Simple SIP Voice Paging Service for IP Emergency Call Phone

AddPac Technology

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Contents

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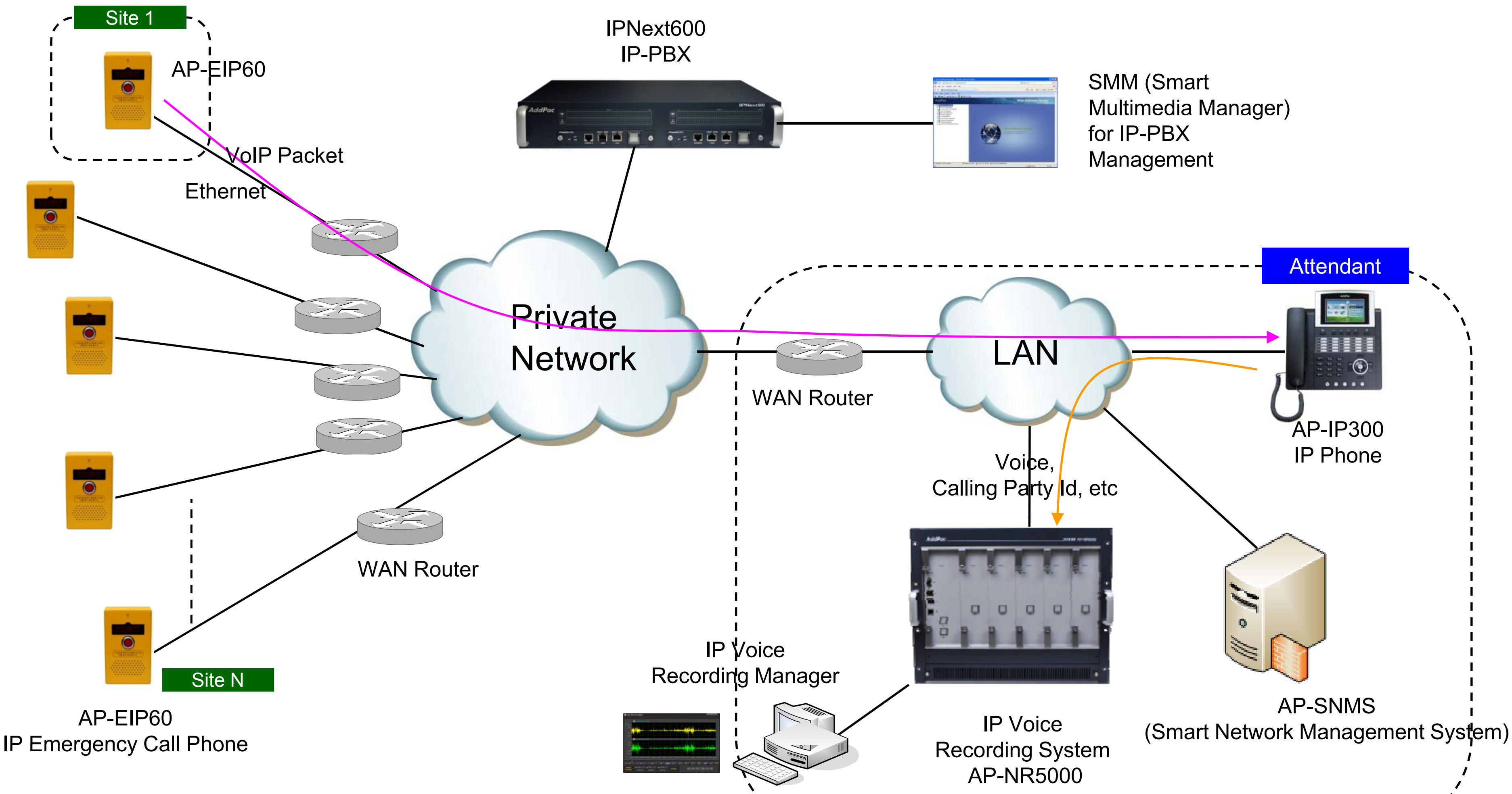
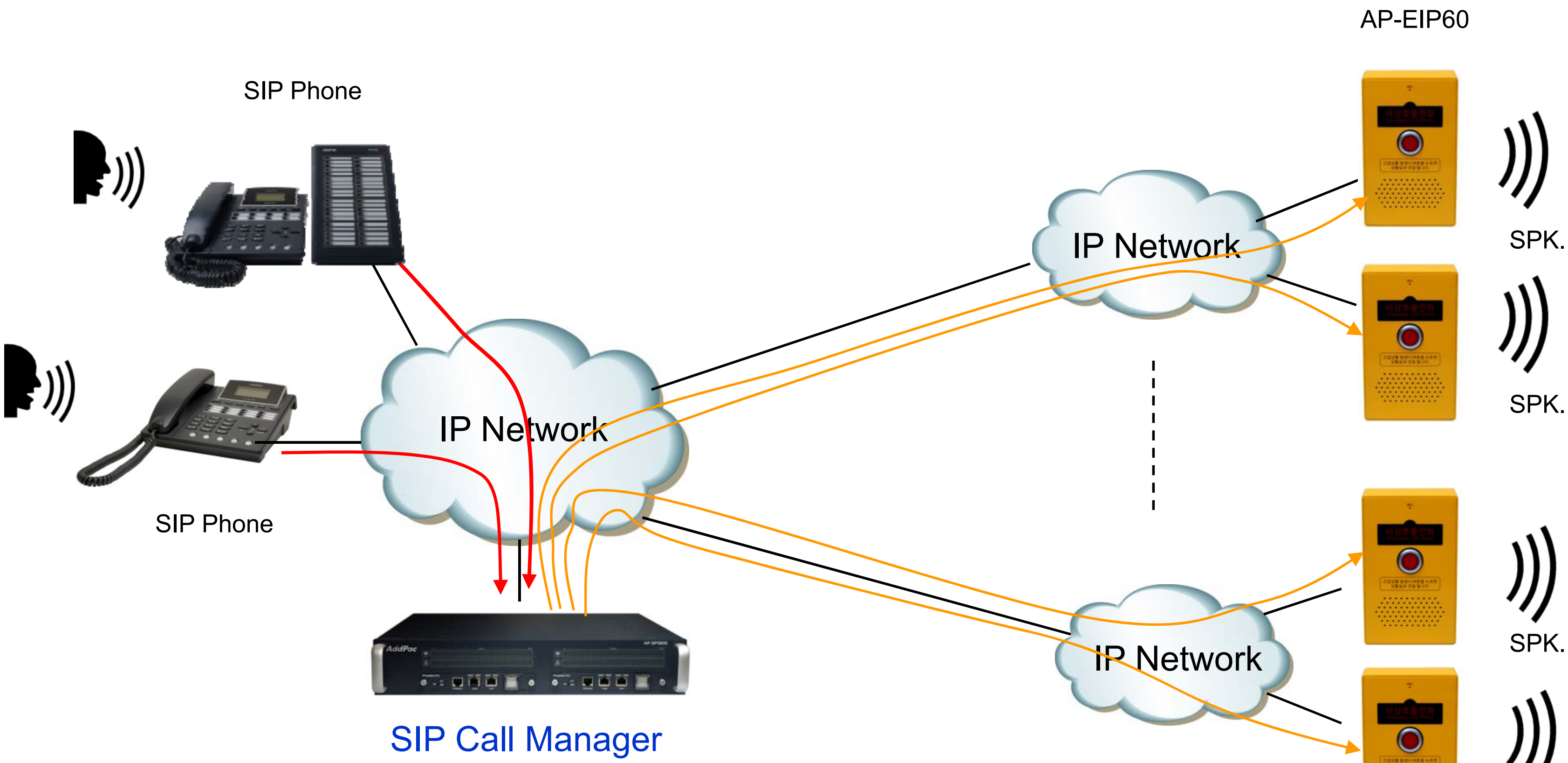
  - SIP Emergency Call Phone: AP-EIP70, AP-EIP60
  - SIP Phones for Paging Service : AP-IP300, AP-IP120
  - Web based Smart Multimedia Manager for SIP Call Manager

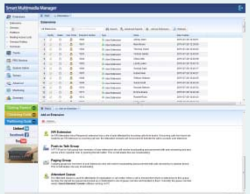
SIP Emergency Call Service Network Diagram



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
	 IPNext600	 AP-EIP70	 AP-BCR5000	 AP-IP300 + AP-PT20x2
	 IPNext180	 AP-EIP60		 AP-IP120 + AP-PT20x2

SIP Call Manager based Simple Paging Service Overview

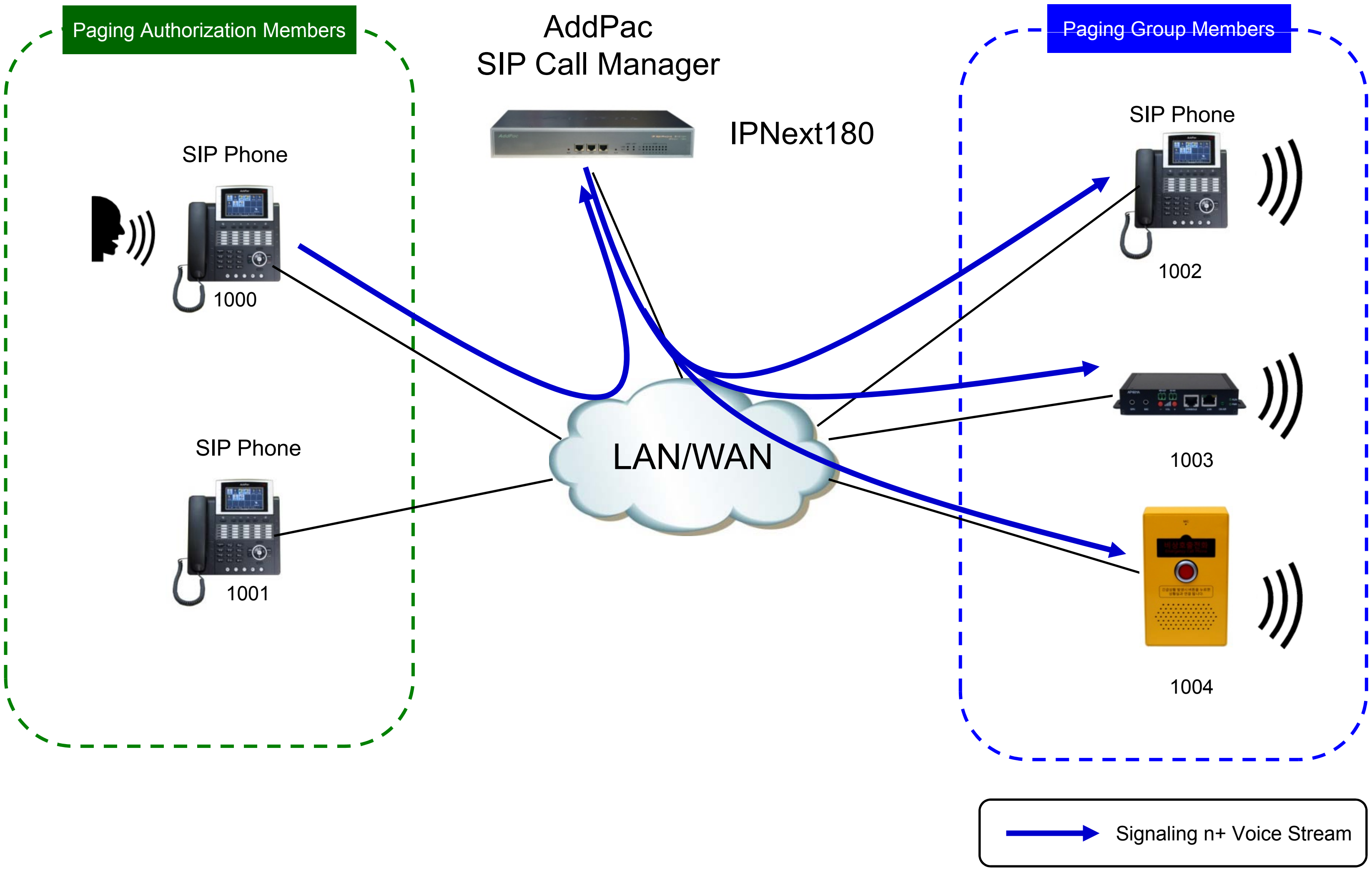
Contents

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

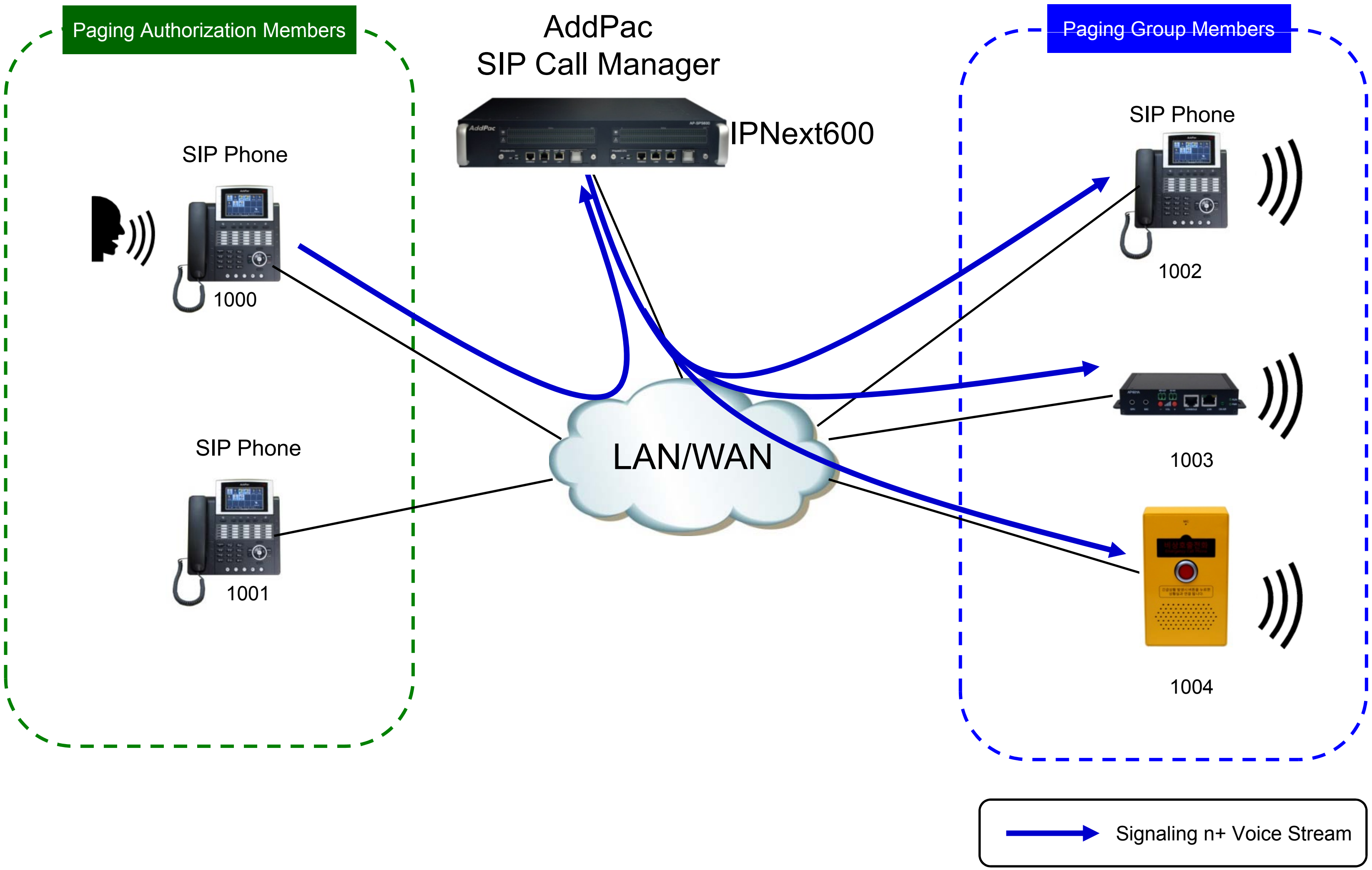
Paging Service Features

- 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(μ-law, A-law)
- External Broadcasting Server for Large Capacity (Option)

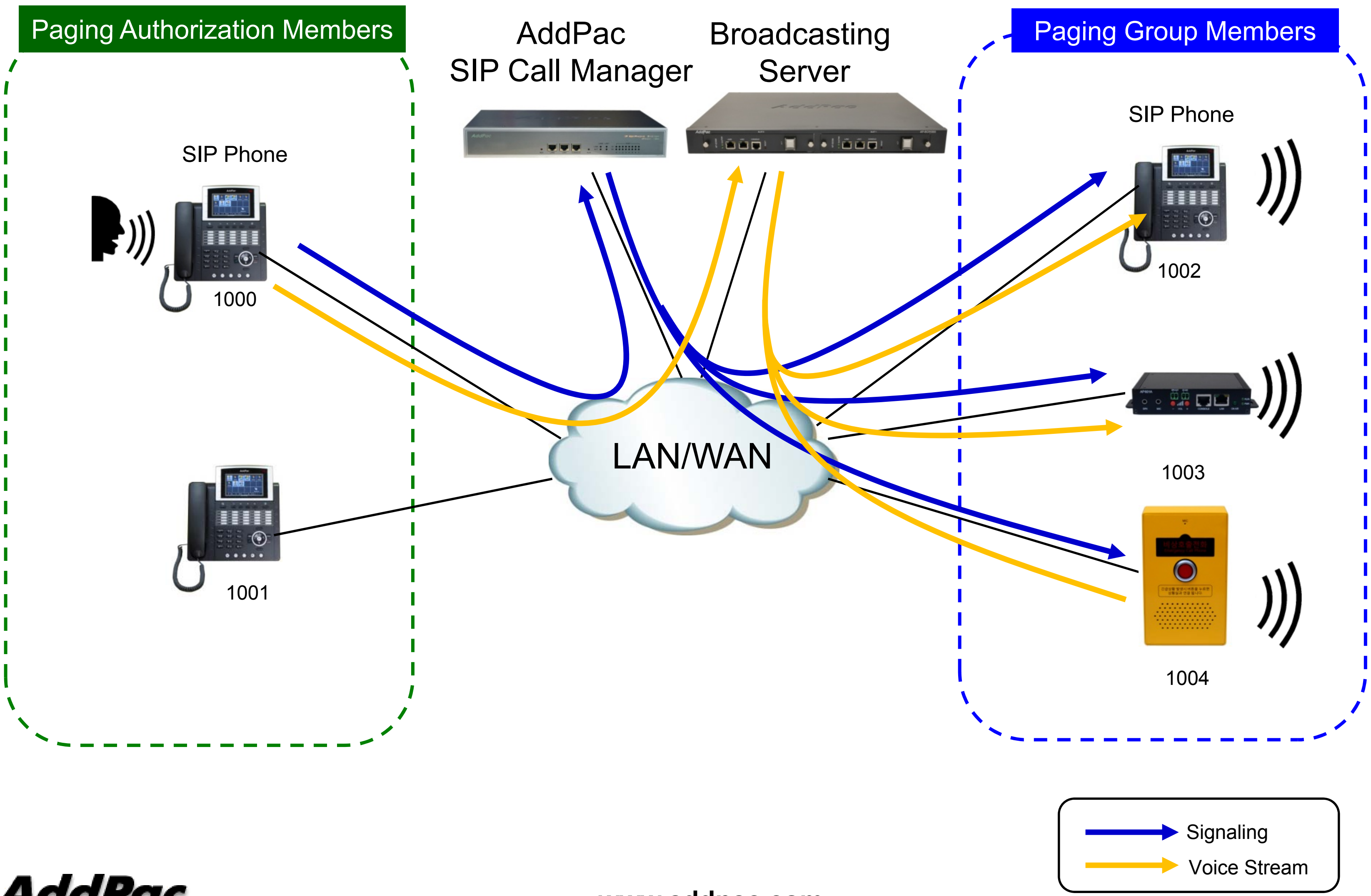
Paging Service Model (Small Scale)



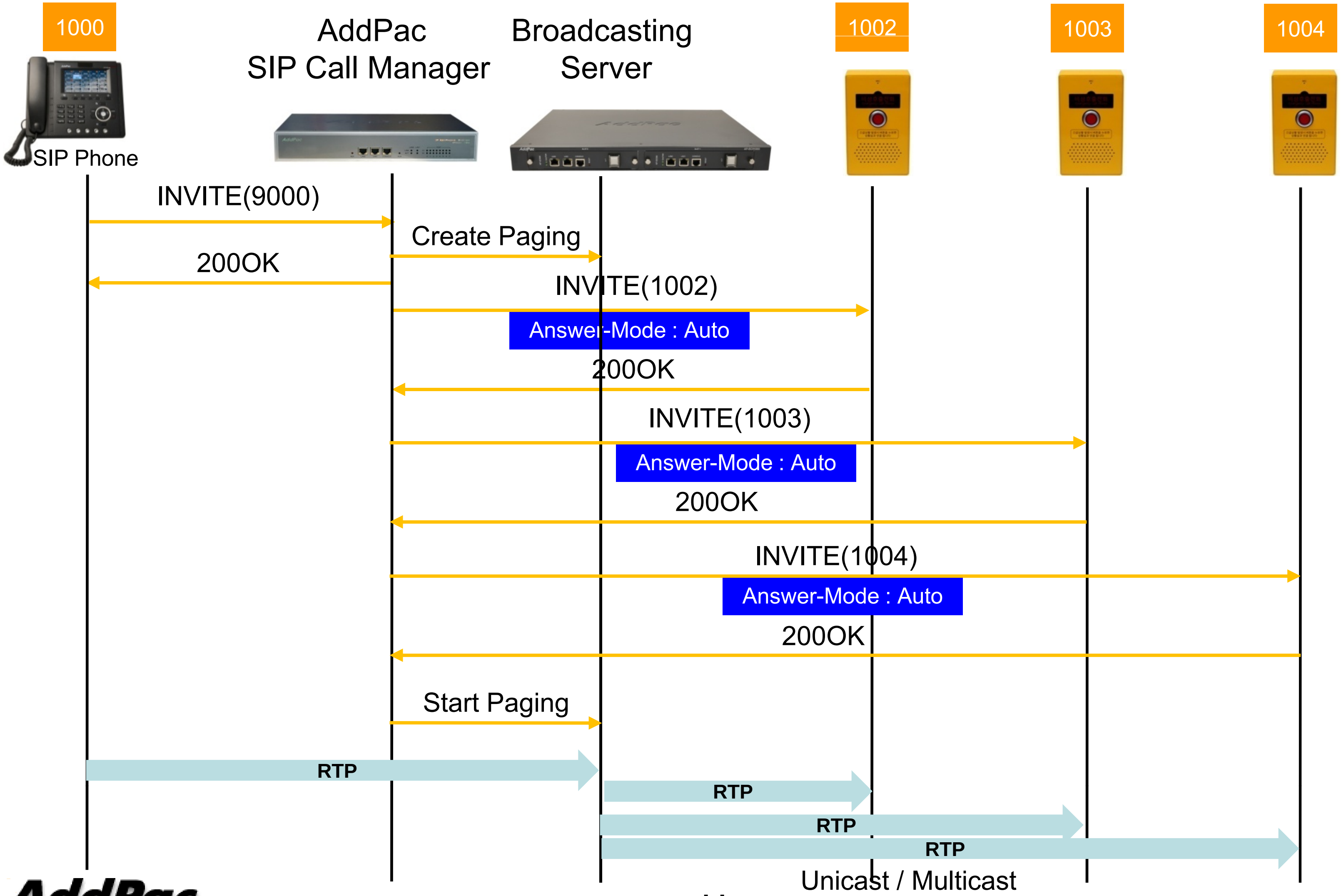
Paging Service Model (Medium Scale)



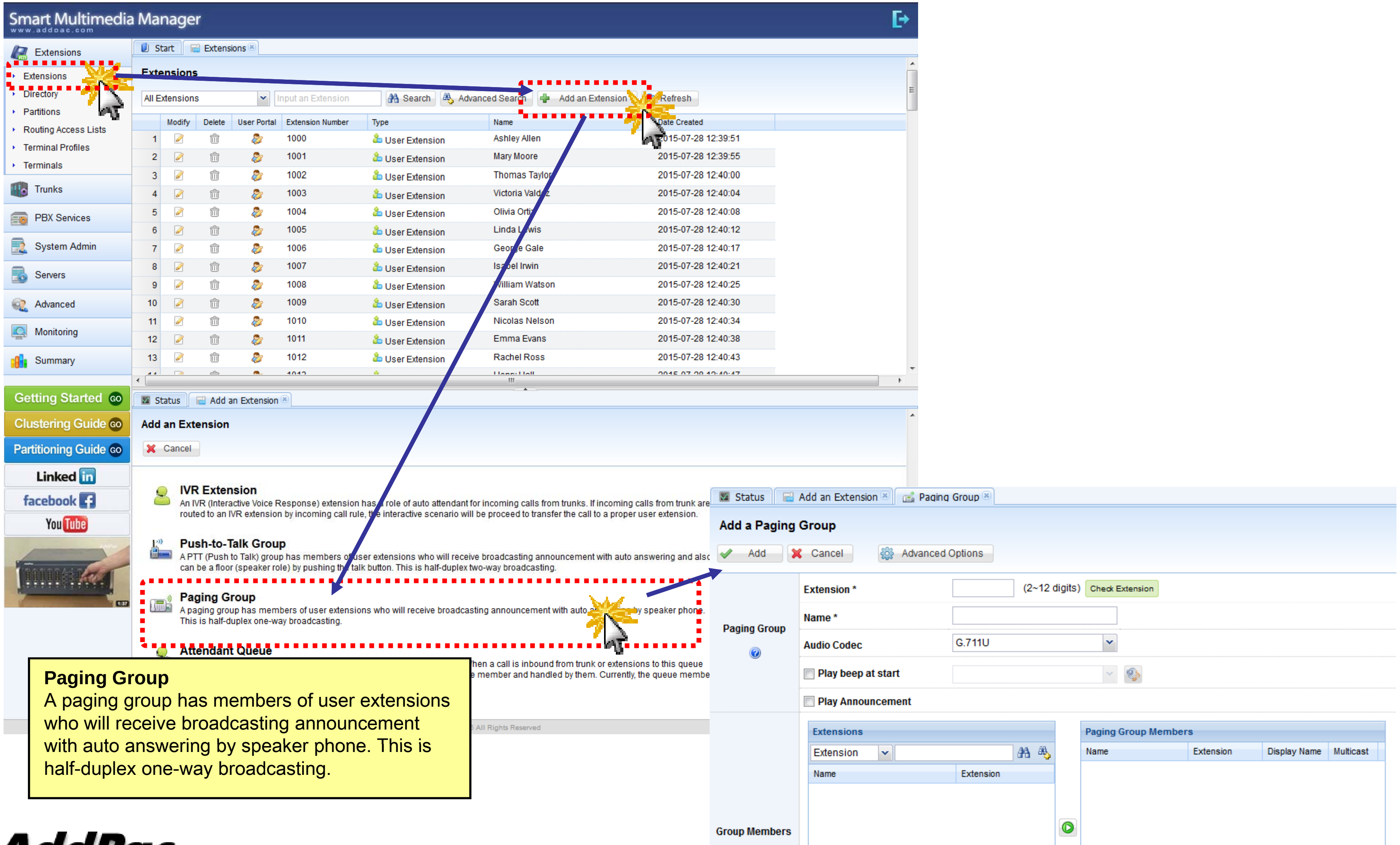
Paging Service Model (Large Scale)



Paging Group Signaling Flow

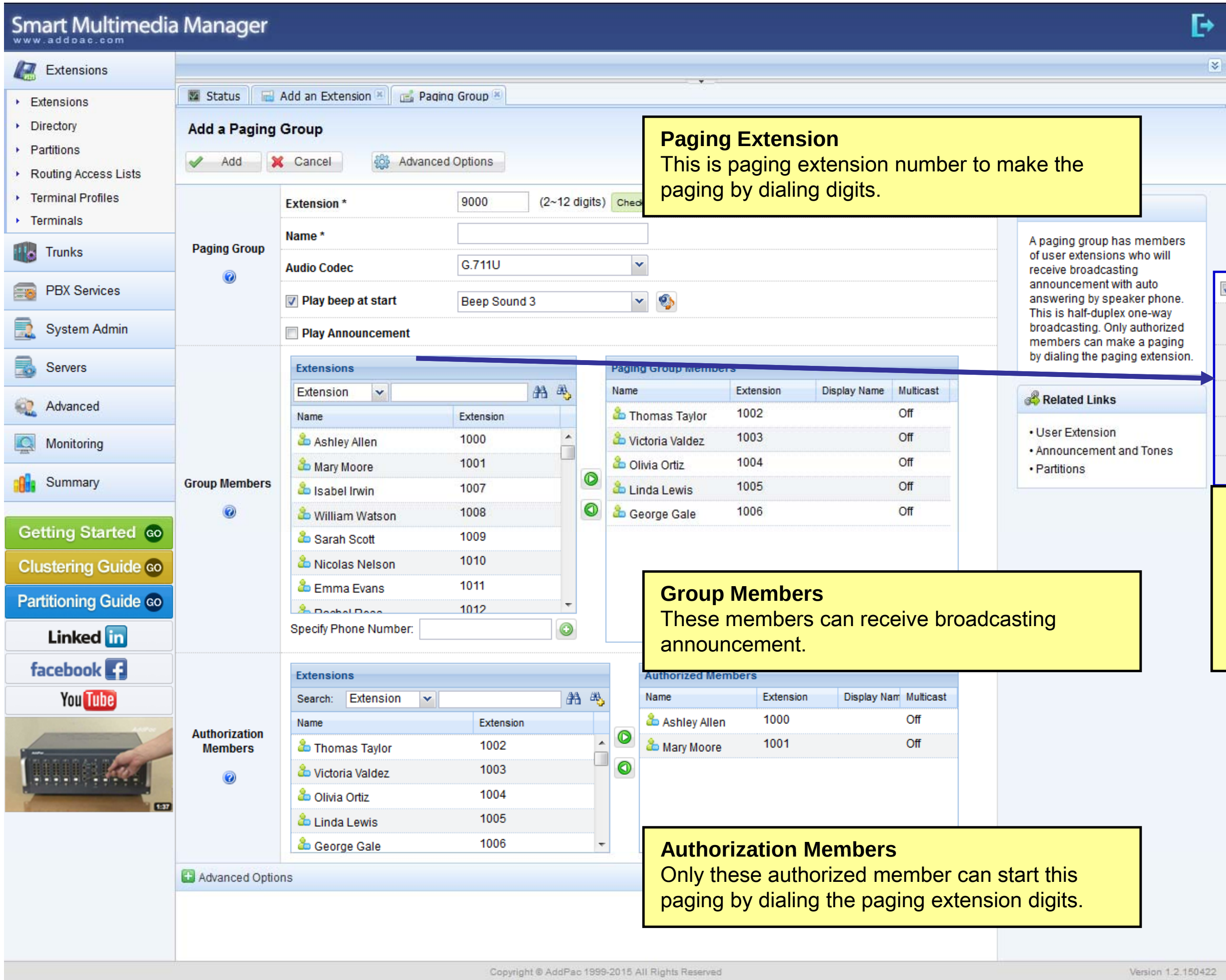


Extension – Paging Group
(WSMM SIP Call Manager Program)



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.

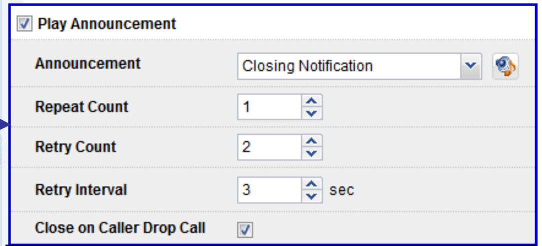
Paging Group Configuration (WSMM SIP Call Manager Program)



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.

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

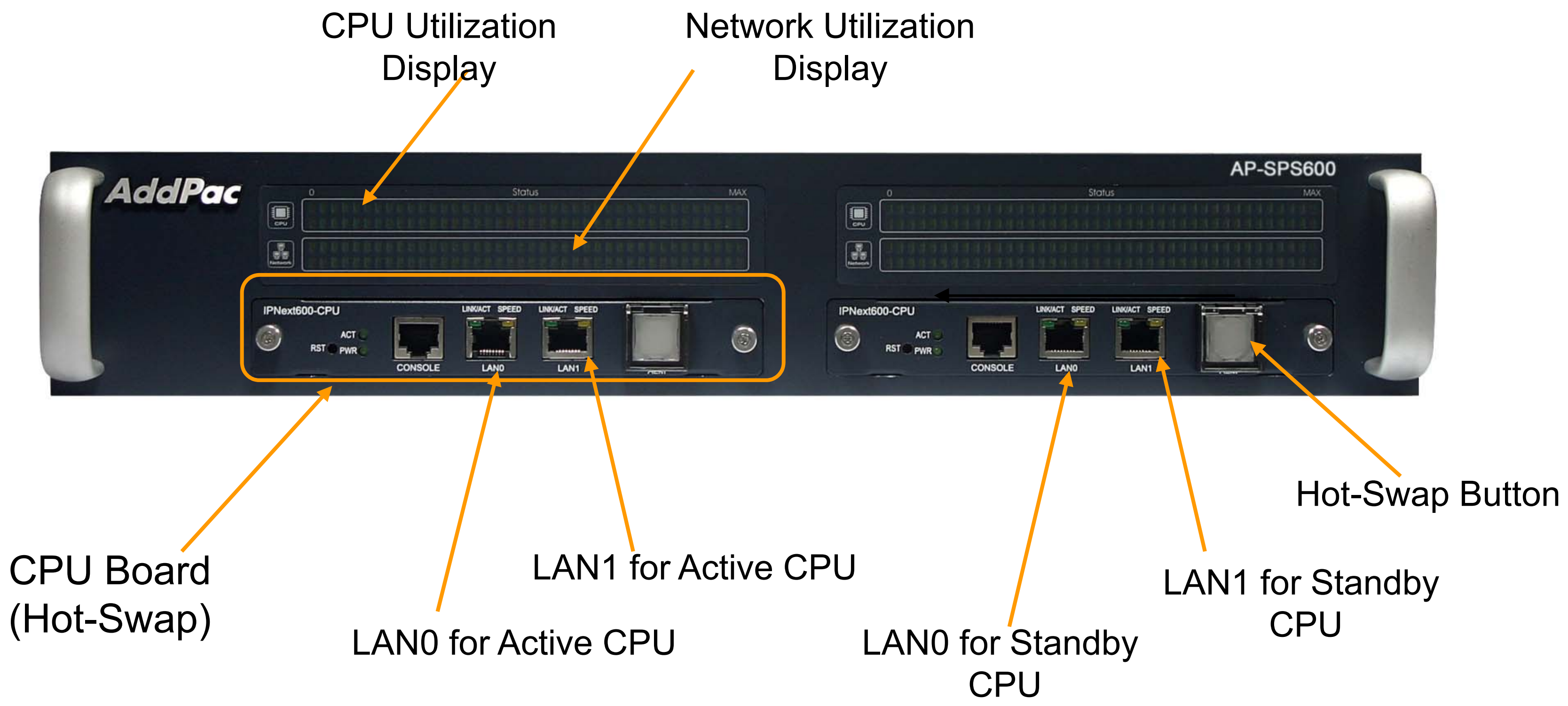
IPNext600 SIP Call Manager

- 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

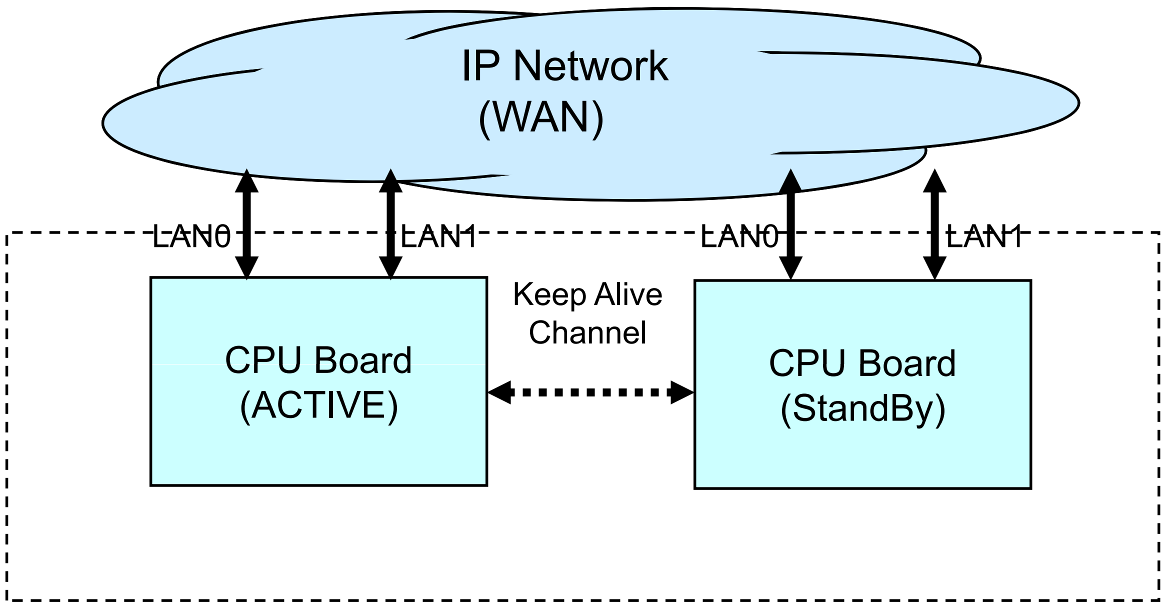
PSU Module B

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

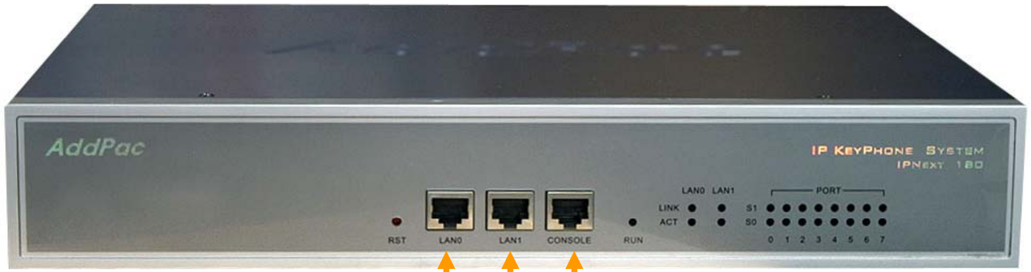
Hardware Specification

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

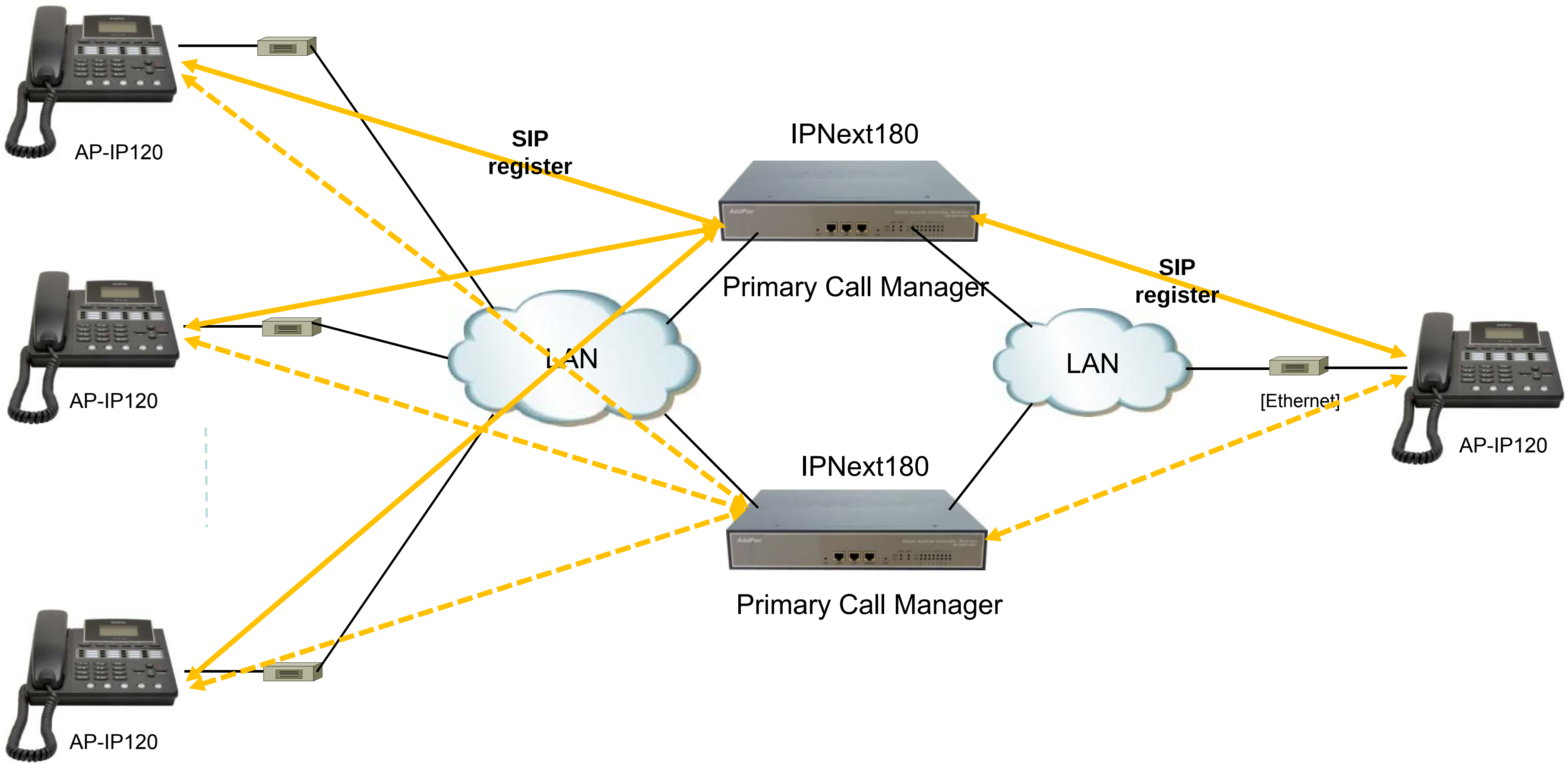
- **VoIP Interface Module**

AP-N1-FXS8		8-Port FXS Voice Processing Module (8 x RJ11)
AP-N1-FXO8		8-Port FXO Voice Processing Module (8 x RJ11)
AP-N1-FXO4S4		4-Port FXO and 4-Port FXS Voice Processing Module (8 x RJ11)
AP-N1-E1T1		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

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 TM 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-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

Back Side

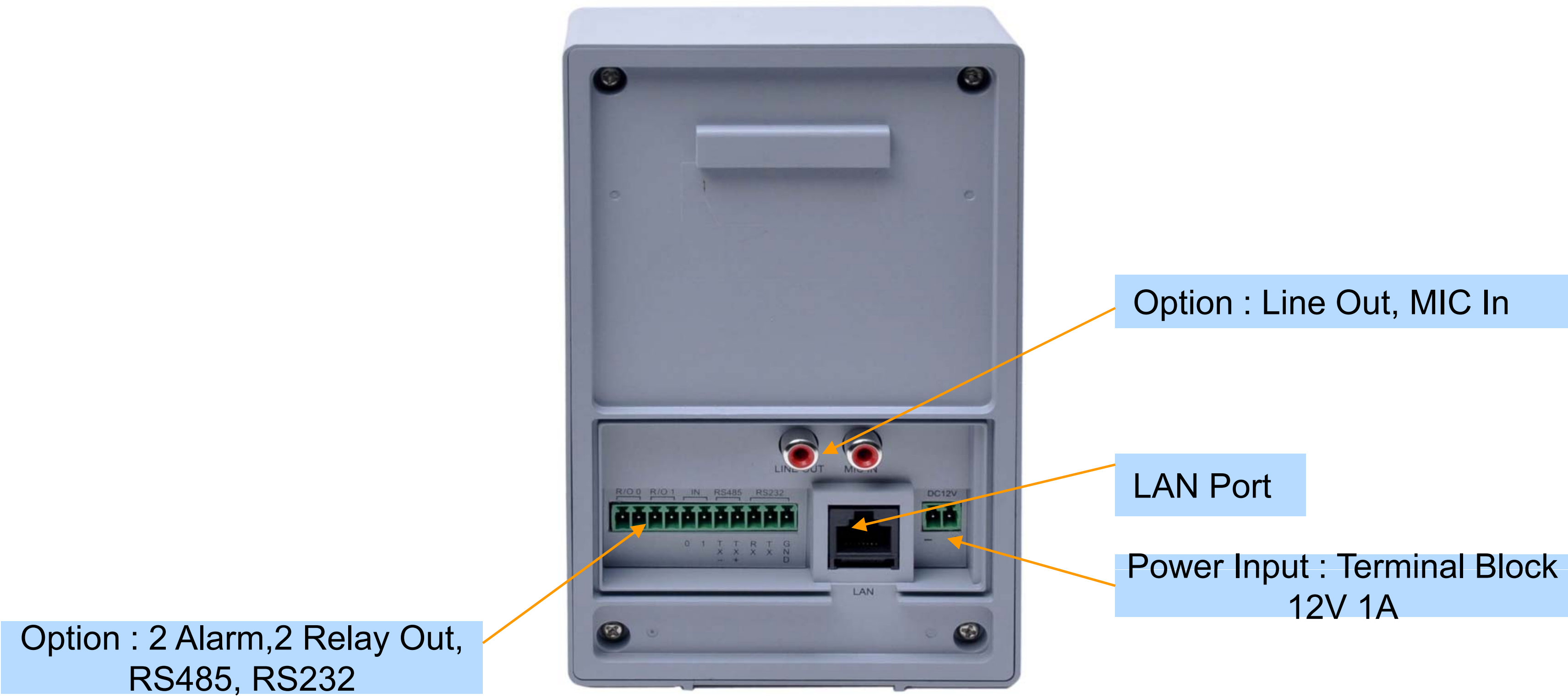
Backside Wall Mount Bracket

Rubber Cover

Hardware Specification

AP-EIP70 SIP Emergency Call Phone

Back Side



Hardware Specification

AP-EIP70 SIP Emergency Call Phone

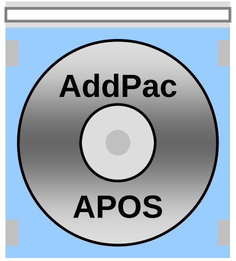
Power Supply

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

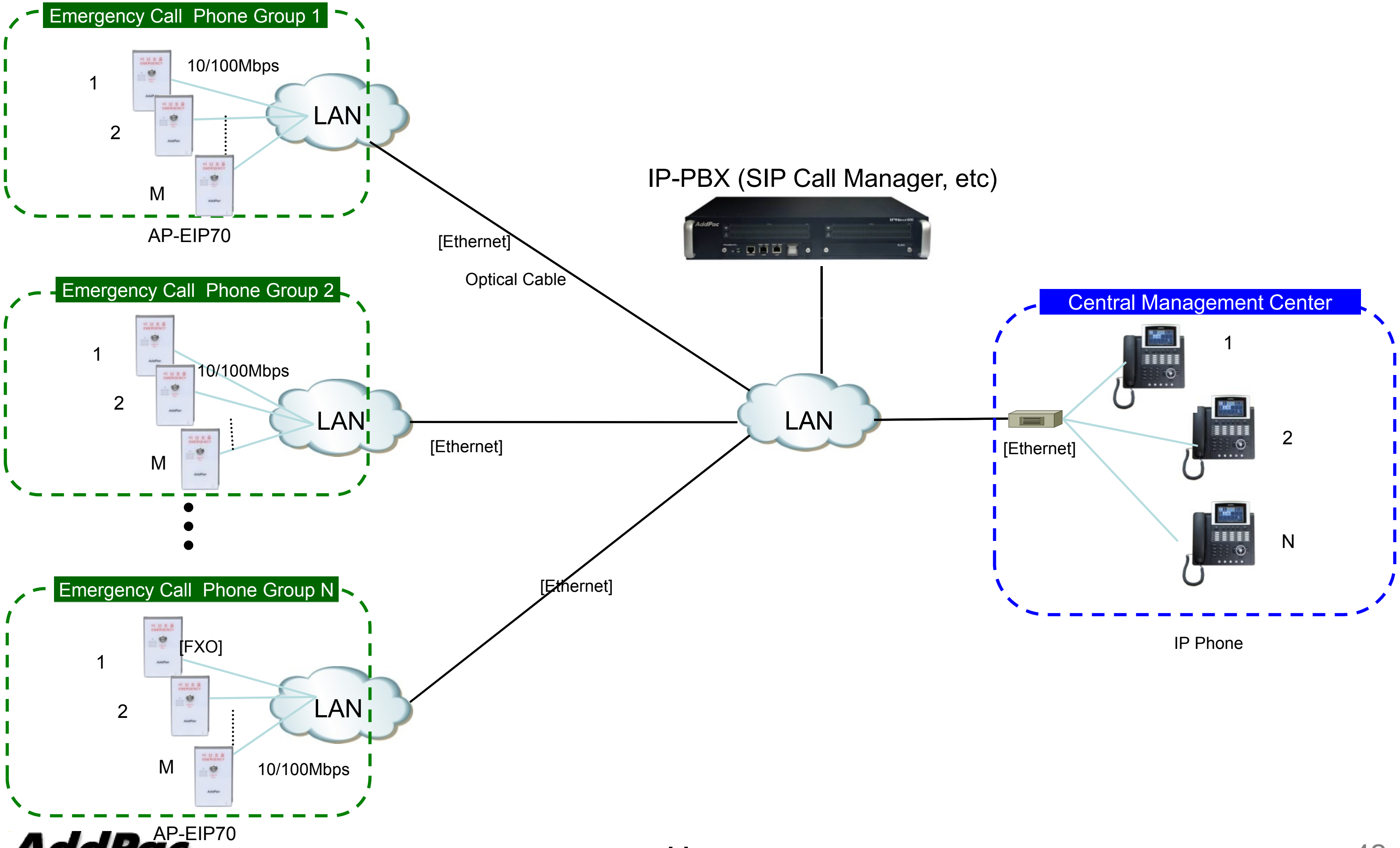
Network Service and Features

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

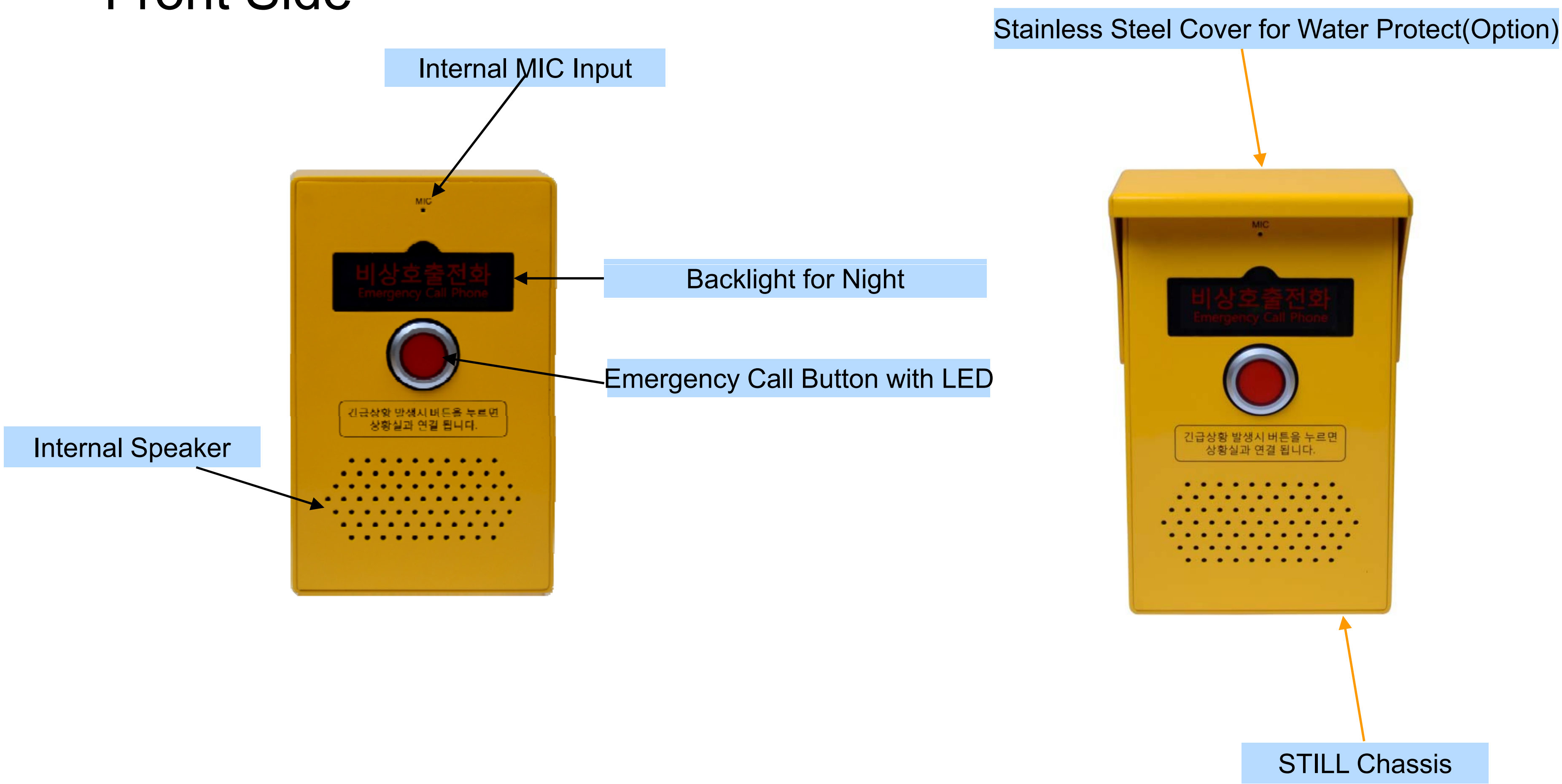
AP-EIP60 SIP Emergency Call Phone

- 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

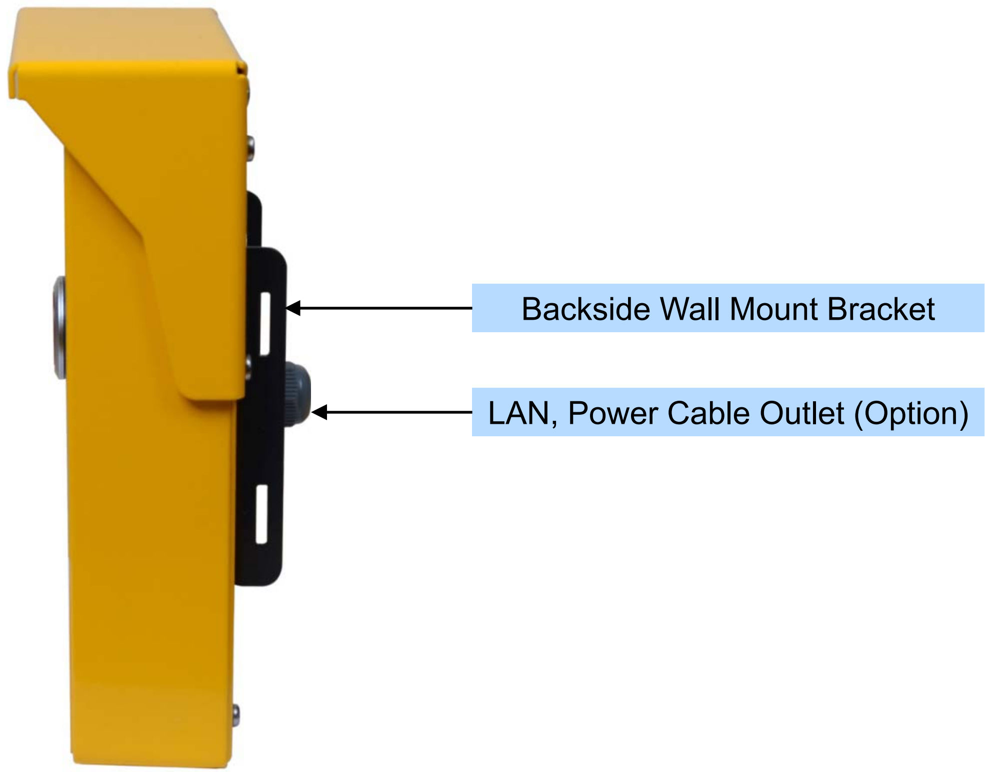
Front Side



Hardware Specification

AP-EIP60 SIP Emergency Call Phone

Back Side



Power Supply

Terminal Block

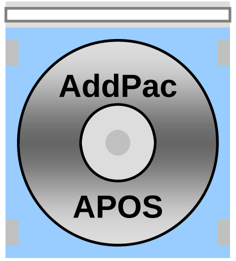
12V 1A Power Adaptor

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

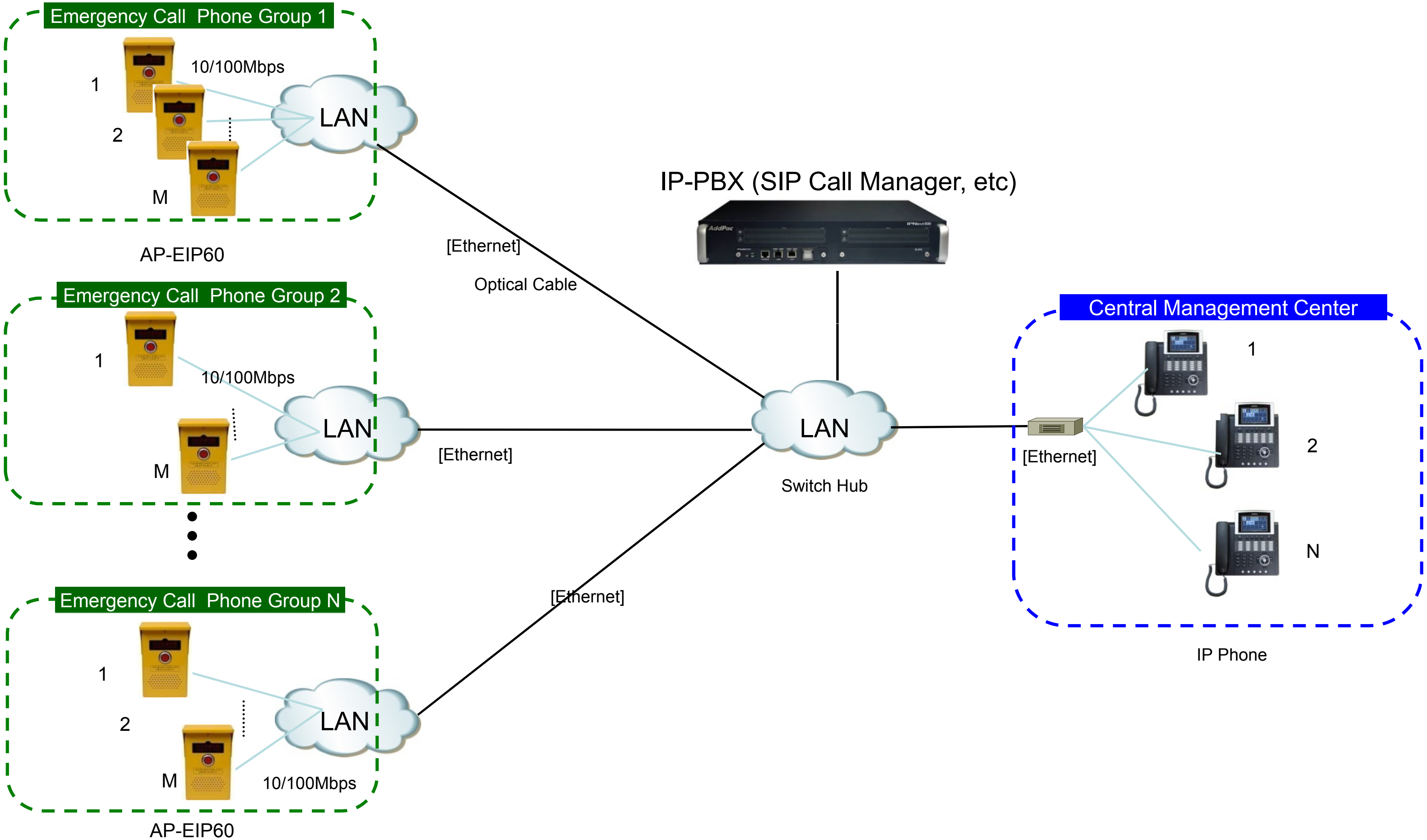
Network Service and Features

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

Pac Technology

AP-IP300 AP-PT20 (40 Speed Dial Key)

Sales and Marketing

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

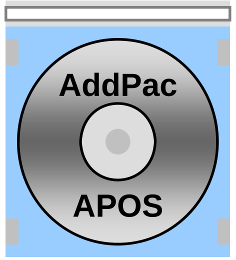
AP-IP300 SIP Broadcasting Phone

- 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

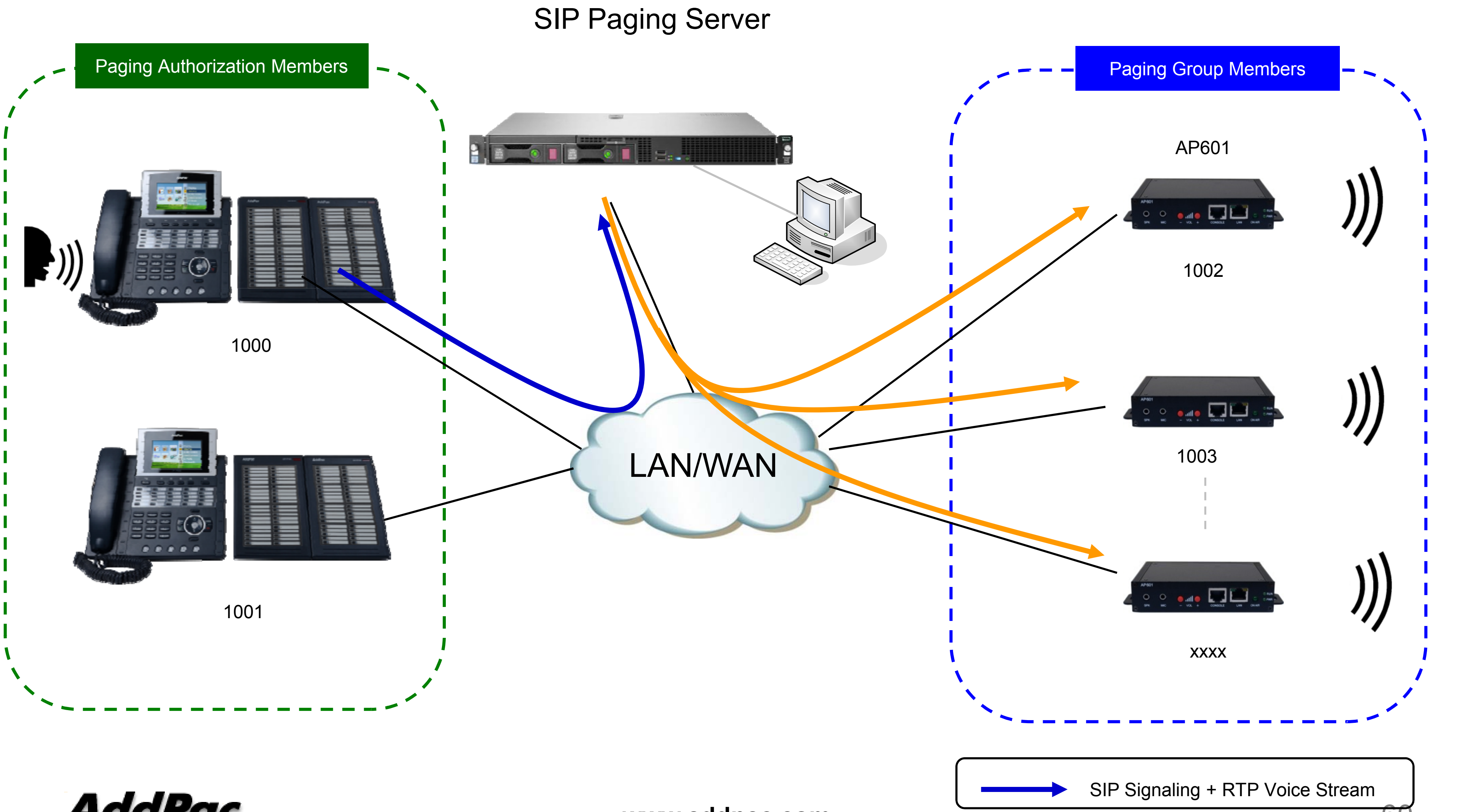
AP-IP300 SIP Broadcasting Phone

- 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)

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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

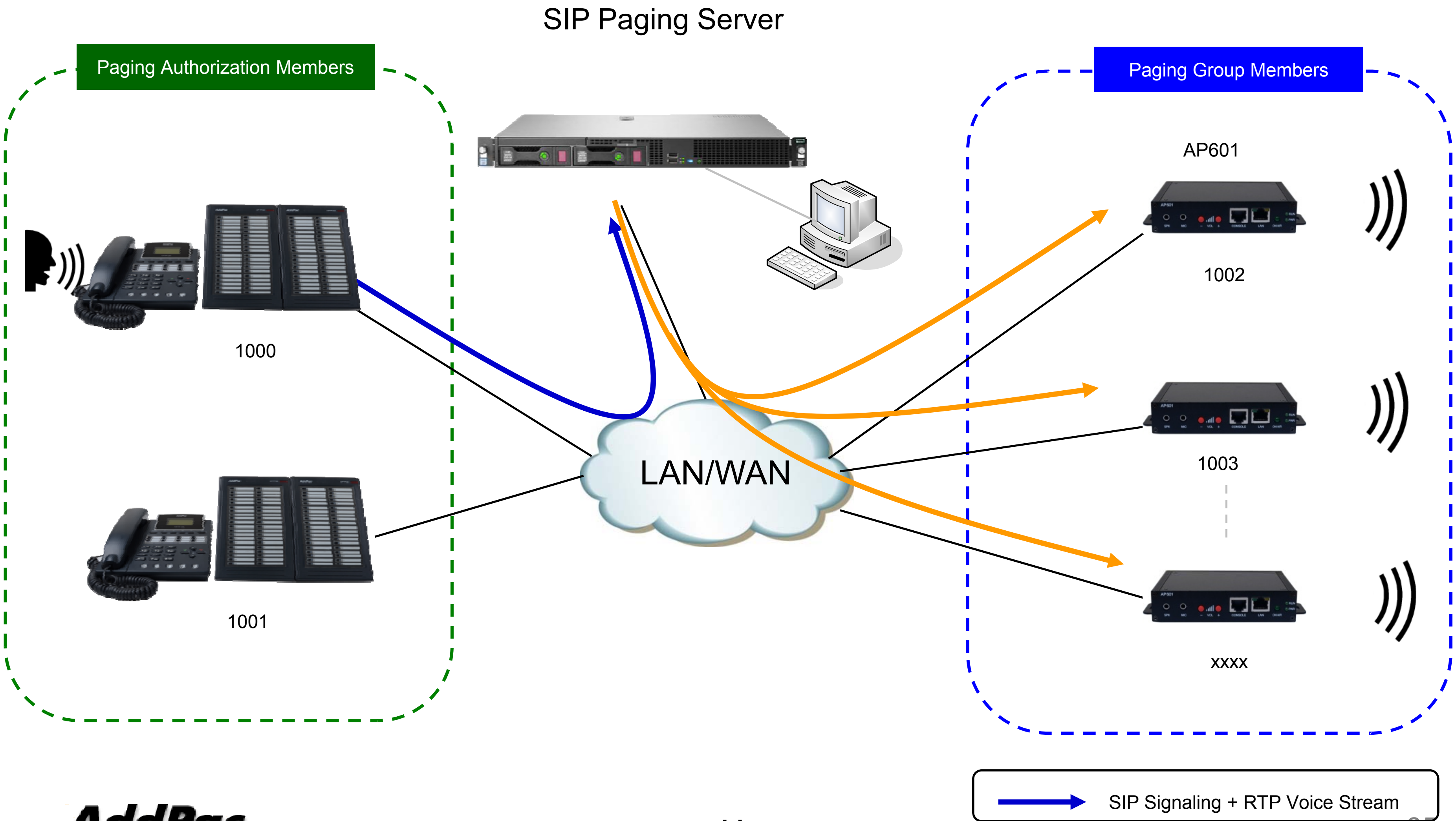
AP-IP120 SIP Broadcasting Phone	Basic Specifications
CPU	RISC Microprocessor
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
PSTN Backup Port (Optional)	1-Port PSTN Backup Port(RJ-11)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	16Mbyte High-speed SDRAM
Power Requirement	External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

Network interface Configurations

Back Side

AP-IP120 SIP Broadcasting Phone

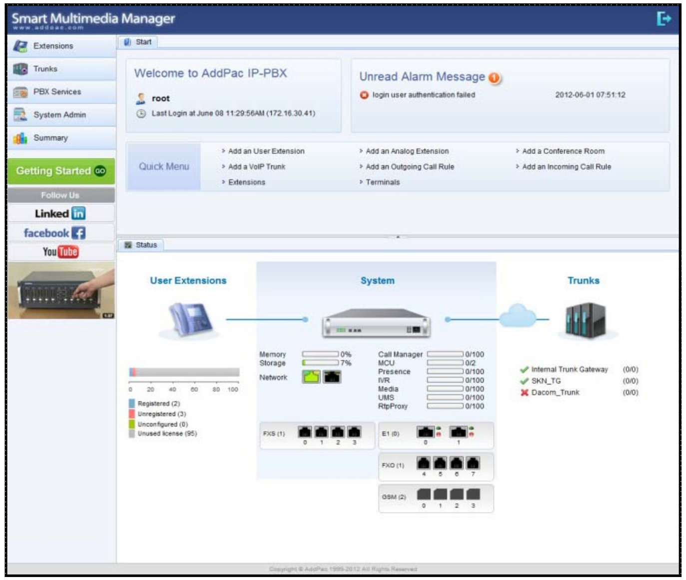
Application Area



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)

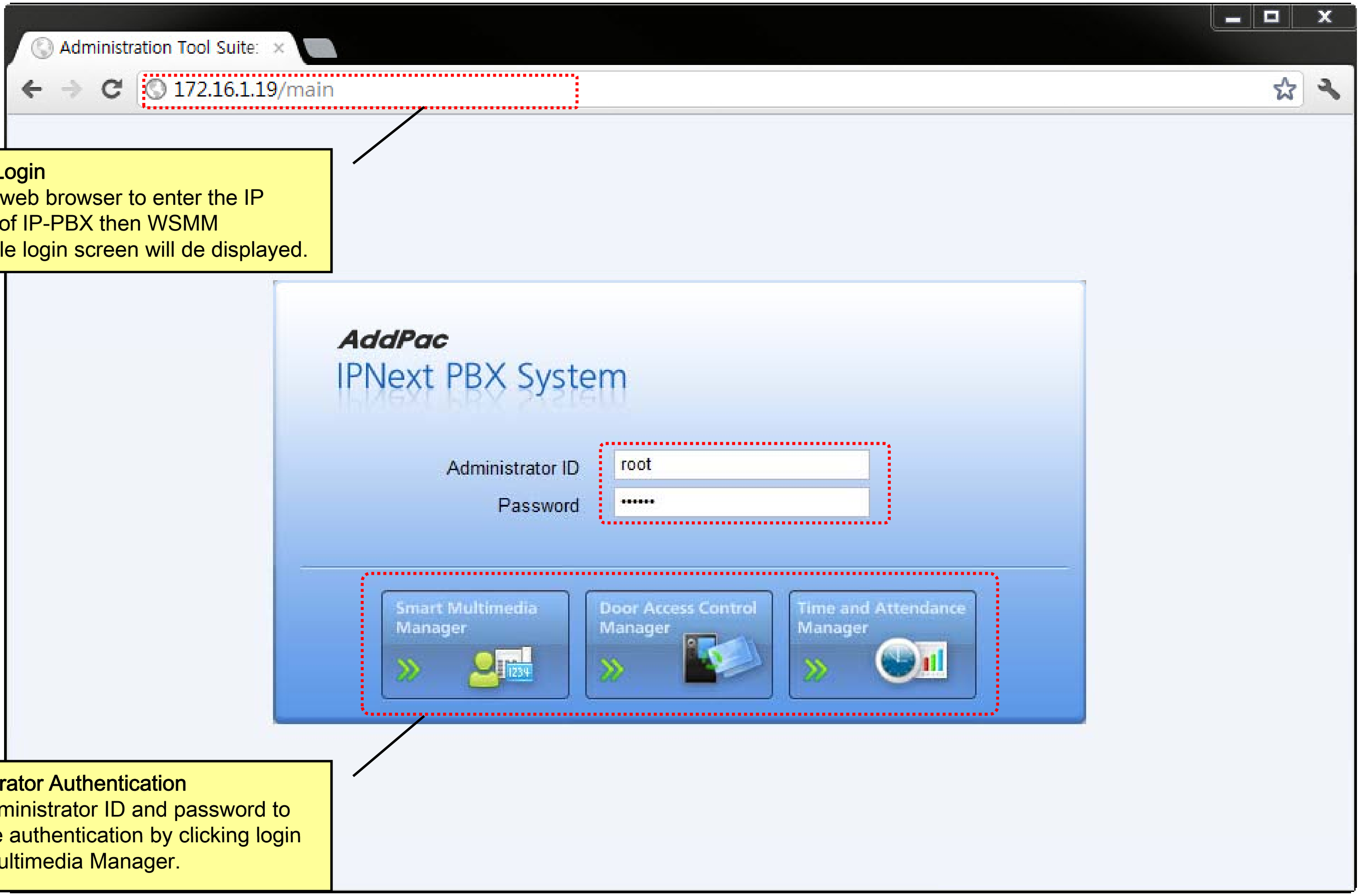
- Simple Menu and Easy Configuration
- Provides Built-In IVR Scenario Editor and Service Configuration
- Provides easy-to-user IP-PBX System API Services and ways to integration with 3rd party systems
- Integrated voice line management such as FXS, FXO, E1, GSM, 3G
- User portal to configure personal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

System Requirement

WSMM (Web based Smart Multimedia Manager)

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

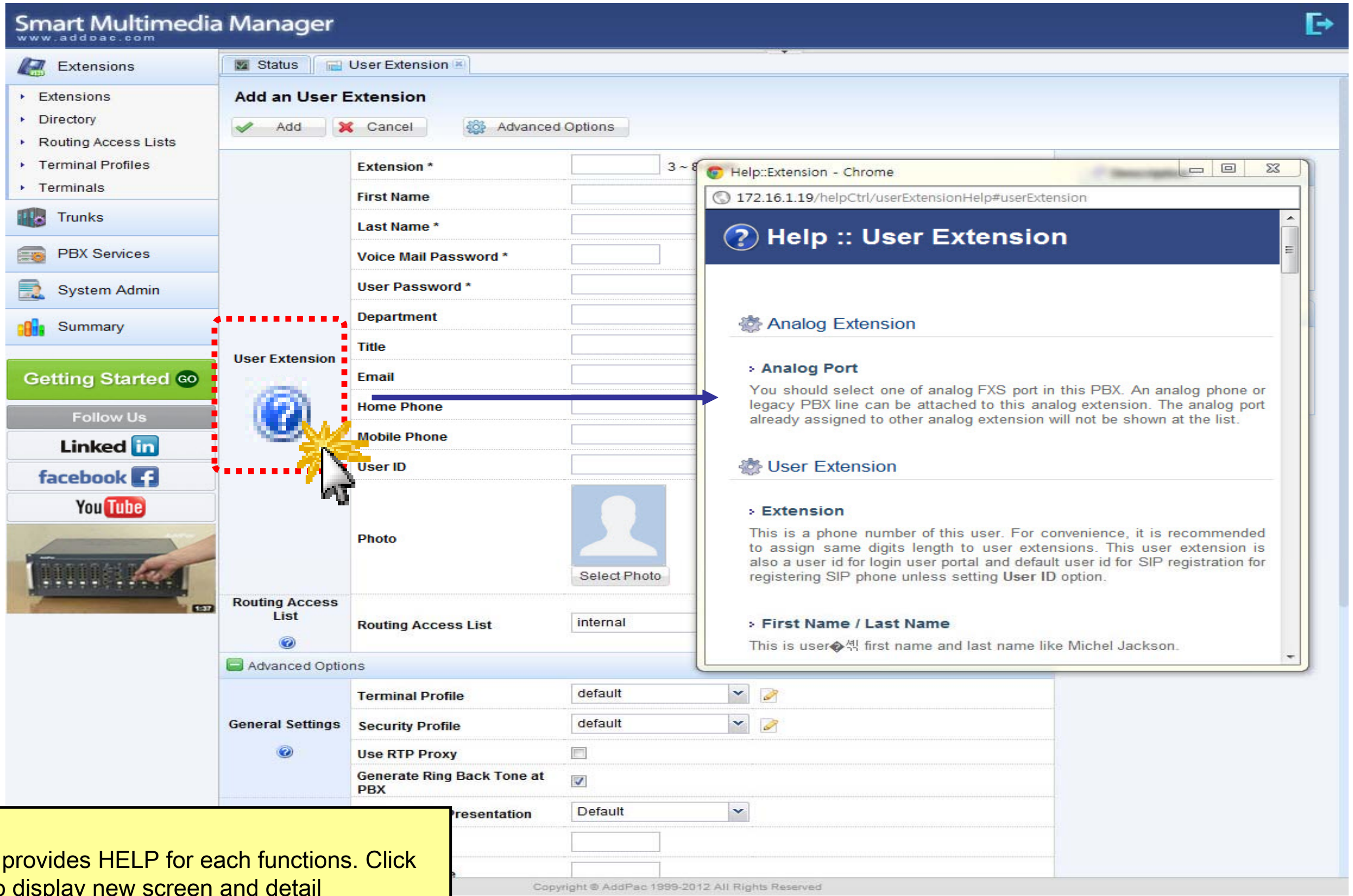
Login



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

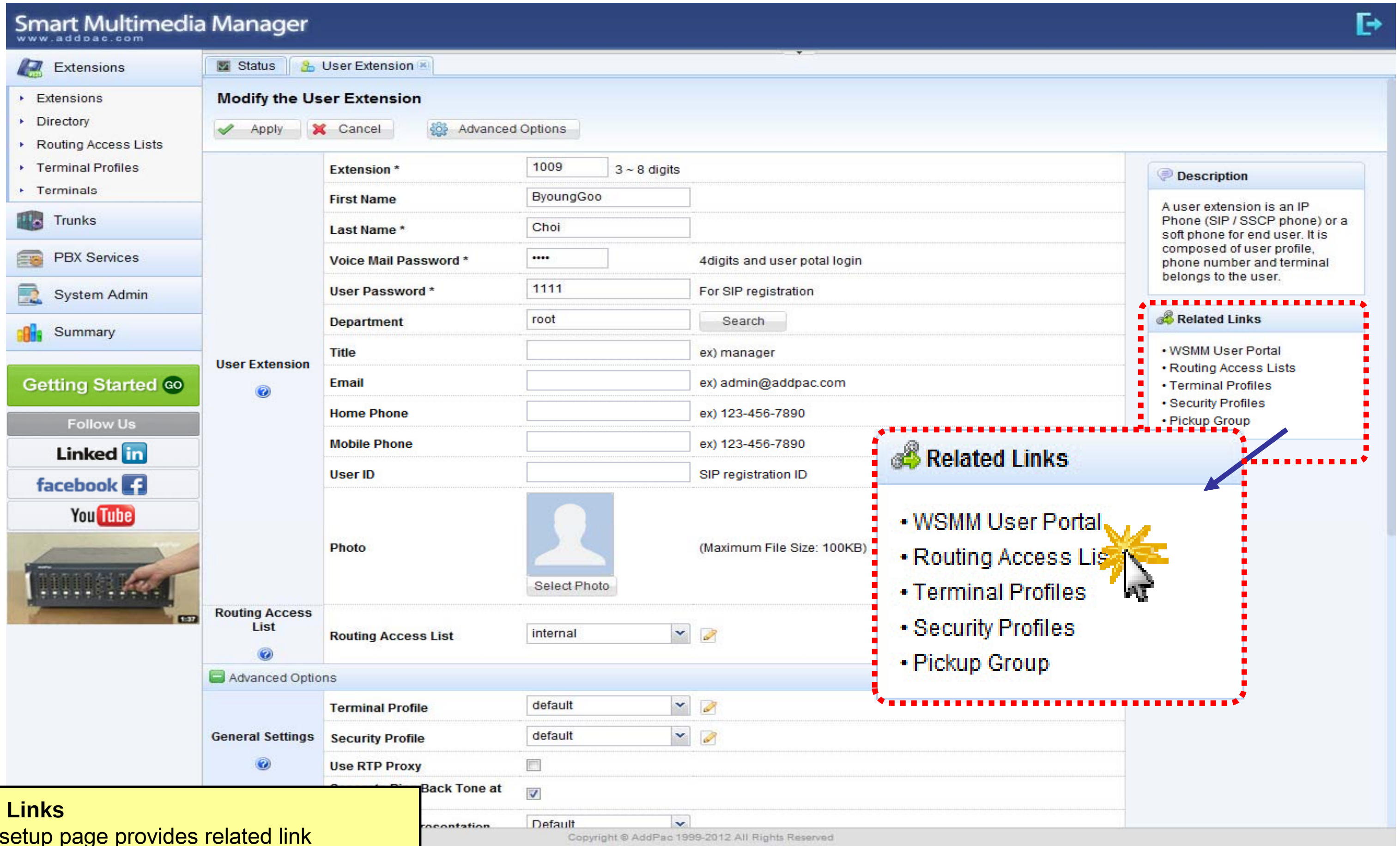
Administrator Authentication
Enter administrator ID and password to complete authentication by clicking login Smart Multimedia Manager.

Help



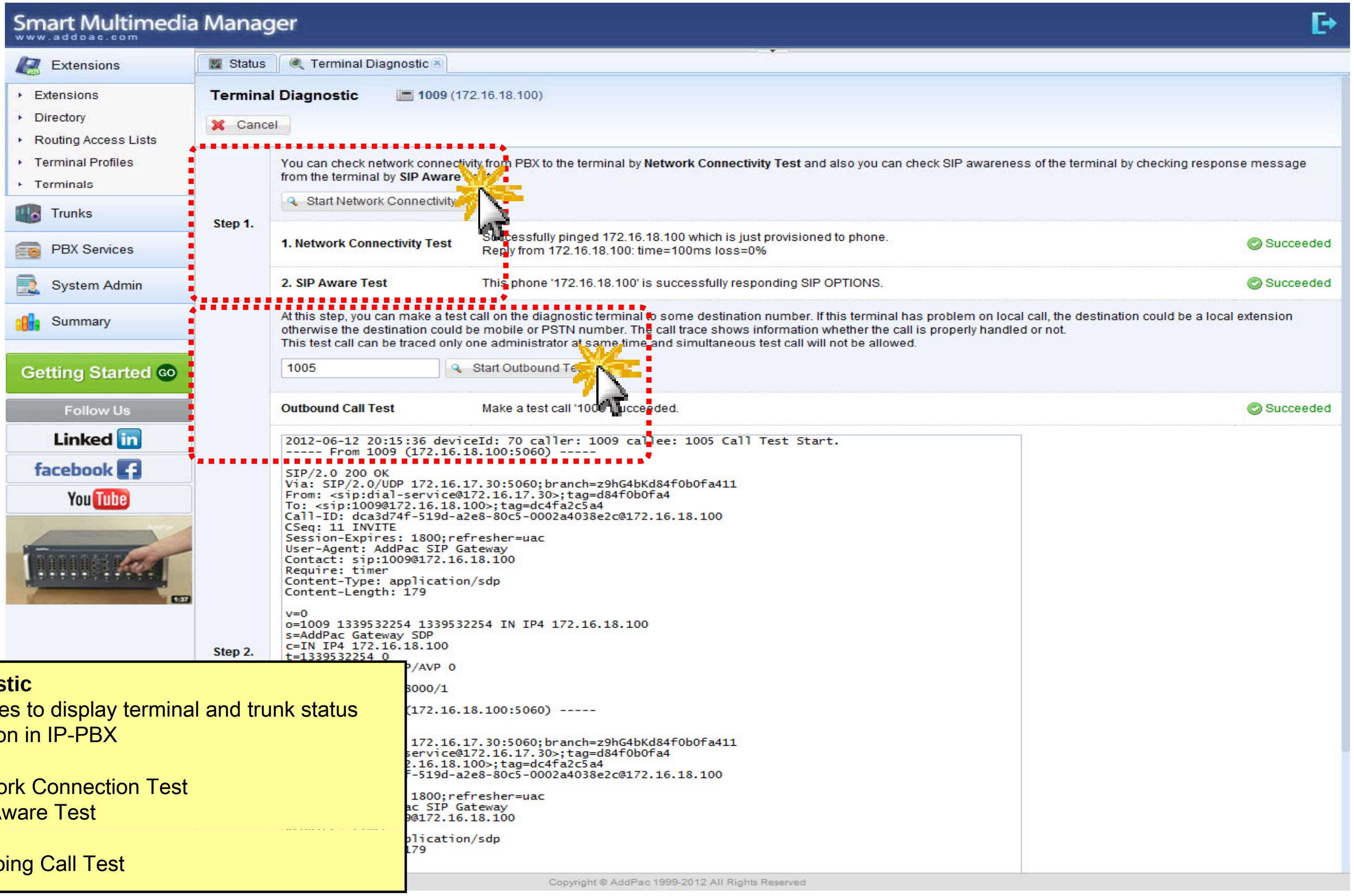
HELP
WSMM provides HELP for each functions. Click HELP to display new screen and detail description of setup is clearly explained in homepage.

Related Links



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

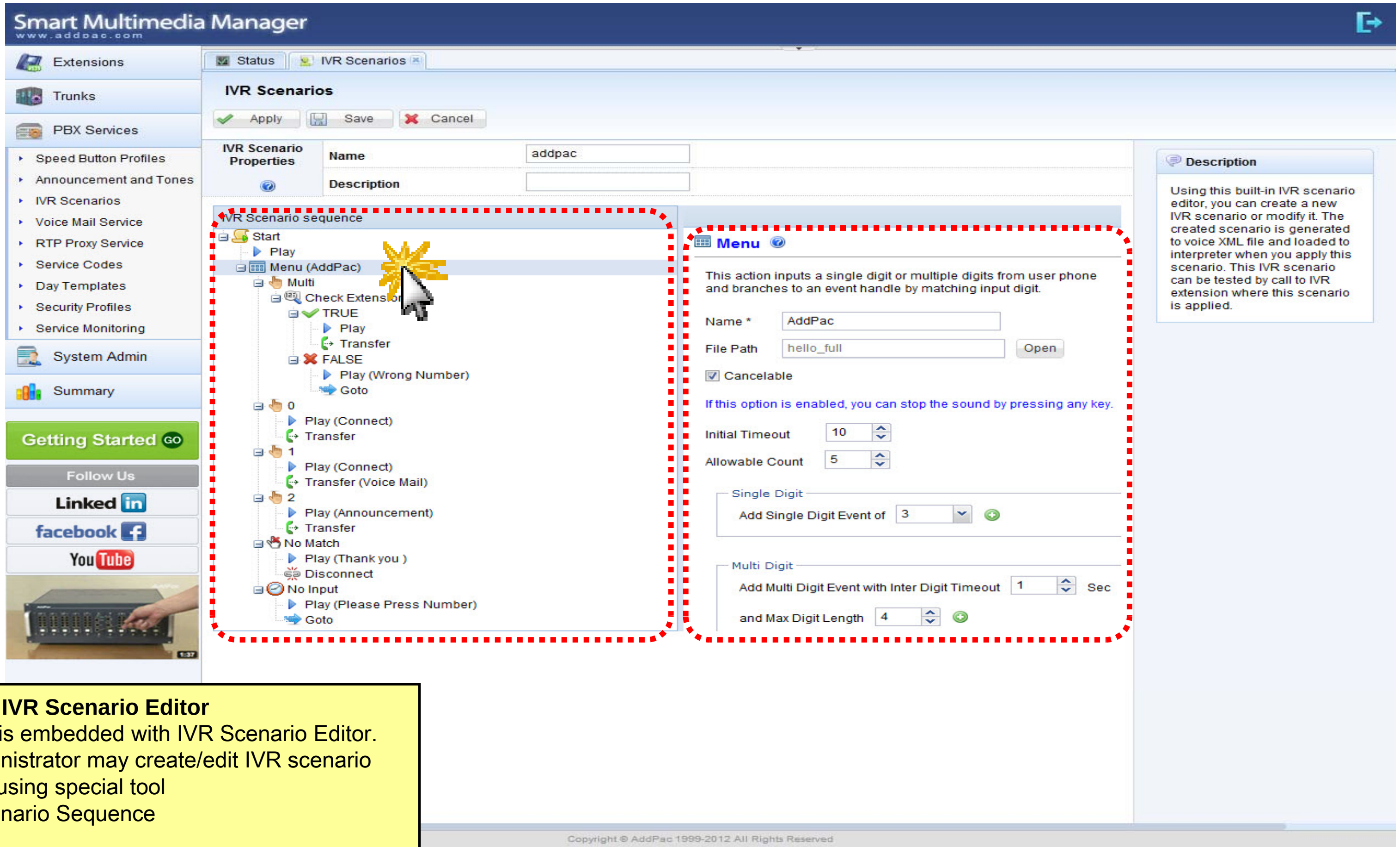
Diagnostic



Diagnostic
It provides to display terminal and trunk status inspection in IP-PBX
Step 1.
- Network Connection Test
- SIP Aware Test

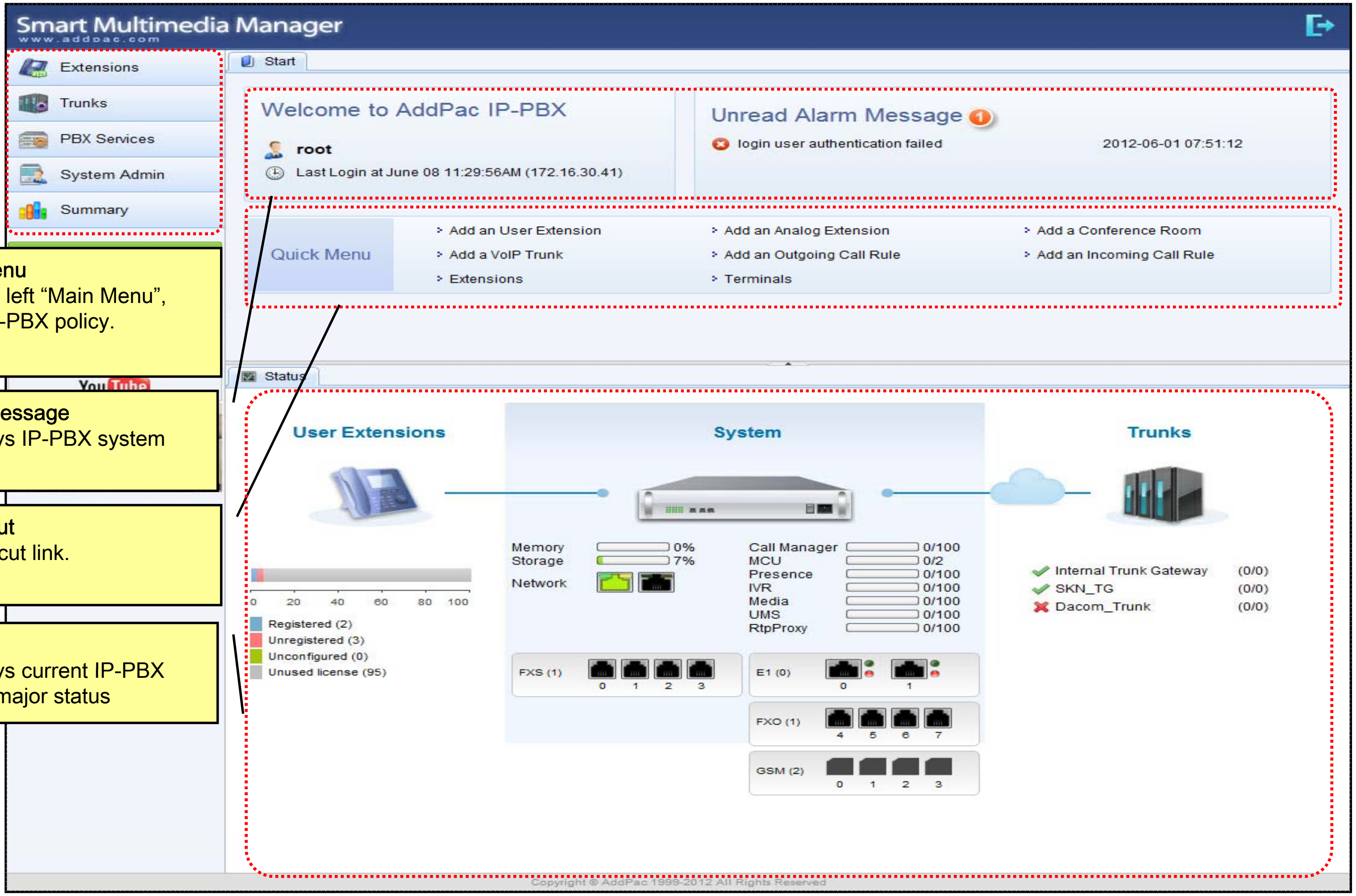
Step 2.
- Outgoing Call Test

Built-in IVR Scenario Editor



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



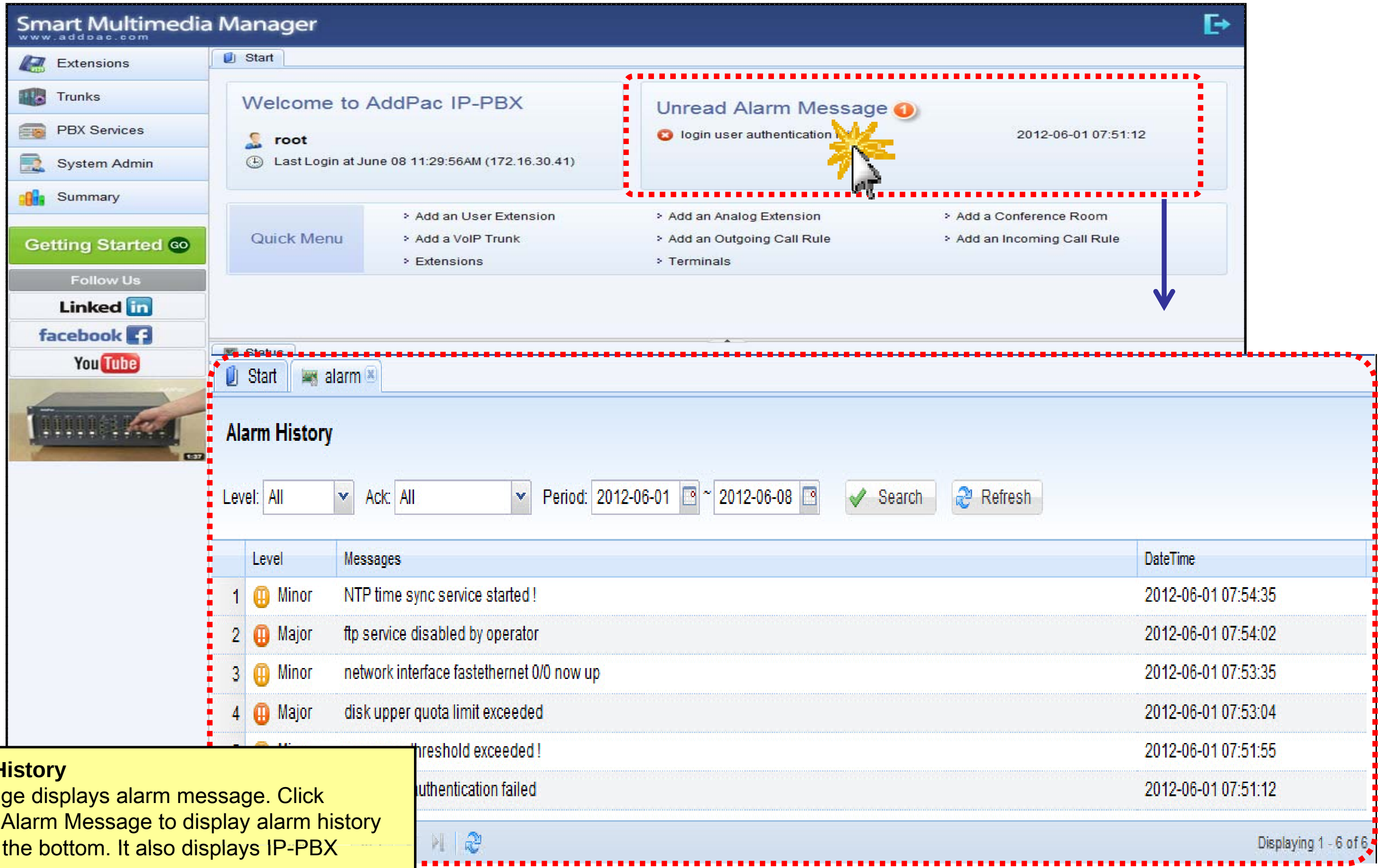
Main Menu
Through left "Main Menu", setup IP-PBX policy.

Alarm Message
It displays IP-PBX system errors

Short Cut
A short cut link.

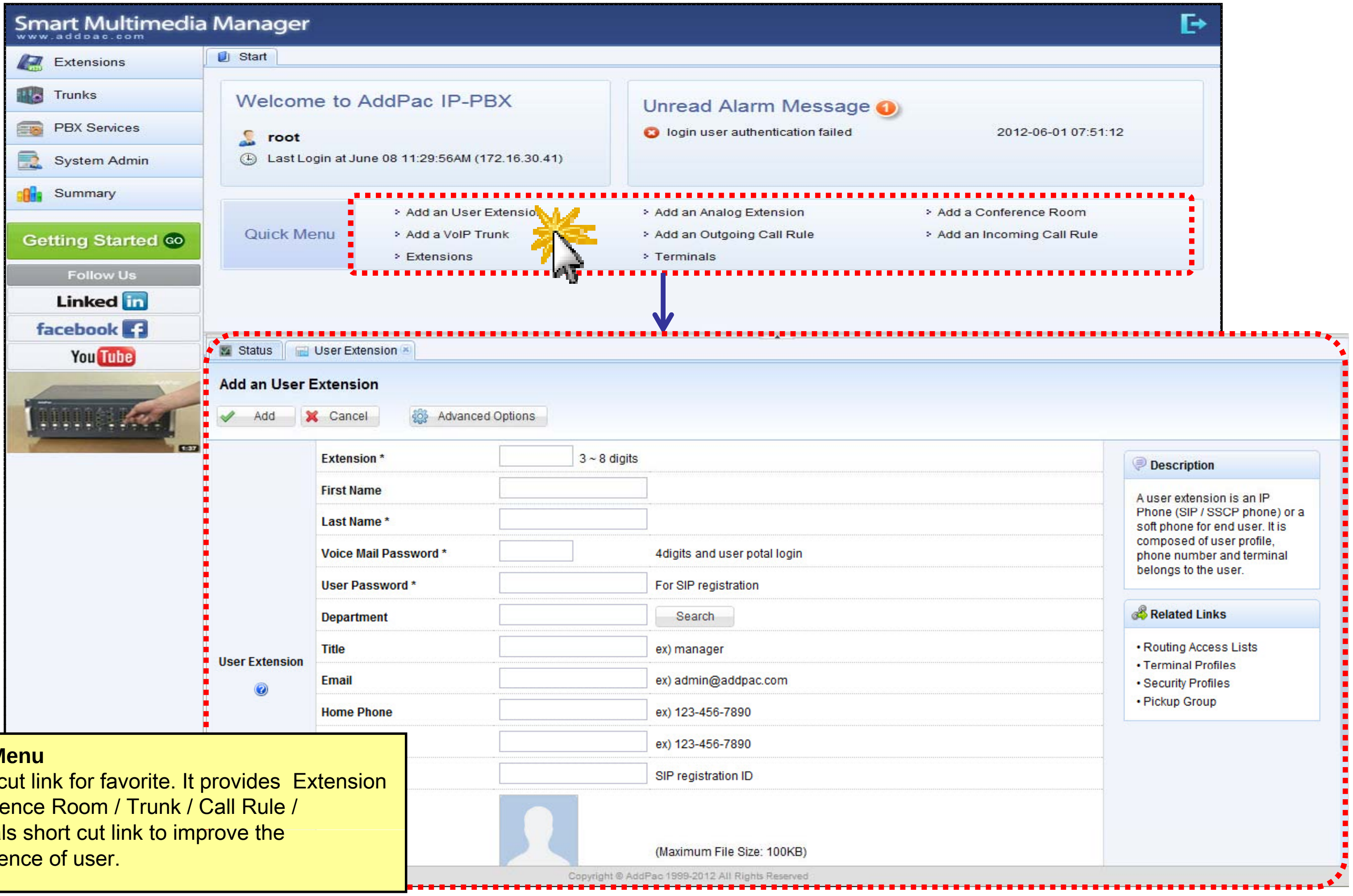
Status
It displays current IP-PBX system major status

Main - Alarm History



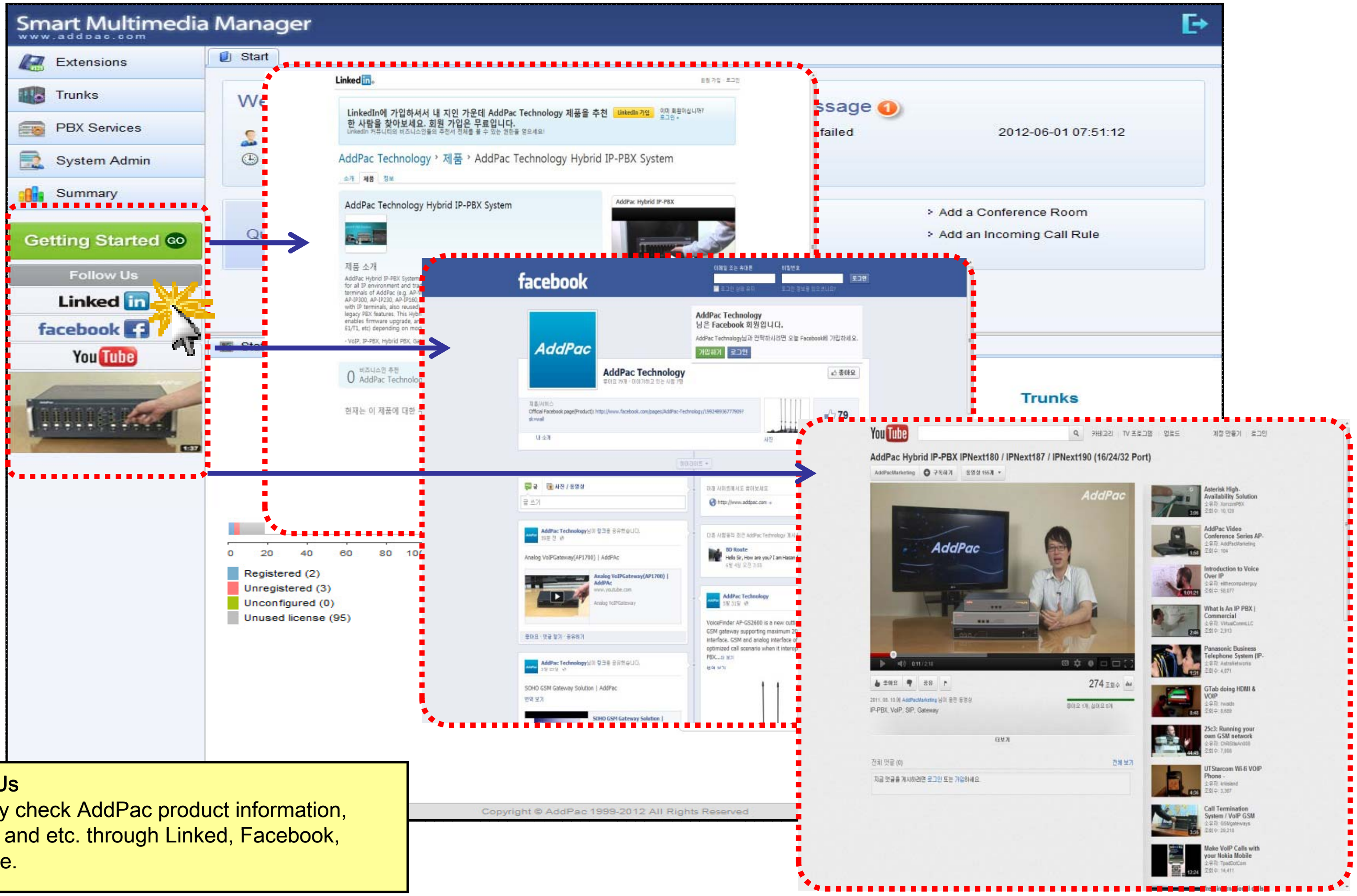
Alarm History
Main page displays alarm message. Click Unread Alarm Message to display alarm history page at the bottom. It also displays IP-PBX system errors.

Main – Quick Menu



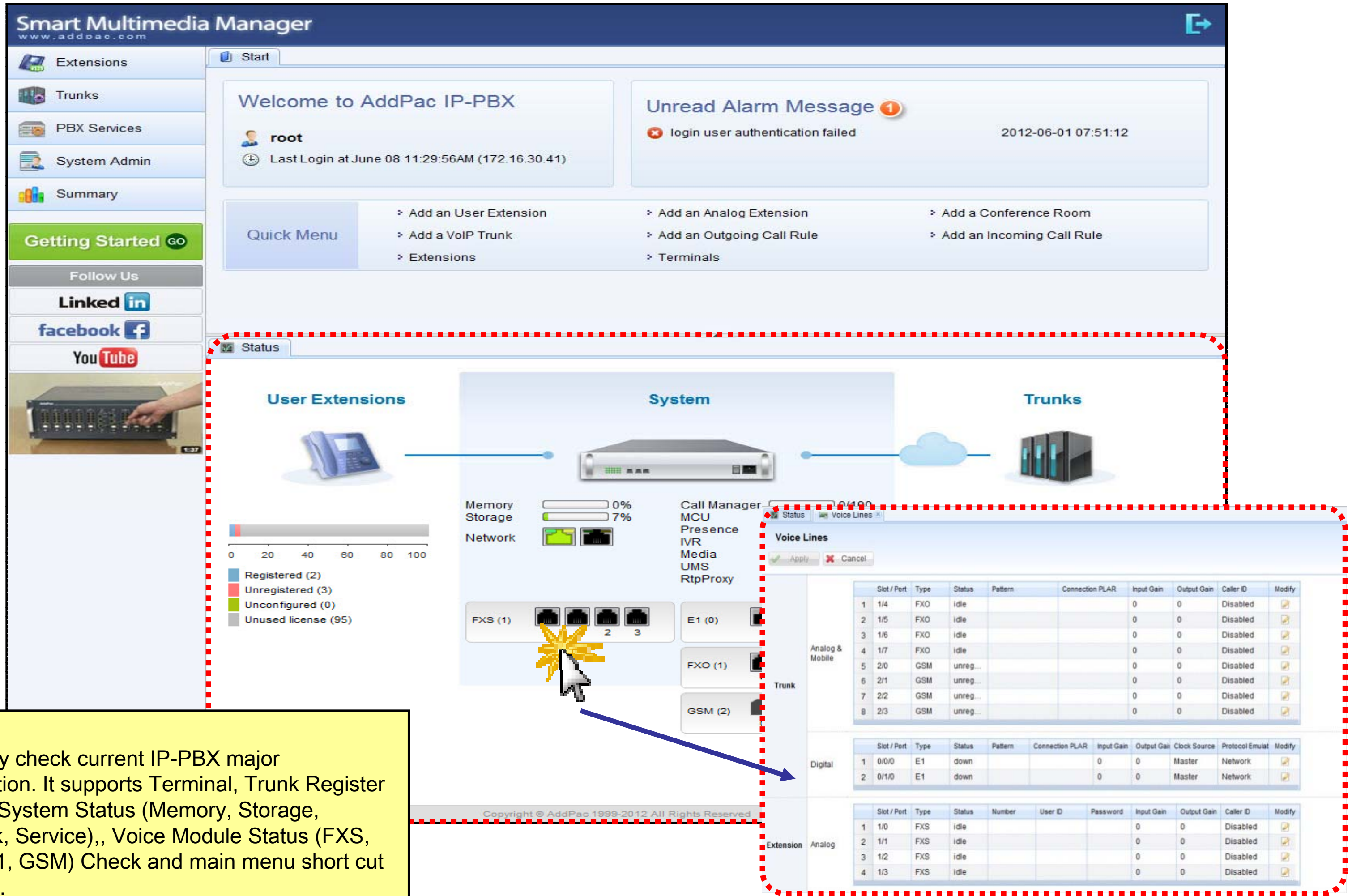
Quick Menu
A short cut link for favorite. It provides Extension / Conference Room / Trunk / Call Rule / Terminals short cut link to improve the convenience of user.

Main – Follow Us



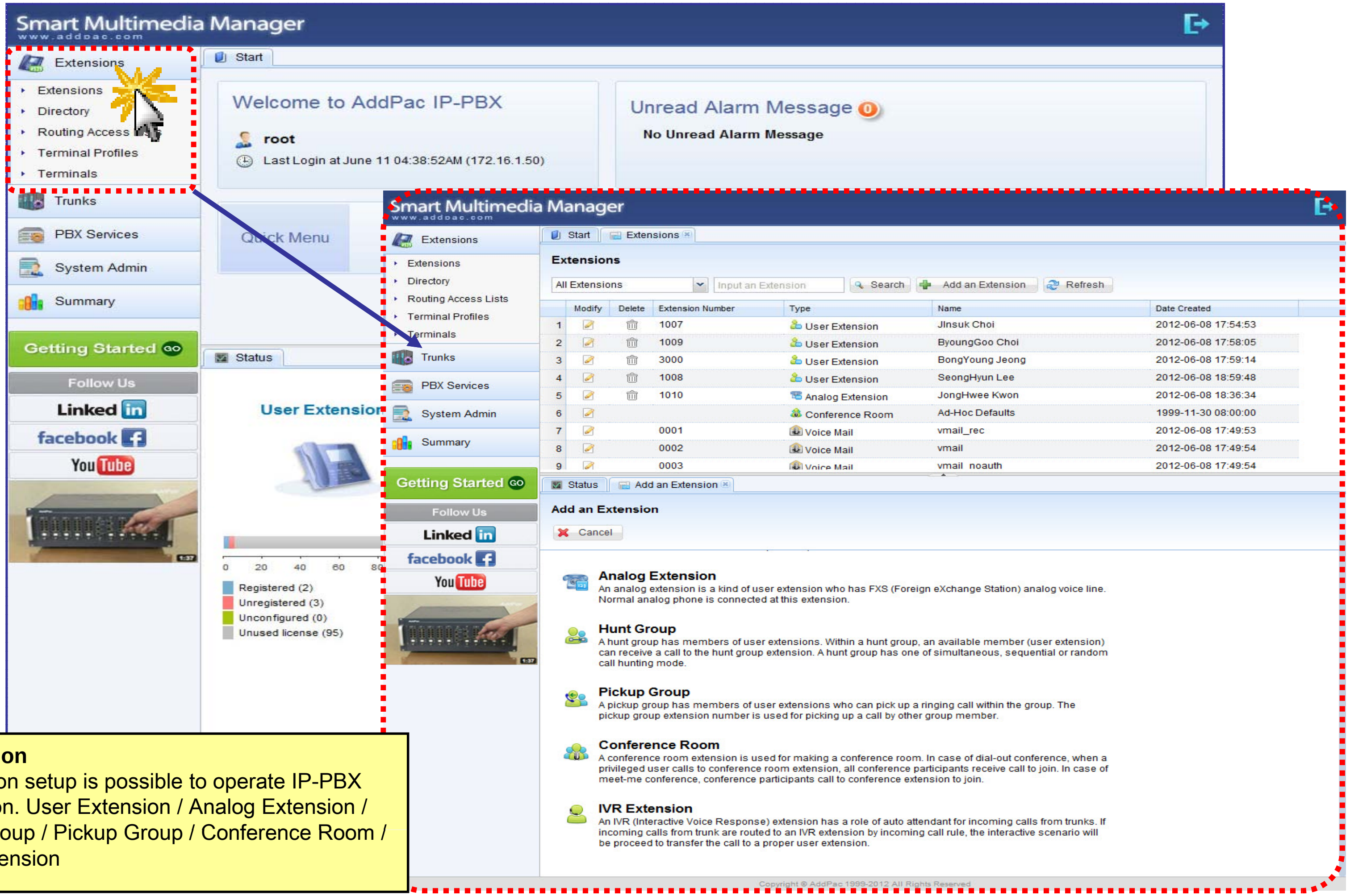
Follow Us
You may check AddPac product information, solution and etc. through Linked, Facebook, YouTube.

Main – Status Monitoring



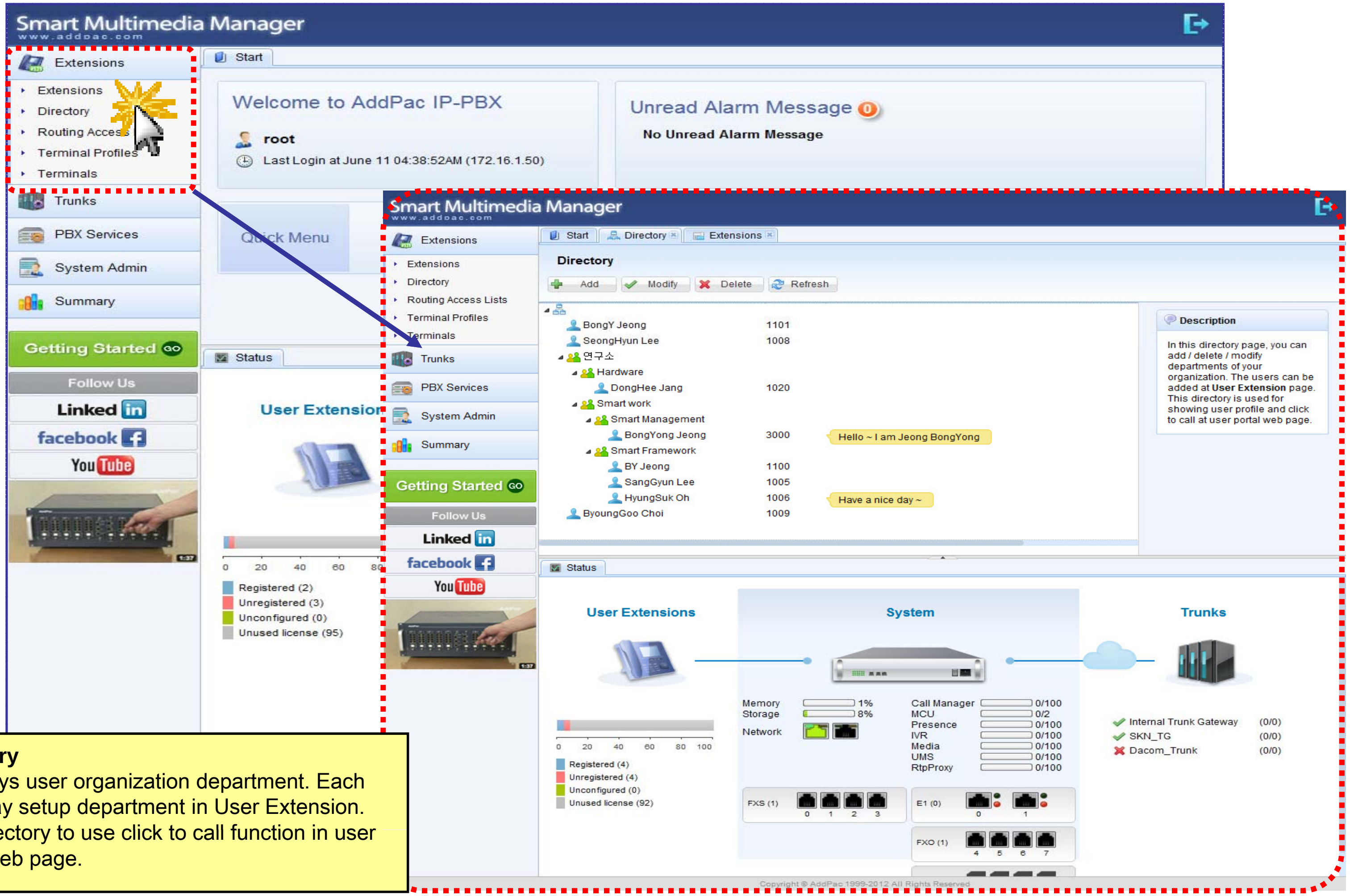
Status
You may check current IP-PBX major information. It supports Terminal, Trunk Register Status, System Status (Memory, Storage, Network, Service),, Voice Module Status (FXS, FXO, E1, GSM) Check and main menu short cut function.

Extension - Extensions



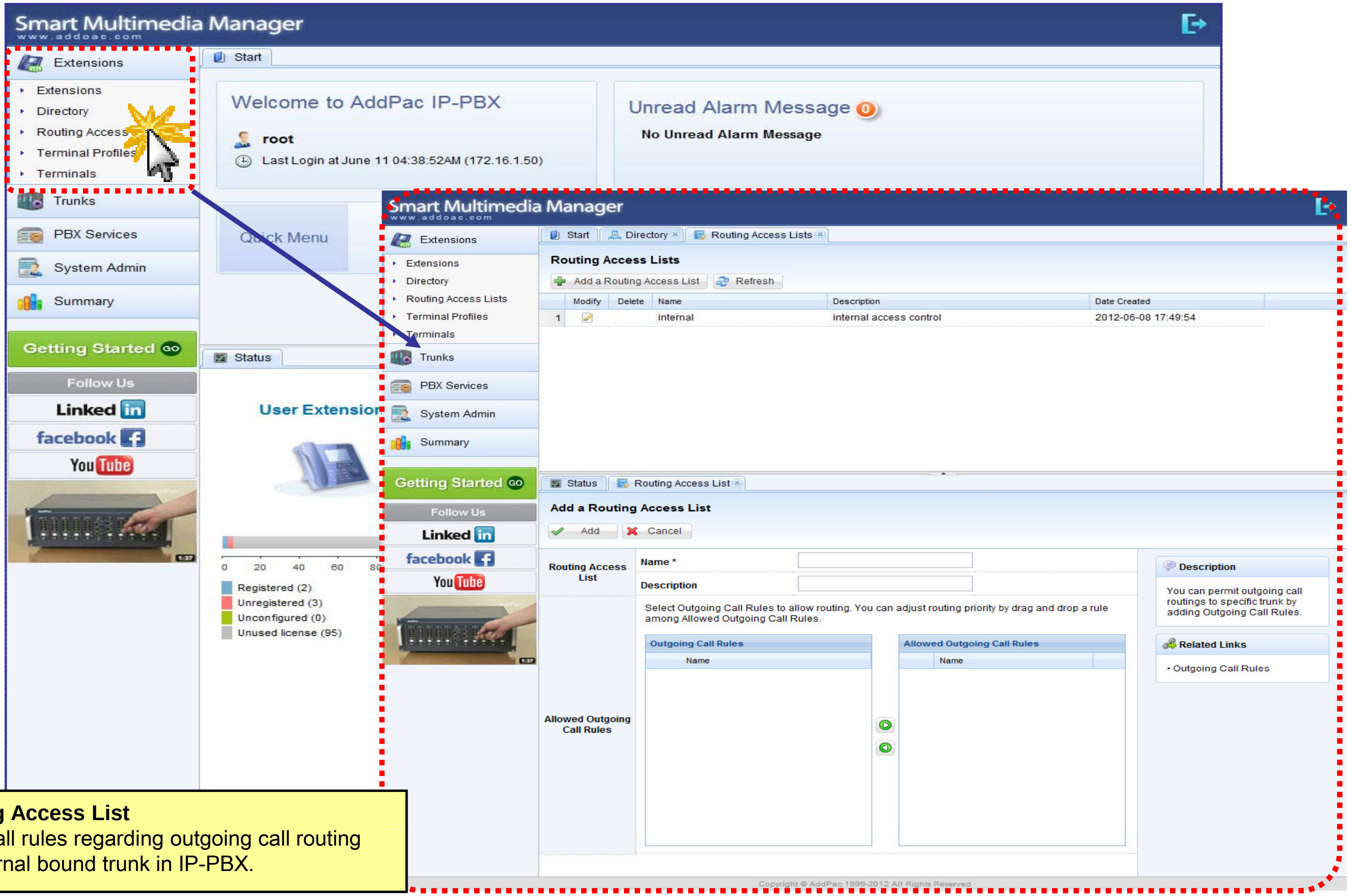
Extension
Extension setup is possible to operate IP-PBX operation. User Extension / Analog Extension / Hunt Group / Pickup Group / Conference Room / IVR Extension

Extension - Directory



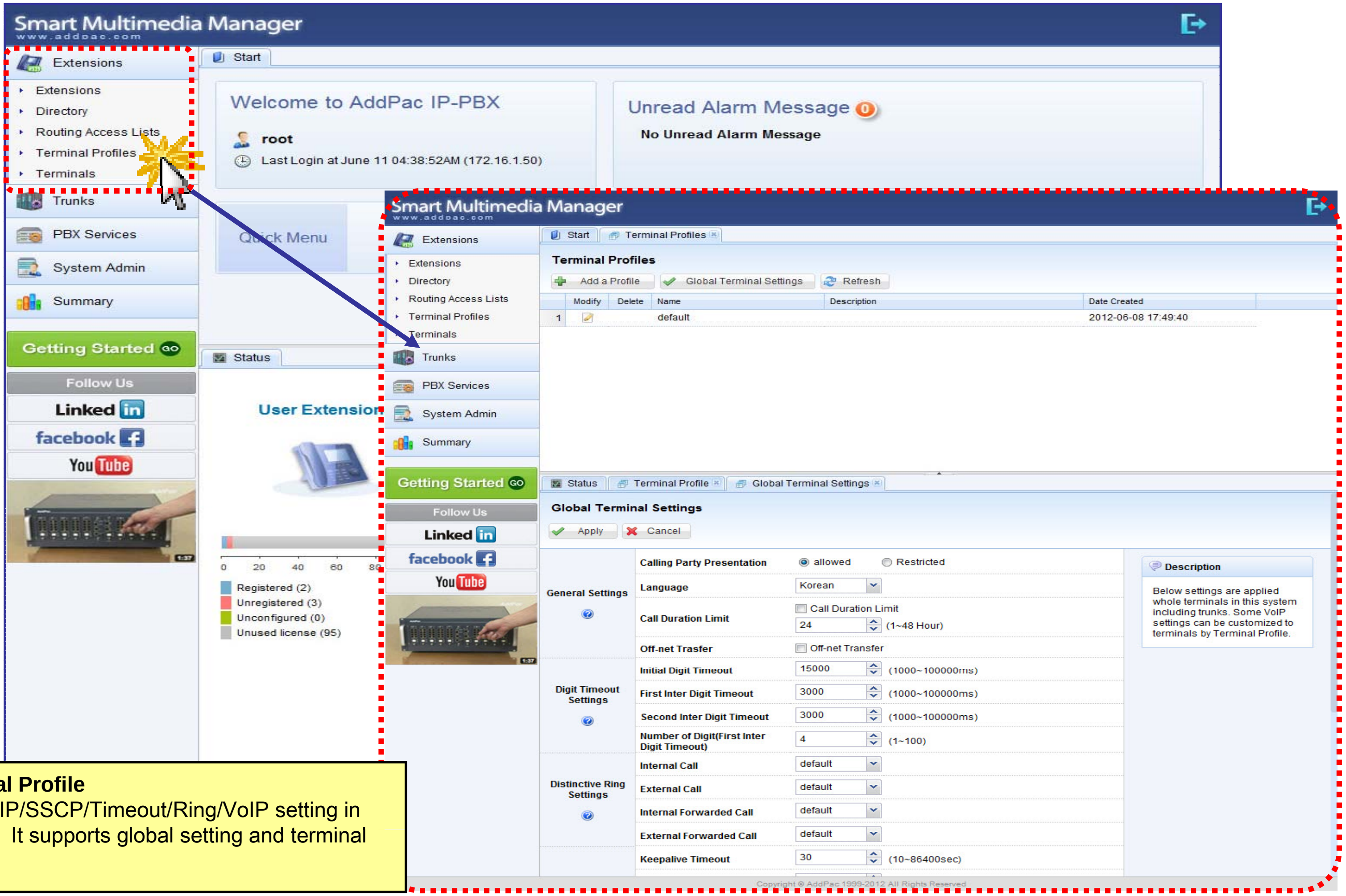
Directory
It displays user organization department. Each user may setup department in User Extension. Use directory to use click to call function in user portal web page.

Extension - Routing Access List



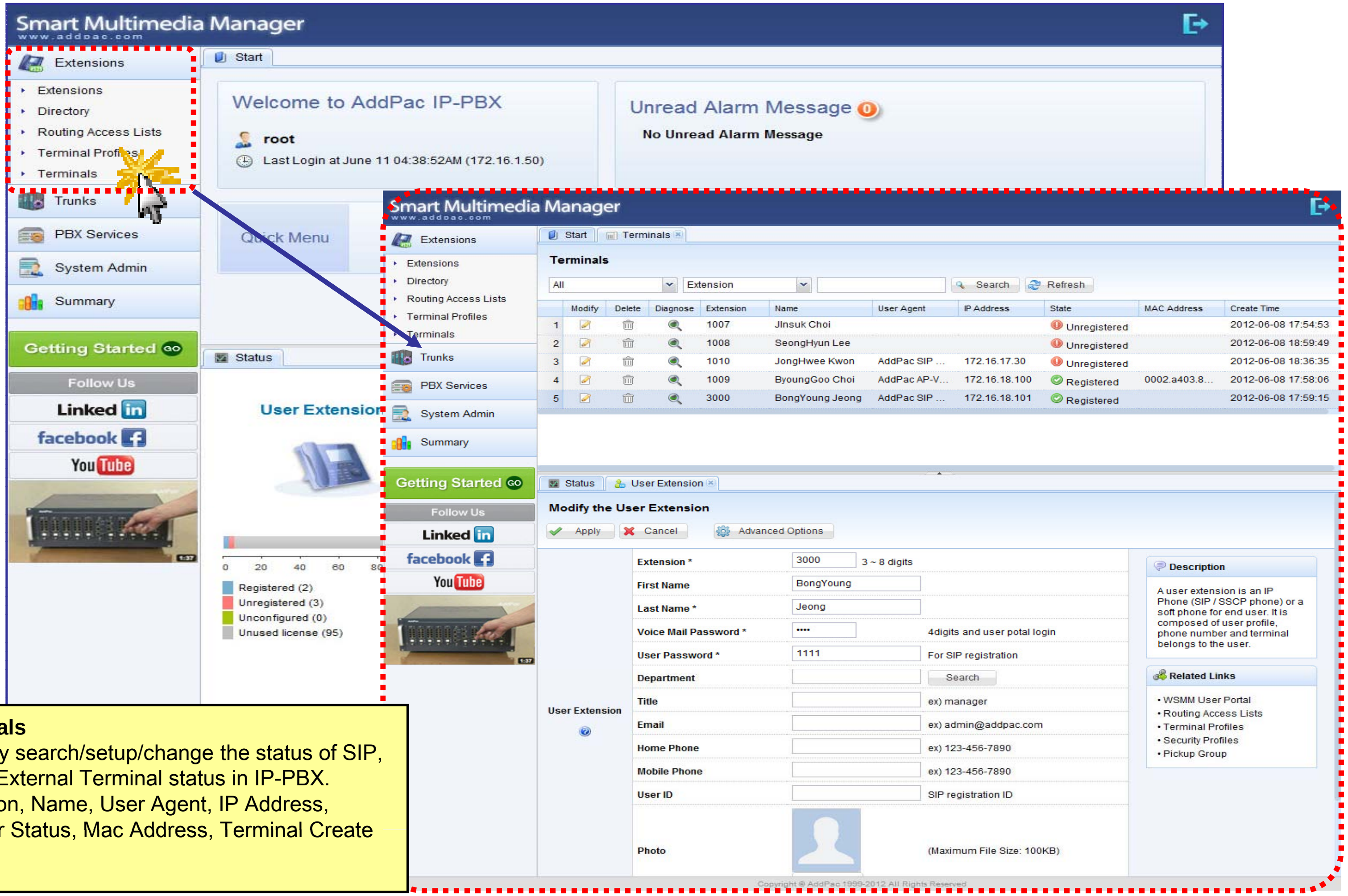
Routing Access List
Apply call rules regarding outgoing call routing for external bound trunk in IP-PBX.

Extension - Terminal Profile



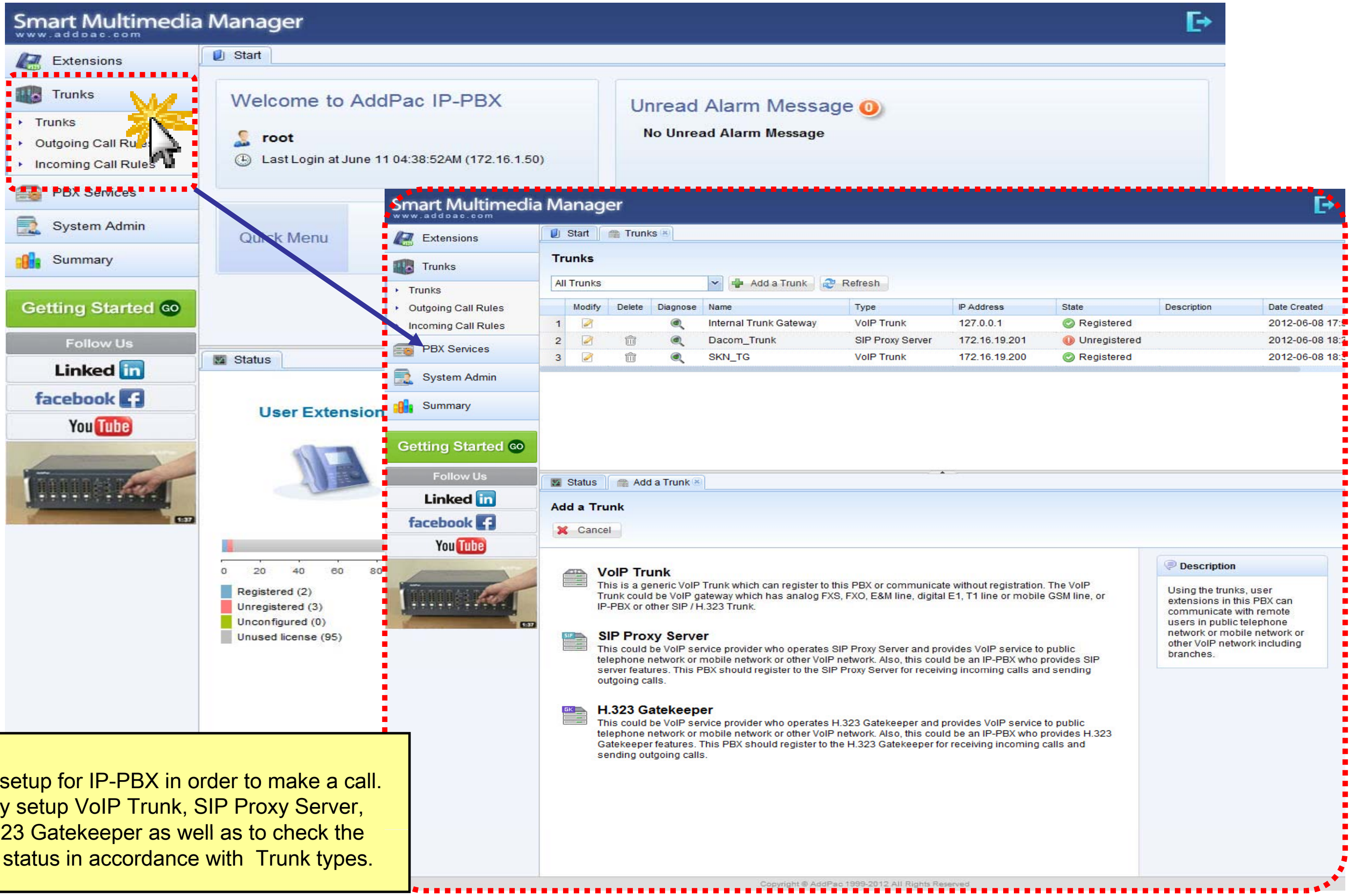
Terminal Profile
Setup SIP/SSCP/Timeout/Ring/VoIP setting in IP-PBX. It supports global setting and terminal profile.

Extension - Terminals



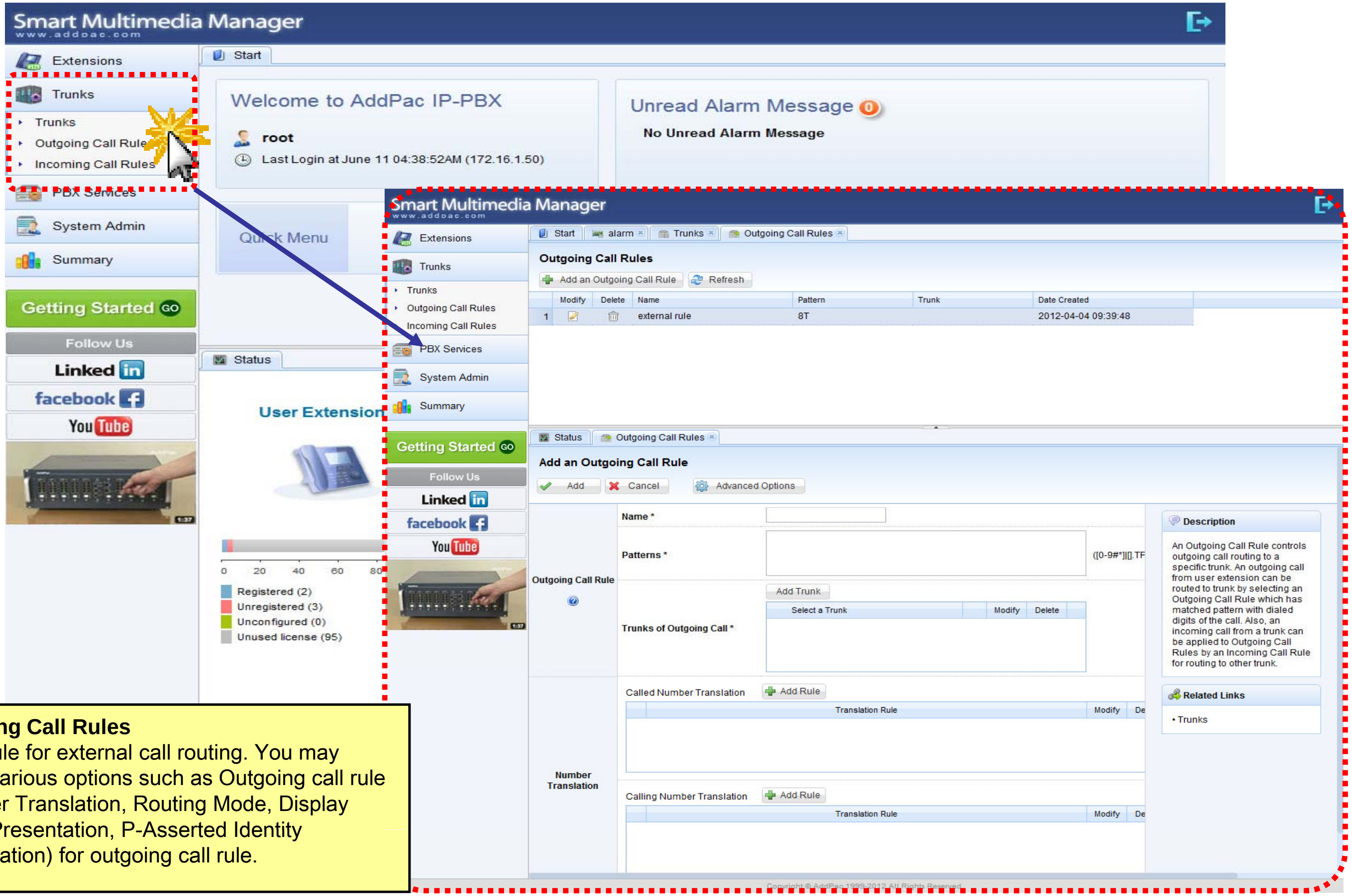
Terminals
You may search/setup/change the status of SIP, SSCP, External Terminal status in IP-PBX. Extension, Name, User Agent, IP Address, Register Status, Mac Address, Terminal Create Time

Trunk - Trunks



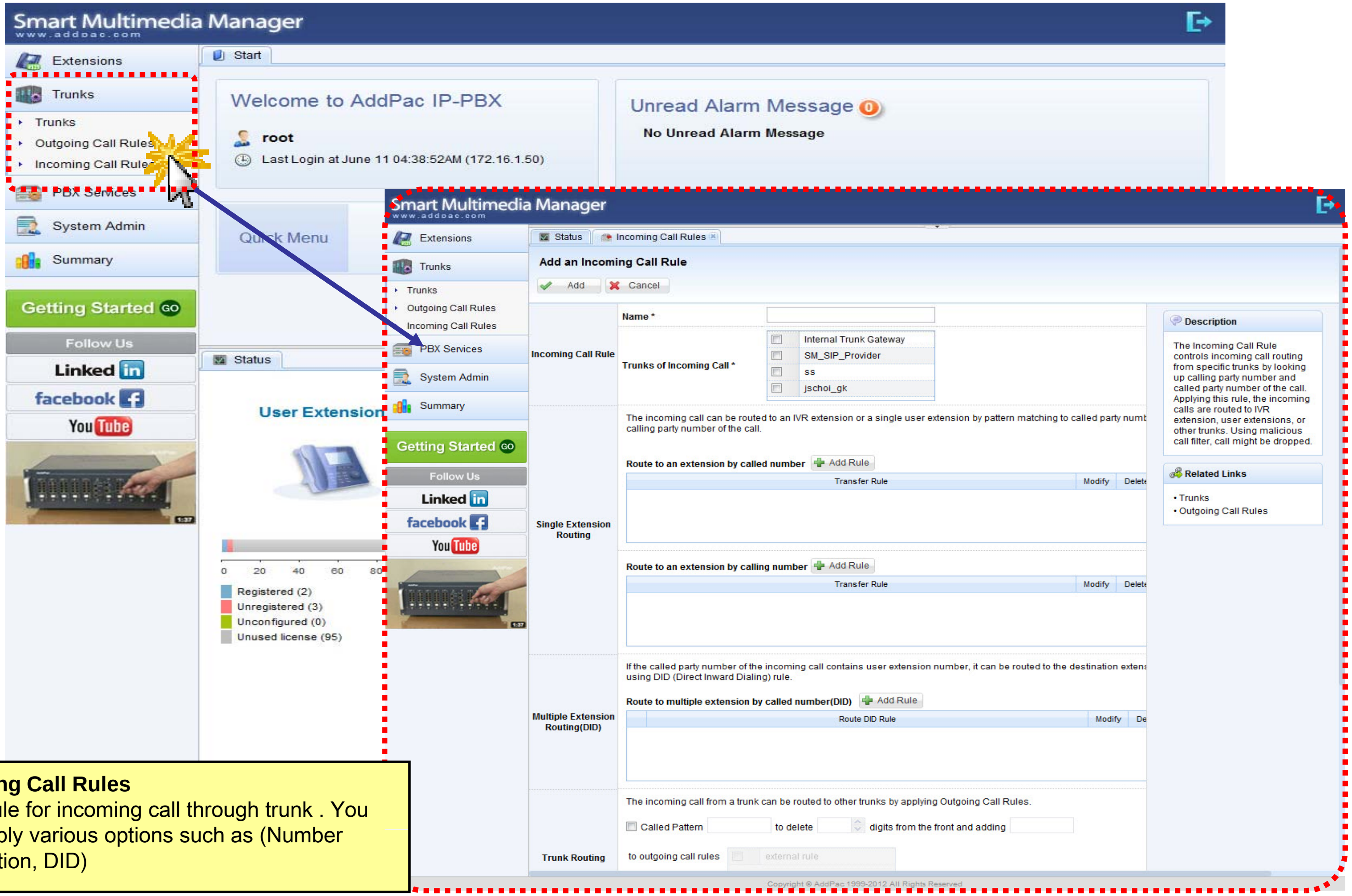
Trunk
A trunk setup for IP-PBX in order to make a call. You may setup VoIP Trunk, SIP Proxy Server, and H.323 Gatekeeper as well as to check the register status in accordance with Trunk types.

Trunk - Outgoing Call Rules



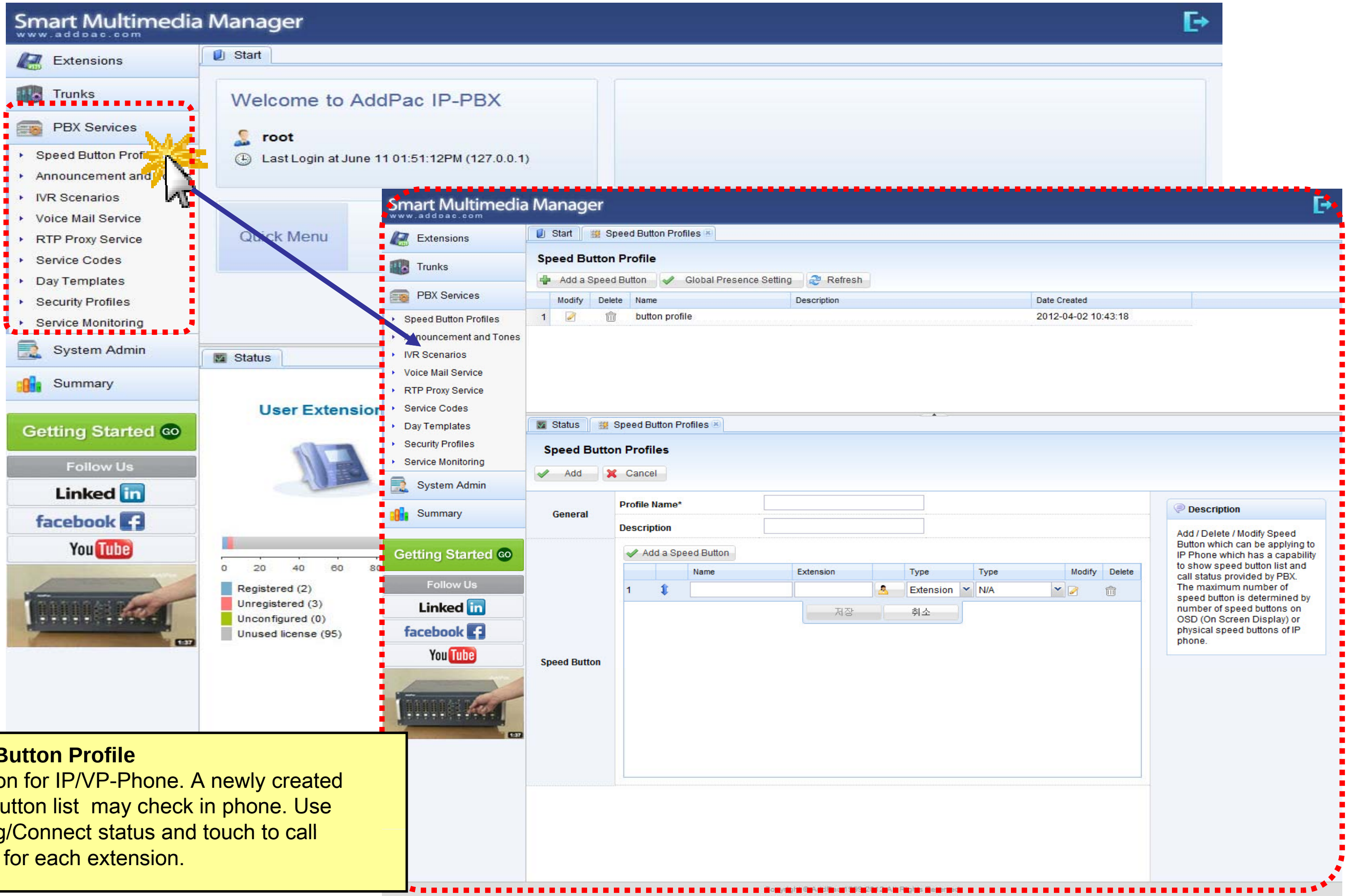
Outgoing Call Rules
A call rule for external call routing. You may apply various options such as Outgoing call rule (Number Translation, Routing Mode, Display Name Presentation, P-Asserted Identity Presentation) for outgoing call rule.

Trunk - Incoming Call Rules



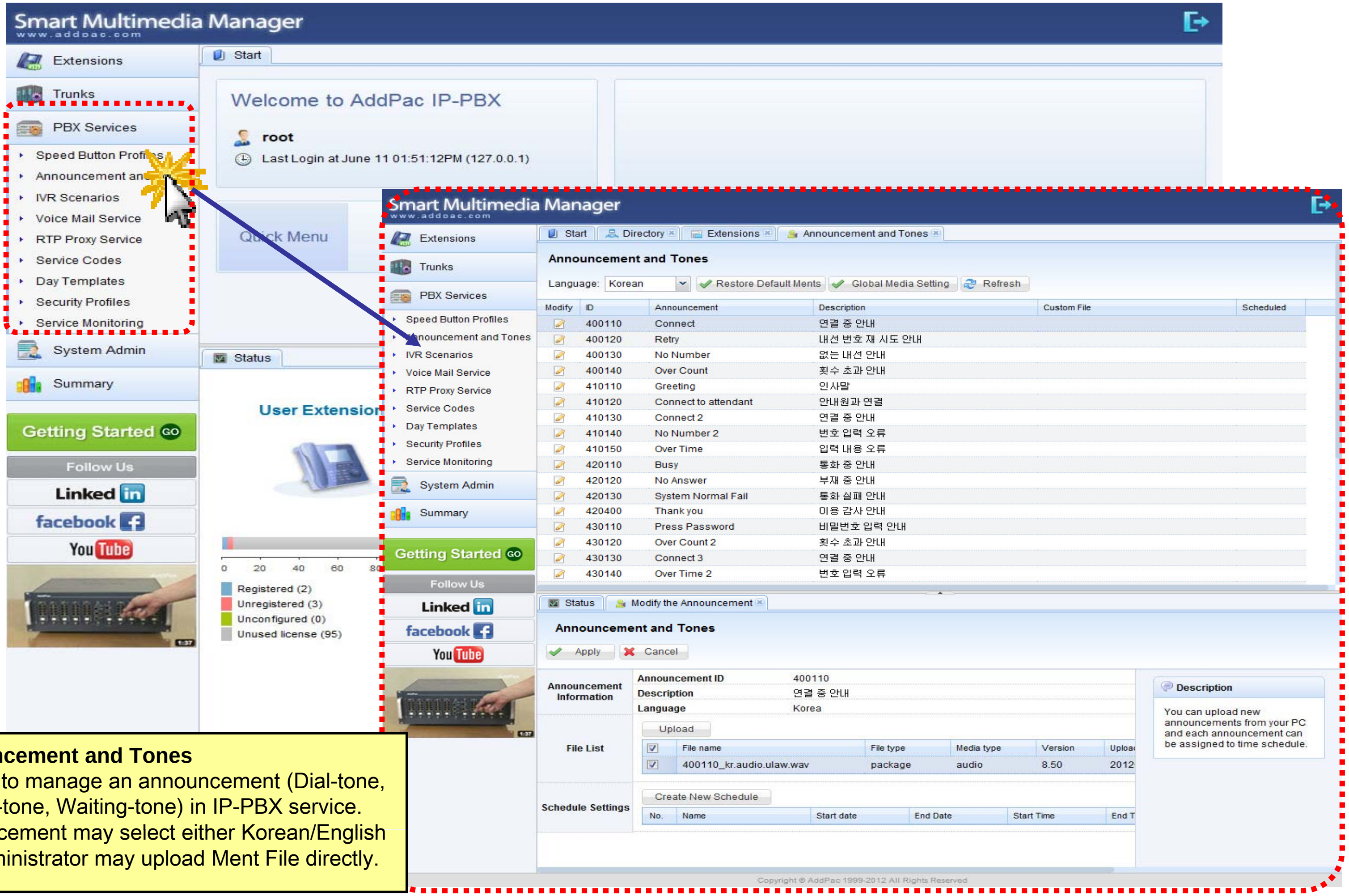
Incoming Call Rules
A call rule for incoming call through trunk . You may apply various options such as (Number Translation, DID)

PBX Service - Speed Button Profiles



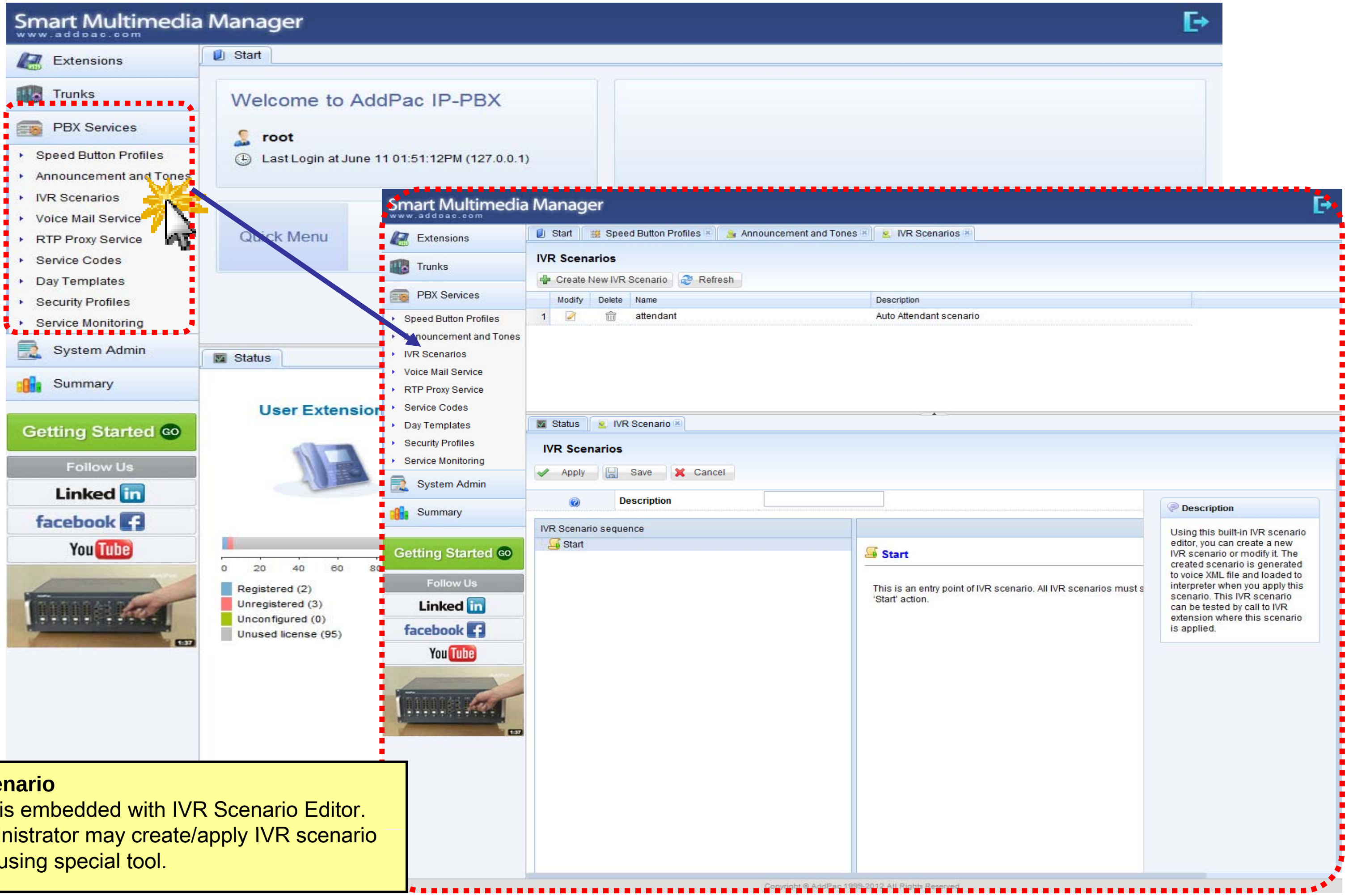
Speed Button Profile
A function for IP/VP-Phone. A newly created speed button list may check in phone. Use idle/Ring/Connect status and touch to call function for each extension.

PBX Service - Announcement and Tones



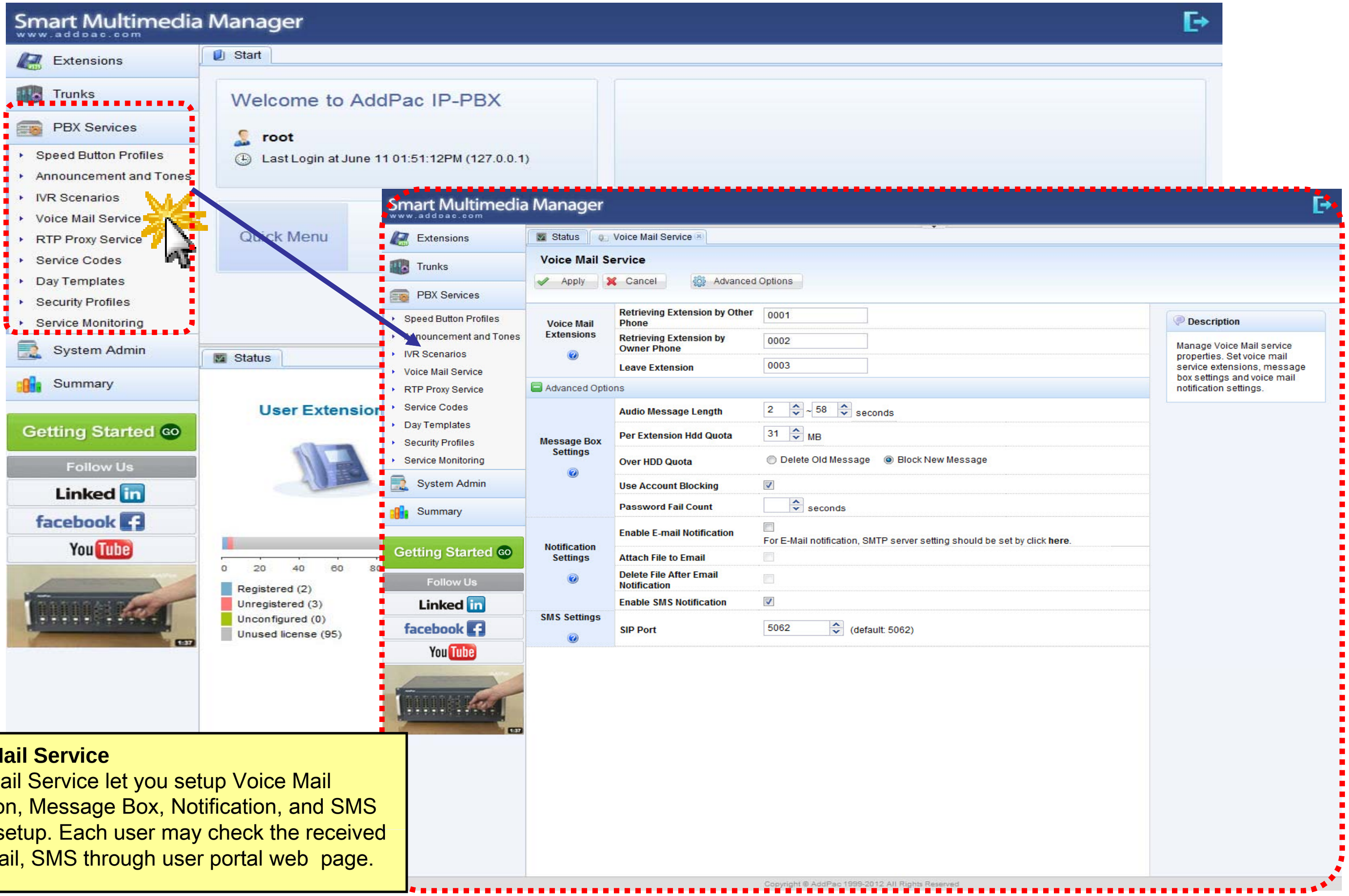
Announcement and Tones
A setup to manage an announcement (Dial-tone, Consult-tone, Waiting-tone) in IP-PBX service. Announcement may select either Korean/English and administrator may upload Ment File directly.

PBX Service - IVR Scenarios



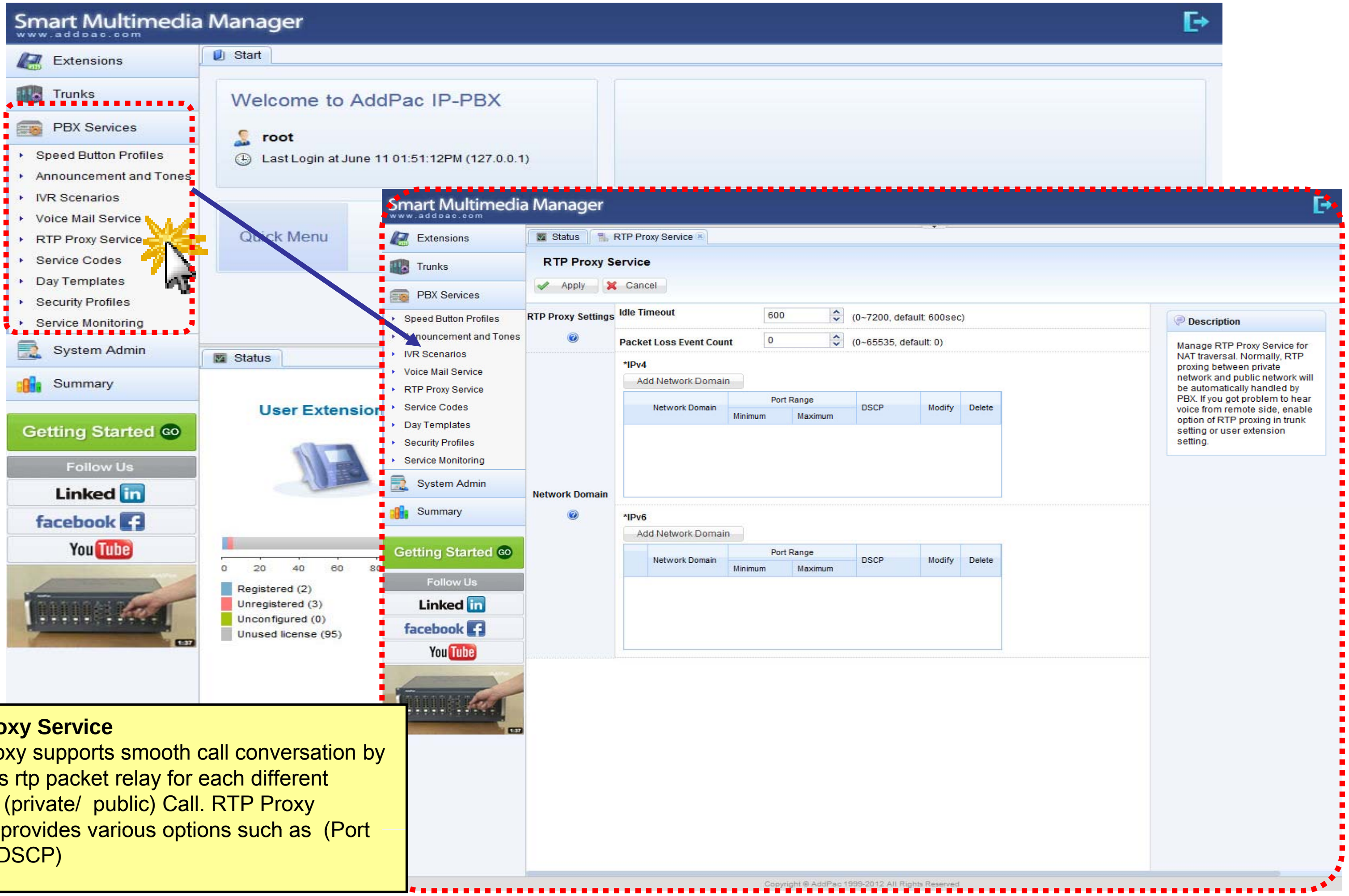
IVR Scenario
WSMM is embedded with IVR Scenario Editor.
An administrator may create/apply IVR scenario without using special tool.

PBX Service - Voice Mail Services



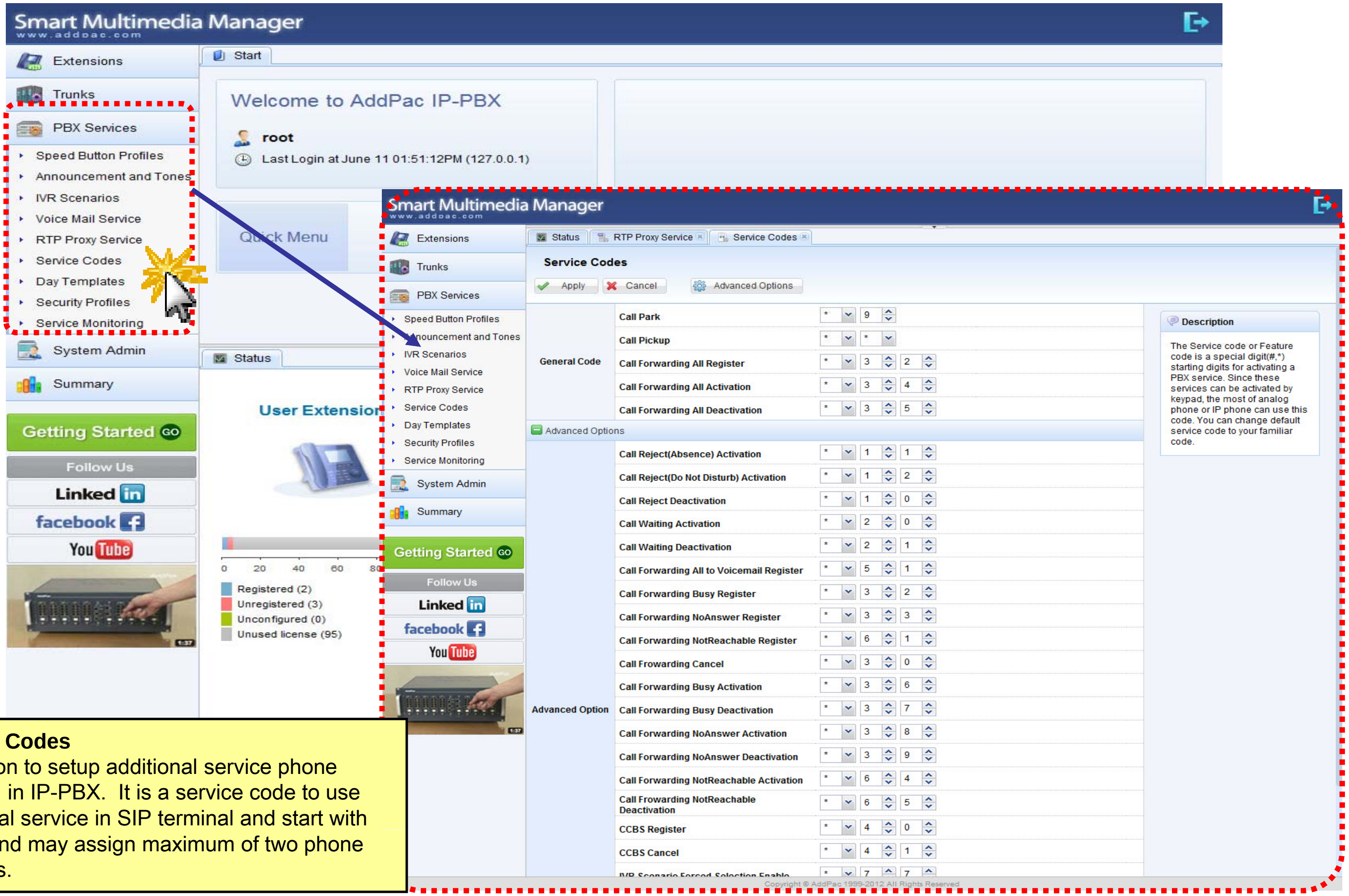
Voice Mail Service
Voice Mail Service let you setup Voice Mail Extension, Message Box, Notification, and SMS related setup. Each user may check the received voice-mail, SMS through user portal web page.

PBX Service - RTP Proxy Service



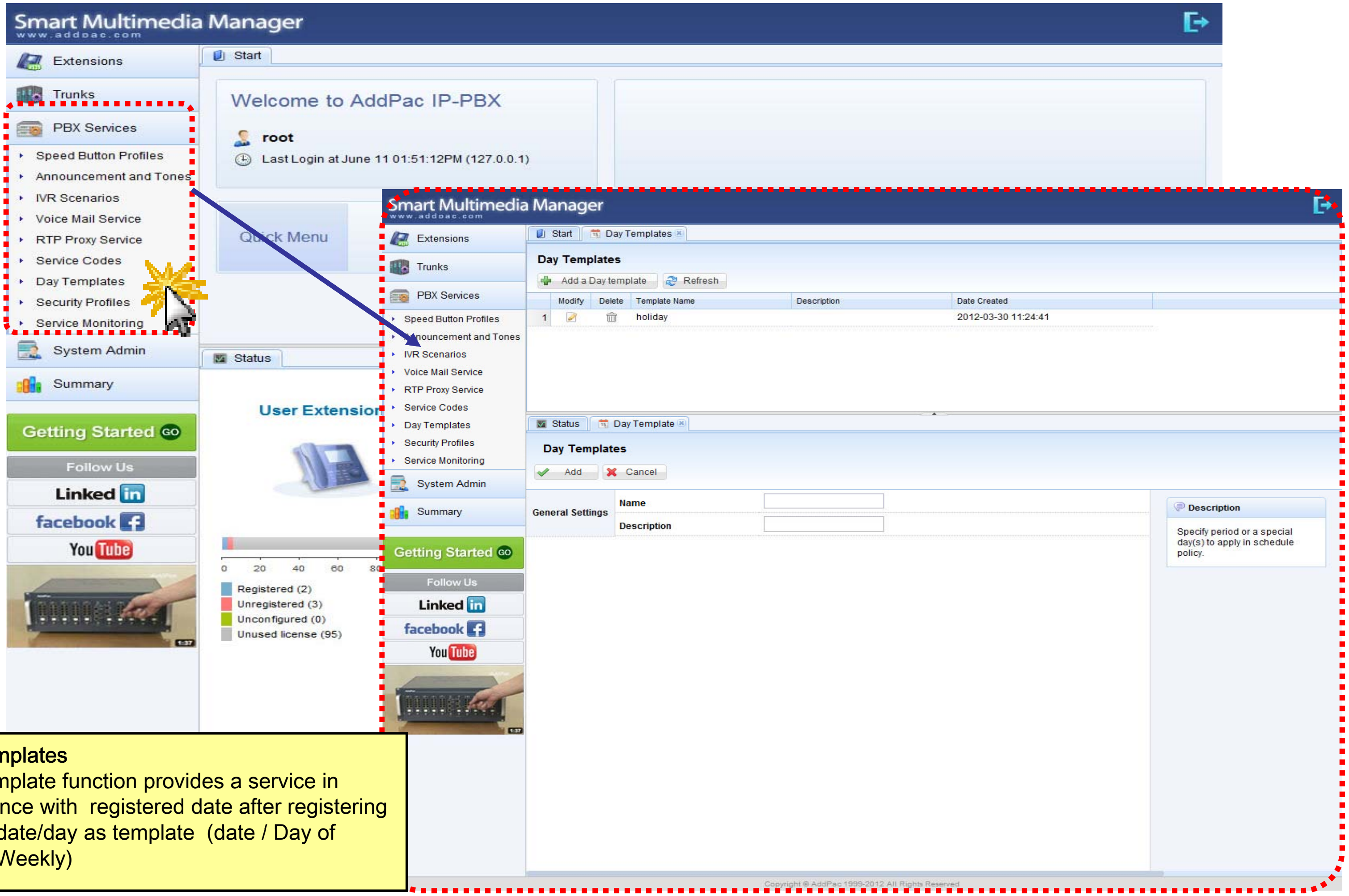
RTP Proxy Service
RTP Proxy supports smooth call conversation by acting as rtp packet relay for each different network (private/ public) Call. RTP Proxy Service provides various options such as (Port range / DSCP)

PBX Service - Service Codes



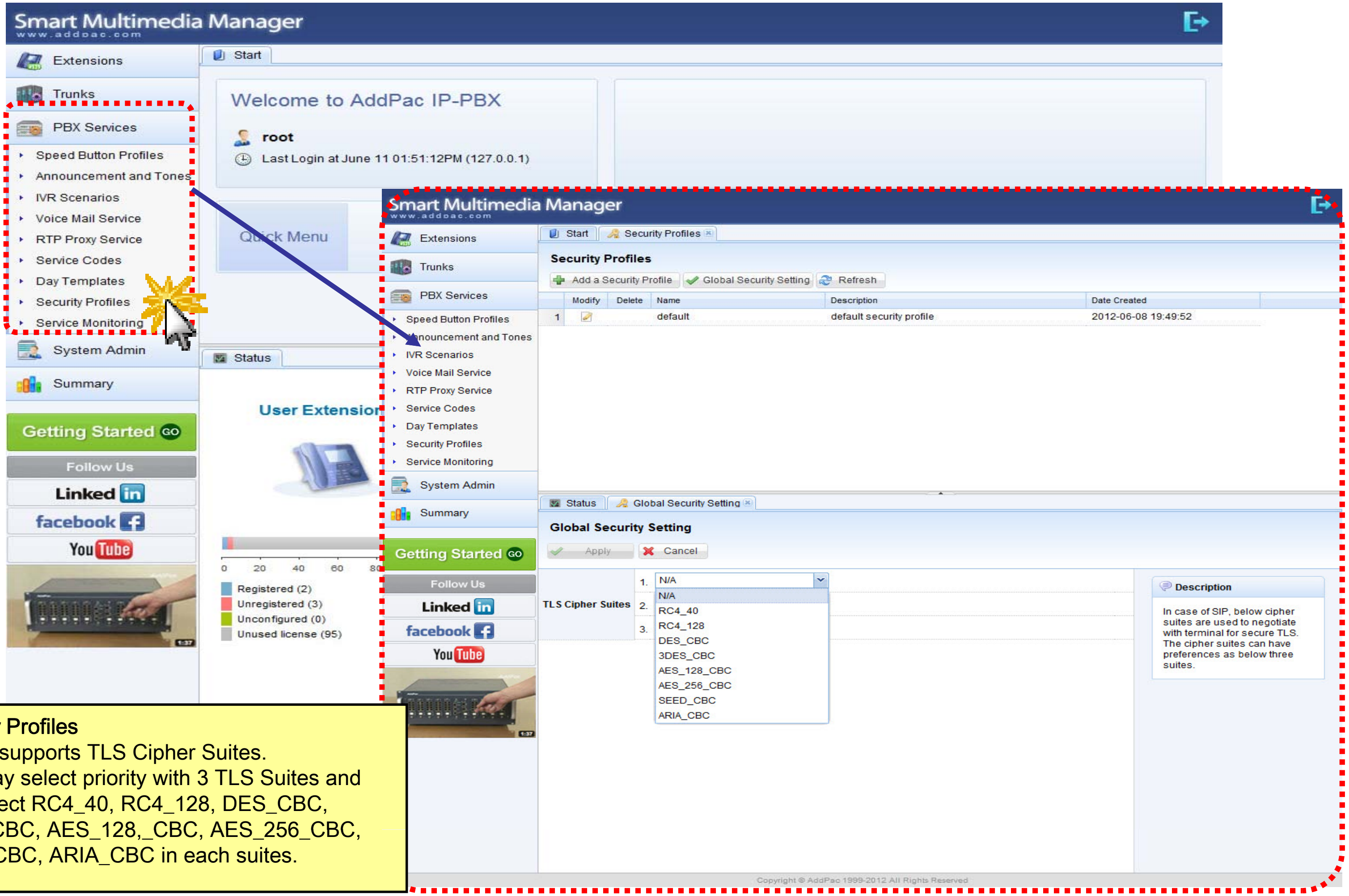
Service Codes
A function to setup additional service phone number in IP-PBX. It is a service code to use additional service in SIP terminal and start with # or * and may assign maximum of two phone numbers.

PBX Service - Day Templates



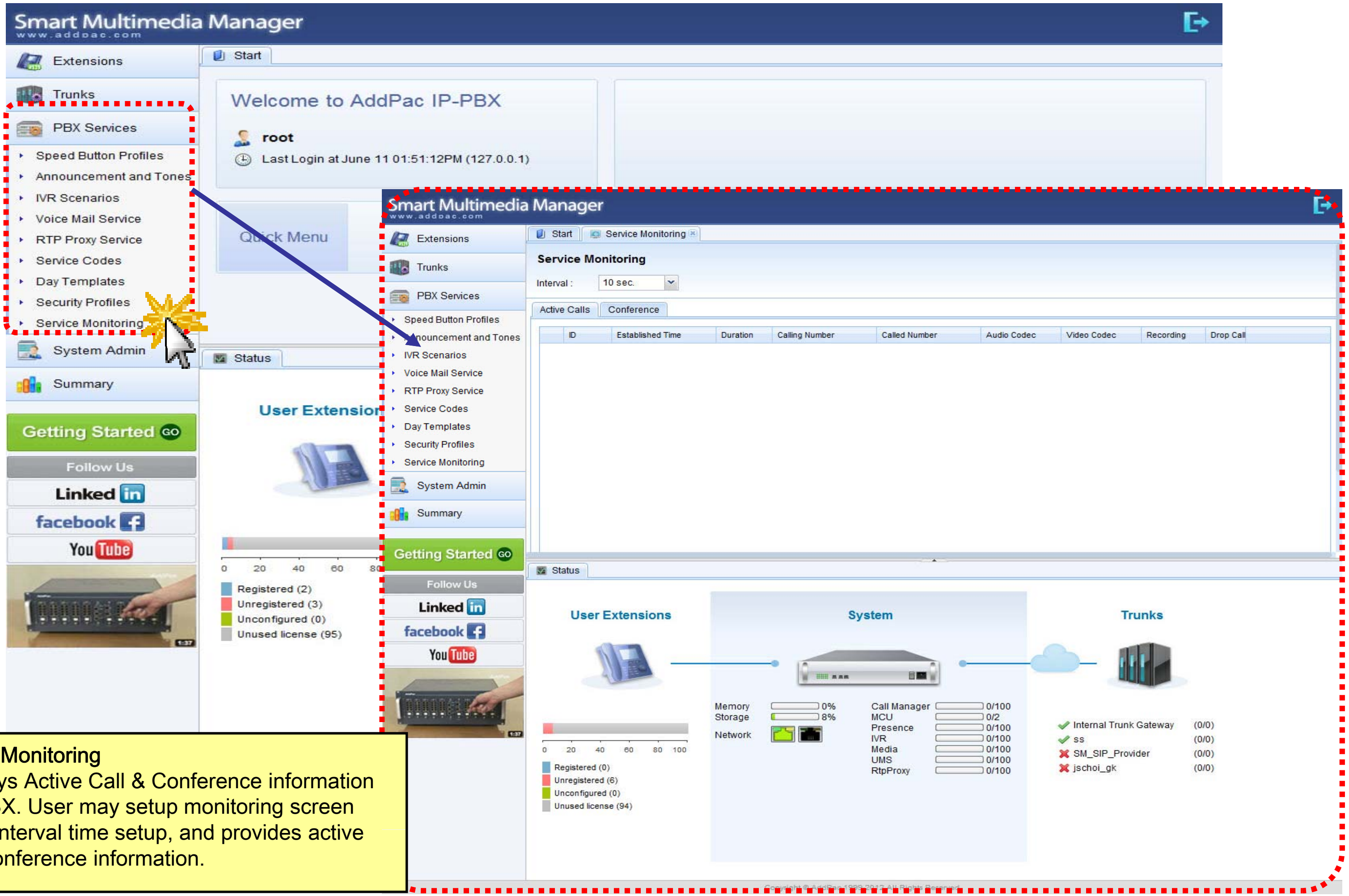
Day Templates
Day Template function provides a service in accordance with registered date after registering special date/day as template (date / Day of Week / Weekly)

PBX Service - Security Profiles



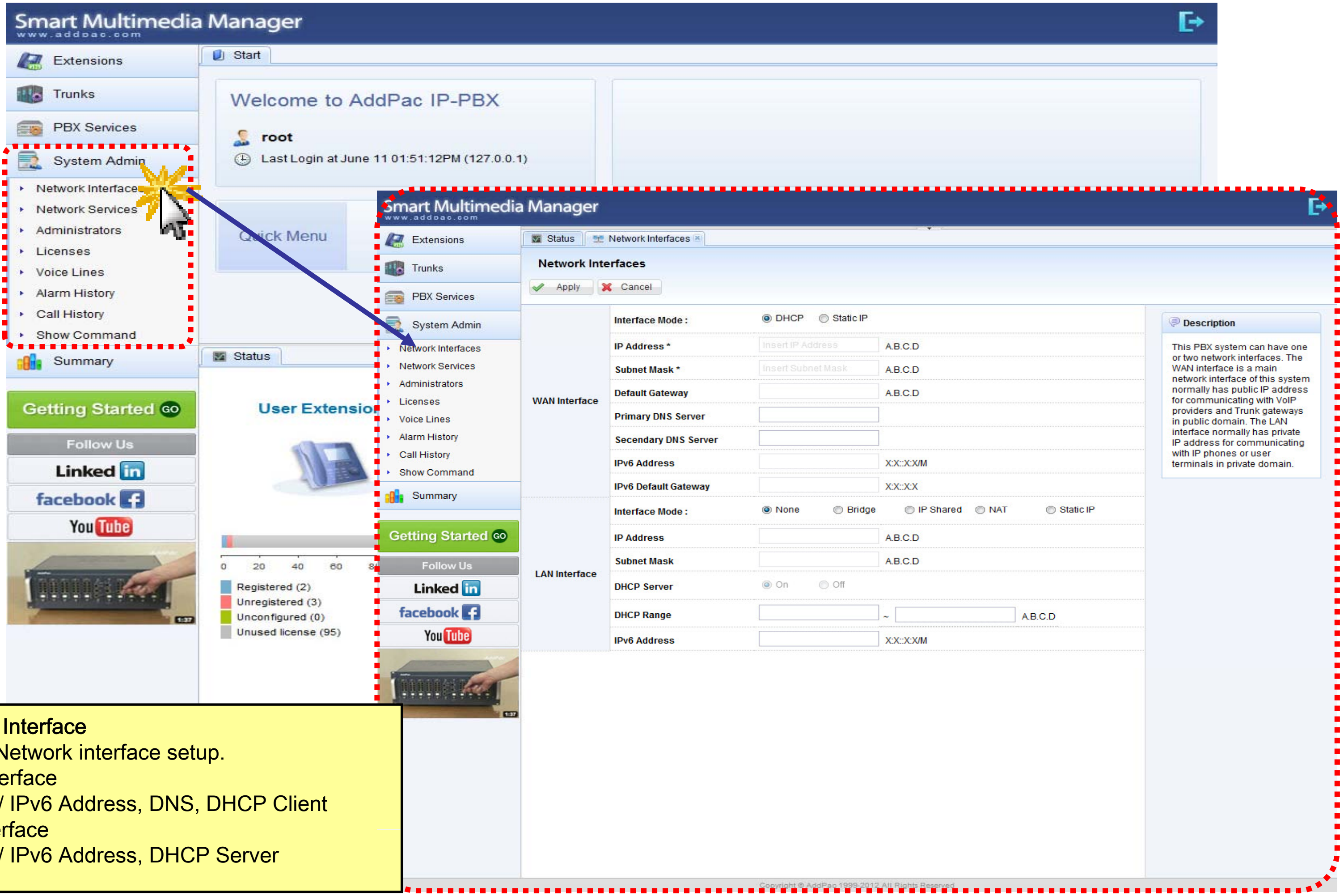
Security Profiles
IP-PBX supports TLS Cipher Suites.
User may select priority with 3 TLS Suites and may select RC4_40, RC4_128, DES_CBC, 3DES_CBC, AES_128,_CBC, AES_256_CBC, SEED_CBC, ARIA_CBC in each suites.

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.

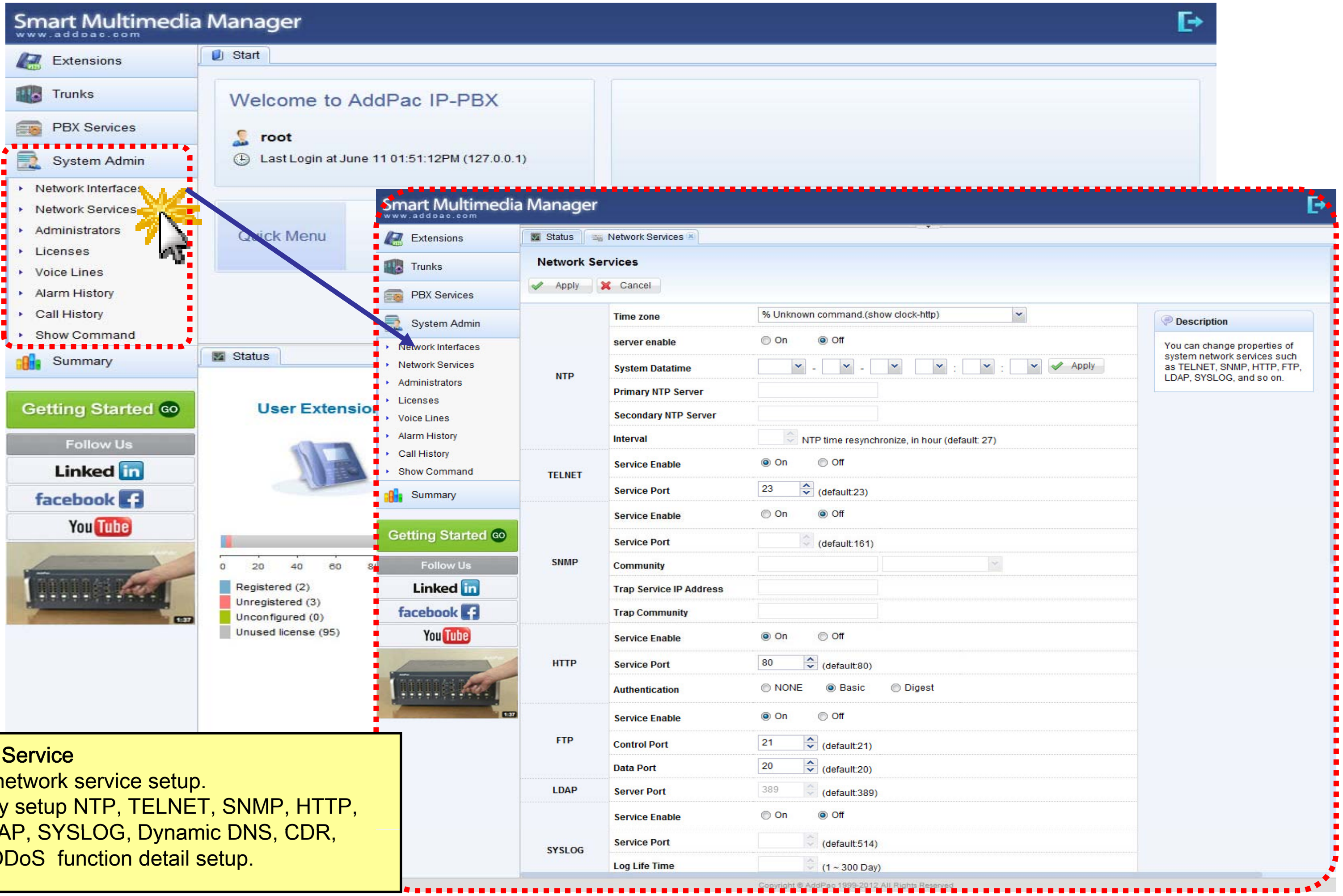
System Admin - Network Interface



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

LAN Interface
- IPv4 / IPv6 Address, DHCP Server

System Admin - Network Services

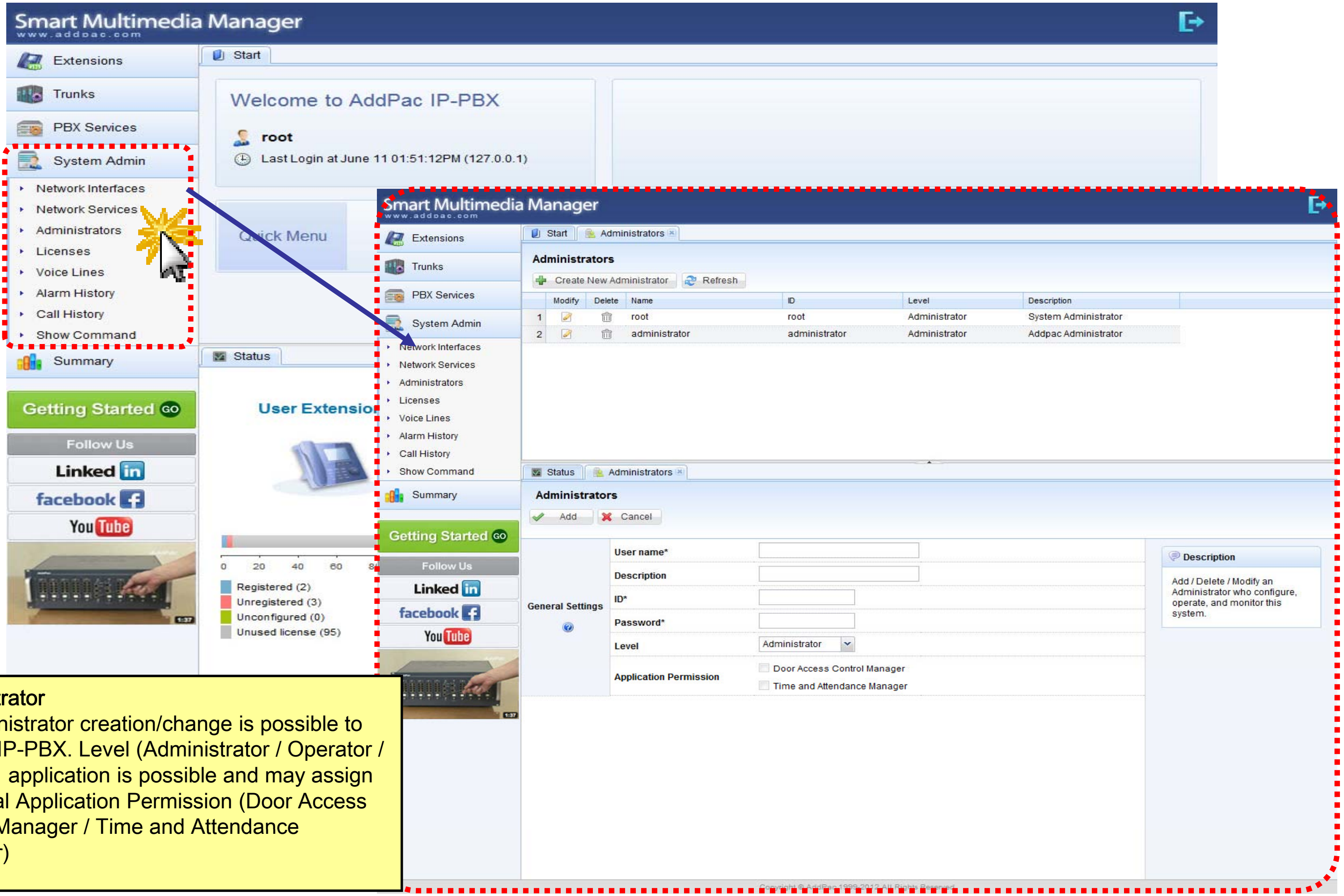


Network Service

IP-PBX network service setup.

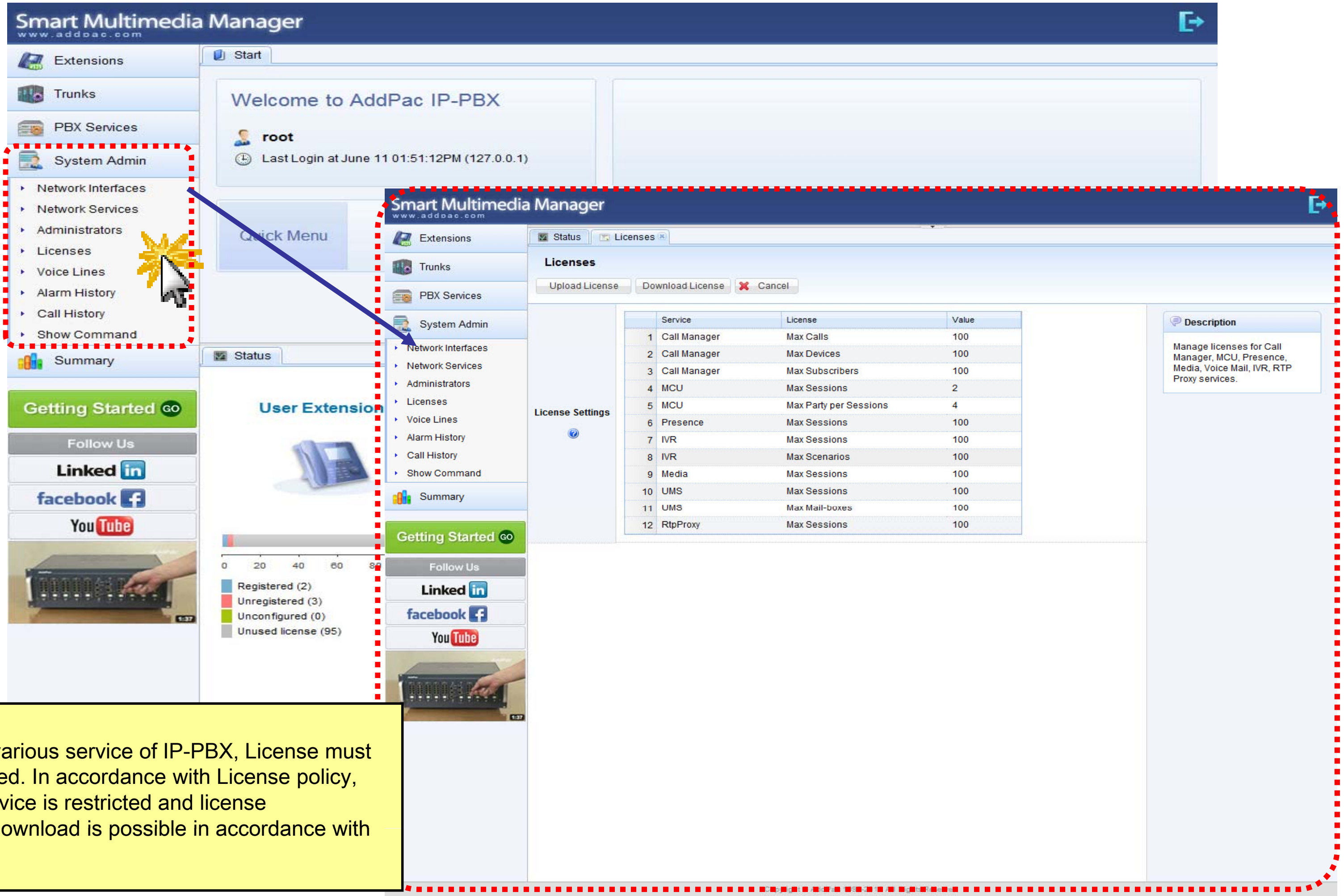
User may setup NTP, TELNET, SNMP, HTTP, FTP, LDAP, SYSLOG, Dynamic DNS, CDR, SMTP, DDoS function detail setup.

System Admin - Administrators



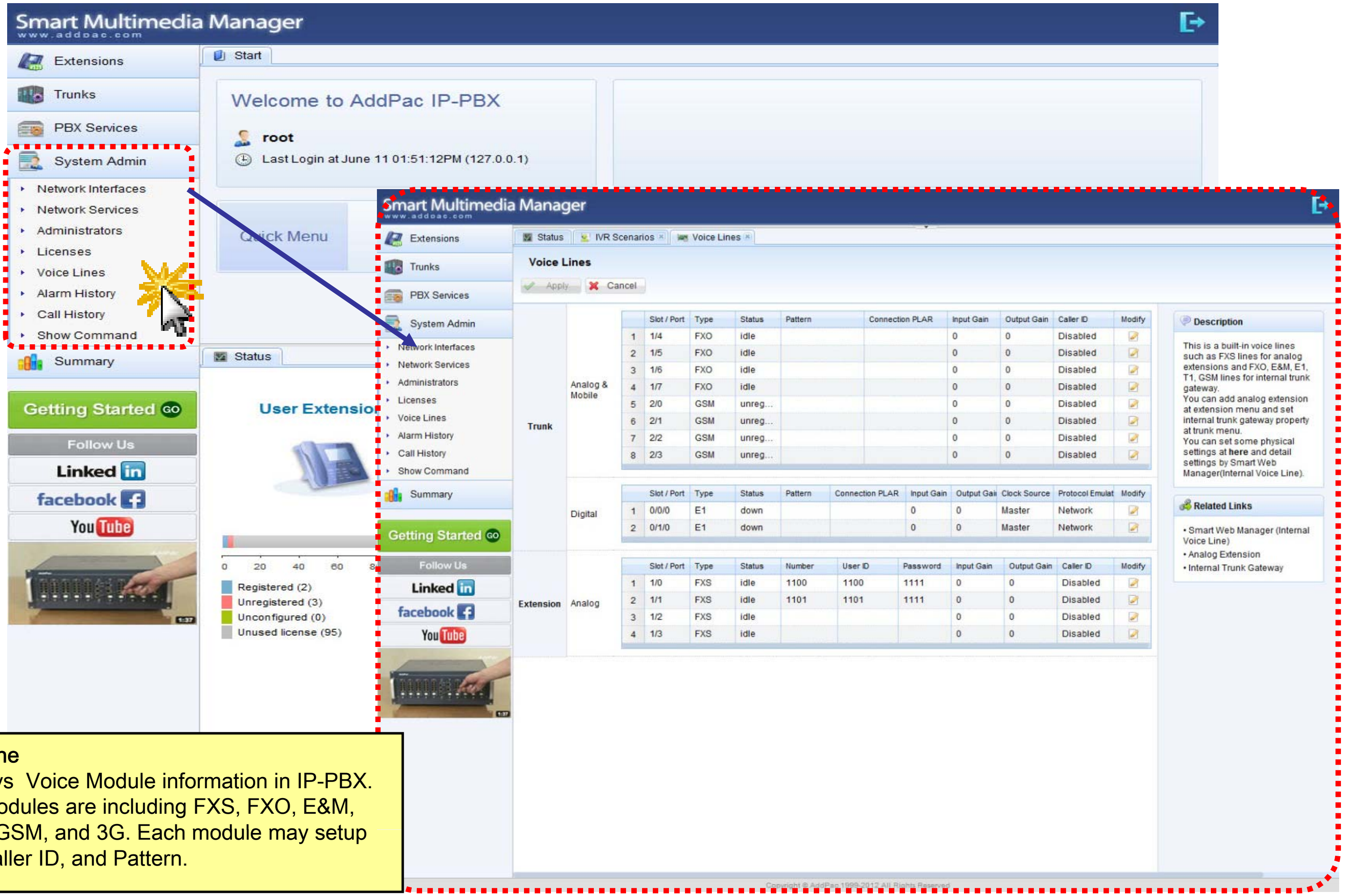
Administrator
An administrator creation/change is possible to operate IP-PBX. Level (Administrator / Operator / Monitor) application is possible and may assign additional Application Permission (Door Access Control Manager / Time and Attendance Manager)

System Admin - Licenses



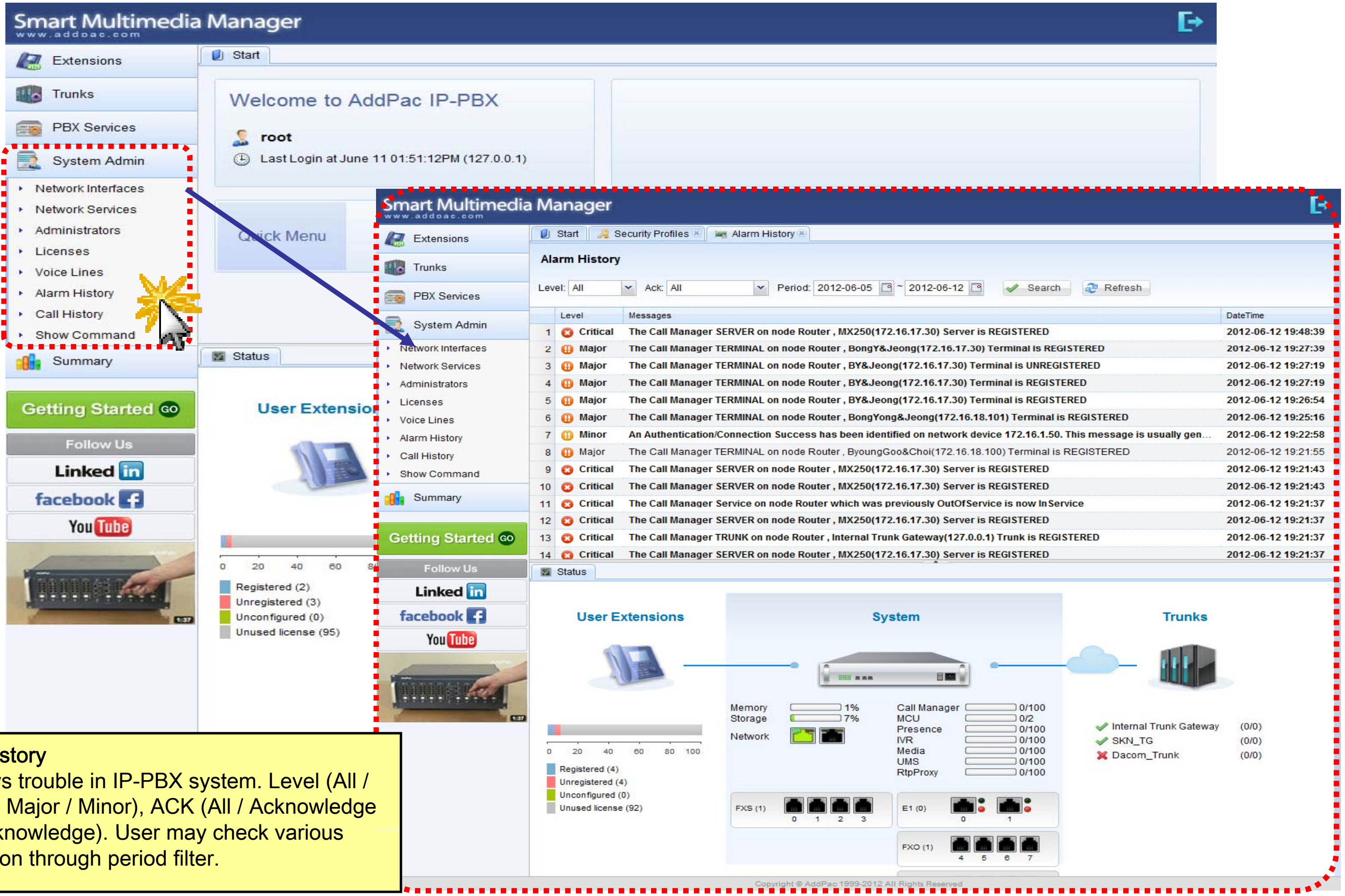
License
To use various service of IP-PBX, License must be created. In accordance with License policy, Max Service is restricted and license upload/download is possible in accordance with policy.

System Admin - Voice Lines



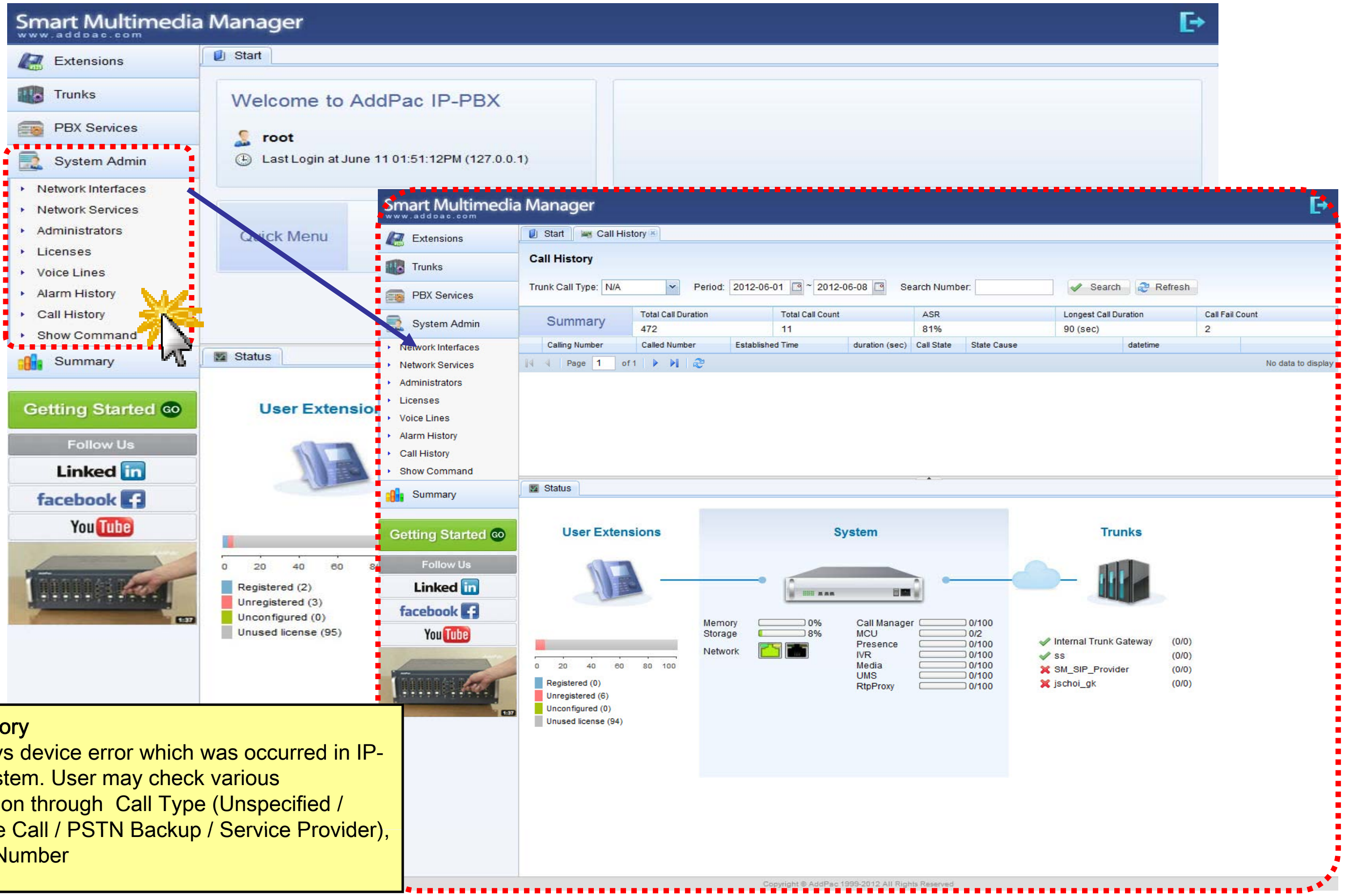
Voice Line
It displays Voice Module information in IP-PBX.
Voice modules are including FXS, FXO, E&M, E1, T1, GSM, and 3G. Each module may setup Gain, Caller ID, and Pattern.

System Admin - Alarm History



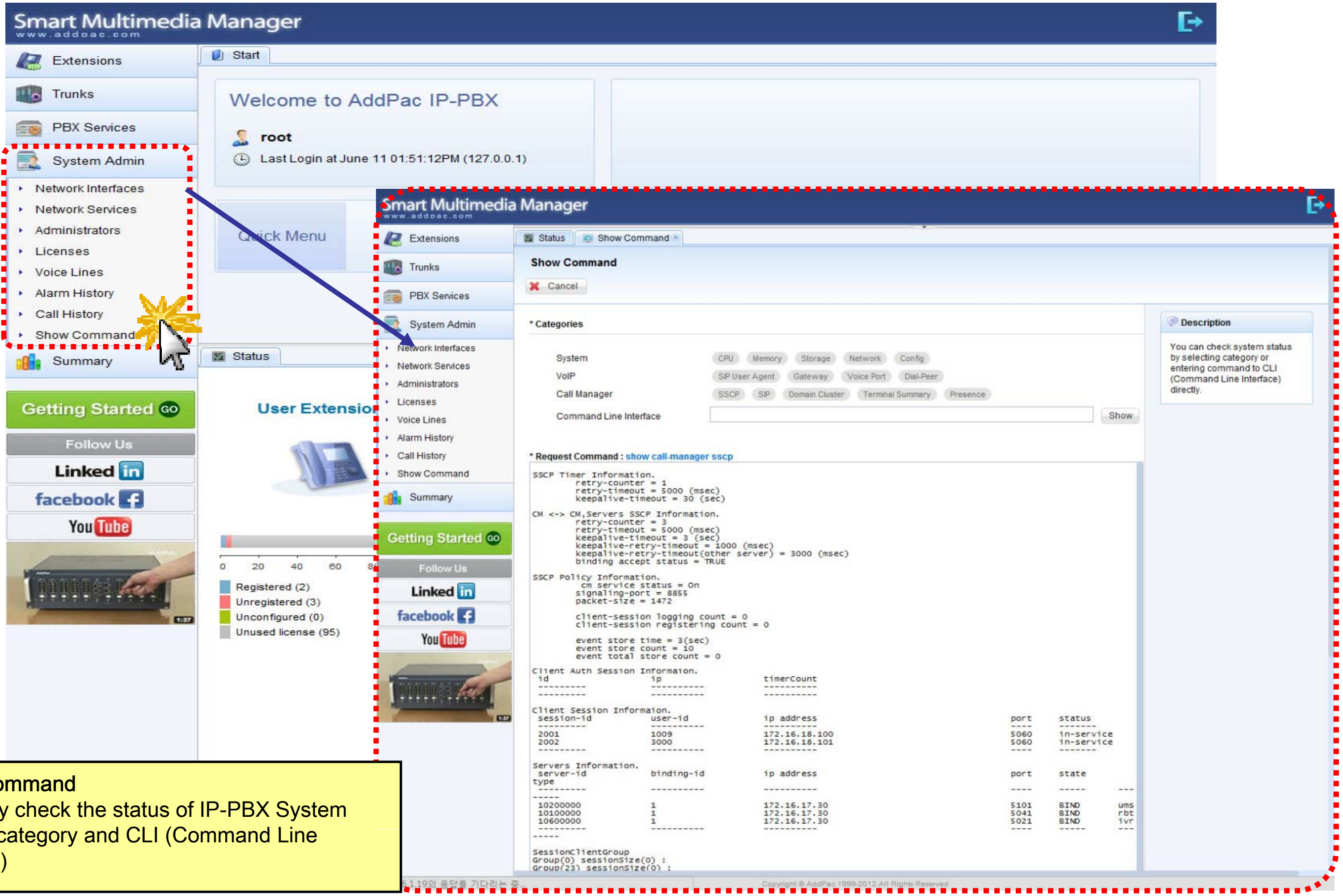
Alarm History
It displays trouble in IP-PBX system. Level (All / Critical / Major / Minor), ACK (All / Acknowledge / Not Acknowledge). User may check various information through period filter.

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



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

Thank you!

AddPac Technology Co., Ltd.
Sales and Marketing

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