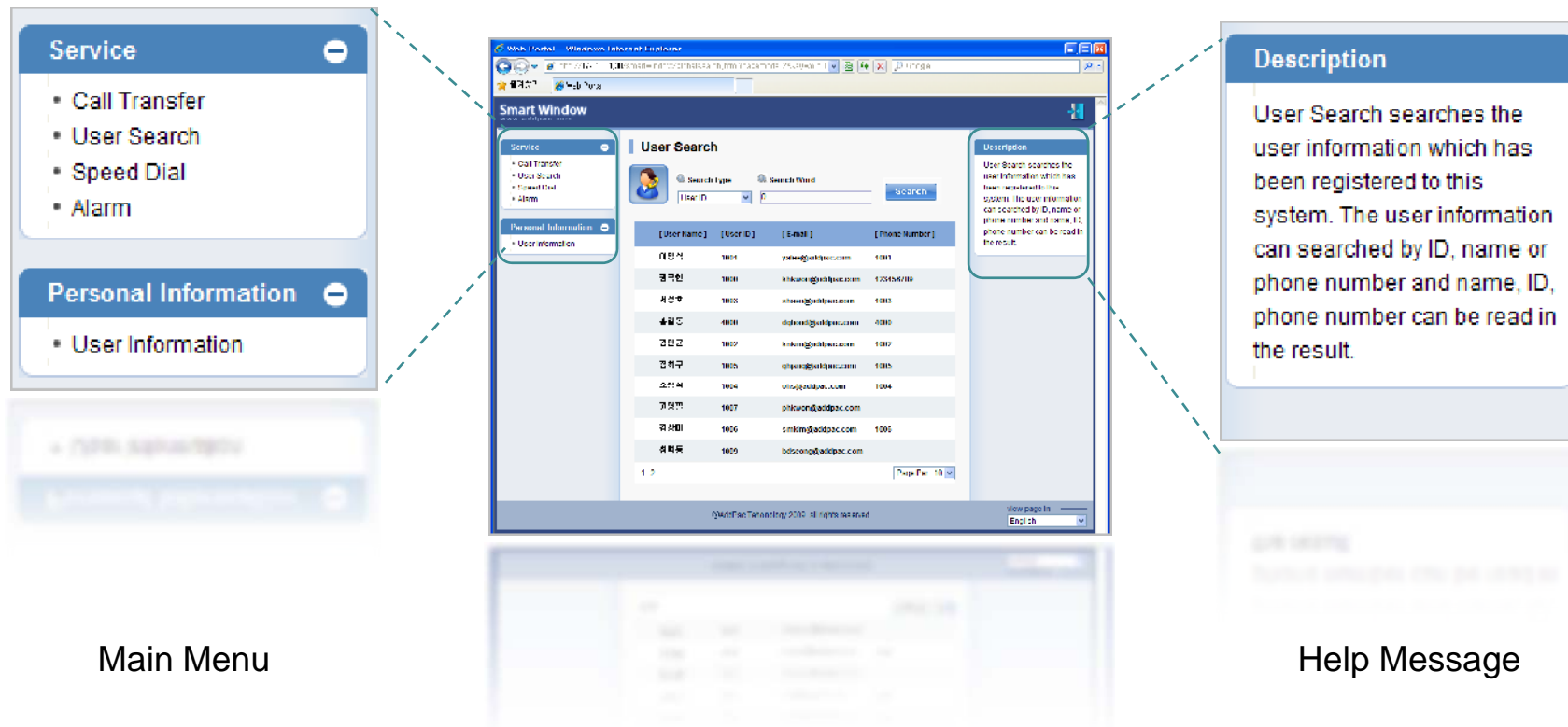
## **Service Features**

- 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.





# IP Video Phones for IPNext IP-PBX



# IP Video Phones for IPNext 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 IPNext IP-PBX

# IP Phones for IPNext 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



# Web Smart Multimedia Manager (WSMM)

# 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 3<sup>rd</sup> 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, Routing Access List, Terminal Profile, Security Profile, Use RTP Proxy, Generate Ring Back Tone at PBX, and Presentation. A red dashed box highlights a help icon in the left sidebar, with an arrow pointing to a help window titled "Help :: User Extension". The help window contains the following text:

**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.

## 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**  
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 **Succeeded**

**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
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
```

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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

**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

**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**

**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)

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

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

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# 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 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, showing a list of system events. A yellow callout box provides instructions on how to access the alarm history from the main page.

**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. A 'Quick Menu' section is highlighted with a red dashed box, containing links for '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', and 'Terminals'. A blue arrow points from the 'Add an User Extension' link to a detailed form titled 'Add an User Extension'. This form includes fields for Extension (3-8 digits), First Name, Last Name, Voice Mail Password (4 digits), User Password (for SIP), Department, Title (e.g., manager), Email (e.g., admin@addpac.com), and Home Phone (e.g., 123-456-7890). 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' 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 and a video post. 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

**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 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	

**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 is titled 'Smart Multimedia Manager' and shows a 'Welcome to AddPac IP-PBX' message. 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 'Extensions' table lists various extension types and their details.

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 | Directory | Extensions

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)

**Directory**

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	

**User Extension**

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

**System**

Memory Storage: 1%  
MCU: 8%  
Network: [Icon]

Call Manager: 0/100  
IVR: 0/100  
Media: 0/100  
UNIS: 0/100  
RtpProxy: 0/100

Trunks: Internal Trunk Gateway (0/0), SKN\_TG (0/0), Dacom\_Trunk (0/0)

**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

**Smart Multimedia Manager**  
www.addpac.com

Start | Directory | Routing Access Lists

### Routing Access Lists

+ Add a Routing Access List Refresh

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

Status | Routing Access List

#### 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	

Allowed Outgoing Call Rules

Description

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

Related Links

- Outgoing Call Rules

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**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' tab in the main content area.

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

Setting	Value
Calling Party Presentation	allowed
Language	Korean
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
Keepalive Timeout	30 (10~86400sec)

A 'Description' box on the right side of the settings area 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 left of the screenshot, a yellow box 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

**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

	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**

Extension \*  3 ~ 8 digits

First Name

Last Name \*

Voice Mail Password \*  4digits and user portal login

User Password \*  For SIP registration

Department  Search

Title

Email

Home Phone

Mobile Phone

User 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



# Trunk - Trunks

**Smart Multimedia Manager**  
www.addpac.com

Welcome to AddPac IP-PBX  
root  
Last Login at June 11 04:38:52AM (172.16.1.50)

Unread Alarm Message  
No Unread Alarm Message

**Trunks**

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**

Cancel

**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.

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

**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

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

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

**Smart Multimedia Manager**  
www.addpac.com

Start alarm Trunks Outgoing Call Rules

Outgoing Call Rules

Add an Outgoing Call Rule Refresh

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

Add an Outgoing Call Rule

Add Cancel Advanced Options

Name \*

Patterns \*

Trunks of Outgoing Call \*

Called Number Translation Add Rule

Number Translation

Calling Number Translation Add Rule

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

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## 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

**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

**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

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:

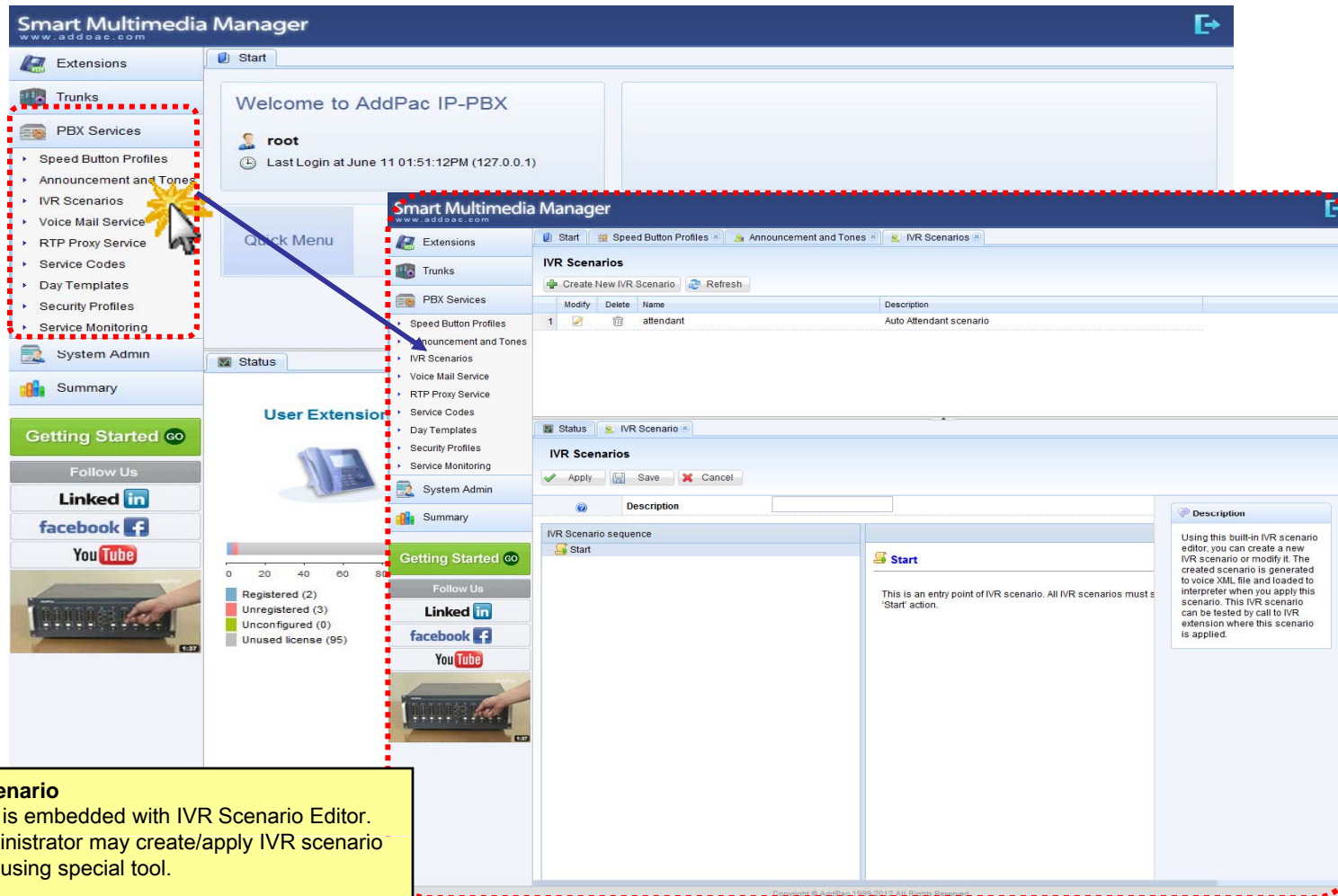
- Announcement ID:** 400110
- Description:** 연결 중 안내
- Language:** Korea
- File List:**

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 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 'IVR Scenarios' menu item, with a yellow starburst icon and a blue arrow pointing to the main content area. The main content area is titled 'Smart Multimedia Manager' and shows a 'Welcome to AddPac IP-PBX' message for user 'root'. Below this is a 'Quick Menu' and a 'User Extension' section with a status bar showing: Registered (2), Unregistered (3), Unconfigured (0), and Unused license (95). The 'IVR Scenarios' section is active, displaying a table with one entry: 'attendant' with description 'Auto Attendant scenario'. Below the table are buttons for 'Apply', 'Save', and 'Cancel'. The 'IVR Scenarios' editor is open, showing a 'Start' action in the sequence. 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 voice XML files and loaded to be interpreted by the system.

**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 'PBX Services' highlighted. A red dashed box encloses the 'PBX Services' menu and the 'Voice Mail Service' configuration page. A yellow starburst icon is positioned over the 'Voice Mail Service' menu item. The main content area shows the 'Voice Mail Service' configuration page with various settings such as 'Retrieving Extension by Other Phone', 'Audio Message Length', 'Per Extension Hdd Quota', and 'SMS Settings'. A 'Description' box on the right explains the purpose of the settings.

**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

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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  
Voice Mail Service

Apply Cancel Advanced Options

Voice Mail Service

Voice Mail Extensions

- Retrieving Extension by Other Phone: 0001
- Retrieving Extension by Owner Phone: 0002
- Leave Extension: 0003

Advanced Options

Audio Message Length: 2 - 58 seconds

Per Extension Hdd Quota: 31 MB

Over HDD Quota:  Delete Old Message  Block New Message

Use Account Blocking:

Password Fail Count: seconds

Enable E-mail Notification:   
For E-Mail notification, SMTP server setting should be set by click here.

Attach File to Email:

Delete File After Email Notification:

Enable SMS Notification:

SMS Settings

SIP Port: 5062 (default: 5062)

Description

Manage Voice Mail service properties. Set voice mail service extensions, message box settings and voice mail notification settings.

User Extension

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

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**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  
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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)

**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.

**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)

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# 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. The left sidebar contains a 'PBX Services' menu with 'Service Codes' highlighted. The main content area shows the 'Service Codes' configuration page with various settings for call forwarding, call reject, and call waiting. A 'Description' box on the right explains that service codes are special digits used to activate PBX services.

Service Code	Digit 1	Digit 2
Call Park	*	9
Call Pickup	*	*
Call Forwarding All Register	*	3 2
Call Forwarding All Activation	*	3 4
Call Forwarding All Deactivation	*	3 5
Call Reject(Absence) Activation	*	1 1
Call Reject(Do Not Disturb) Activation	*	1 2
Call Reject Deactivation	*	1 0
Call Waiting Activation	*	2 0
Call Waiting Deactivation	*	2 1
Call Forwarding All to Voicemail Register	*	5 1
Call Forwarding Busy Register	*	3 2
Call Forwarding NoAnswer Register	*	3 3
Call Forwarding NotReachable Register	*	6 1
Call Forwarding Cancel	*	3 0
Call Forwarding Busy Activation	*	3 6
Call Forwarding Busy Deactivation	*	3 7
Call Forwarding NoAnswer Activation	*	3 8
Call Forwarding NoAnswer Deactivation	*	3 9
Call Forwarding NotReachable Activation	*	6 4
Call Forwarding NotReachable Deactivation	*	6 5
CCBS Register	*	4 0
CCBS Cancel	*	4 1
IVR Scenario Speed Selection Enable	*	7 7

# PBX Service - Day Templates

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with 'PBX Services' expanded to show 'Day Templates'. The main content area shows the 'Day Templates' configuration page, which includes a table of existing templates and a form for adding a new one.

Modify	Delete	Template Name	Description	Date Created
		holiday		2012-03-30 11:24:41

**Day Templates**

General Settings

Name	<input type="text"/>
Description	<input type="text"/>

Description: 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 web interface. The left navigation menu has 'PBX Services' expanded to show 'Security Profiles'. The main content area displays the 'Security Profiles' configuration page, which includes a table with one profile named 'default'. Below this is the 'Global Security Setting' section, which has a dropdown menu for 'TLS Cipher Suites' showing options like 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 at the bottom left explains that IP-PBX supports TLS Cipher Suites and lists the available options.

**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

**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.

The screenshot shows the 'Service Monitoring' page with the following details:

- Service Monitoring Settings:** Interval: 10 sec.
- Active Calls Table:**

ID	Established Time	Duration	Calling Number	Called Number	Audio Codec	Video Codec	Recording	Drop Call
- System Status Dashboard:**
  - User Extensions:** 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).

# System Admin - Network Interface

**Smart Multimedia Manager**  
www.addpac.com

Extensions  
Trunks  
PBX Services  
System Admin

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

Summary

Getting Started GO

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User Extension

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

Smart Multimedia Manager

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

IPv6 Default Gateway

LAN Interface

Interface Mode:  None  Bridge  IP Shared  NAT  Static IP

IP Address

Subnet Mask

DHCP Server  On  Off

DHCP Range  ~  A.B.C.D

IPv6 Address

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.

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**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

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

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User Extension

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

Getting Started GO

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**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	- : : (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
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)
	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 System Admin interface. The left sidebar contains a navigation menu with the following items: Extensions, Trunks, PBX Services, System Admin (highlighted with a red dashed box), Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, Show Command, and Summary. A mouse cursor is pointing at the 'Administrators' link. The main content area displays the 'Administrators' page, which includes 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

The form for creating or editing an administrator includes the following fields:

- User name\*
- Description
- ID\*
- Password\*
- Level (Administrator)
- Application Permission (Door Access Control Manager, Time and Attendance Manager)

**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

**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

**Licenses**

Upload License Download License Cancel

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

**Description**  
Manage licenses for Call Manager, MCU, Presence, Media, Voice Mail, IVR, RTP Proxy services.

**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)

Getting Started GO

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Smart Multimedia Manager

www.addpac.com

Extensions Trunks PBX Services System Admin

IVR Scenarios 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

**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 Resources: 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)

**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

**Smart Multimedia Manager**  
www.addpac.com

Start | Call History

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)

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Registered (0)  
Unregistered (6)  
Unconfigured (0)  
Unused license (94)

Smart Multimedia Manager

www.addpac.com

Start | Call History

Call History

Trunk Call Type: NIA | Period: 2012-06-01 ~ 2012-06-08 | Search Number: | Search | Refresh

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

Calling Number	Called Number	Established Time	duration (sec)	Call State	State Cause	datetime
No data to display						

Page 1 of 1

Status

User Extensions

System

Trunks

Memory 0%  
Storage 8%  
Network

Call Manager 0/100  
MCU 0/2  
Presence 0/100  
IVR 0/100  
Media 0/100  
UMS 0/100  
RtpProxy 0/100

Internal Trunk Gateway (0/0)  
ss (0/0)  
SM\_SIP\_Provider (0/0)  
Jschoi\_gk (0/0)

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## 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**  
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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
- Summary

**Show 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)



# Thank you!

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