

# Simple SIP Voice Paging Service for IP Emergency Call Phone

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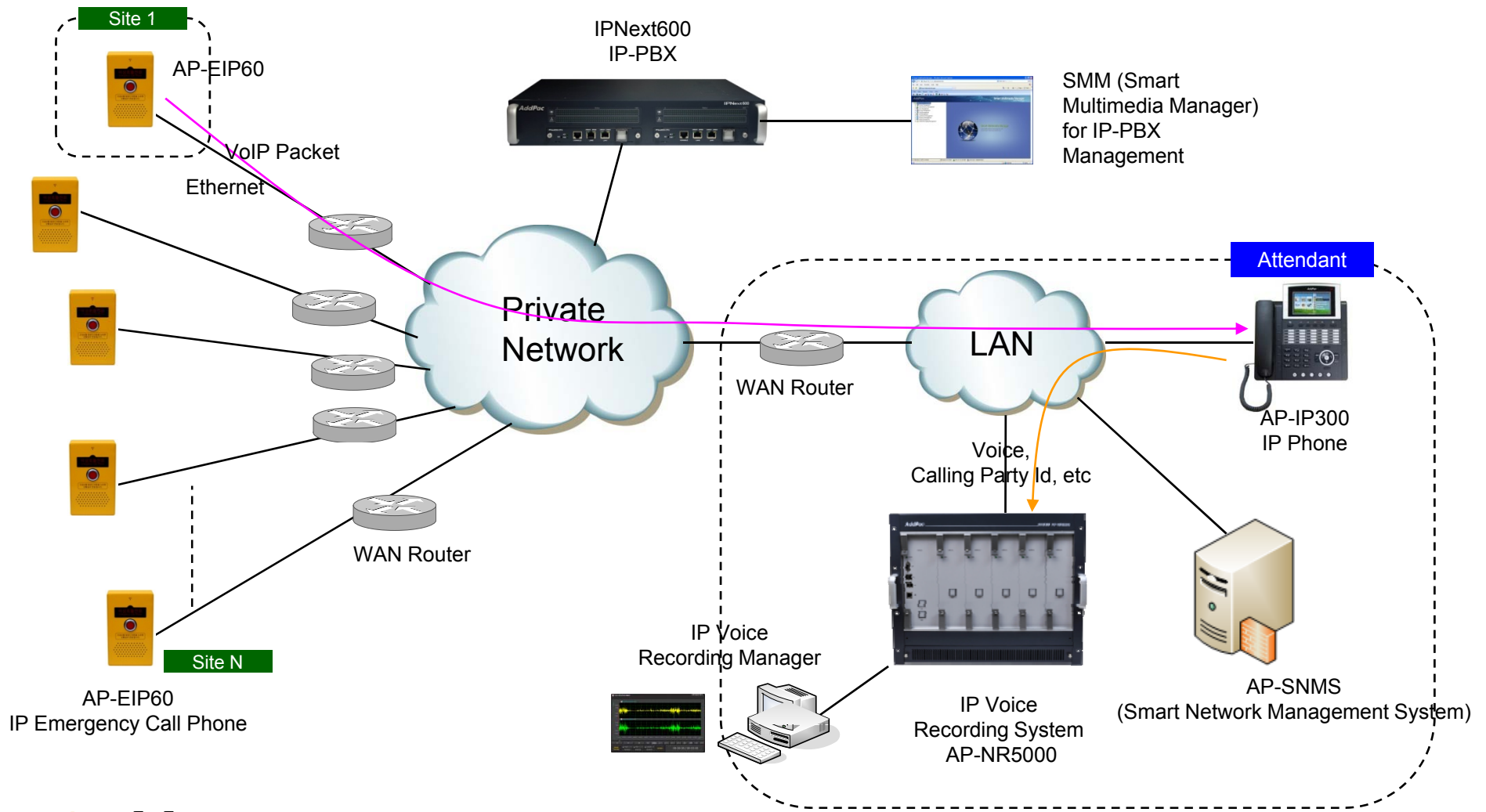
Sales and Marketing

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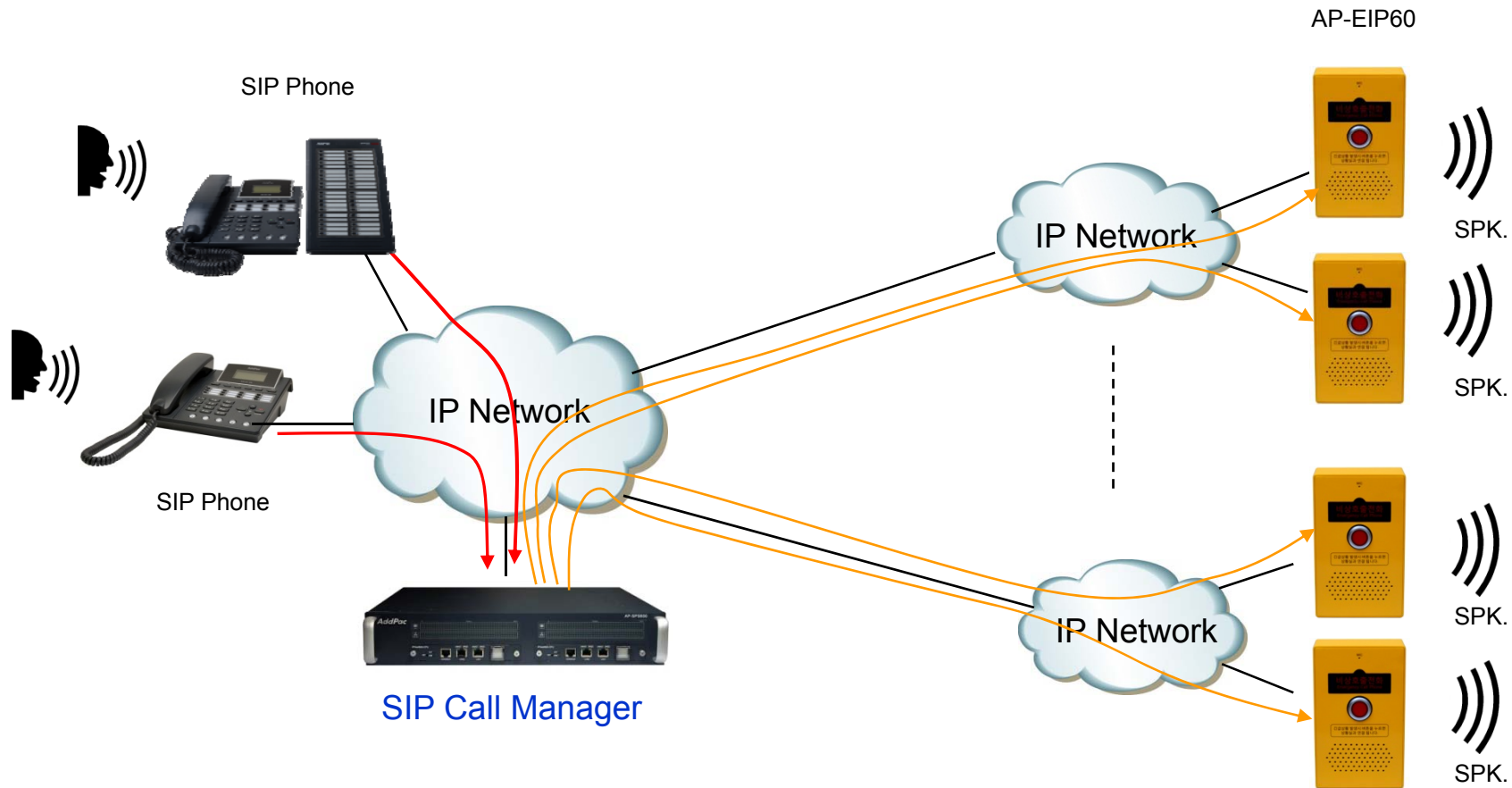
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  - SIP Emergency Call Phone: AP-EIP70, AP-EIP60
  - SIP Phones for Paging Service : AP-IP300, AP-IP120
  - Web based Smart Multimedia Manager for SIP Call Manager

# SIP Emergency Call Service Network Diagram



# SIP Paging Call Service Network Diagram



# SIP Paging Product Solution

Web Smart Multimedia Manager for Call Manager	SIP Call Manager	IP Emergency Call Phone	Broadcasting Router (Relay Server)	SIP Phone for Paging Service
	 <p>IPNext600</p>	 <p>AP-EIP70</p>	 <p>AP-BCR5000</p>	 <p>AP-IP300 + AP-PT20x2</p>
	 <p>IPNext180</p>	 <p>AP-EIP60</p>		 <p>AP-IP120 + AP-PT20x2</p>



# SIP Call Manager based Simple Paging Service Overview

# Contents

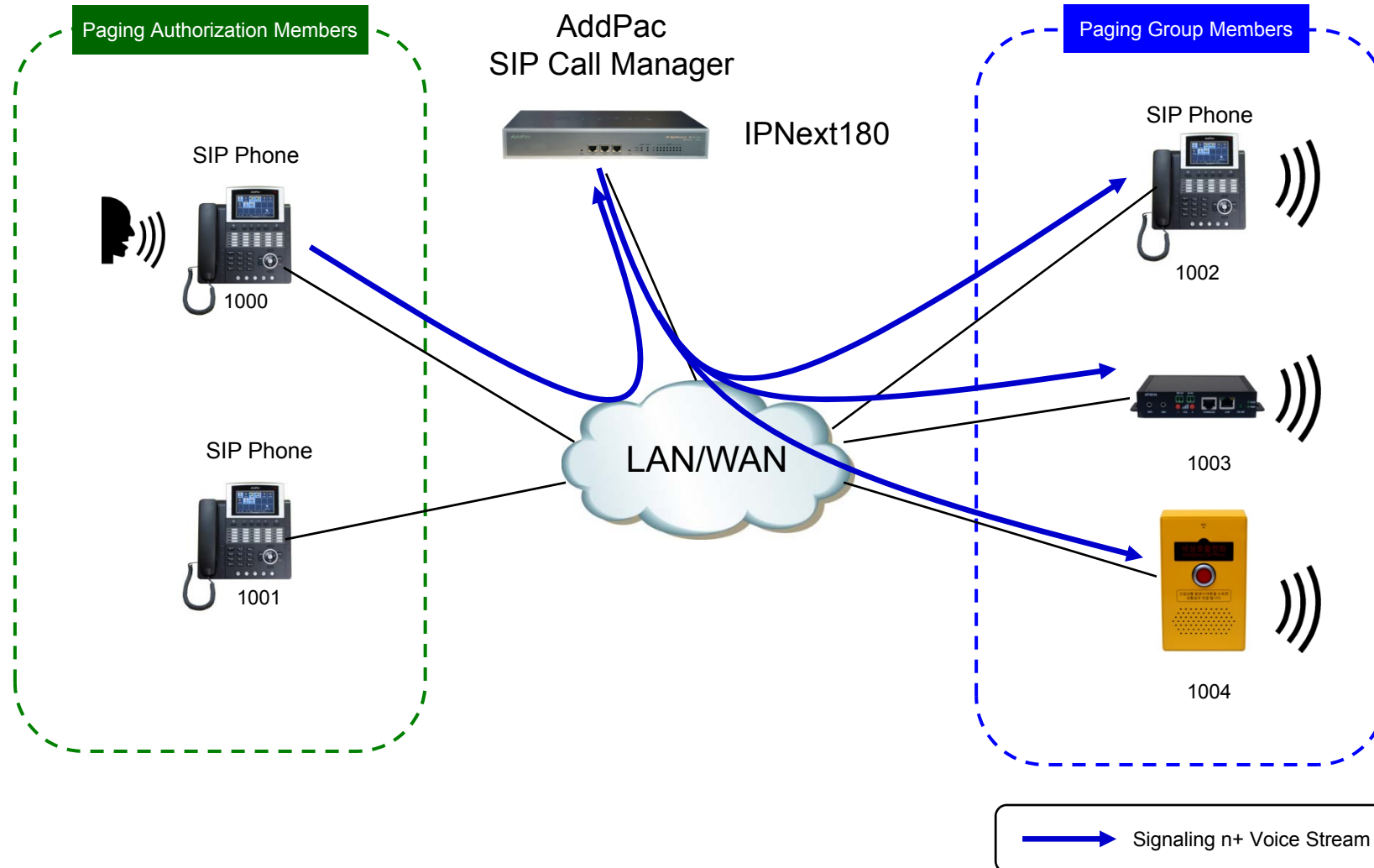
- Paging Service Features
- Paging Service Model
  - Small Scale
  - Large Scale
- Paging Group Signal Flow
- Extension- Paging Group
- Paging Group Configuration

# Paging Service Features

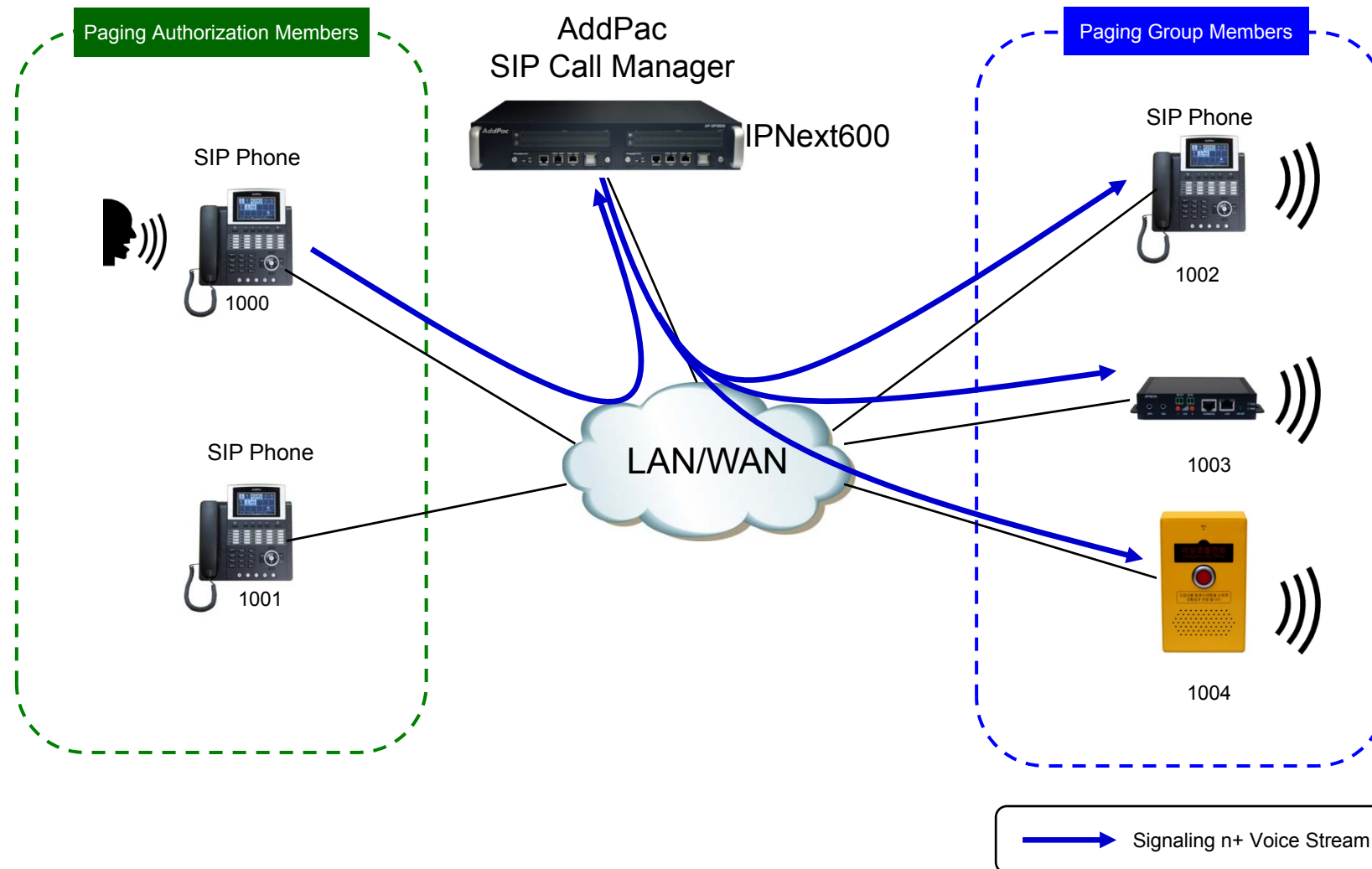
- Voice broadcasting service over IP
- SIP Answer-Mode(RFC5373) call signaling
- Unicast and Multicast Broadcasting Scheme
- Support Pre-built audio media broadcasting and scheduling
- Supported Audio Codec : G.711( $\mu$ -law, A-law)
- External Broadcasting Server for Large Capacity (Option)



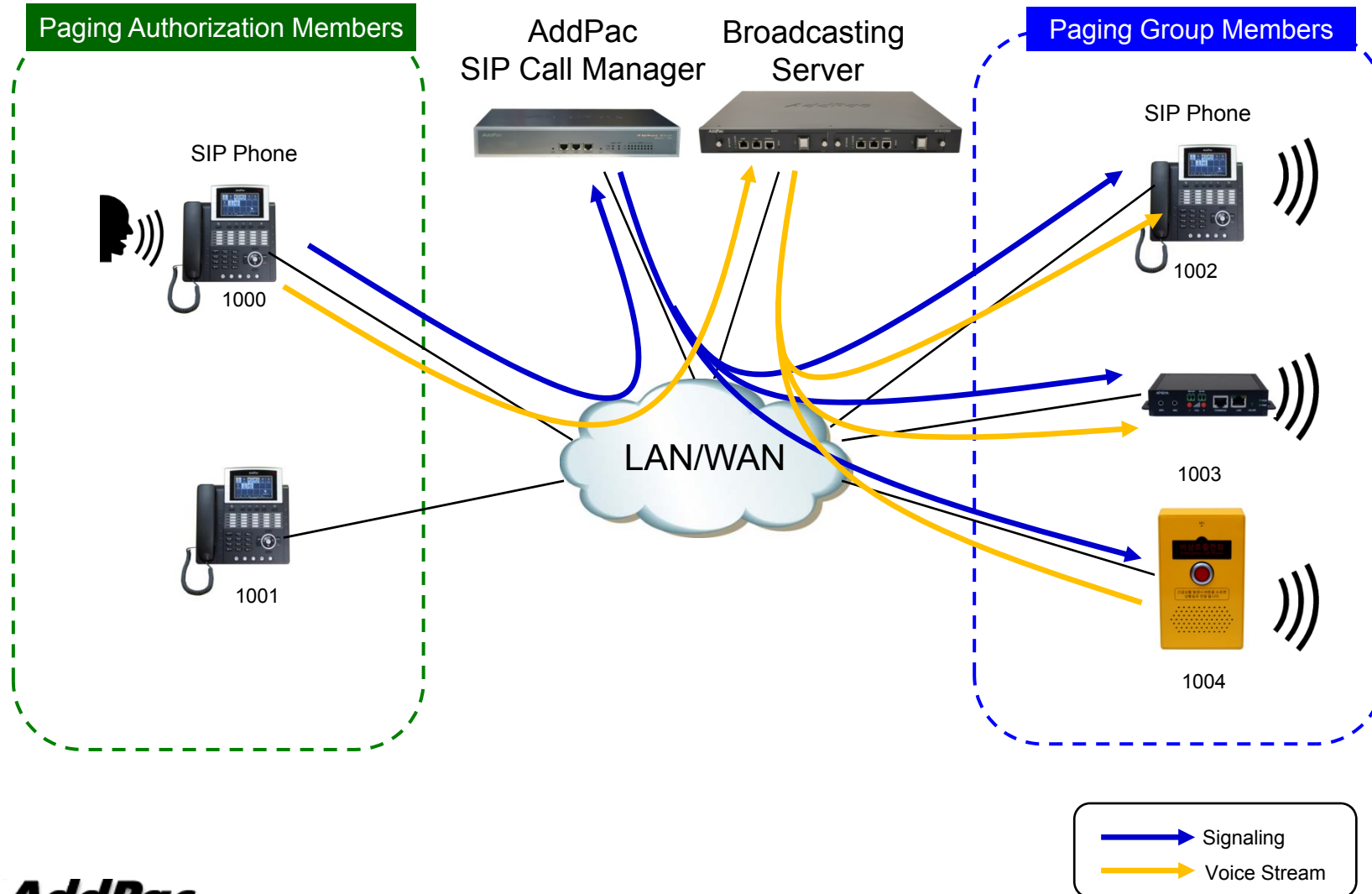
# Paging Service Model (Small Scale)



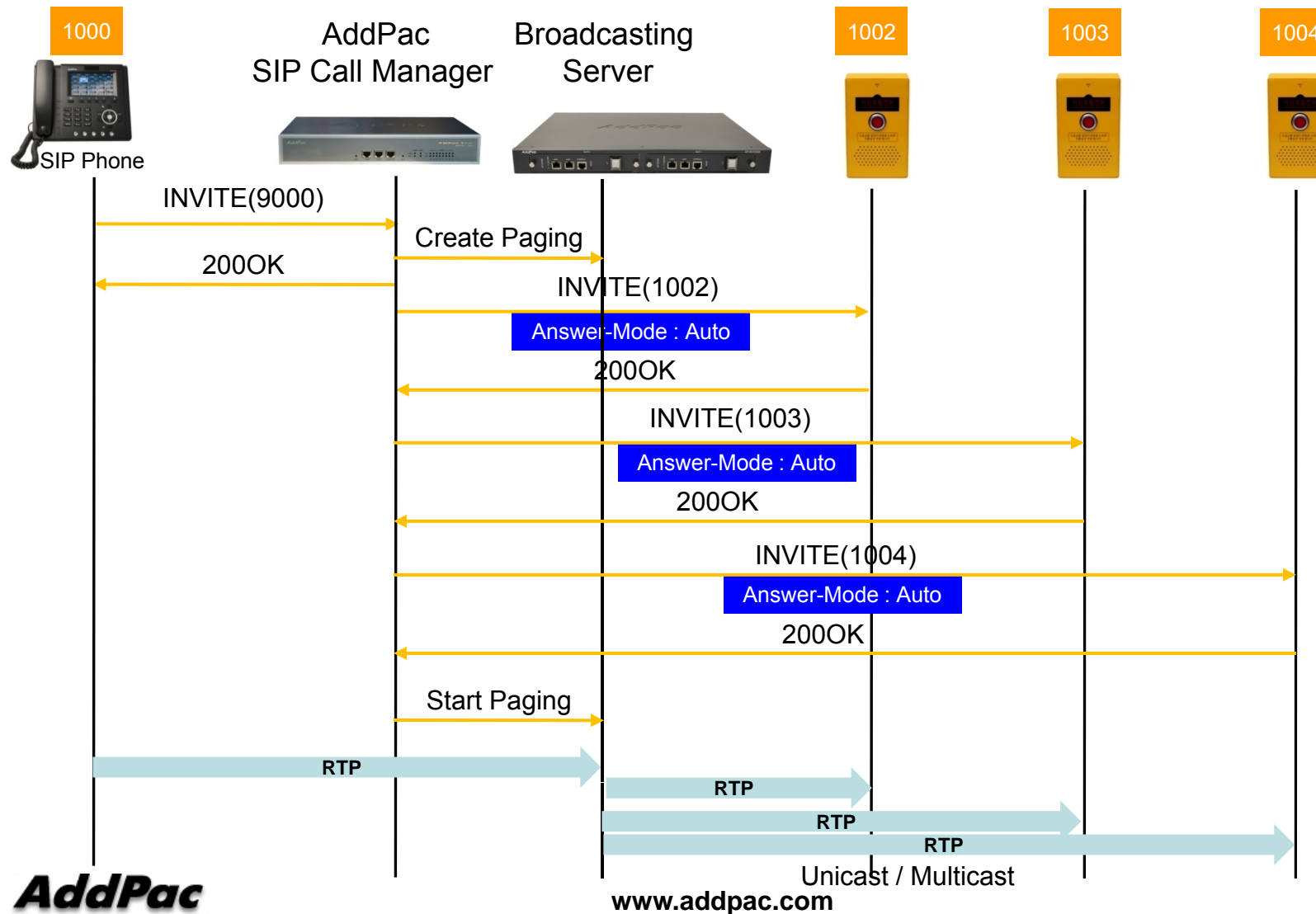
# Paging Service Model (Medium Scale)



# Paging Service Model (Large Scale)



# Paging Group Signaling Flow



# Extension – Paging Group (WSMM SIP Call Manager Program)

**Smart Multimedia Manager**  
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**Extensions**

Modify	Delete	User Portal	Extension Number	Type	Name	Date Created
1			1000	User Extension	Ashley Allen	2015-07-28 12:39:51
2			1001	User Extension	Mary Moore	2015-07-28 12:39:55
3			1002	User Extension	Thomas Taylor	2015-07-28 12:40:00
4			1003	User Extension	Victoria Valdez	2015-07-28 12:40:04
5			1004	User Extension	Olivia Ortiz	2015-07-28 12:40:08
6			1005	User Extension	Linda Lewis	2015-07-28 12:40:12
7			1006	User Extension	George Gale	2015-07-28 12:40:17
8			1007	User Extension	Isabel Irwin	2015-07-28 12:40:21
9			1008	User Extension	William Watson	2015-07-28 12:40:25
10			1009	User Extension	Sarah Scott	2015-07-28 12:40:30
11			1010	User Extension	Nicolas Nelson	2015-07-28 12:40:34
12			1011	User Extension	Emma Evans	2015-07-28 12:40:38
13			1012	User Extension	Rachel Ross	2015-07-28 12:40:43

**Add an Extension**

**Add a Paging Group**

**Paging Group**

**Extension \*** (2~12 digits)

**Name \***

**Audio Codec** G.711U

Play beep at start

Play Announcement

**Extensions**

Extension	Name
Extension	Extension

**Paging Group Members**

Name	Extension	Display Name	Multicast
------	-----------	--------------	-----------

**Group Members**

**Getting Started** **Clustering Guide** **Partitioning Guide**

**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.

**Push-to-Talk Group**  
A PTT (Push to Talk) group has members of user extensions who will receive broadcasting announcement with auto answering and also can be a floor (speaker role) by pushing the talk button. This is half-duplex two-way broadcasting.

**Paging Group**  
A paging group has members of user extensions who will receive broadcasting announcement with auto answering by speaker phone. This is half-duplex one-way broadcasting.

**Attendant Queue**  
When a call is inbound from trunk or extensions to this queue member and handled by them. Currently, the queue member

**Paging Group**  
A paging group has members of user extensions who will receive broadcasting announcement with auto answering by speaker phone. This is half-duplex one-way broadcasting.

# Paging Group Configuration (WSMM SIP Call Manager Program)

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**Add a Paging Group**

Extension \* 9000 (2-12 digits)  Check

Name \*

Audio Codec G.711U

Play beep at start Beep Sound 3

Play Announcement

**Group Members**

Name	Extension
Ashley Allen	1000
Mary Moore	1001
Isabel Irwin	1007
William Watson	1008
Sarah Scott	1009
Nicolas Nelson	1010
Emma Evans	1011
Debra Ross	1012

Specify Phone Number:

**Paging Group Members**

Name	Extension	Display Name	Multicast
Thomas Taylor	1002		Off
Victoria Valdez	1003		Off
Olivia Ortiz	1004		Off
Linda Lewis	1005		Off
George Gale	1006		Off

**Authorization Members**

Name	Extension	Display Name	Multicast
Ashley Allen	1000		Off
Mary Moore	1001		Off

**Play Announcement**

Play Announcement

Announcement Closing Notification

Repeat Count 1

Retry Count 2

Retry Interval 3 sec

Close on Caller Drop Call

**Related Links**

- User Extension
- Announcement and Tones
- Partitions

**Paging Extension**  
This is paging extension number to make the paging by dialing digits.

**Group Members**  
These members can receive broadcasting announcement.

**Authorization Members**  
Only these authorized member can start this paging by dialing the paging extension digits.

**Play Announcement**  
If enabled, group members will hear announcement at broadcasting. The announcement can be selected among announcement files and can be uploaded at Announcements and Tones menu.

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# SIP Call Manager

# IPNext600 SIP Call Manager



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# Product Overview

## IPNext600 SIP Call Manager

- SIP Application Server, Proxy, Registrar and Location Server
- Standalone SIP Paging & Broadcasting Service Support
- Legacy IP-PBX Clone Mode Support (Trunk, etc)
- RTP(Real-time Transport Protocol) Support for Unicast and Multicast Paging Service
- Internal & External RTP Routing Service Support
- Paging Service Support via SIP IP Phone
- Dual System Redundancy Architecture
  - Two(2) Fast Ethernet Interface / System
- High Performance RISC Architecture
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- Dual Redundancy Power Module

# Hardware Specification

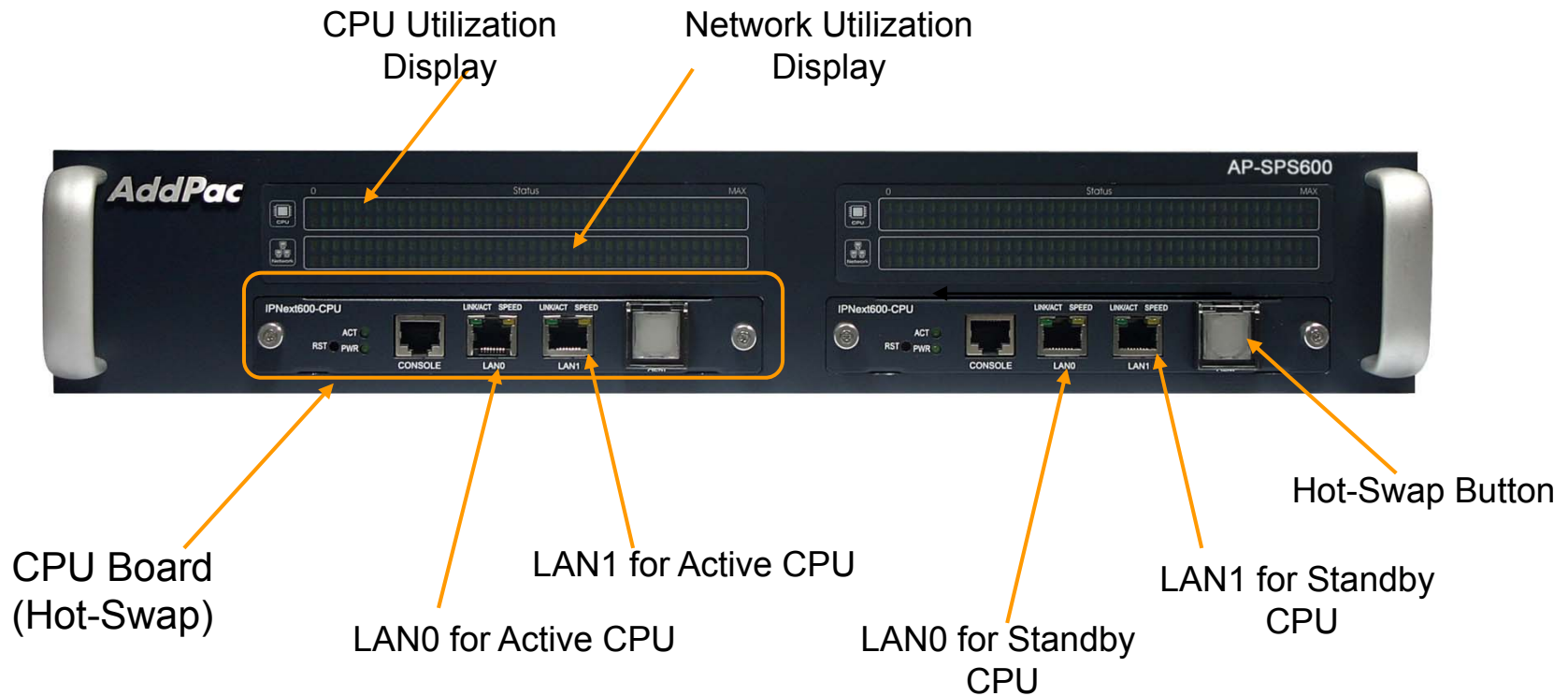
## IPNext600 SIP Call Manager

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

# Hardware Specification

IPNext600 SIP Call Manager

## Front Side



# Hardware Specification

IPNext600 SIP Call Manager

## Back Side

Dual Power Supply  
Modules  
(Hot-Swap)



PSU Module A

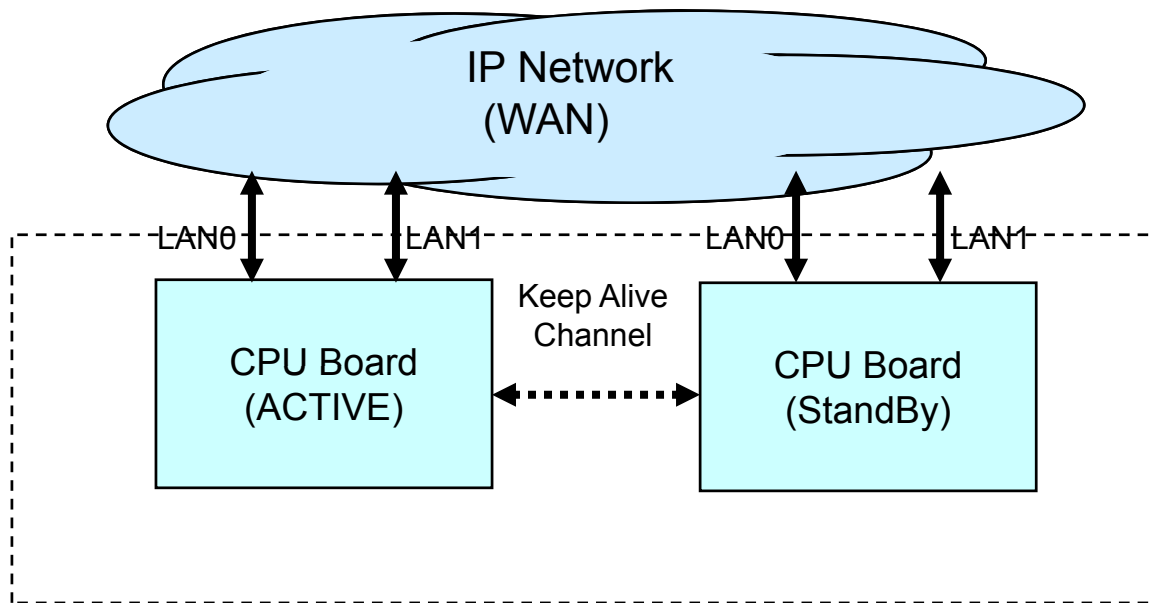
PSU Module B

Power On/Off Switch  
for System

# System Redundancy Features

IPNext600 SIP Call Manager

System Block Diagram



# IPNext180 SIP Call Manager



# Product Overview

## IPNext180 SIP Call Manager

- IP based Advanced Hybrid IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Small Office
- PSTN Interface (FXO, FXS) 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 Key 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

IPNext180 SIP Call Manager

64bit  
CPU

- RISC Microprocessor Computing Power
- Main Chassis
  - Fixed Network Interface
    - Two(2) 10/100Mbps Fast Ethernet
    - One(1) RS-232C Console (RJ45)
  - Two(2) VoIP Module Slots for FXS, FXO etc





# Hardware Specification

IPNext180 SIP Call Manager

64bit  
CPU

DSP

## Front Side



LAN1 (10/100Mbps)

LAN0(10/100Mbps)

Console Port

# Hardware Specification

IPNext180 SIP Call Manager

64bit  
CPU

DSP

## Back Side

PSTN Interface Module



Power Inlet

Power Switch



Hot-Swap Switch & LAMP Indication





# Hardware Specification

IPNext180 SIP Call Manager

64bit  
CPU

Audio  
Codec

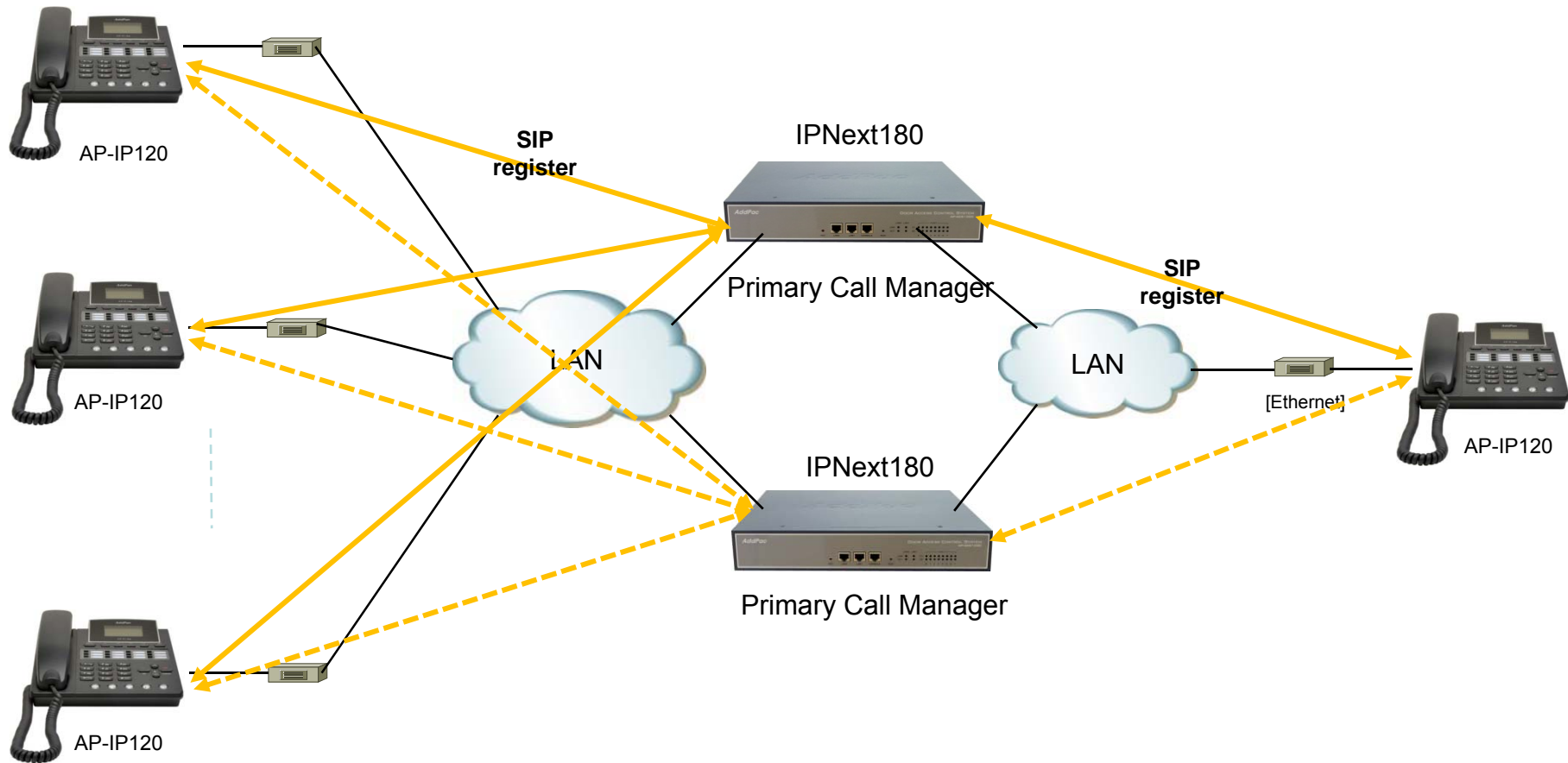
- **VoIP Interface Module**

<b>AP-N1-FXS8</b>	 A black 8-port FXS voice processing module with RJ11 ports and a handset jack.	8-Port FXS Voice Processing Module (8 x RJ11)
<b>AP-N1-FXO8</b>	 A black 8-port FXO voice processing module with RJ11 ports and a handset jack.	8-Port FXO Voice Processing Module (8 x RJ11)
<b>AP-N1-FXO4S4</b>	 A black 8-port voice processing module with 4 FXO and 4 FXS ports.	4-Port FXO and 4-Port FXS Voice Processing Module (8 x RJ11)
<b>AP-N1-E1T1</b>	 A black 1-port VoIP digital E1/T1 interface module with an RJ45 port and a handset jack.	1-Port VoIP Digital E1/T1 Interface Module(1xRJ45)

# System Redundancy Application

IPNext180 SIP Call Manager

## Active – Standby Duplication Scheme





# SIP Emergency Call Phone

# AP-EIP70

## SIP Emergency Call Phone



# Contents

- Product Overview
- Hardware Specification
- Software Service
- Audio & Voice Service and Features
- Network Service and Features
- Application Area



# Product Overview

## AP-EIP70 IP Emergency Call Phone

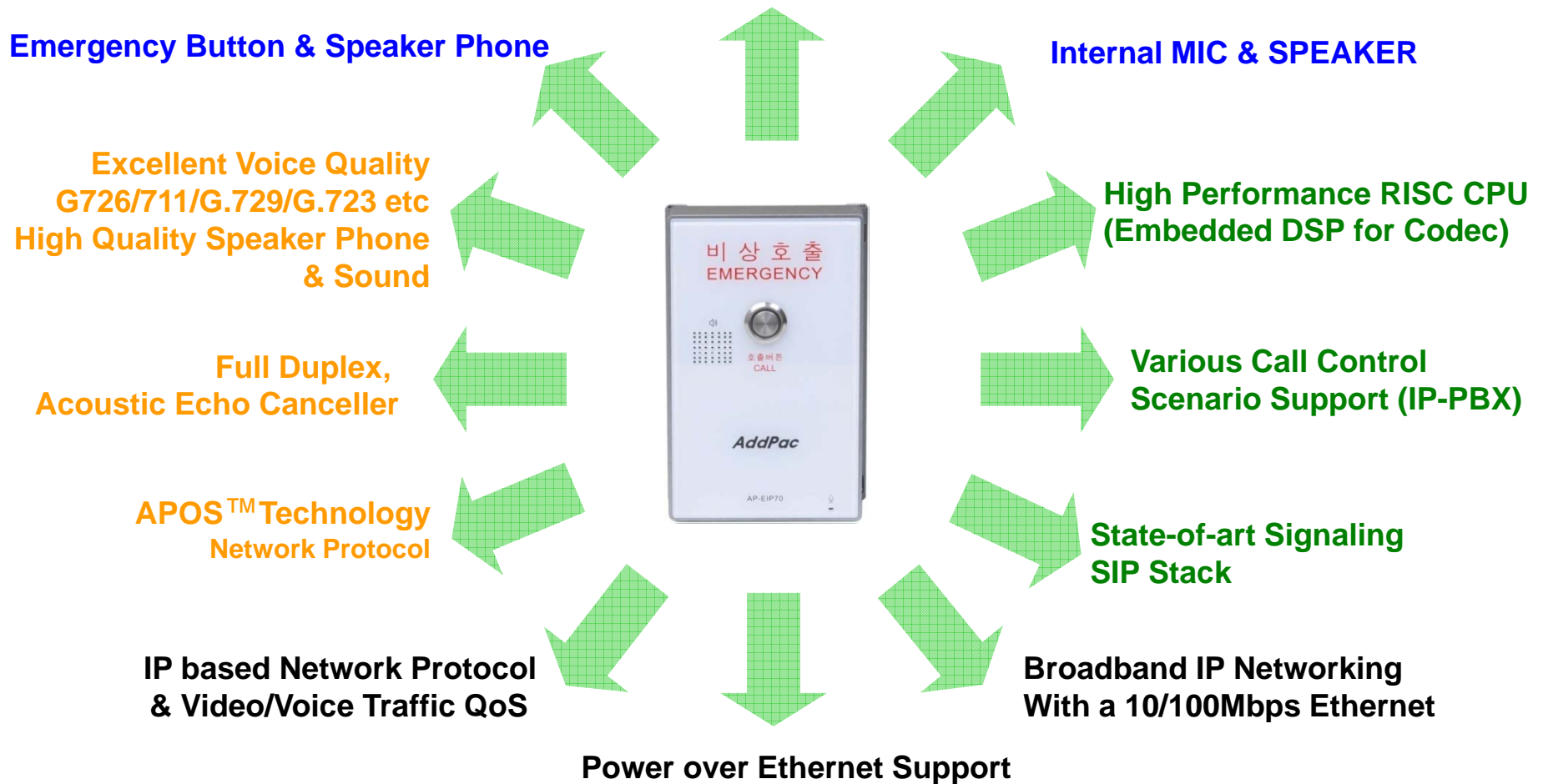
- High Performance IP Emergency Call Phone Solution
- Full Duplex Voice Communication
- One(1) 10/100Mbps Fast Ethernet
- High Quality Speaker Phone Features
- SIP VoIP Signaling Stack Embedded
- Powerful Acoustic Echo Canceller Chip Embedded
- Optional External Audio In/Out Interface Support for Noisy Street Installation
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- Advanced Voice QoS Mechanism
- PoE(Power over Ethernet) Support (Option)



# Product Highlights

AP-EIP70 IP Emergency Call Phone

## Emergency Call IP Phone Solution



# Hardware Specification

## AP-EIP70 SIP Emergency Call Phone

- RISC+DSP Microprocessor Computing Power
- Audio and Voice Interface
  - Internal MIC
  - Internal Speaker
- Emergency Call Button & LAMP
- Network Interface
  - One(1) 10/100Mbps Fast Ethernet
- Option : Alarm & Relay Out, RS232/RS485 Interface
- Option : External RCA Audio Line Out and MIC In (back side)
- Power Supply
  - Power over Ethernet (Option)
  - External Power Adaptor

# Hardware Specification

## AP-EIP70 SIP Emergency Call Phone

### Front Side



# Hardware Specification

AP-EIP70 SIP Emergency Call Phone

## Back Side



Backside Wall Mount Bracket



Rubber Cover

# Hardware Specification

AP-EIP70 SIP Emergency Call Phone

## Back Side



Option : Line Out, MIC In

LAN Port

Power Input : Terminal Block  
12V 1A

Option : 2 Alarm, 2 Relay Out,  
RS485, RS232

# Hardware Specification

## AP-EIP70 SIP Emergency Call Phone

### Power Supply

Terminal Block



12V 1A Power Adaptor

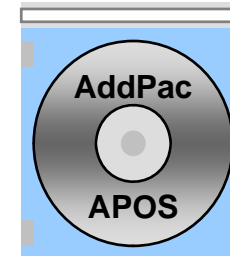
Example



# Software Service

## AP-EIP70 SIP Emergency Call Phone

- Built-in AddPac APOS Internetworking Software
  - Scalability, Functionality, and Stability Features
  - Audio Traffic QoS Control
- Programmable Video, Audio, and Voice Services
  - Audio, and Voice Codec
- Firmware Upgradeable Architecture
- Industry Standard IP based Network Protocol Features



# Network Service and Features

## AP-EIP70 SIP Emergency Call Phone

- Network Managements
  - Standard SNMP Agent (MIB v2) Support
  - Remote Management using Console, Telnet
  - Web based Management using HTTP Server Interface
- Security Functions
  - Standard & Extended IP Access List
  - Enable/Disable for Specific Network Protocols
  - Multi-level User Account Management
  - Auto-disconnect for Telnet/Console Sessions
  - PPP User Authentication Supports (PAP & CHAP)
- Operation & Managements
  - System Performance Analysis for Process, CPU, Connection Interface
  - Debugging, System Auditing, and Diagnostics Support
  - System Booting and Auto-rebooting with Watchdog Feature
  - System Managements with Data Logging
  - IP Traffic Statistics with Accounting



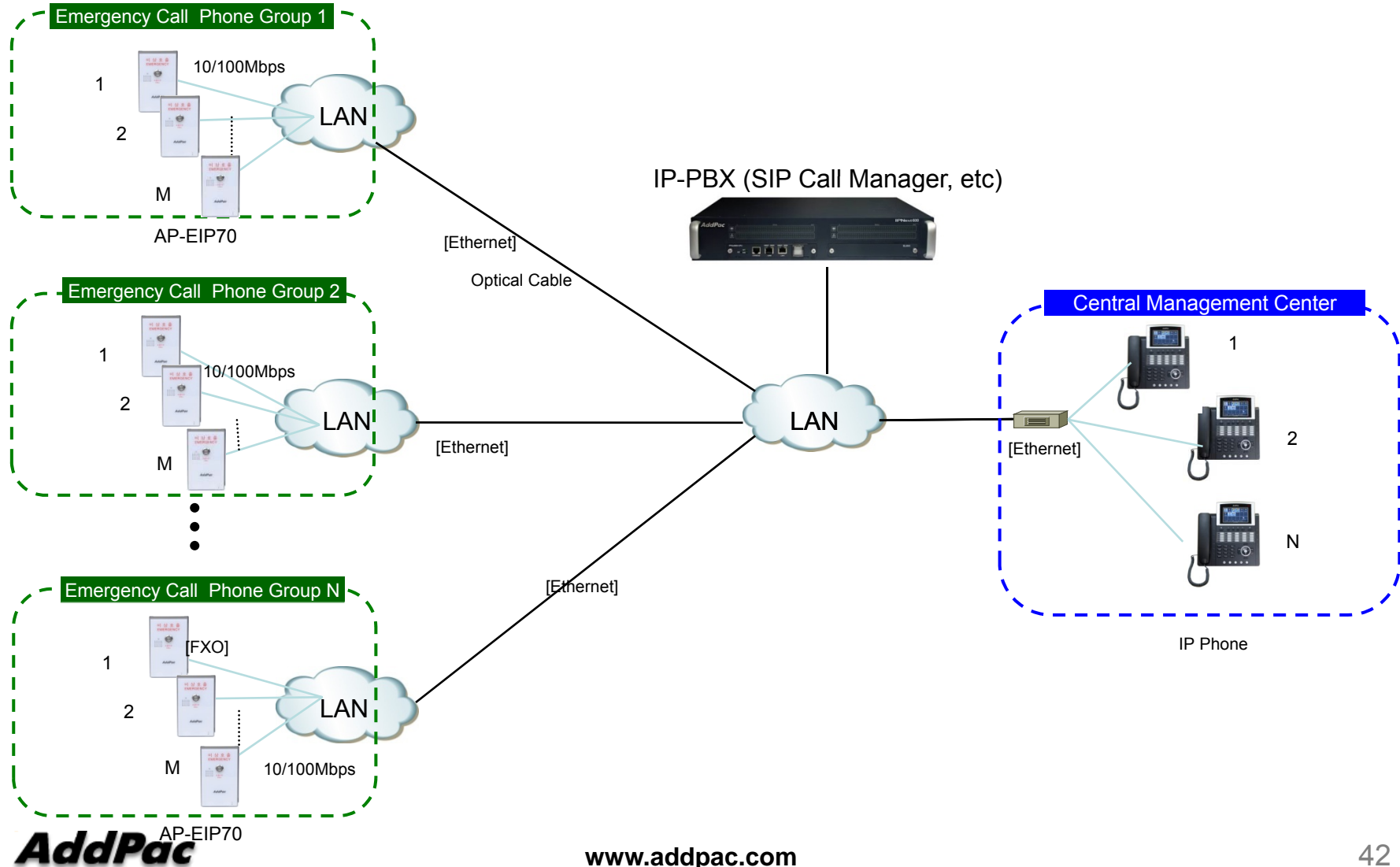
# Network Service and Features

## AP-EIP70 SIP Emergency Call Phone

- Network Managements
  - DHCP Server & Relay Functions
  - Network Address Translation (NAT) Function
  - Port Address Translation (PAT) Function
  - Transparent Bridging (IEEE Standard) Function
    - Spanning Tree Bridging Protocol Support
    - Remote Bridging Support
    - Concurrent Routing and Bridging Support
  - Cisco Style Command Line Interface (CLI)
  - Network time Protocol (NTP) Support

# Emergency Call Center Application

## AP-EIP70 SIP Emergency Call Phone



# AP-EIP60

## SIP Emergency Call Phone



# Contents

- Product Overview
- Hardware Specification
- Software Service
- Audio & Voice Service and Features
- Network Service and Features
- Application Area



# Product Overview

## AP-EIP60 SIP Emergency Call Phone

- High Performance SIP Emergency Call Phone Solution
- SIP Emergency Call Phone Solution for Outdoor Application
- Water Resistance Function Support
- Full Duplex Voice Communication
- One(1) 10/100Mbps Fast Ethernet
- PoE(Power over Ethernet) Support
- High Quality Speaker Phone Features
- SIP VoIP Signaling Stack Embedded
- Powerful Acoustic Echo Canceller Chip Embedded
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- Advanced Voice QoS Mechanism
- Die-Casting STILL Chassis (Option)

# Product Highlights

## AP-EIP60 SIP Emergency Call Phone

### Outdoor Emergency Call SIP Phone Solution

Emergency Button & Speaker Phone

Internal MIC & SPEAKER

Excellent Voice Quality  
G726/711/G.729/G.723 etc  
High Quality Speaker Phone  
& Sound

High Performance RISC CPU  
(Embedded DSP for Codec)

Full Duplex,  
Acoustic Echo Canceller

Various Call Control  
Scenario Support (IP-PBX)

APOS™ Technology  
Network Protocol

State-of-art Signaling  
SIP Stack

IP based Network Protocol  
& Video/Voice Traffic QoS

Broadband IP Networking  
With a 10/100Mbps Ethernet

Power over Ethernet Support



# Hardware Specification

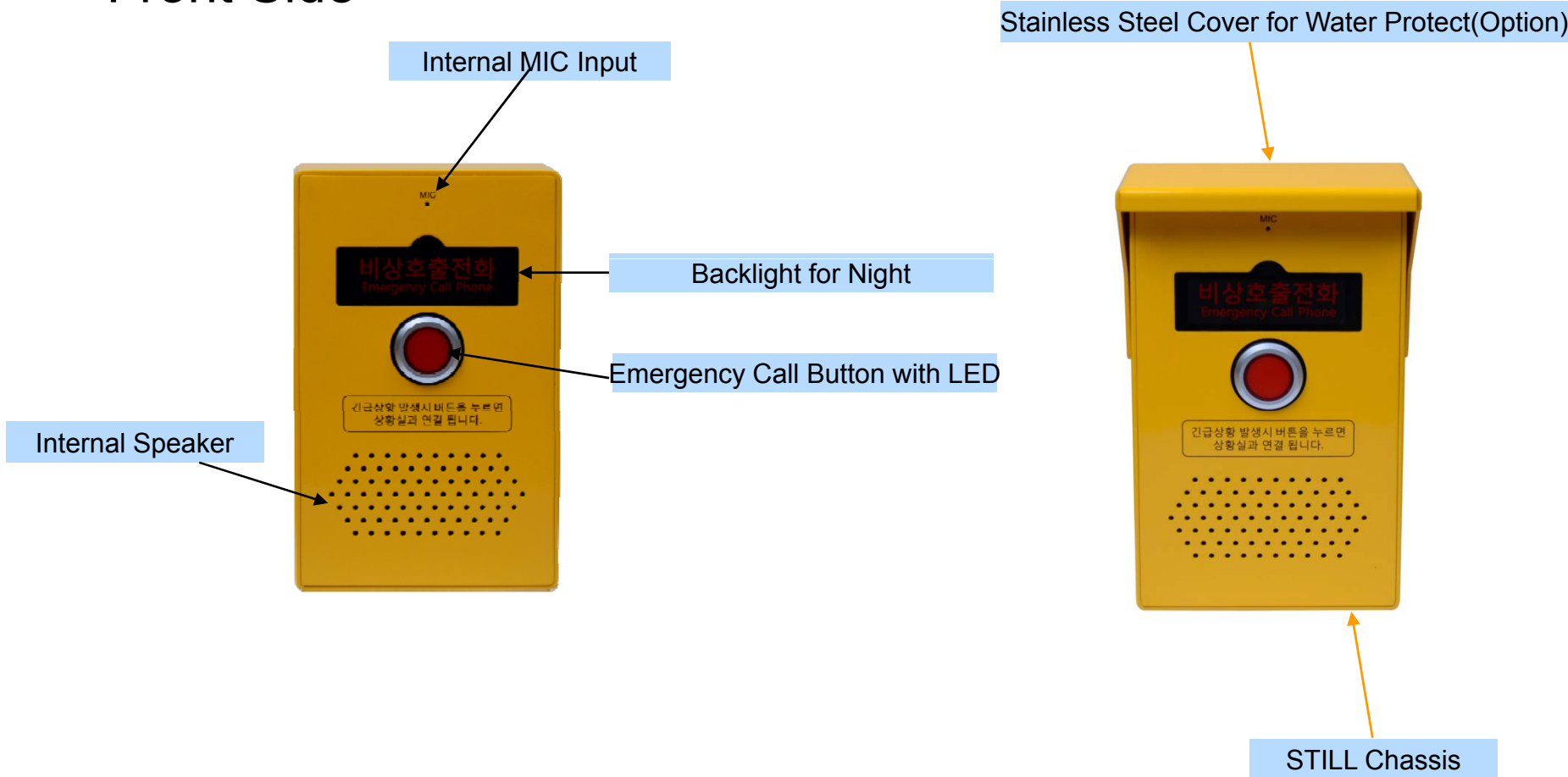
## AP-EIP60 SIP Emergency Call Phone

- RISC+DSP Microprocessor Computing Power
- Audio and Voice Interface
  - Internal MIC
  - Internal Speaker
- Emergency Call Button & LAMP
- Network Interface
  - One(1) 10/100Mbps Fast Ethernet
- Power Supply
  - Power over Ethernet (Option)
  - External Power Adaptor
- Cable (LAN, Power) Outlet (Option : Bottom or Backside)
- Die-Casting STILL Chassis

# Hardware Specification

## AP-EIP60 SIP Emergency Call Phone

### Front Side

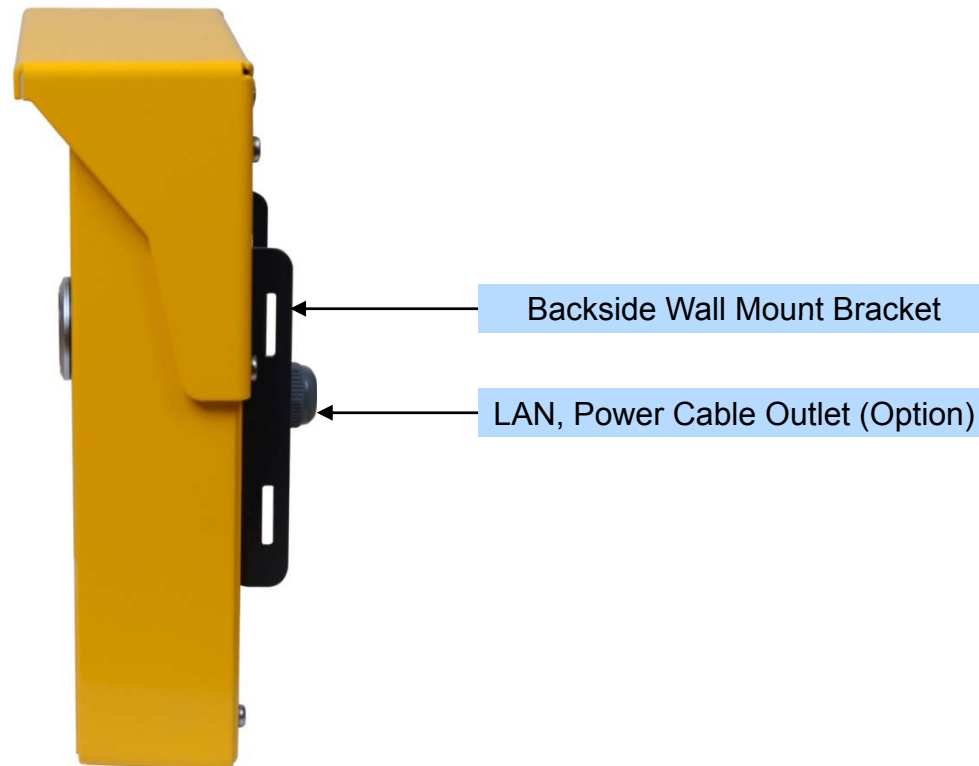




# Hardware Specification

AP-EIP60 SIP Emergency Call Phone

## Back Side



# Hardware Specification

## AP-EIP60 SIP Emergency Call Phone

### Power Supply

Terminal Block



12V 1A Power Adaptor

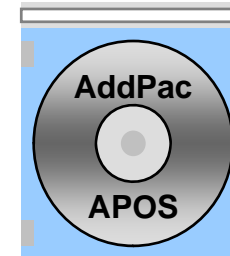
Example



# Software Service

## AP-EIP60 SIP Emergency Call Phone

- Built-in AddPac APOS Internetworking Software
  - Scalability, Functionality, and Stability Features
  - Audio Traffic QoS Control
- Programmable Video, Audio, and Voice Services
  - Audio, and Voice Codec
- Firmware Upgradeable Architecture
- Industry Standard IP based Network Protocol Features



# Network Service and Features

## AP-EIP60 SIP Emergency Call Phone

- Network Managements
  - Standard SNMP Agent (MIB v2) Support
  - Remote Management using Console, Telnet
  - Web based Management using HTTP Server Interface
- Security Functions
  - Standard & Extended IP Access List
  - Enable/Disable for Specific Network Protocols
  - Multi-level User Account Management
  - Auto-disconnect for Telnet/Console Sessions
  - PPP User Authentication Supports (PAP & CHAP)
- Operation & Managements
  - System Performance Analysis for Process, CPU, Connection Interface
  - Debugging, System Auditing, and Diagnostics Support
  - System Booting and Auto-rebooting with Watchdog Feature
  - System Managements with Data Logging
  - IP Traffic Statistics with Accounting

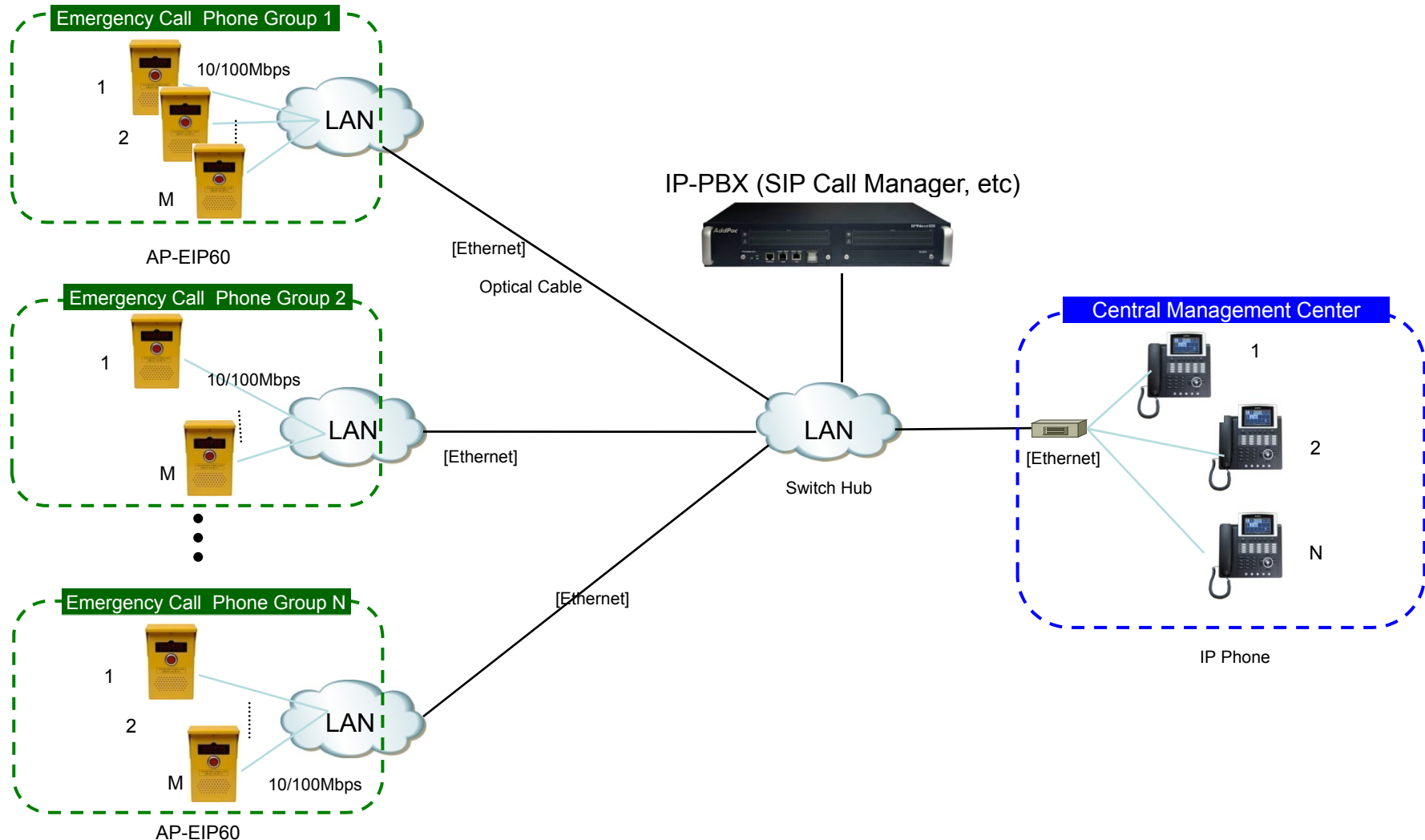
# Network Service and Features

## AP-EIP60 SIP Emergency Call Phone

- Network Managements
  - DHCP Server & Relay Functions
  - Network Address Translation (NAT) Function
  - Port Address Translation (PAT) Function
  - Transparent Bridging (IEEE Standard) Function
    - Spanning Tree Bridging Protocol Support
    - Remote Bridging Support
    - Concurrent Routing and Bridging Support
  - Cisco Style Command Line Interface (CLI)
  - Network time Protocol (NTP) Support

# Emergency Call Center Application

## AP-EIP60 SIP Emergency Call Phone





# SIP Phones for Paging Service

# AP-IP300

## SIP Broadcasting Phone



AP-IP300

AP-PT20 (40 Speed Dial Key)  
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# Product Overview

## AP-IP300 SIP Broadcasting Phone

- Premium IP Phone Solution
- SIP, H.323 Dual VoIP Signaling Stack
- SIP Paging Service Solution
- 25 Speed-Dial Button for Group Paging Service
- External Speed-Dial Extend Pack Support (AP-PT20, etc)
- Various VoIP Voice Codec Support (G.711,G.726, G.729A,G.7231.1,etc)
- High-performance Audio, and Voice Service
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection

# Hardware Specification

## AP-IP300 SIP Broadcasting Phone

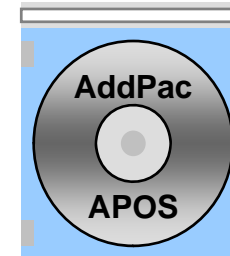
- RISC+DSP Microprocessor Computing Power (Dual Processor Architecture)
- High Quality 4.3 Inch Color LCD Panel
- 25 Speed Dial Key & User Presence Indication LED
- Optional PSTN Backup Interface
  - FXO Interface
- High quality Audio and Voice Interface
  - Stereo Audio Input Connector
  - Stereo Audio Output Connector
- Network Interface
  - Two(2) 10/100Mbps Fast Ethernet
- USB Host Mode Interface
  - USB Memory(Flash, HDD), USB Keyboard, USB Mouse, USB Wifi
- Power Supply
  - Power over Ethernet
  - External Power Adaptor (5V, 3A)



# Software Service

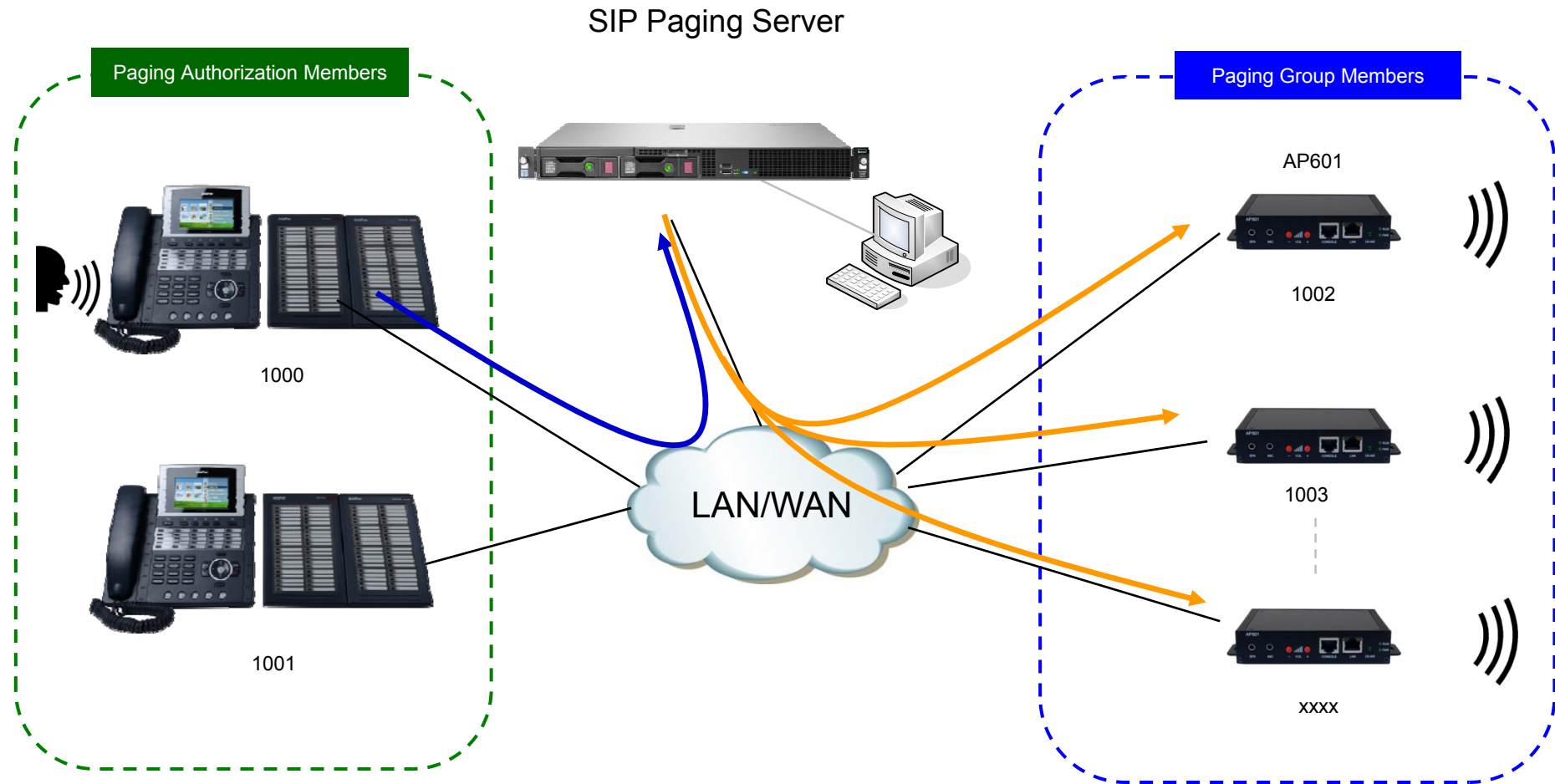
## AP-IP300 SIP Broadcasting Phone

- Built-in AddPac APOS Internetworking Software
  - Scalability, Functionality, and Stability Features
  - Audio Traffic QoS Control
- Programmable Video, Audio, and Voice Services
  - Audio, and Voice Codec
- Firmware Upgradeable Architecture
- Industry Standard IP based Network Protocol Features



# AP-IP300 SIP Broadcasting Phone

## Application Area



# AP-IP120

## SIP Broadcasting Phone



AP-IP120

AP-PT20 (40 Speed Dial Key)

# Product Overview

## AP-IP120 SIP Broadcasting Phone

- IP Phone Solution
- SIP, H.323 Dual VoIP Signaling Stack
- SIP Paging Service Solution
- 12 Speed-Dial Key with Presence Indication Lamp
- External Speed-Dial Extend Pack Support (AP-PT20, etc)
- Various VoIP Voice Codec Support (G.711,G.726, G.729A,G.7231.1,etc)
- High-performance Audio, and Voice Service
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection

# Hardware Specification

## AP-IP120 SIP Broadcasting Phone

- RISC+DSP Microprocessor Computing Power (Dual Processor Architecture)
- Optional PSTN Backup (FXO) Interface
- Optional PoE (Power over Ethernet)
- High quality Audio and Voice Interface
  - Stereo Audio Input Connector
  - Stereo Audio Output Connector
- Network Interface
  - Two(2) 10/100Mbps Fast Ethernet
- LCD Window : Graphic LCD (4 Line Text)
- 12 Speed-Dial Key with Presence Indication LAMP
- Power Supply
  - External Power Adaptor (5V, 2A)

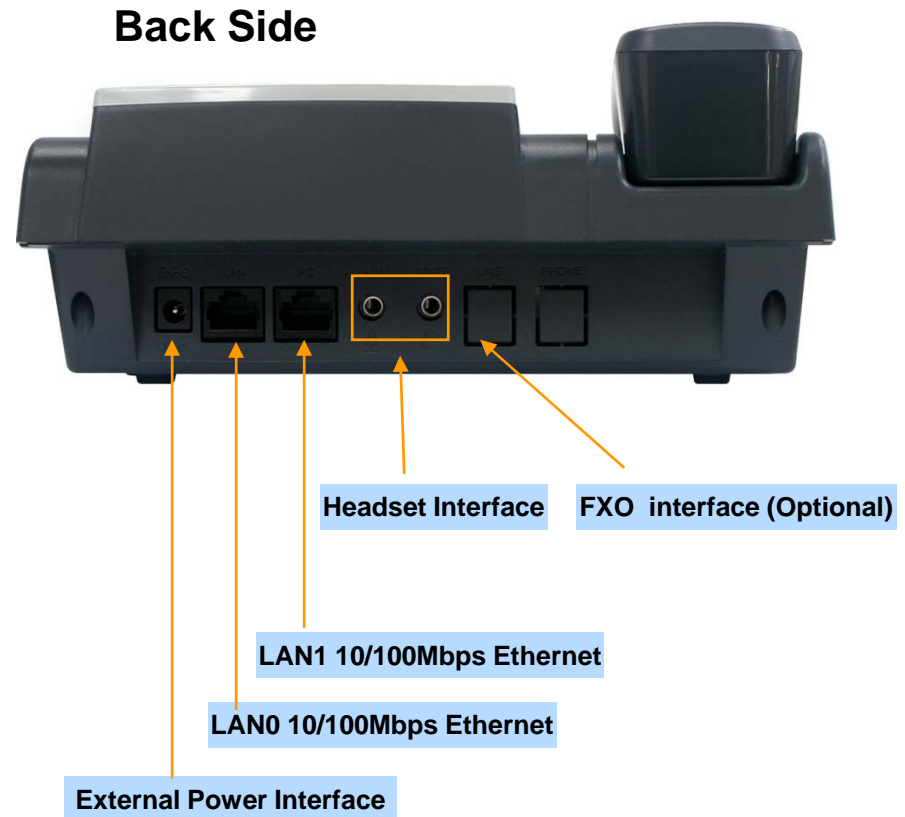
# Hardware Specification

## AP-IP120 SIP Broadcasting Phone

### Hardware Specifications

AP-IP120 SIP Broadcasting Phone	Basic Specifications
CPU	RISC Microprocessor
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
PSTN Backup Port (Optional)	1-Port PSTN Backup Port(RJ-11)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	16Mbyte High-speed SDRAM
Power Requirement	External Power Supply Adaptor / VAC 110~220V, 50/60Hz, 10Watt(5V,2A)
	Power over Ethernet (option)
Operating Temperature	0°C ~ 45°C (32 °F ~ 122°F)
Storage Temperature	-40°C ~ 85°C (-40°C ~ 185°F)
Relative Humidity	5% ~ 95% (Non-condensing)
Dimensions	H x W x D ( 70mm x 200mm x 210mm)
Weight (g)	1Kg

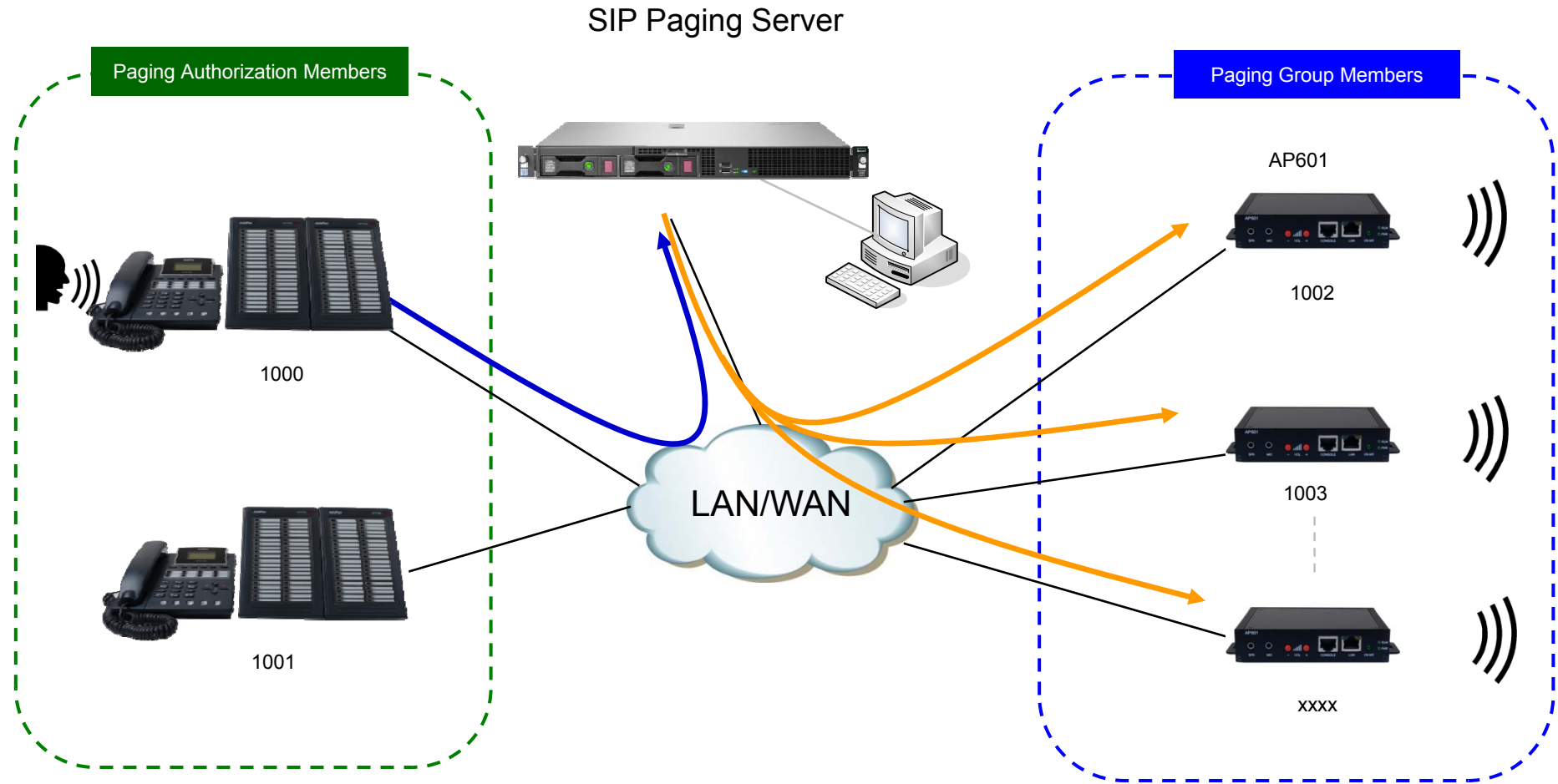
### Network interface Configurations



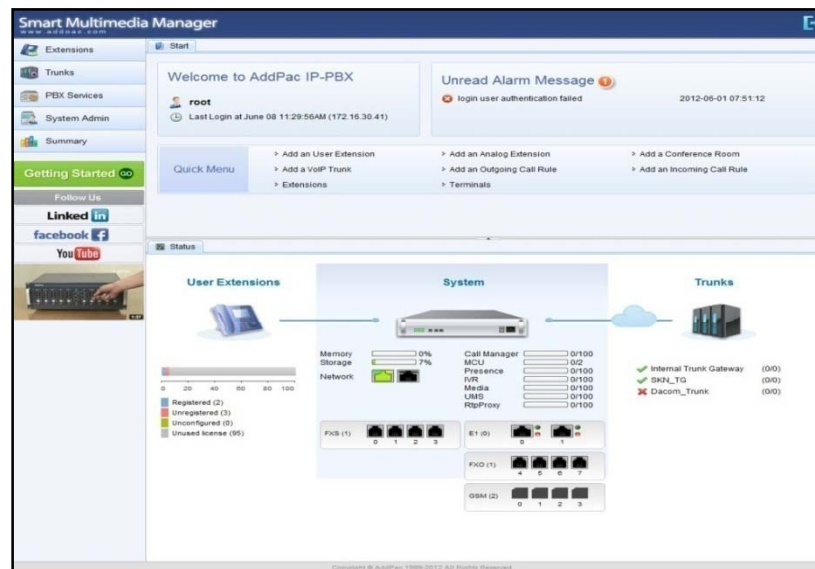


# AP-IP120 SIP Broadcasting Phone

## Application Area



# 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

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

**Add an User Extension**

Extension \*  
First Name  
Last Name \*  
Voice Mail Password \*  
User Password \*  
Department  
Title  
Email  
Home Phone  
Mobile Phone  
User ID  
Photo  
Routing Access List  
Routing Access List: internal

**Help :: User Extension**

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

**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

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

Step 2.

```

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
dial-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
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

**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 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' regarding a failed login. A 'Quick Menu' 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

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

Extensions  
Trunks  
PBX Services  
System Admin  
Summary

Getting Started GO

Follow Us  
 LinkedIn  
 facebook  
 YouTube

Start

Message failed 2012-06-01 07:51:12

Add a Conference Room  
Add an Incoming Call Rule

Trunks

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

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**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 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%  
Storage: 7%  
Network: [Icons]

Call Manager  
MCU  
Presence  
IVR  
Media  
UMS  
RtpProxy

FXS (1) [Icons] E1 (0) [Icon]  
FXO (1) [Icon]  
GSM (2) [Icon]

**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	[Icon]
2 1/5	FXO	Idle			0	0	Disabled	[Icon]
3 1/6	FXO	Idle			0	0	Disabled	[Icon]
4 1/7	FXO	Idle			0	0	Disabled	[Icon]
5 2/0	GSM	unreg...			0	0	Disabled	[Icon]
6 2/1	GSM	unreg...			0	0	Disabled	[Icon]
7 2/2	GSM	unreg...			0	0	Disabled	[Icon]
8 2/3	GSM	unreg...			0	0	Disabled	[Icon]

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	[Icon]
2 0/1/0	E1	down			0	0	Master	Network	[Icon]

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

**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' management page features a table with 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

Below the table is an 'Add an Extension' section with a 'Cancel' button and 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**

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% / 8%
- Network: [Status]
- 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)

**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

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

Quick Menu

Status

User Extension

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

**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

Add a Routing Access List

Add Cancel

Routing Access List

Name \*

Description

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

Outgoing Call Rules

Name
------

Allowed Outgoing Call Rules

Name
------

Description

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

Related Links

- Outgoing Call Rules

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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. On the left, a navigation menu is visible with a red dashed box highlighting the 'Extensions' section, which includes 'Extensions', 'Directory', 'Routing Access Lists', 'Terminal Profiles', and 'Terminals'. A yellow starburst icon is placed over the 'Terminal Profiles' link, with a blue arrow pointing to the 'Terminal Profiles' sub-section in the main content area.

The main content area shows the 'Terminal Profiles' configuration page. At the top, there are buttons for 'Add a Profile', 'Global Terminal Settings', and 'Refresh'. Below this is a table with the following data:

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

Below the table is the 'Global Terminal Settings' section, which includes various configuration options:

- Calling Party Presentation:**  allowed,  Restricted
- 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
- 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.'

At the bottom left, 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

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

Below the table is an 'Add a Trunk' section with a 'Cancel' button and three options:

- 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.

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

**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 for 'Extensions', 'Trunks', 'Outgoing Call Rules', 'Incoming Call Rules', 'PBX Services', 'System Admin', and 'Summary'. The main content area displays the 'Outgoing Call Rules' configuration page, which includes a table of existing rules and a form to add a new rule. The table shows one rule with the name 'external rule', pattern '8T', and date created '2012-04-04 09:39:48'. The form for adding a new rule includes fields for 'Name', 'Patterns', 'Trunks of Outgoing Call', 'Called Number Translation', and 'Number Translation'. A 'Description' box on the right explains that an Outgoing Call Rule controls outgoing call 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**  
www.addpac.com

Start

Welcome to AddPac IP-PBX

Unread Alarm Message  
No Unread Alarm Message

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

Quick Menu

Status

User Extension

Getting Started GO

Follow Us

Linked in

facebook f

YouTube

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

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

Status Incoming Call Rules

Add an Incoming Call Rule

Add Cancel

Name \*

Trunks of Incoming Call \*

- Internal Trunk Gateway
- SM\_SIP\_Provider
- ss
- jschoLgk

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

Route to an extension by called number + Add Rule

Transfer Rule Modify Delete

Single Extension Routing

Route to an extension by calling number + Add Rule

Transfer Rule Modify Delete

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

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

Route DID Rule Modify De

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.



# PBX Service - Announcement and Tones

The screenshot displays the 'Smart Multimedia Manager' web interface. The left sidebar menu is highlighted with a red dashed box, and a yellow starburst icon points to the 'Announcement and Tones' option. The main content area shows the 'Announcement and Tones' configuration page, which includes a table of announcements and a detailed view for a specific 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	Connected 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	Connected 3	연결 중 안내		
	430140	Over Time 2	번호 입력 오류		

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

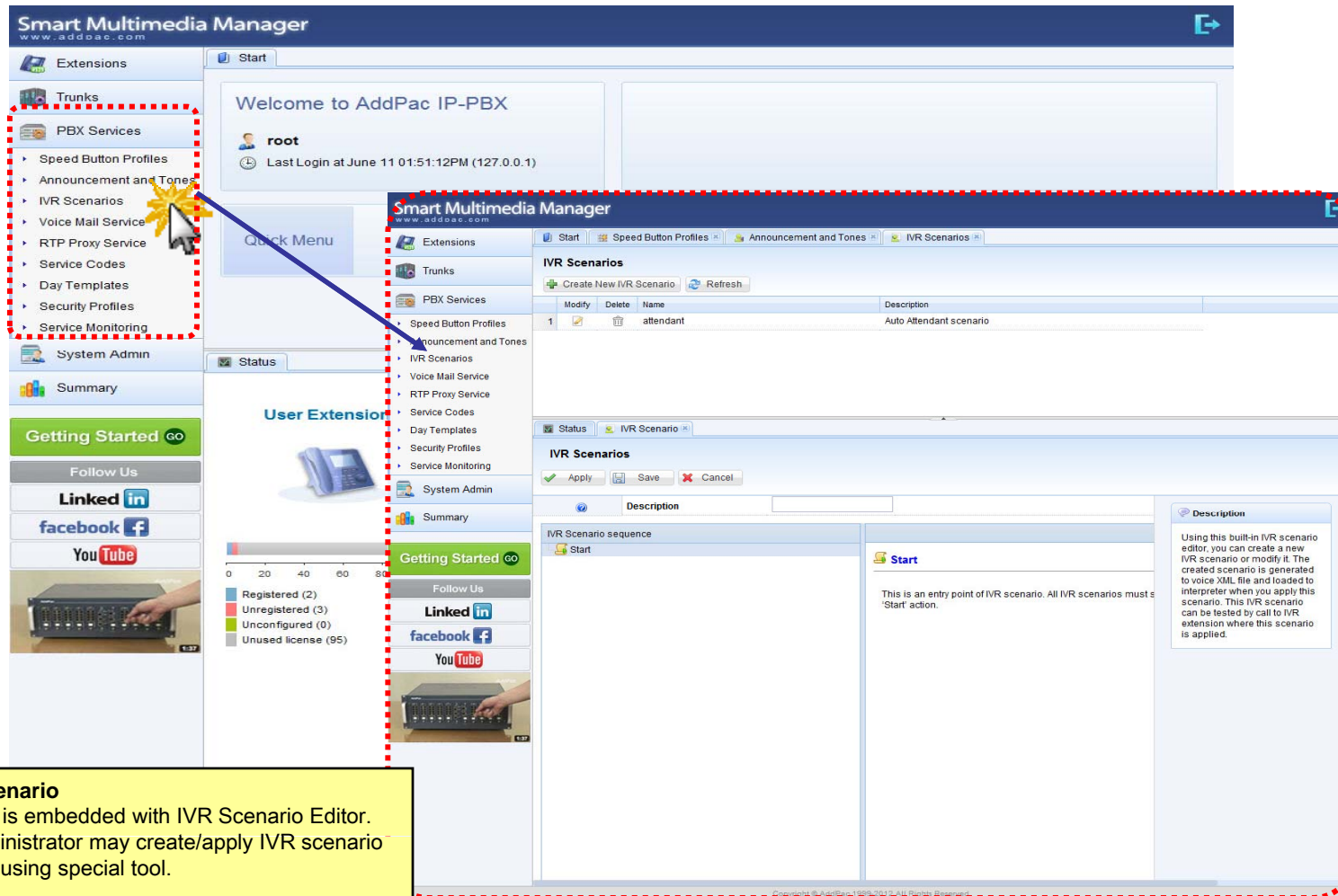
- Announcement ID:** 400110
- Description:** 연결 중 안내
- Language:** Korea
- File List:**

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 yellow starburst highlights the 'IVR Scenarios' menu item, with a blue arrow pointing to the 'IVR Scenarios' tab in the main content area. The main content area shows the 'IVR Scenarios' configuration page, which includes a table of existing scenarios, a 'Create New IVR Scenario' button, and a detailed view of the 'attendant' scenario. The detailed view shows the 'IVR Scenario sequence' with a 'Start' action and a 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.'

Modify	Delete	Name	Description
		attendant	Auto Attendant scenario

**IVR Scenarios**

Apply Save Cancel

**IVR Scenario**

Description

IVR Scenario sequence

Start

**Start**

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

**Description**

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

**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 blue arrow points from the 'Voice Mail Service' menu item to the configuration page. The configuration page includes fields for 'Retrieving Extension by Other Phone', 'Retrieving Extension by Owner Phone', and 'Leave Extension'. It also features 'Advanced Options' for 'Audio Message Length', 'Per Extension Hdd Quota', 'Over HDD Quota', 'Use Account Blocking', 'Password Fail Count', 'Enable E-mail Notification', 'Attach File to Email', 'Delete File After Email Notification', 'Enable SMS Notification', and 'SIP Port'. A 'Description' box on the right explains the purpose of the settings.

**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

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

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

**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
----------------	---------	---------	------	--------	--------

**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.

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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. The left sidebar contains a 'PBX Services' menu with 'Service Codes' highlighted. The main content area displays the 'Service Codes' configuration page, which 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 Reject(Do Not Disturb) Activation', 'Call Waiting Activation', etc. A 'Description' box on the right explains that the service code is a special digit starting with # or \*.

# 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' link in the sidebar and the corresponding 'Day Templates' section in the main content area. A blue arrow points from the sidebar link to the main content area. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root', a 'Quick Menu', and a 'User Extension' status bar. The 'Day Templates' section includes a table with one entry: 'holiday' created on 2012-03-30 11:24:41. Below the table is a form for adding a new day template with fields for Name and Description.

**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.

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

**Global Security Setting**

Apply Cancel

**TLS Cipher Suites**

- N/A
- RC4\_40
- 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.

# 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 'Smart Multimedia Manager' interface. The left sidebar contains a navigation menu with 'Service Monitoring' highlighted. The main content area displays the 'Service Monitoring' page, which includes a table for 'Active Calls' and 'Conference' information. The system status dashboard at the bottom shows various metrics such as 'Registered (2)', 'Unregistered (3)', 'Unconfigured (0)', and 'Unused license (95)'. The 'System' section includes a diagram of the IP-PBX architecture and a list of system components with their respective status indicators.



# 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

The screenshot shows the 'Smart Multimedia Manager' web interface. The left sidebar contains a 'System Admin' menu with 'Network Interface' highlighted. The main content area shows the 'Network Interfaces' configuration page, which is divided into 'WAN Interface' and 'LAN Interface' sections. The WAN interface is configured with DHCP mode, and the LAN interface is configured with 'None' mode. A description on the right explains the roles of WAN and LAN interfaces. A yellow box at the bottom left summarizes the configuration steps for each interface type.

# System Admin - Network Services

**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.

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

# 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, Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, Show Command, and Summary. The main content area displays the 'Administrators' page, which includes a table of 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

The screenshot shows the 'Smart Multimedia Manager' interface. The left sidebar contains a 'System Admin' menu with a red dashed box around it. A blue arrow points from the 'Licenses' option in the menu to the 'Licenses' page. The 'Licenses' page features a table with the following data:

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

A yellow callout box contains the following text:

**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

www.addpac.com

Status IVR Scenarios Voice Lines

Voice Lines

Apply Cancel

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

Trunk

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

Extension

Analog

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

Description

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

Related Links

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

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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 | Security Profiles | Alarm History

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

**Alarm History**

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

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

**System Status**

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

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

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

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

# System Admin - Call History

**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
- Licenses
- Voice Lines
- Alarm History
- Call History
- Show Command

**Show Command**

Categories: CPU, Memory, Storage, Network, Config, SIP User Agent, Gateway, Voice Port, Dial-Peer, SSCP, SP, Domain Cluster, Terminal Summary, Presence

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 Sesion 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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