# Hospital Simple SIP Voice Paging Solution




AddPac

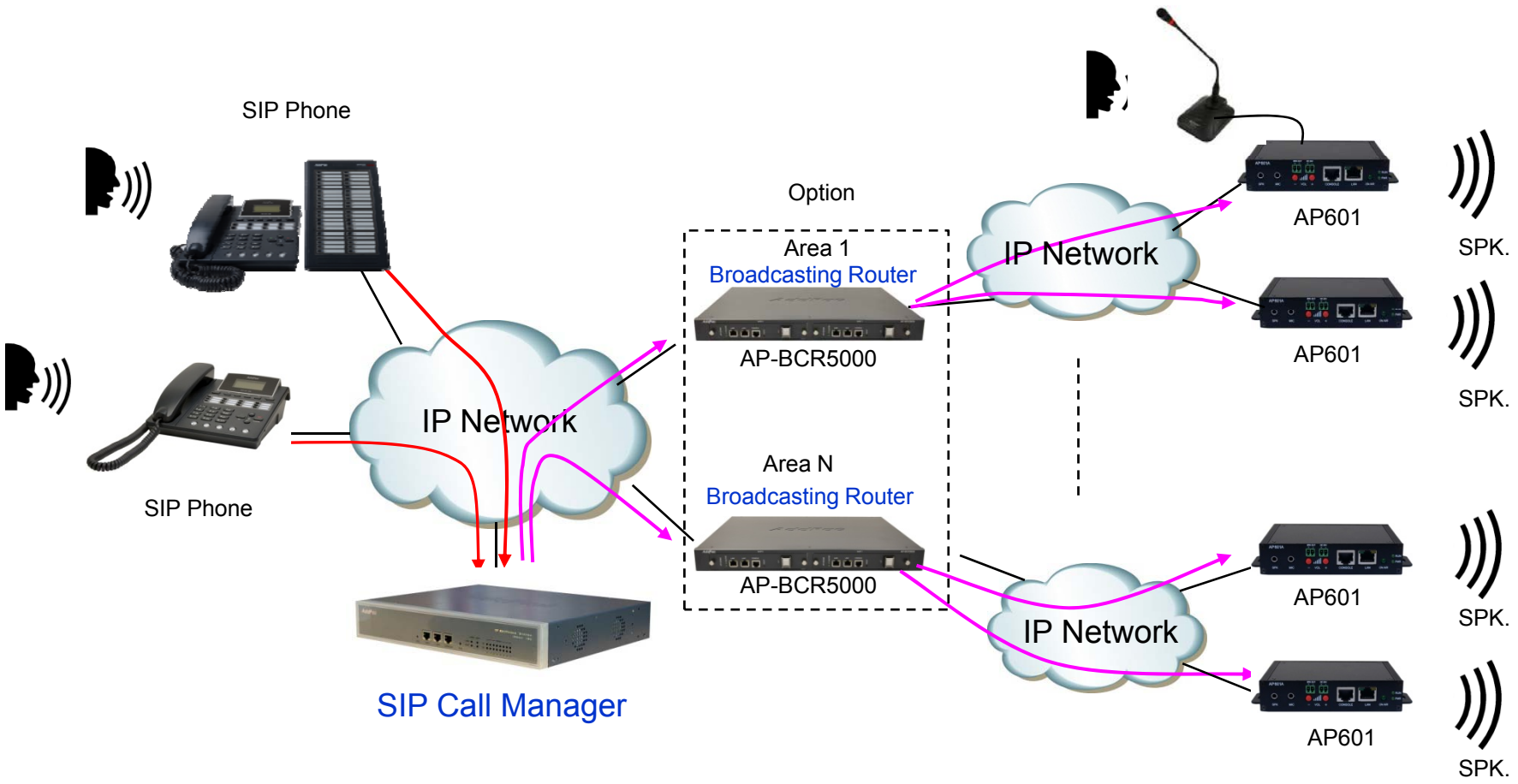
**AddPac Technology**

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# Contents

- SIP Paging Call Service Diagram
- SIP Paging Product Solution
- Simple SIP Paging Service Overview
- Product Specifications
  - SIP Call Manager : IPNext600, IPNext180
  - SIP Paging Terminal: AP1605, AP1602, AP601
  - SIP Phones for Paging Service : AP-IP300, AP-IP120
  - Web based Smart Multimedia Manager for SIP Call Manager

# SIP Paging Call Service Network Diagram



# SIP Paging Product Solution

| Web Smart Multimedia Manager for Call Manager | SIP Call Manager | SIP Paging Terminal | Broadcasting Router (Relay Server) | SIP Phone for Paging Service |
|---|---|---|---|---|
|  |  IPNext600 |  AP1605 |  AP-BCR5000 |  AP-IP300 + AP-PT20x2 |
| IPNext180 | |  AP1602 | |  AP-IP120 + AP-PT20x2 |
| | |  AP601 | | |

# 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

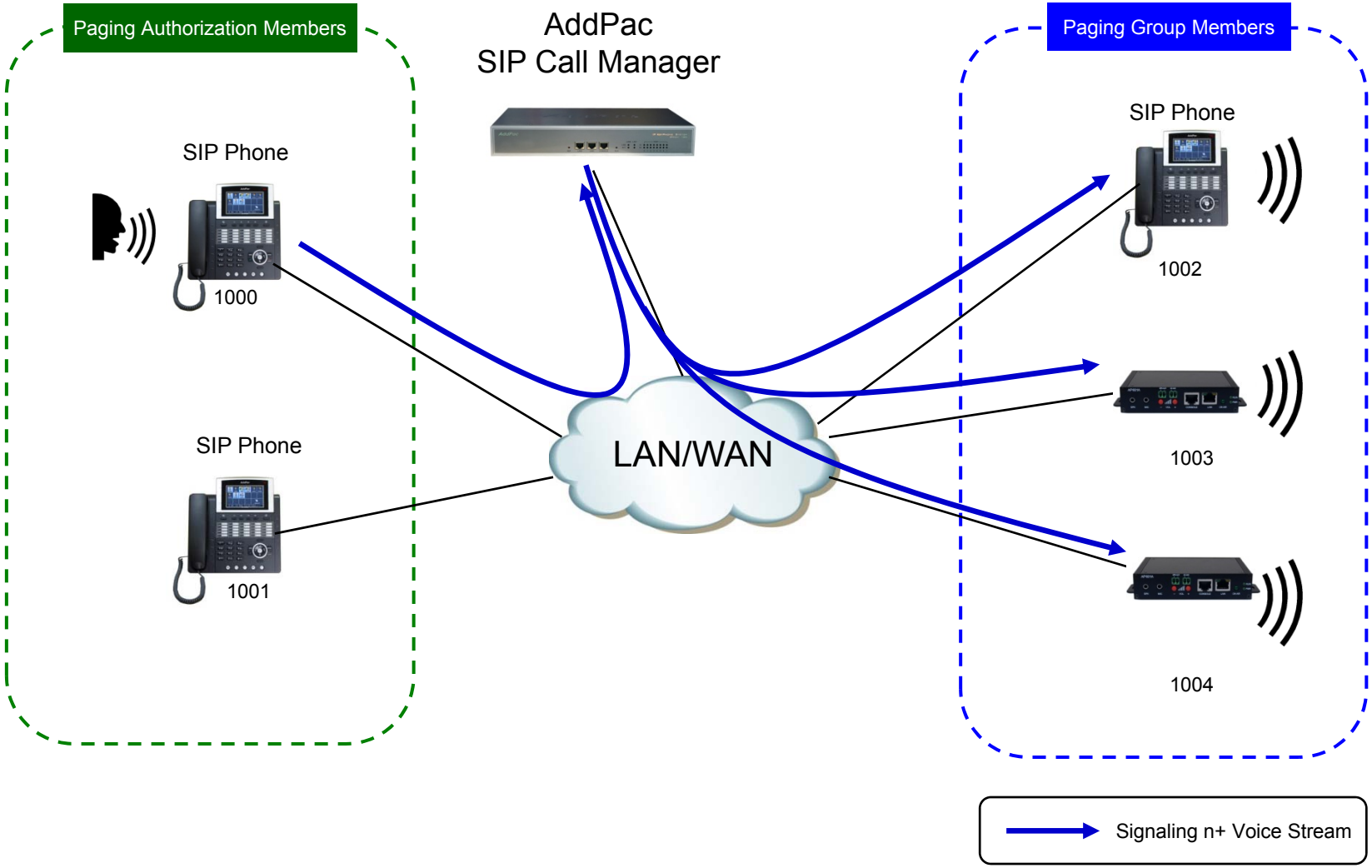




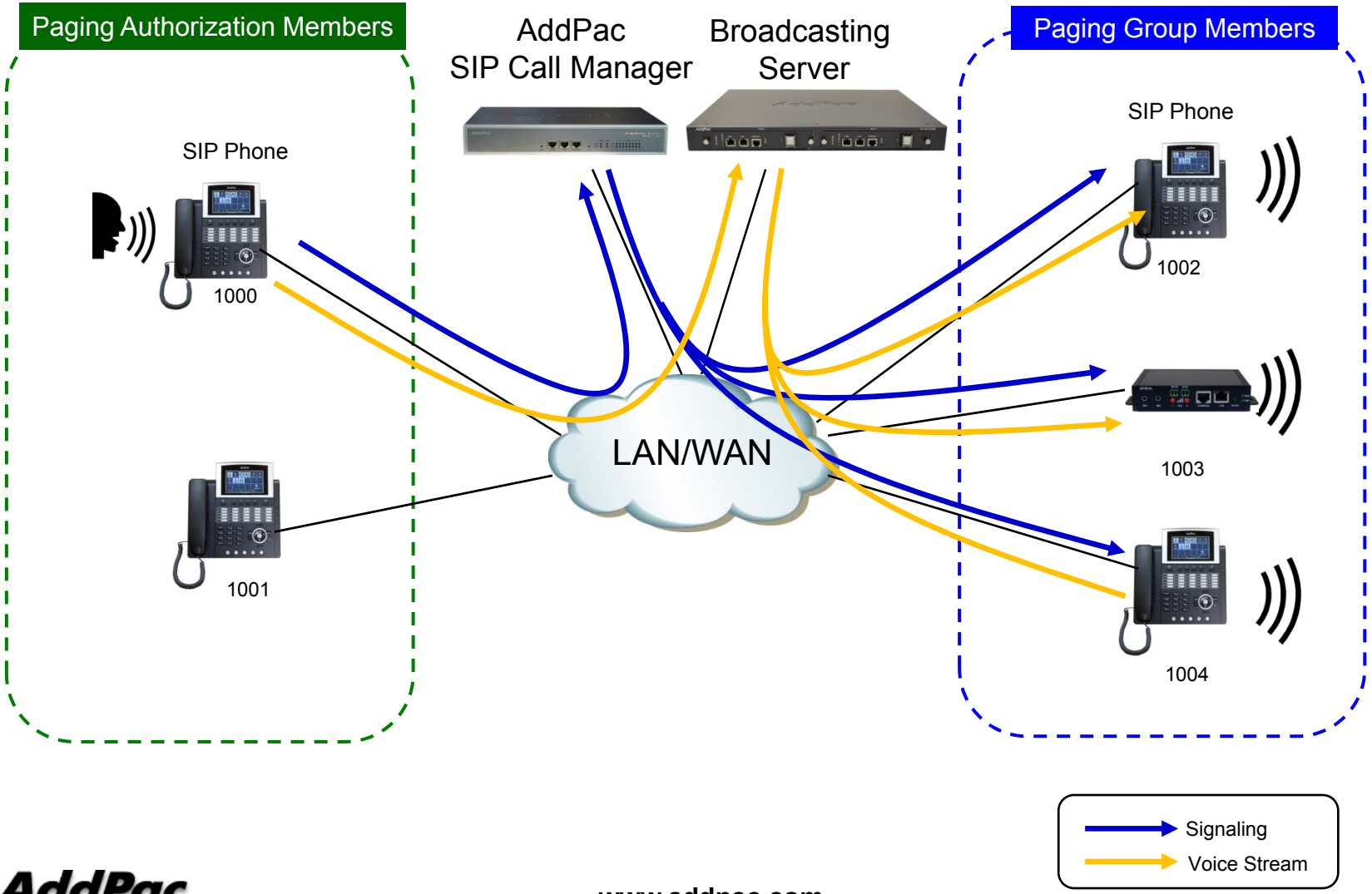
# Paging Service Features

- 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)

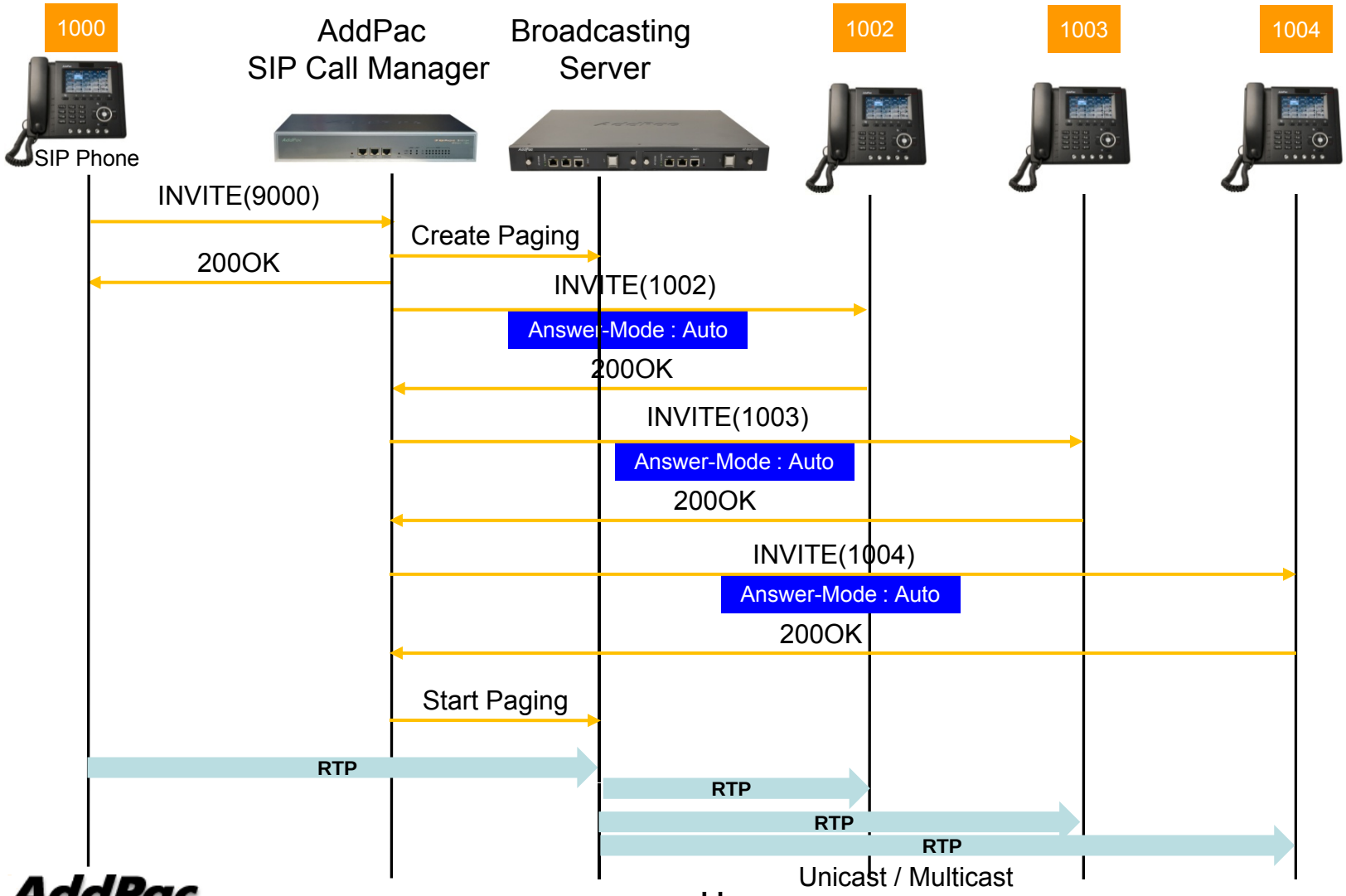
# Paging Service Model (Small Scale)



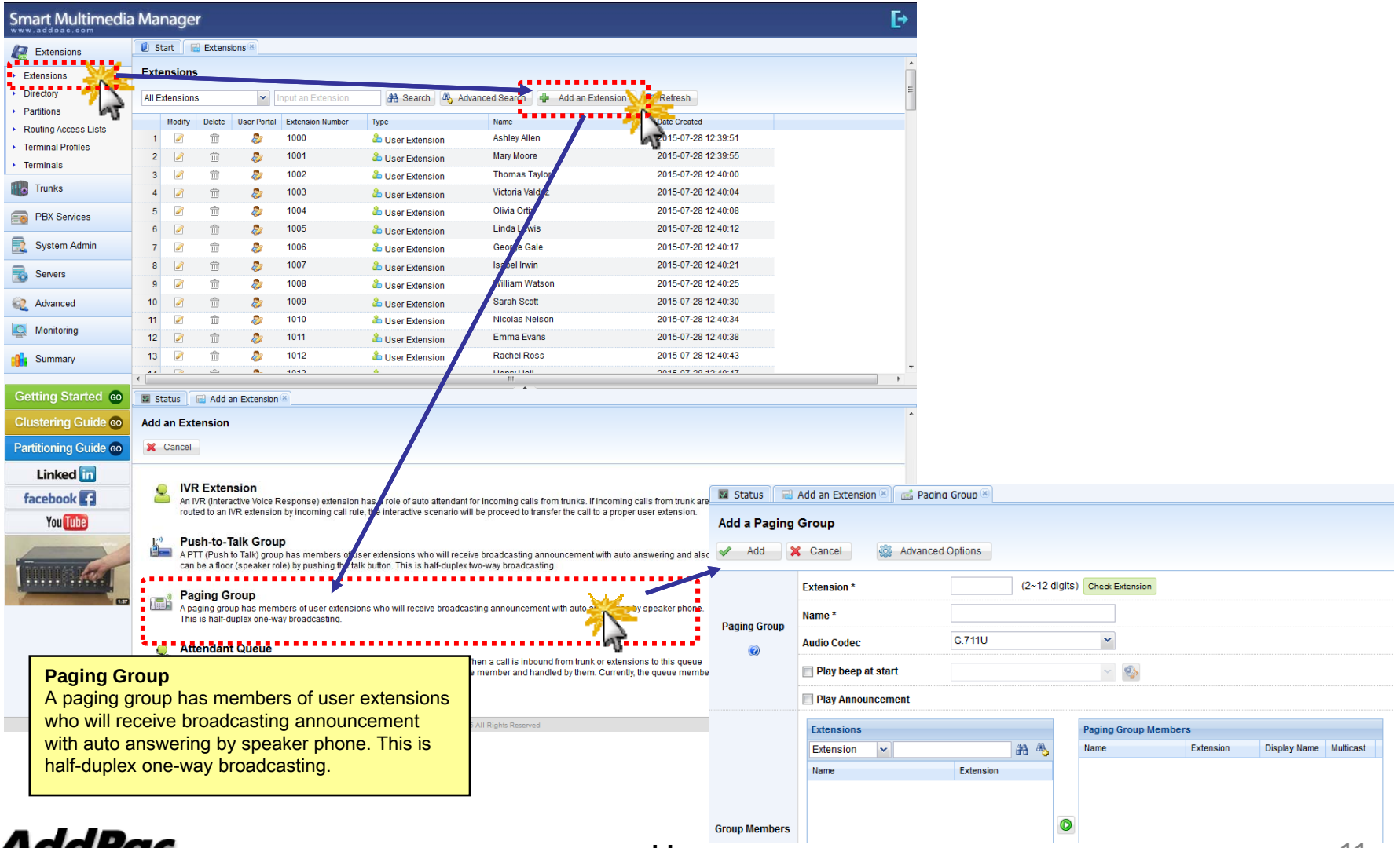
# Paging Service Model (Large Scale)



# Paging Group Signaling Flow

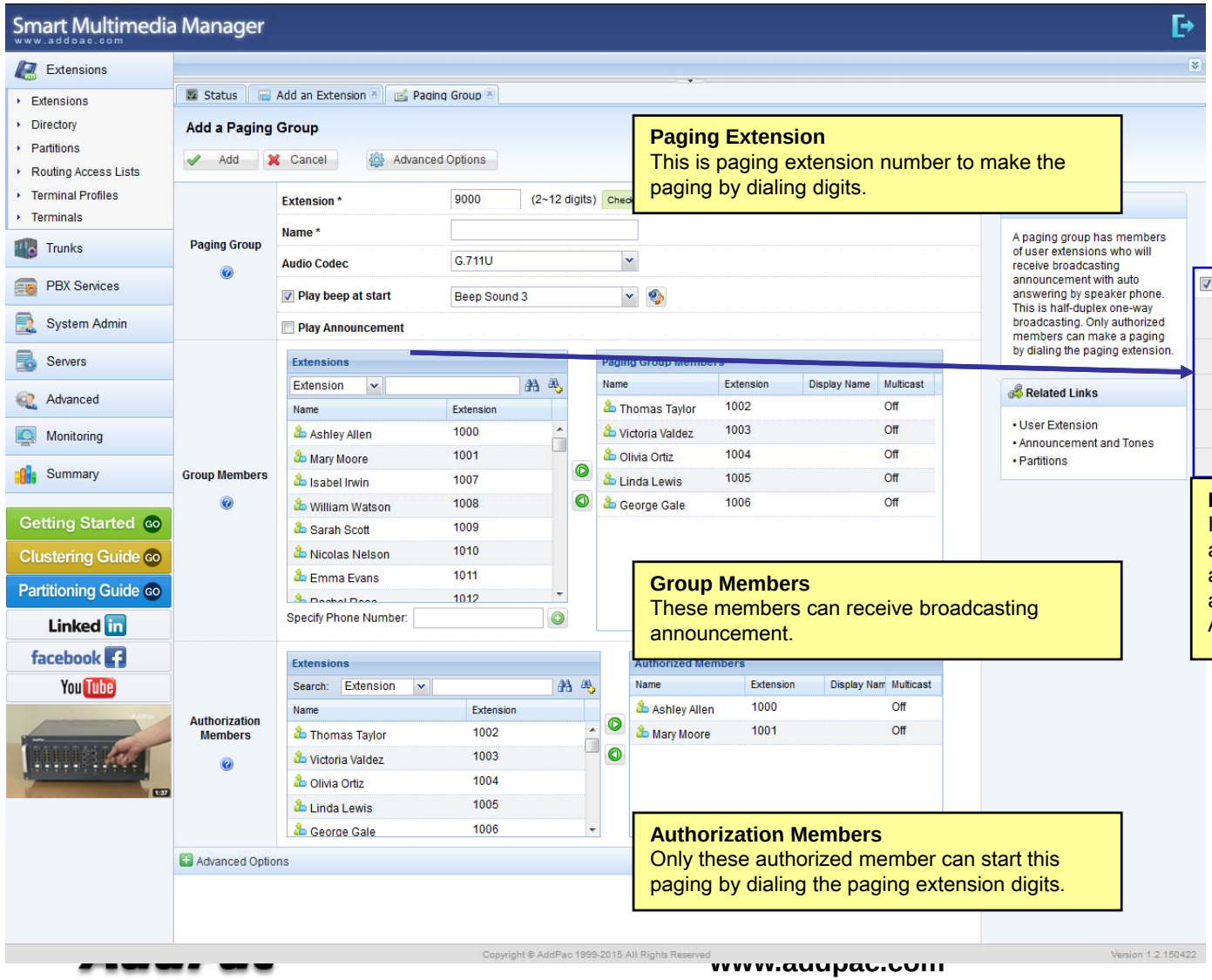


# Extension – Paging Group (WSMM SIP Call Manager Program)




**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)



# 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

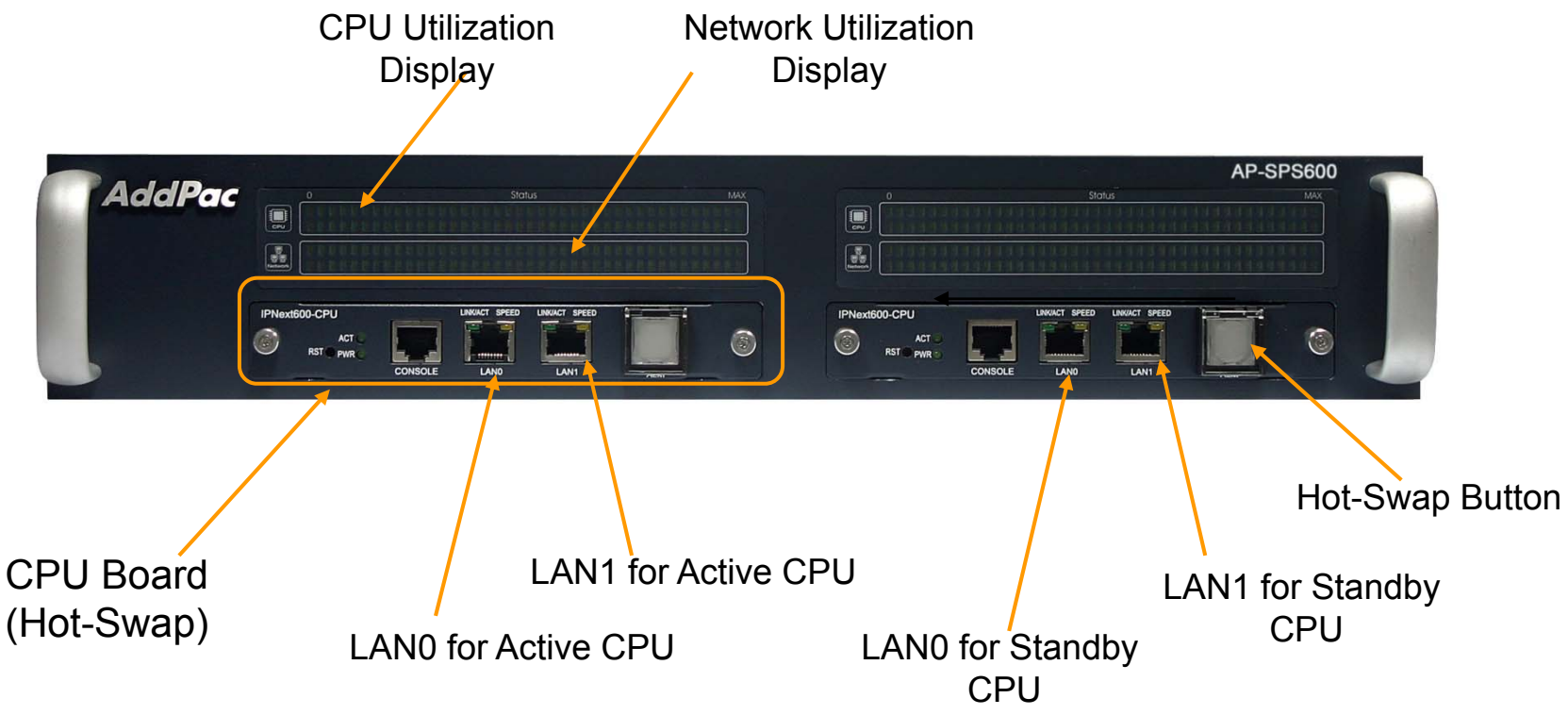
## IPNext600 SIP Call Manager

- 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

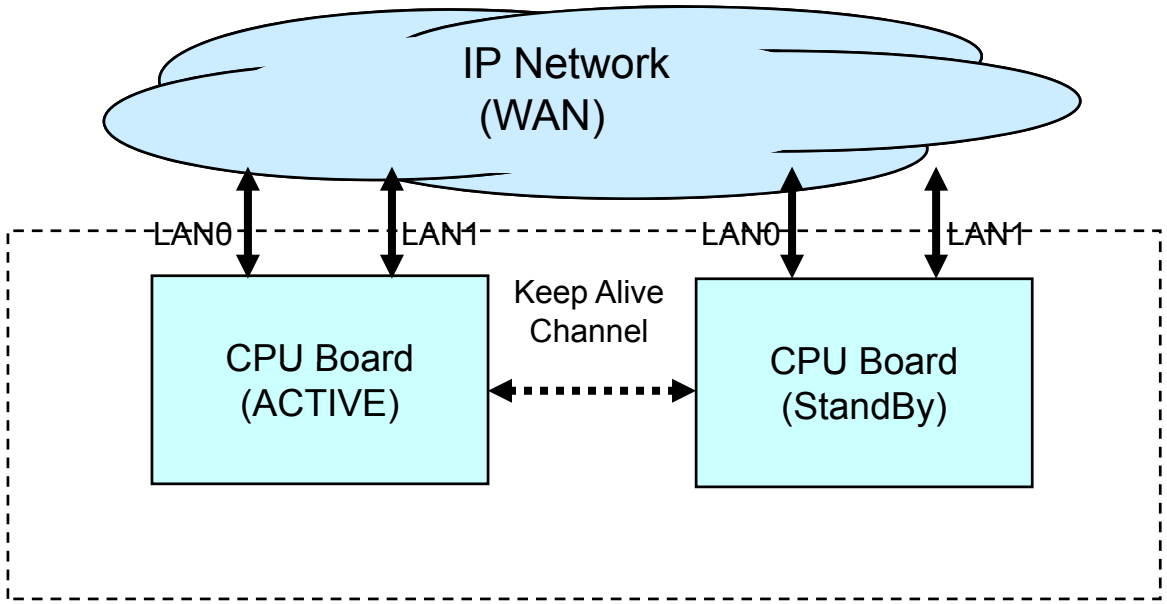
PSU Module B

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



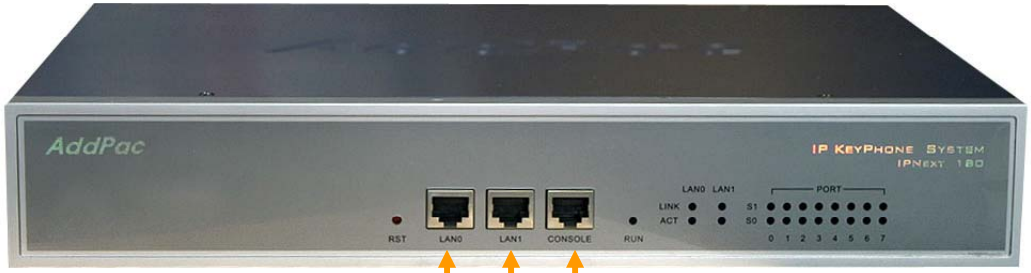
# Hardware Specification

IPNext180 SIP Call Manager

64bit CPU

DSP

## Front Side




**LAN1 (10/100Mbps)**

**LAN0(10/100Mbps)**

**Console Port**

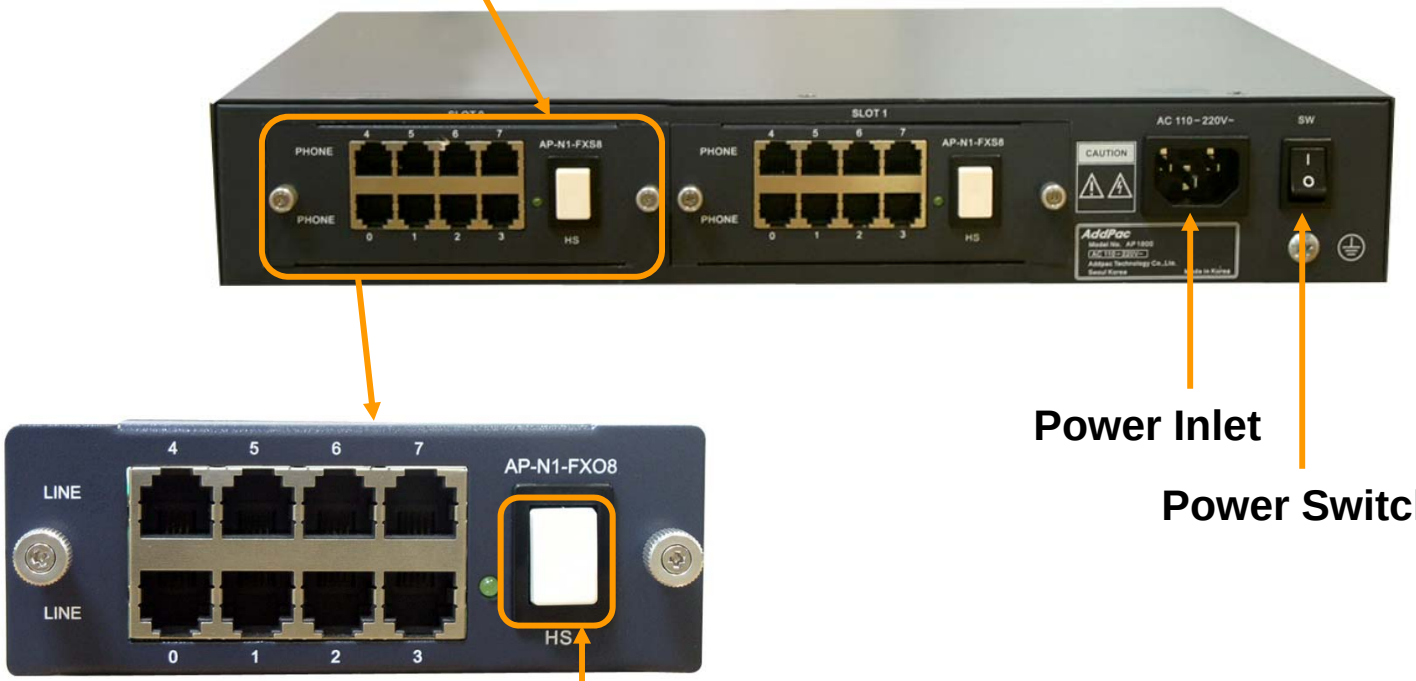
# Hardware Specification

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

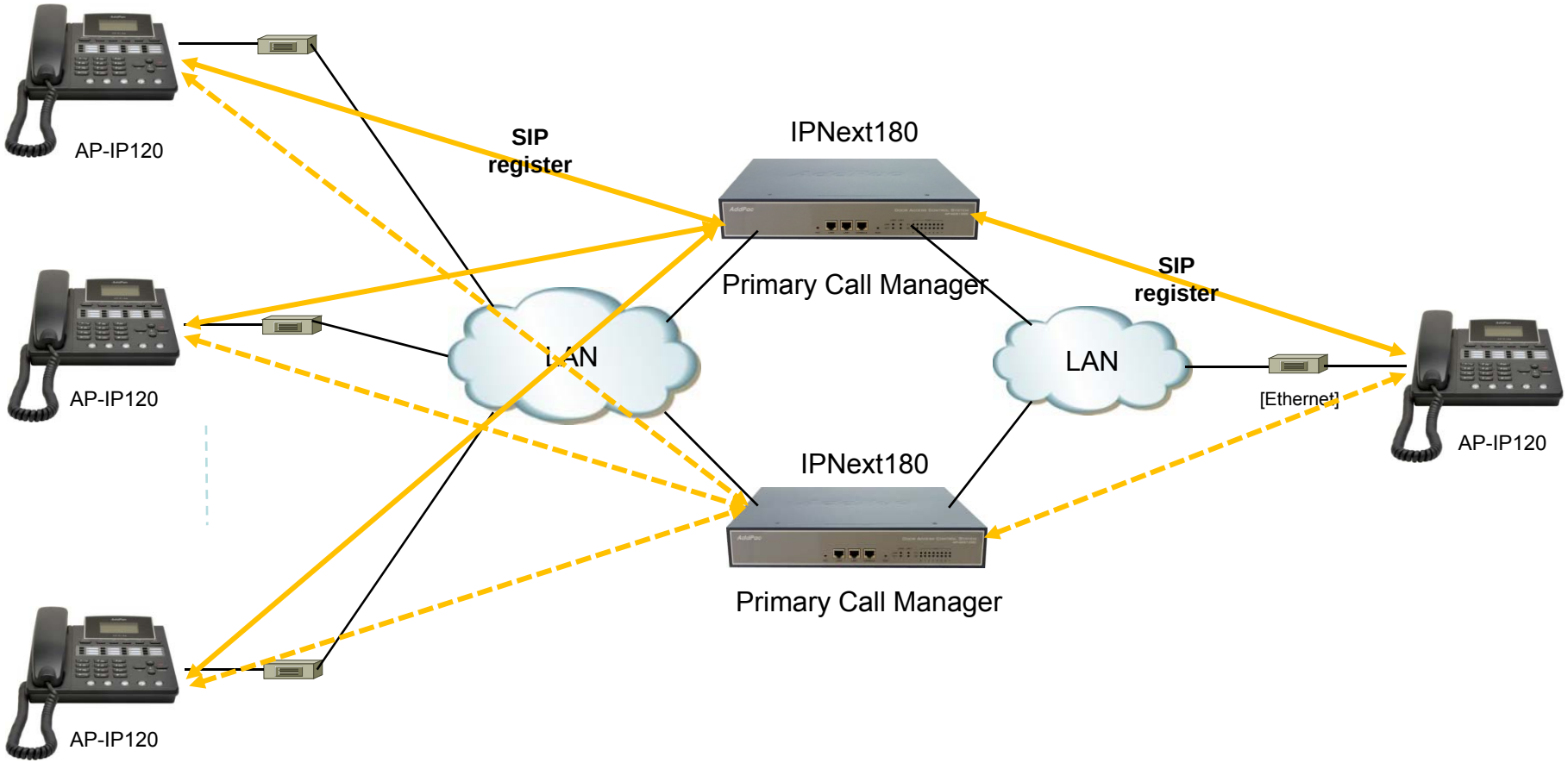
- **VoIP Interface Module**

| | | |
|---|---|---|
| **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 Paging Terminals

# AP601

## SIP Paging Terminal



# Product Overview

## AP601 SIP Paging Terminal

- SIP based Voice Broadcasting Terminal Solution
- Hardware Architecture for Voice Broadcasting Terminal Service
- Remote Broadcasting Service at terminal side
- High Quality Voice Codec Support (High Quality Codec, G.711, G.726, etc)
- RTP/UDP Protocol Support
- Unicast and Multicast Broadcasting Scheme
- SPMS (SIP Paging Management Software) Support
- One(1) channel SPK/MIC Port
- Volume Control Button Support (Up, Down)
- Option (AP601A) : AMP. Built-in
- High-Quality Audio/Voice Service
- Firmware Upgradeable Architecture
- Broadcasting Solution with Outstanding Network Service Capability
- External Power Supply

# Product Highlights

## AP601 SIP Paging Terminal

SIP based Broadcasting Terminal

Unicast & Multicast
Broadcasting Service Scheme

SPMS (SIP Paging
Management Software)
Support

Excellent Voice Quality
for Broadcasting

High Performance RISC CPU
& Programmable DSP




SPK/MIC Audio Interface
Support

Volume Control Button
Support

External Power Supply

Optional Built-in AMP.
(AP601A)

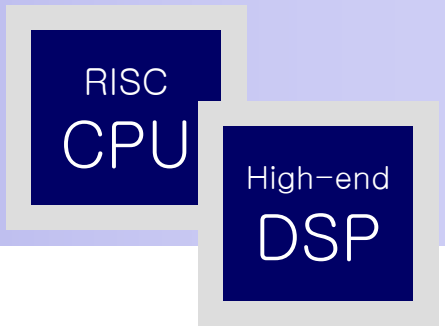
IP based Network Protocol
& Audio/Voice Traffic QoS

Broadband IP Networking
With One(1) 10/100Mbps Fast Ethernet

RS232C Console Interface
Support for Command Line Interface

# Hardware Specification

AP601 SIP Paging Terminal




- RISC Microprocessor Computing Power
- High-end Programmable DSP Hardware Architecture
- High Quality Audio Encoding/Decoding Service
- One(1) 10/100Mbps Fast Ethernet (RJ45)
- One(1) RS-232C Interface (RJ45)
- SPK/MIC Audio Interface Support
- Volume Control Button Support (Up, Down)
- External Power Supply Support
- Option : 30Watt Digital AMP. Built-in (AP601A)

# Hardware Specification

AP601 SIP Paging Terminal

RISC CPU

High-end DSP

## Front Side




Speaker Out

MIC Input

Speaker Volume Control Button (Up, Down)

RS232C Console

Fast Ethernet (LAN)

ON-AIR LED

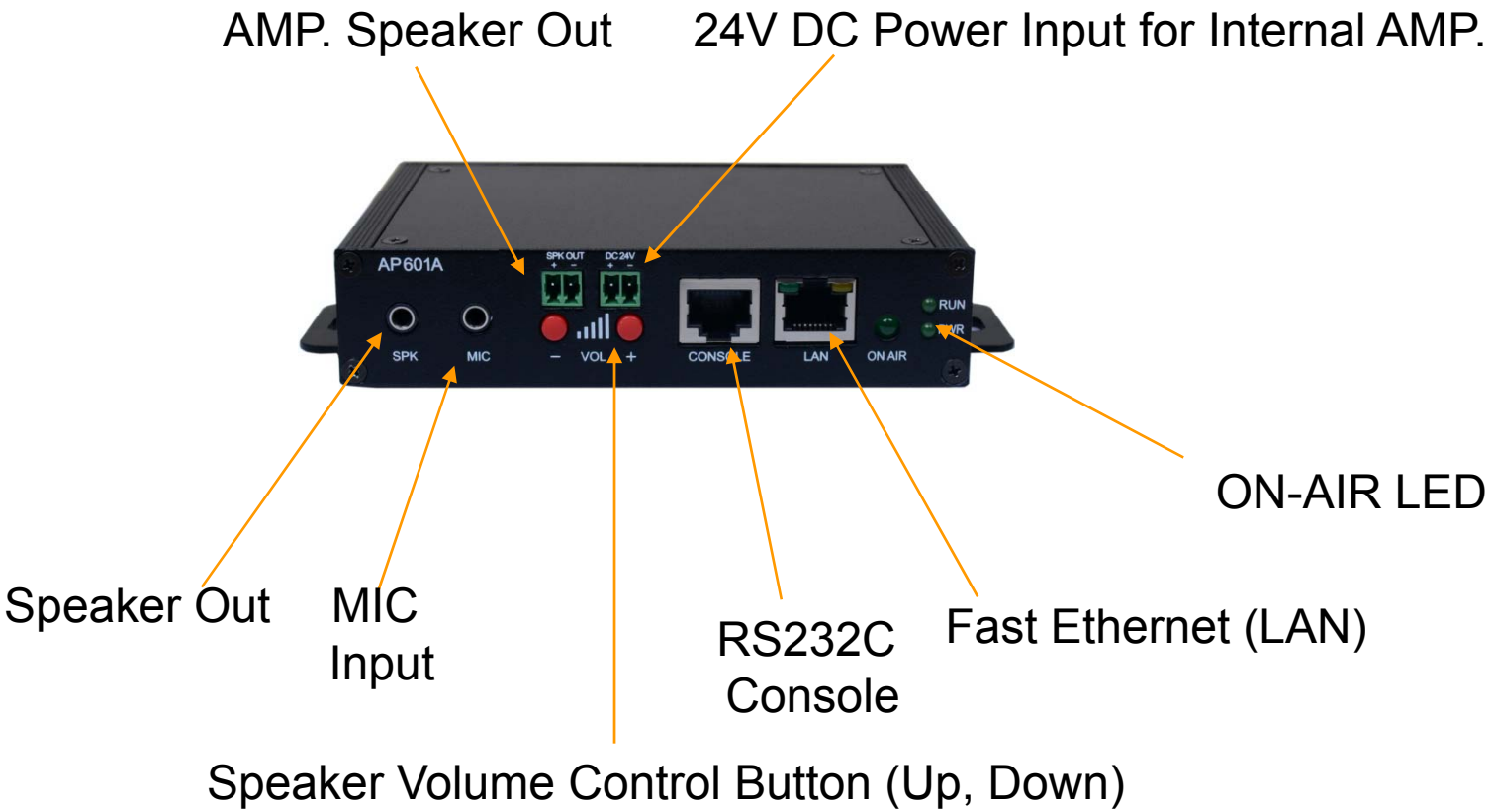
# Hardware Specification

AP601 SIP Paging Terminal

RISC CPU

High-end DSP

## Front Side
## (AMP. Built-In Model : AP601A)



# Hardware Specification

AP601 SIP Paging Terminal

RISC CPU

High-end DSP

## Back Side




External Power Supply Adaptor

Power Switch

# 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

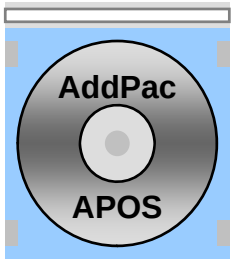
- RISC+DSP Microprocessor Computing Power (Dual Processor Architecture)
- High Quality 4.3 Inch Color LCD Panel
- 25 Speed Dial Key & User Presence Indication LED
- Optional PSTN Backup Interface
  - FXO Interface
- High quality Audio and Voice Interface
  - Stereo Audio Input Connector
  - Stereo Audio 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

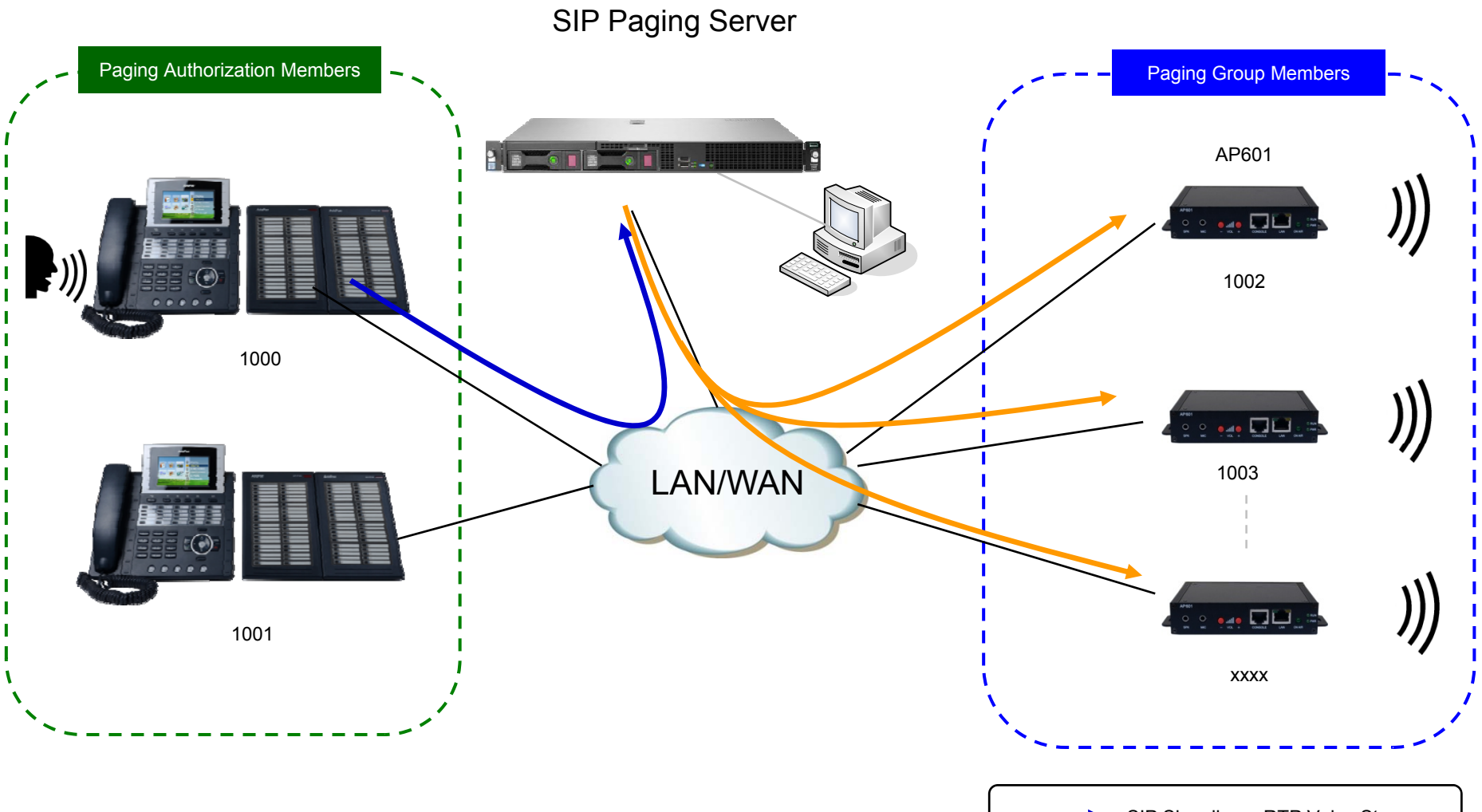
- Built-in AddPac APOS Internetworking Software
  - Scalability, Functionality, and Stability Features
  - Audio Traffic QoS Control
- Programmable Video, Audio, and Voice Services
  - Audio, and Voice Codec
- Firmware Upgradeable Architecture
- Industry Standard IP based Network Protocol Features






# AP-IP300 SIP Broadcasting Phone

## Application Area



# AP-IP120

## SIP Broadcasting Phone



AP-IP120 AP-PT20 (40 Speed Dial Key)

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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

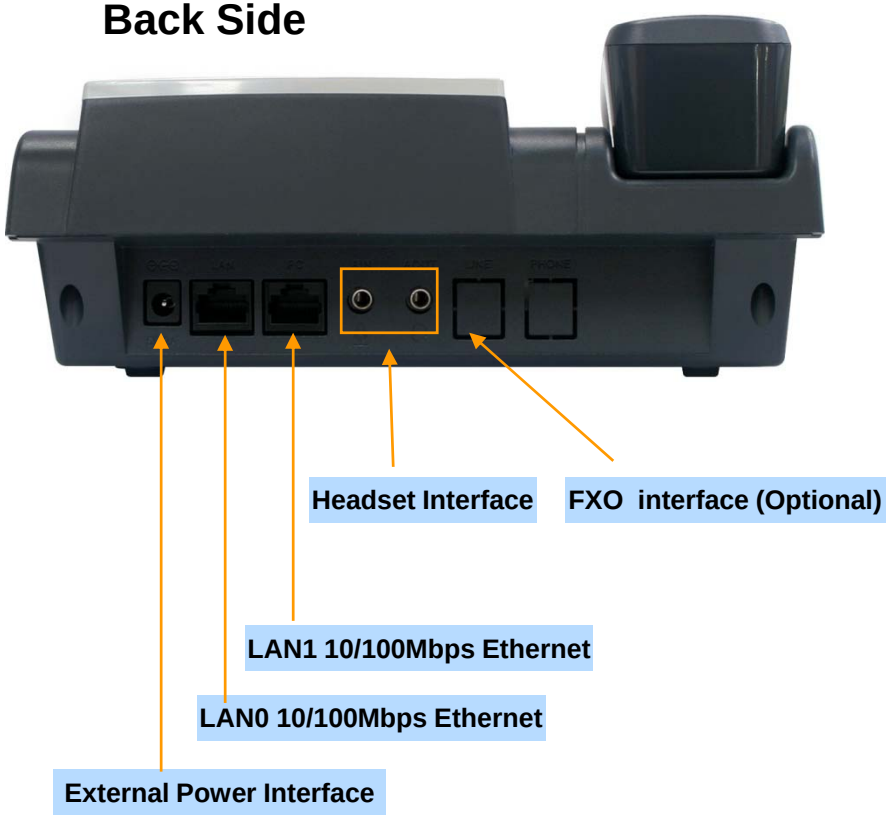
## AP-IP120 SIP Broadcasting Phone

### 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 |

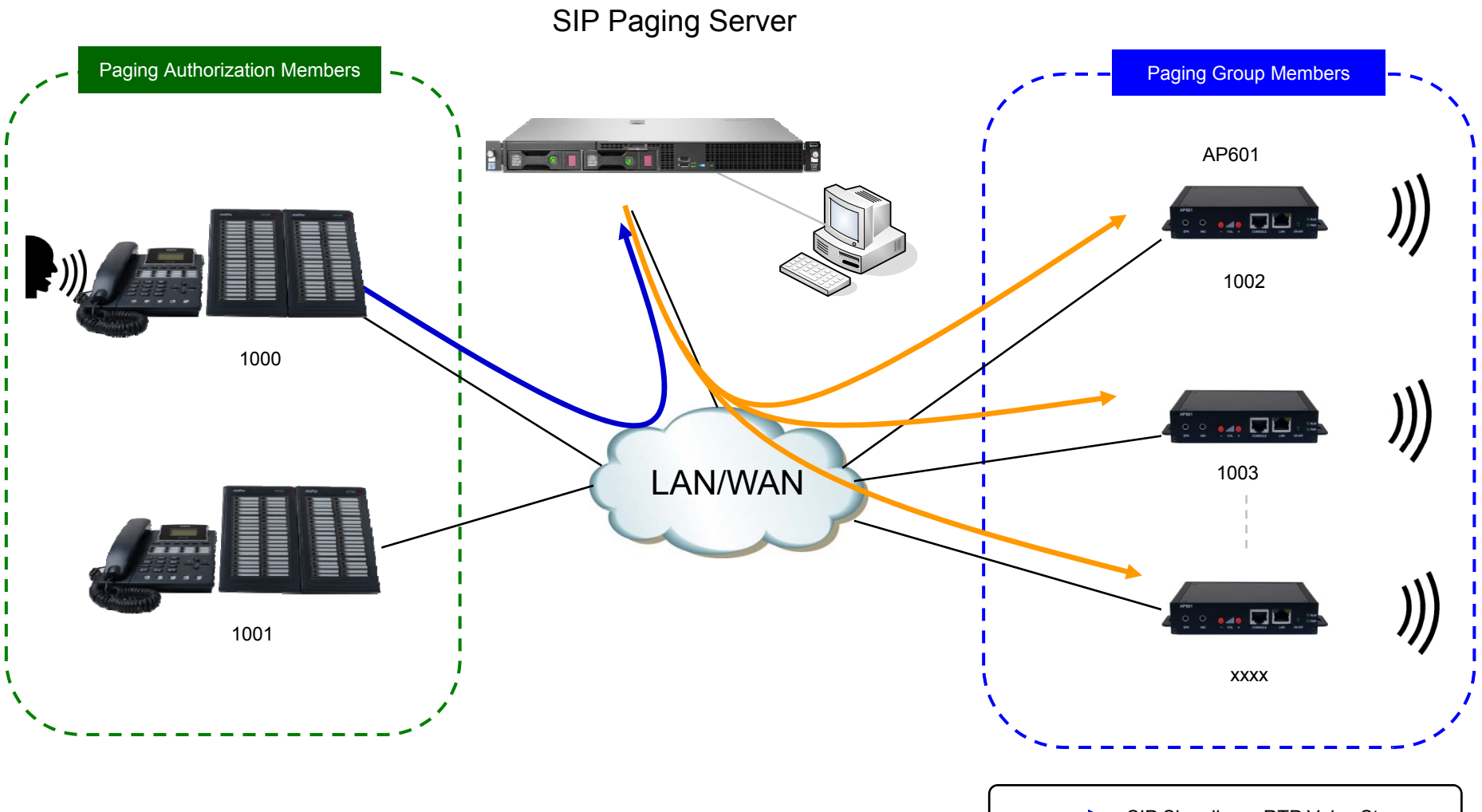
### Network interface Configurations

**Back Side**



# AP-IP120 SIP Broadcasting Phone

## Application Area

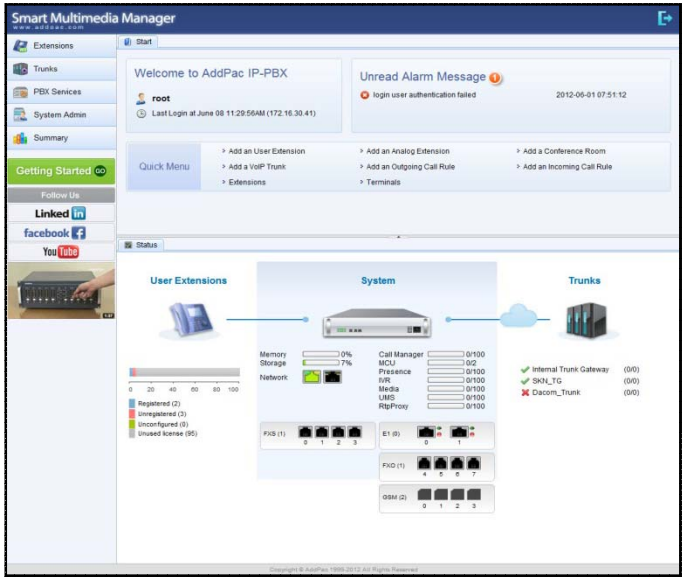


# Web Smart Multimedia Manager (WSMM)



# 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)

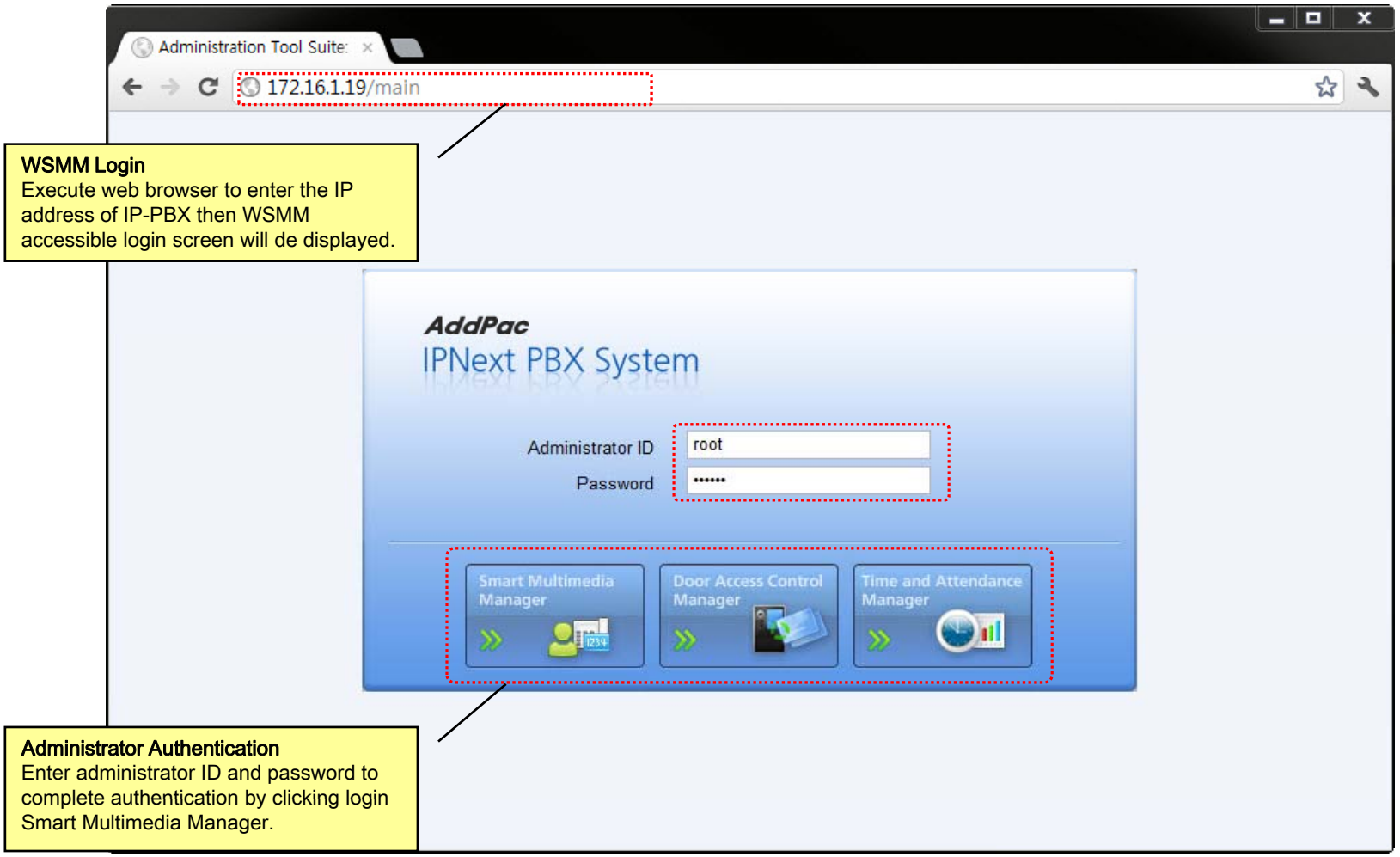
- Simple Menu and Easy Configuration
- Provides Built-In IVR Scenario Editor and Service Configuration
- Provides easy-to-user IP-PBX System API Services and ways to integration with 3rd party systems
- Integrated voice line management such as FXS, FXO, E1, GSM, 3G
- User portal to configure personal 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,... )

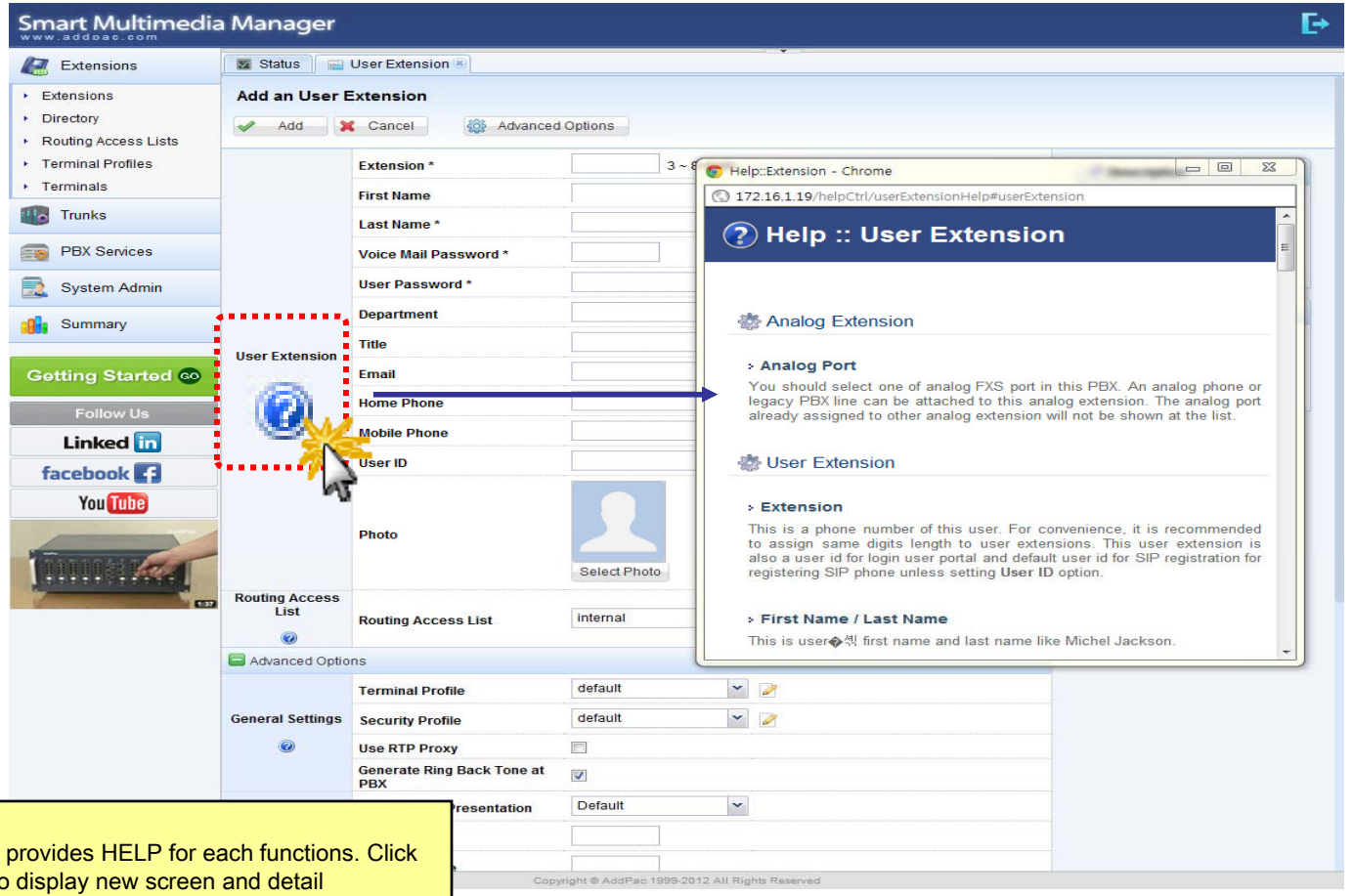
# Login




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

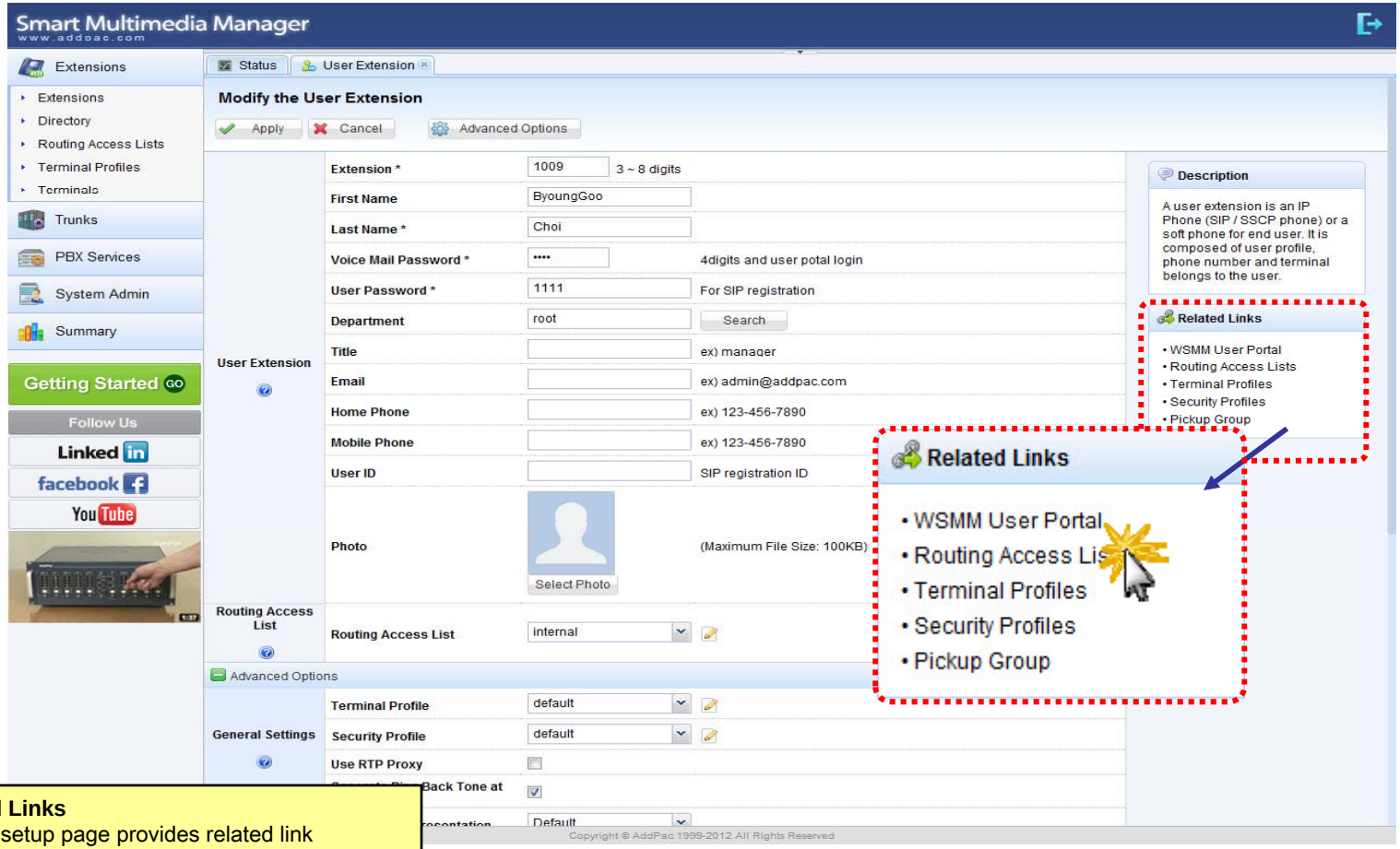
**Administrator Authentication**
Enter administrator ID and password to complete authentication by clicking login Smart Multimedia Manager.

# Help



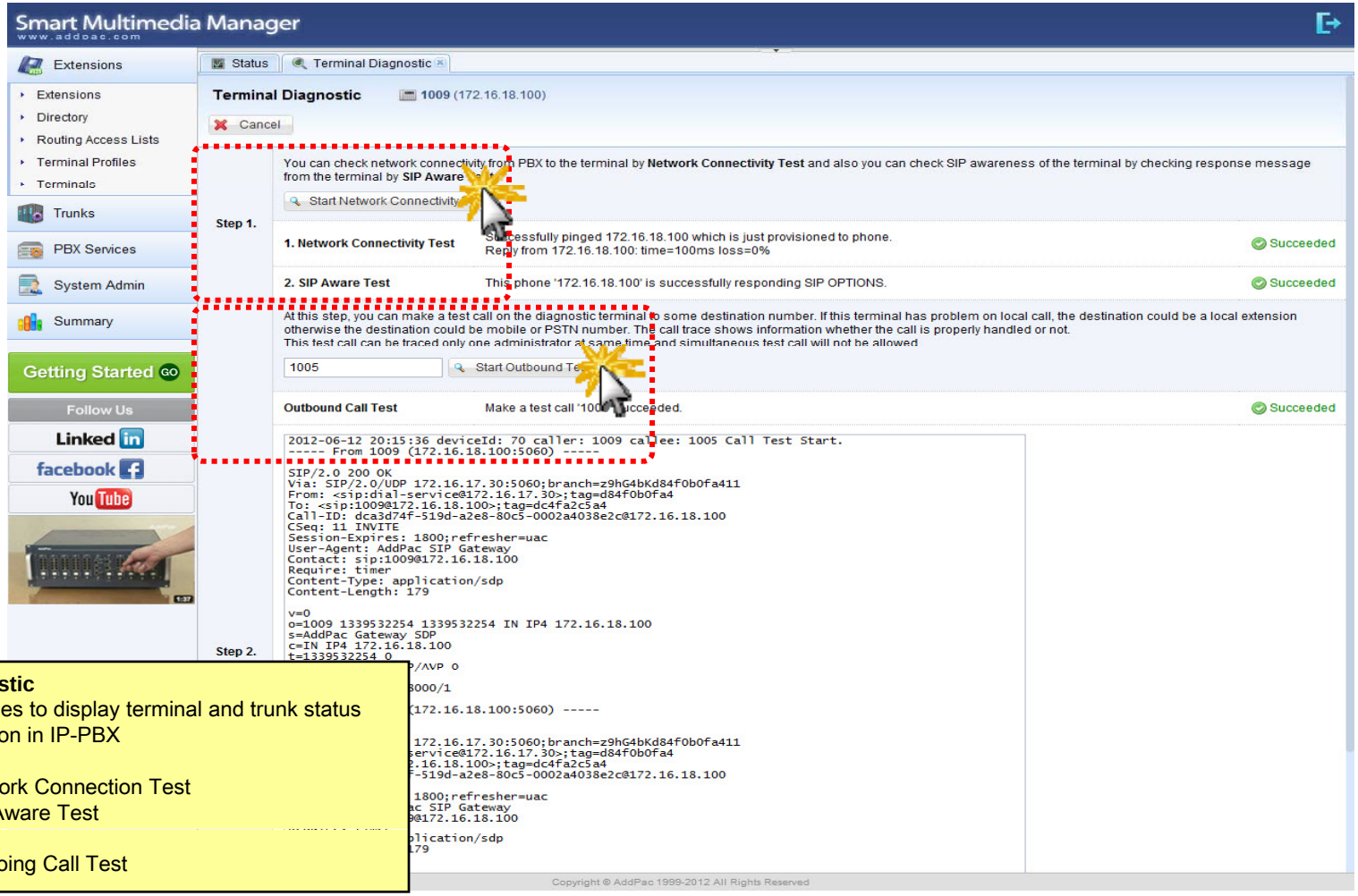

**HELP**
WSMM provides HELP for each functions. Click HELP to display new screen and detail description of setup is clearly explained in homepage.

# Related Links




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

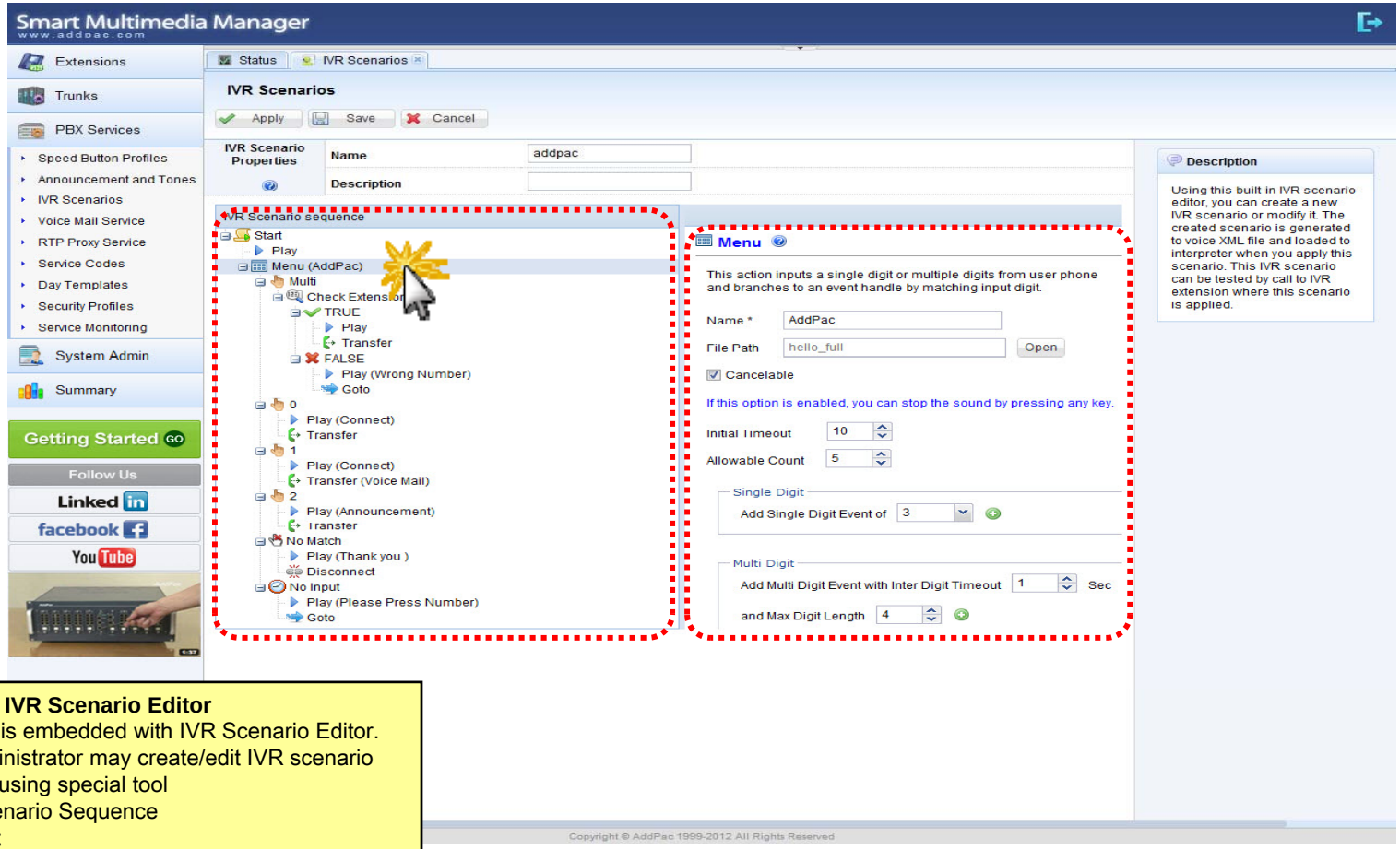
# Diagnostic




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

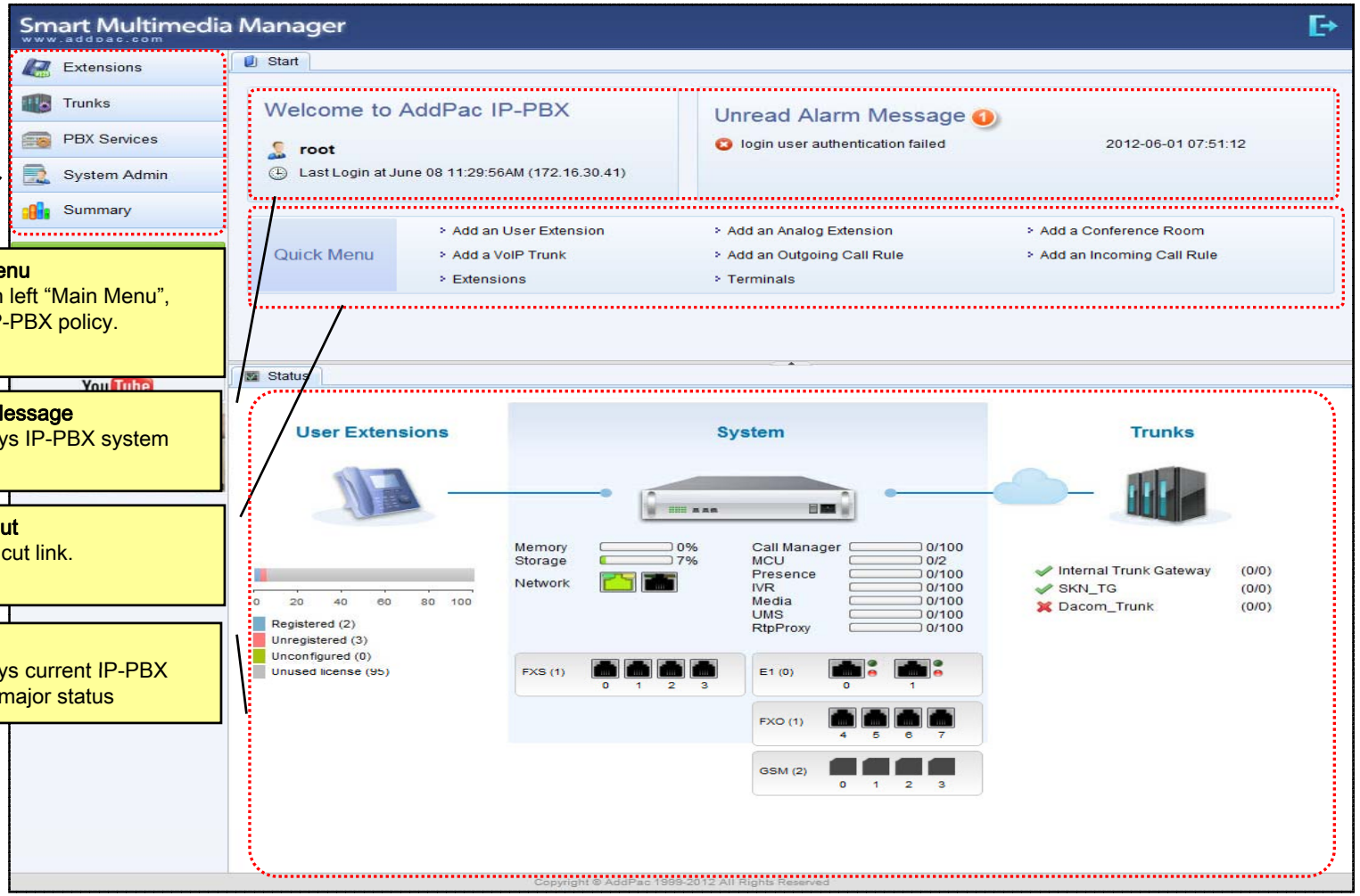
Step 2.
- Outgoing Call Test

# 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



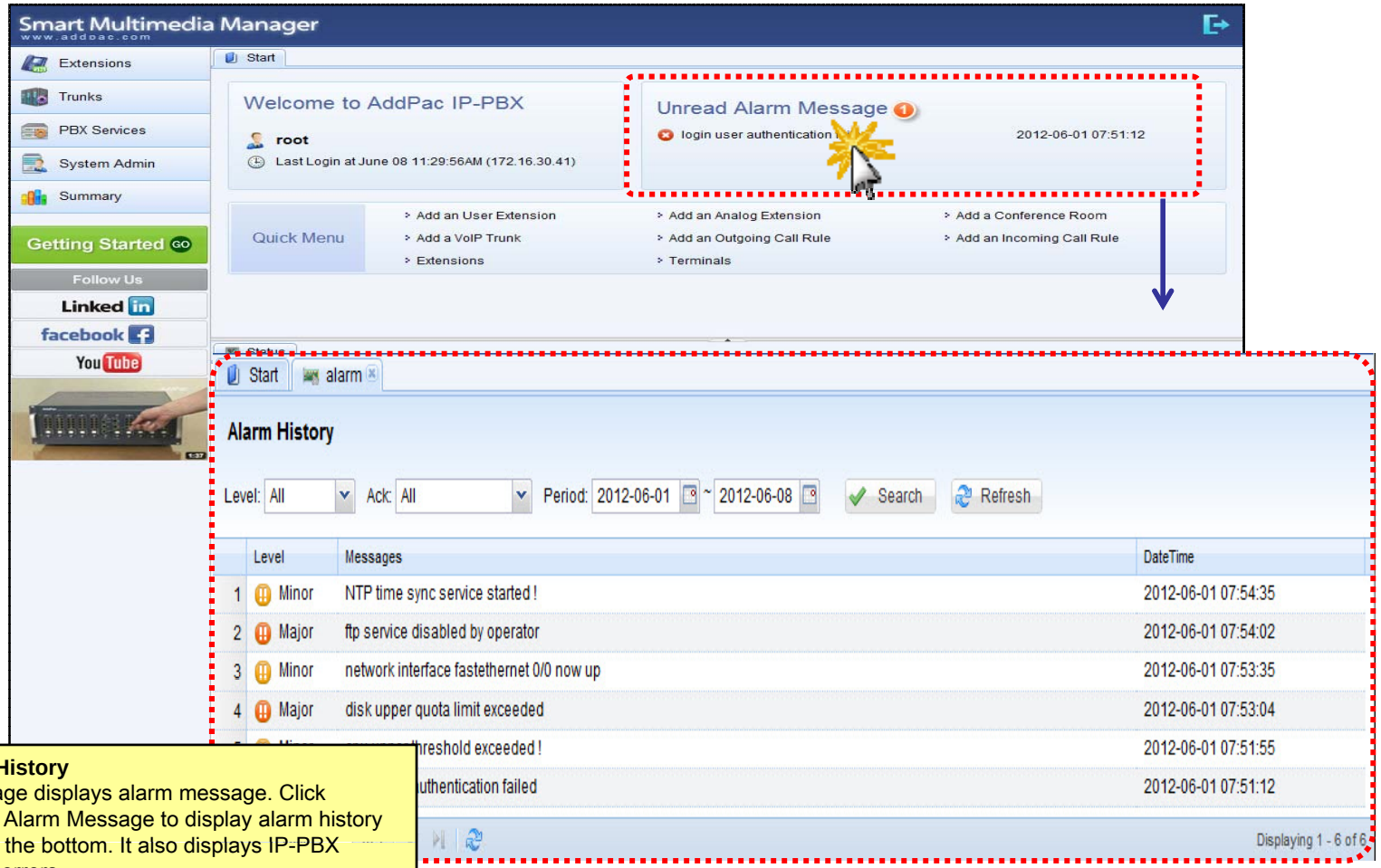

**Main Menu**
Through left "Main Menu", setup IP-PBX policy.

**Alarm Message**
It displays IP-PBX system errors

**Short Cut**
A short cut link.

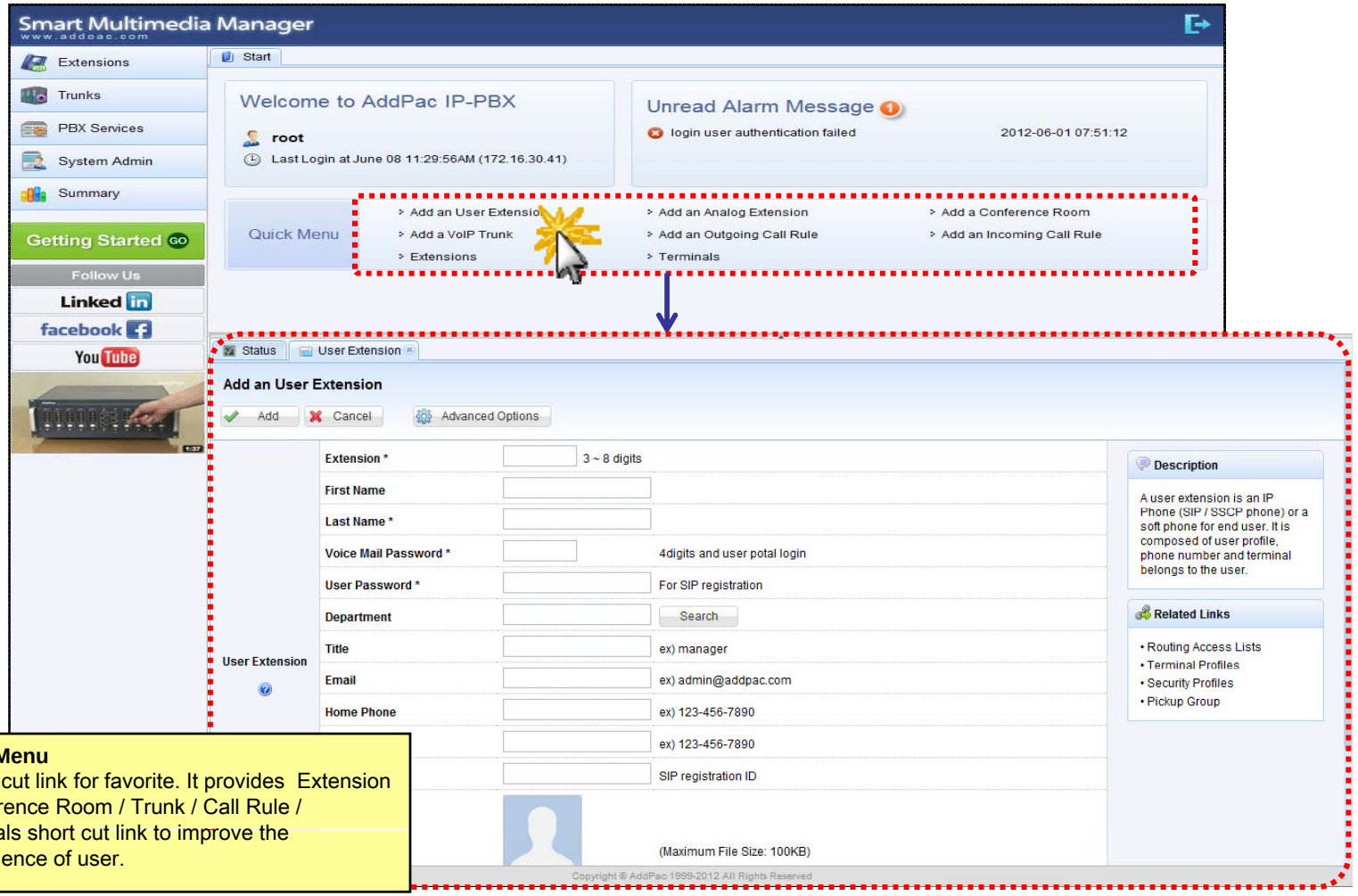
**Status**
It displays current IP-PBX system major status

# Main - Alarm History



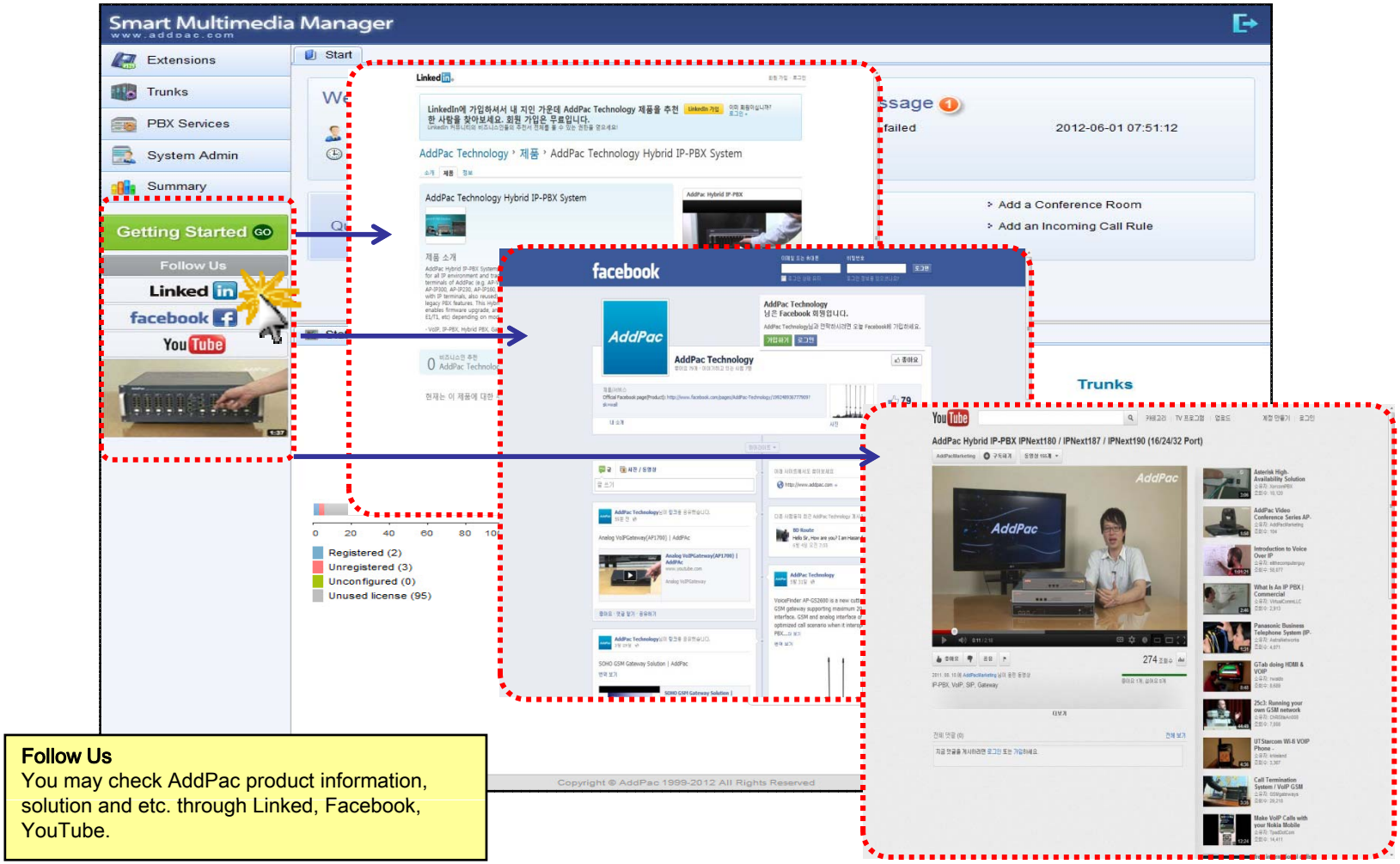

**Alarm History**
Main page displays alarm message. Click Unread Alarm Message to display alarm history page at the bottom. It also displays IP-PBX system errors.

# Main – Quick Menu



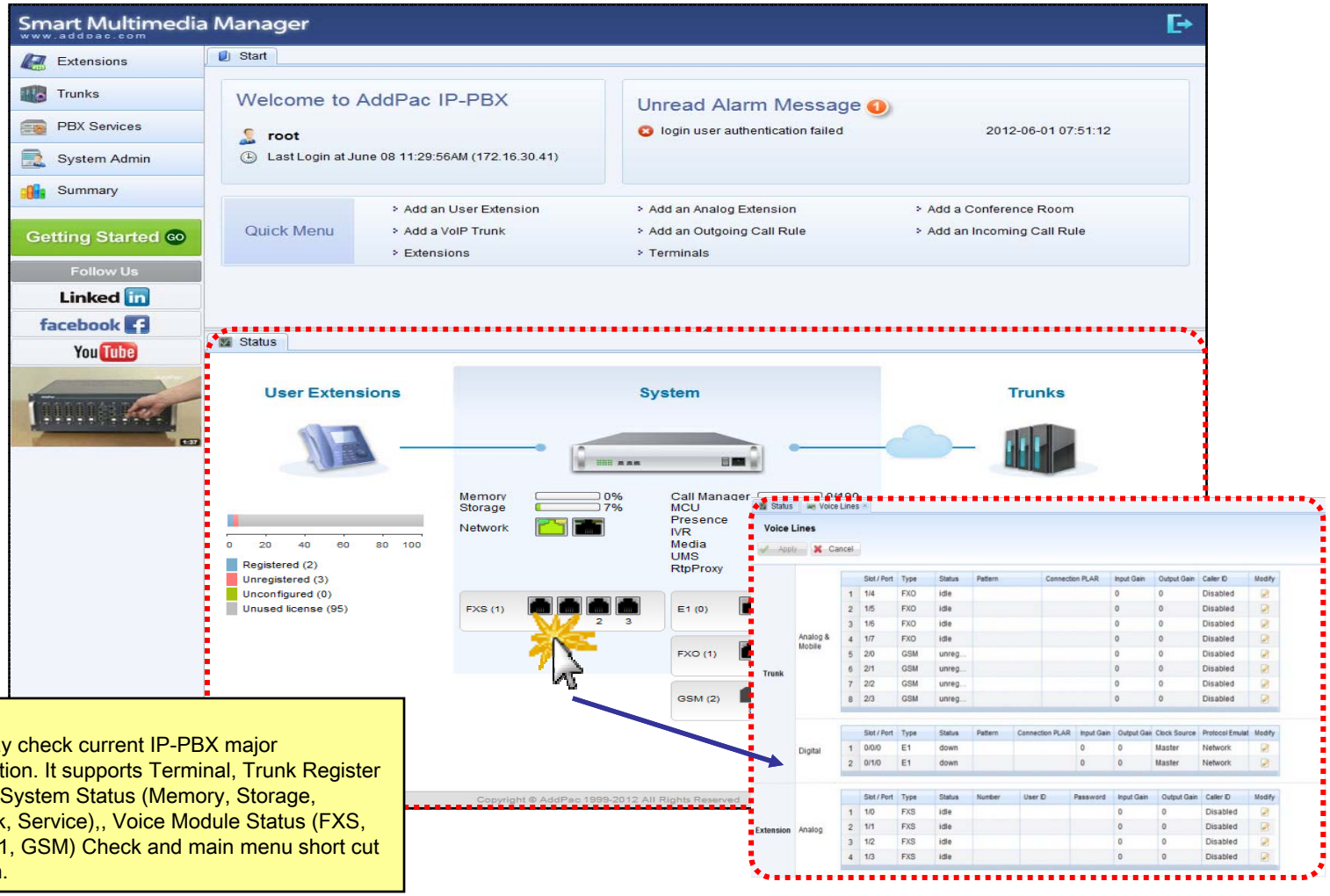

**Quick Menu**
A short cut link for favorite. It provides Extension / Conference Room / Trunk / Call Rule / Terminals short cut link to improve the convenience of user.

# Main – Follow Us



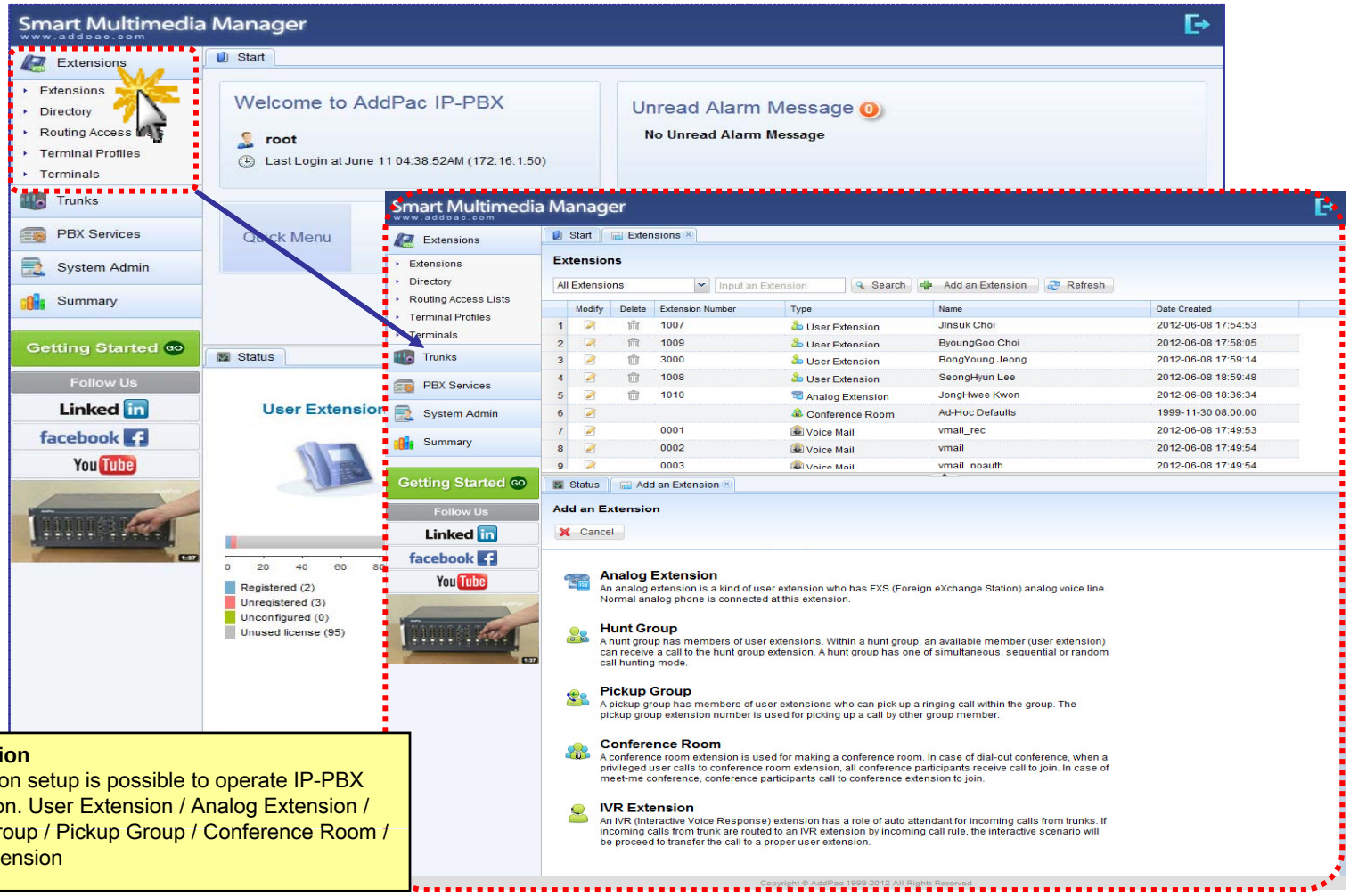

**Follow Us**
You may check AddPac product information, solution and etc. through Linked, Facebook, YouTube.

# Main – Status Monitoring



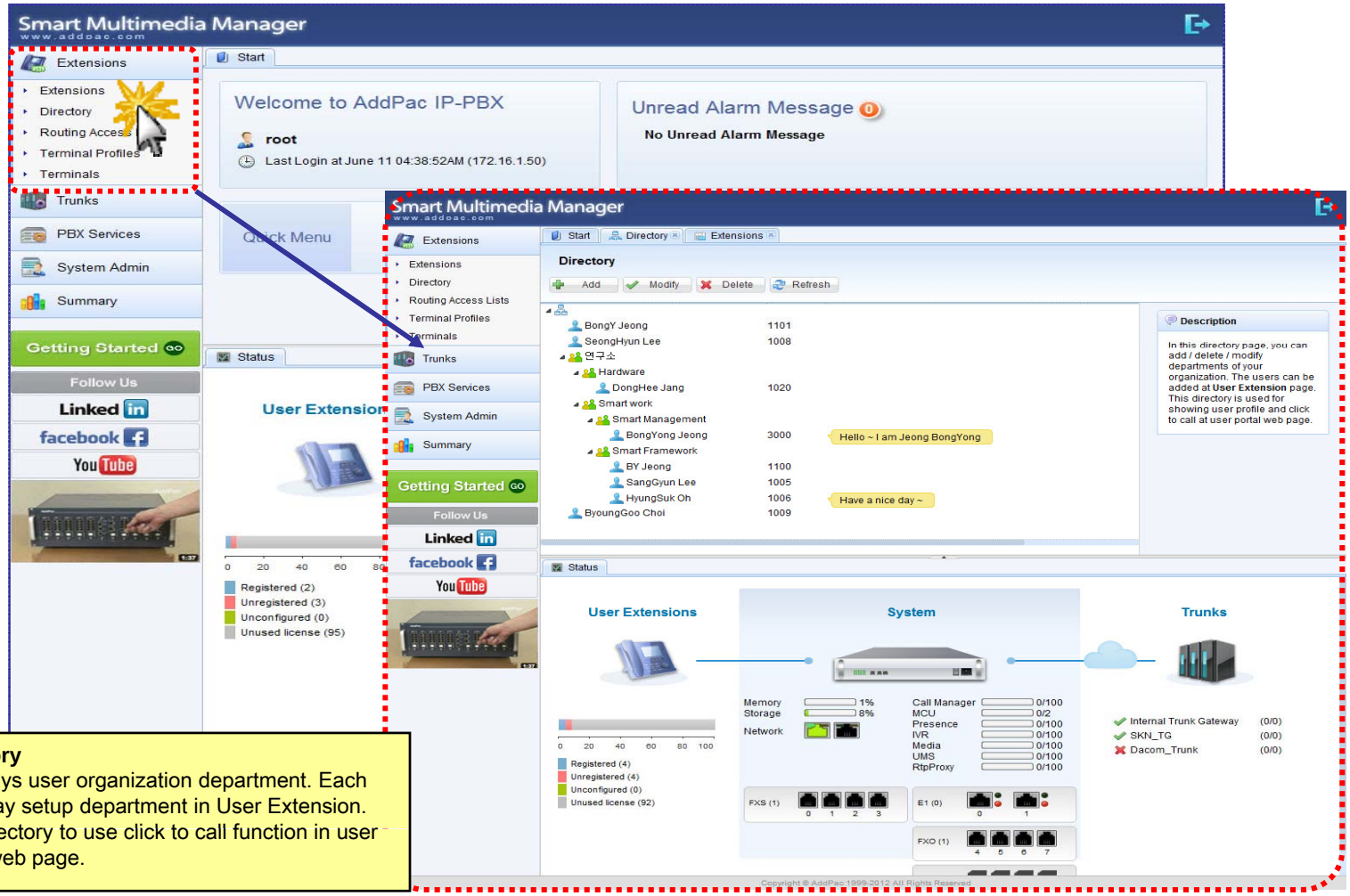

**Status**
You may check current IP-PBX major information. It supports Terminal, Trunk Register Status, System Status (Memory, Storage, Network, Service),, Voice Module Status (FXS, FXO, E1, GSM) Check and main menu short cut function.

# Extension - Extensions



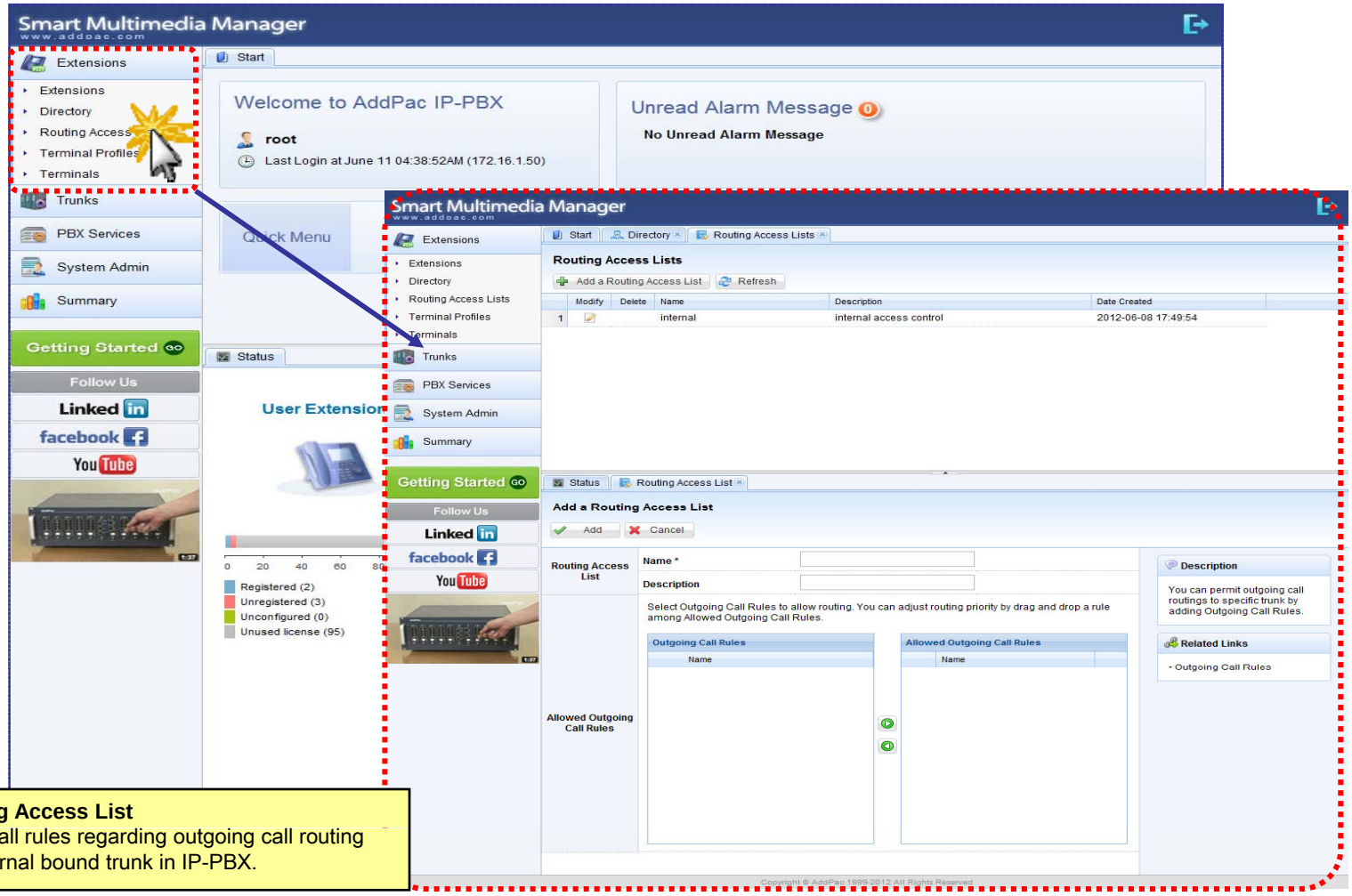

**Extension**
Extension setup is possible to operate IP-PBX operation. User Extension / Analog Extension / Hunt Group / Pickup Group / Conference Room / IVR Extension

# Extension - Directory



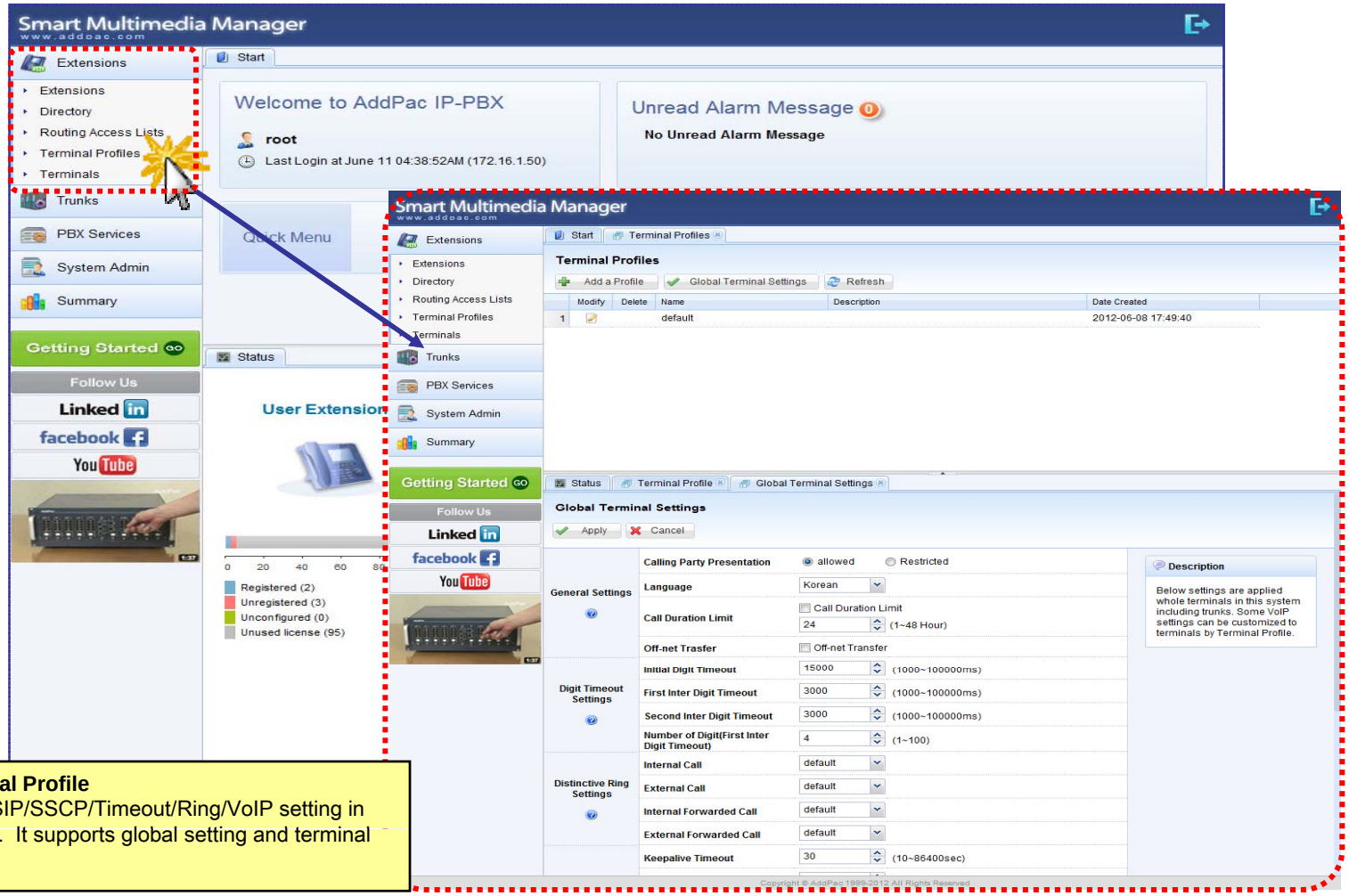

**Directory**
It displays user organization department. Each user may setup department in User Extension. Use directory to use click to call function in user portal web page.

# Extension - Routing Access List



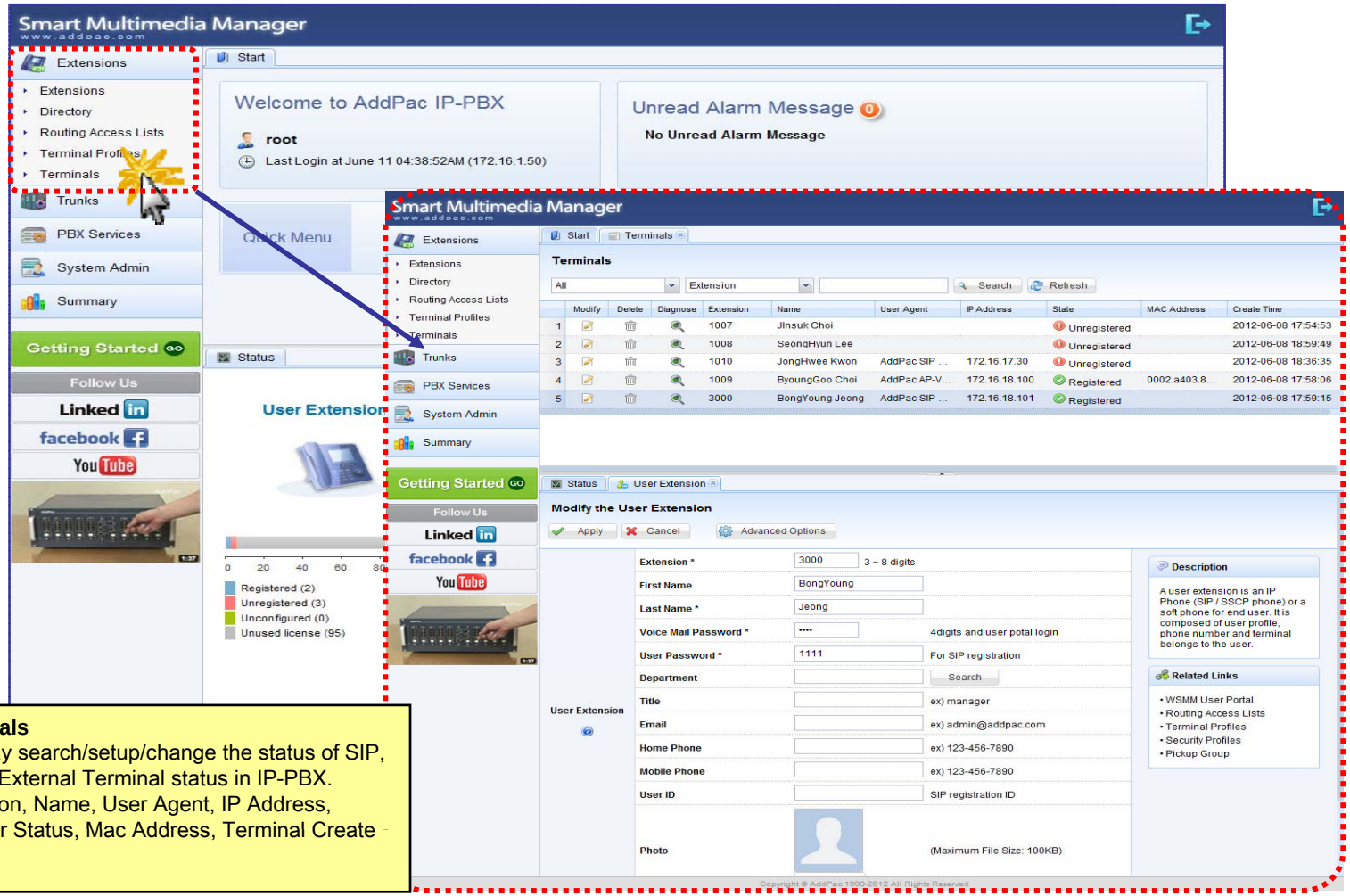

**Routing Access List**
Apply call rules regarding outgoing call routing for external bound trunk in IP-PBX.

# Extension - Terminal Profile



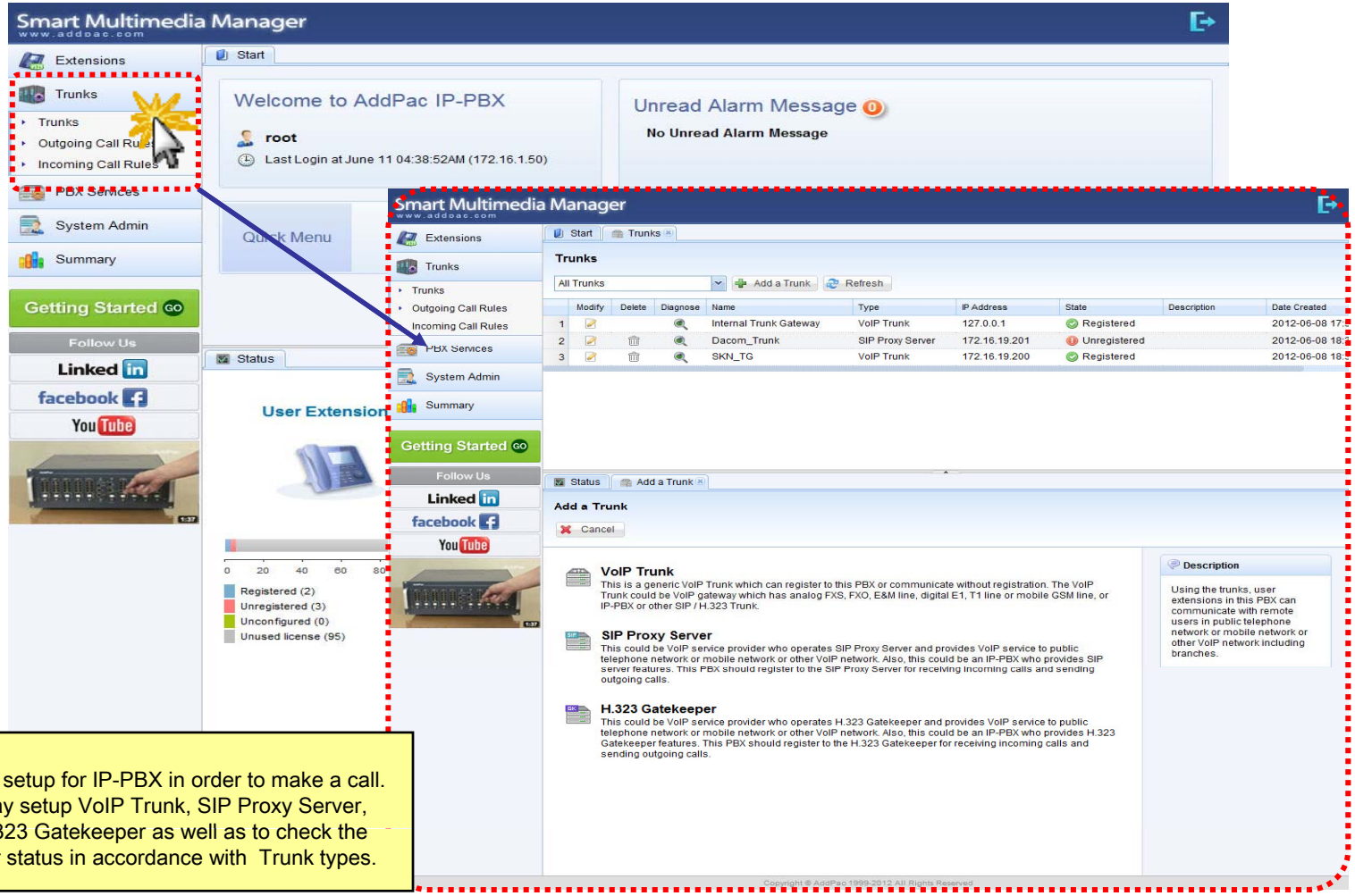

**Terminal Profile**
Setup SIP/SSCP/Timeout/Ring/VoIP setting in IP-PBX. It supports global setting and terminal profile.

# Extension - Terminals



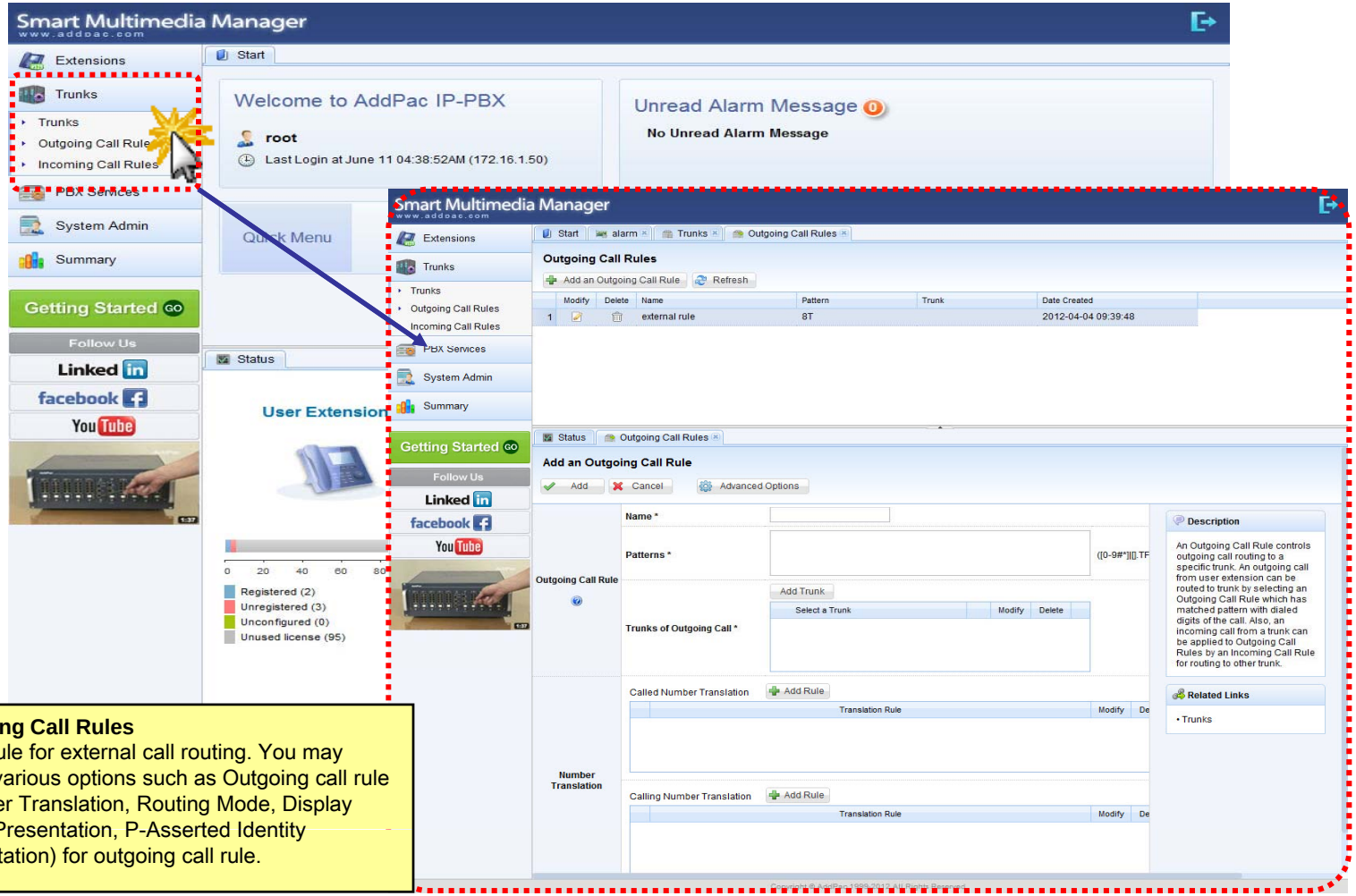

**Terminals**
You may search/setup/change the status of SIP, SSCP, External Terminal status in IP-PBX. Extension, Name, User Agent, IP Address, Register Status, Mac Address, Terminal Create Time

# Trunk - Trunks



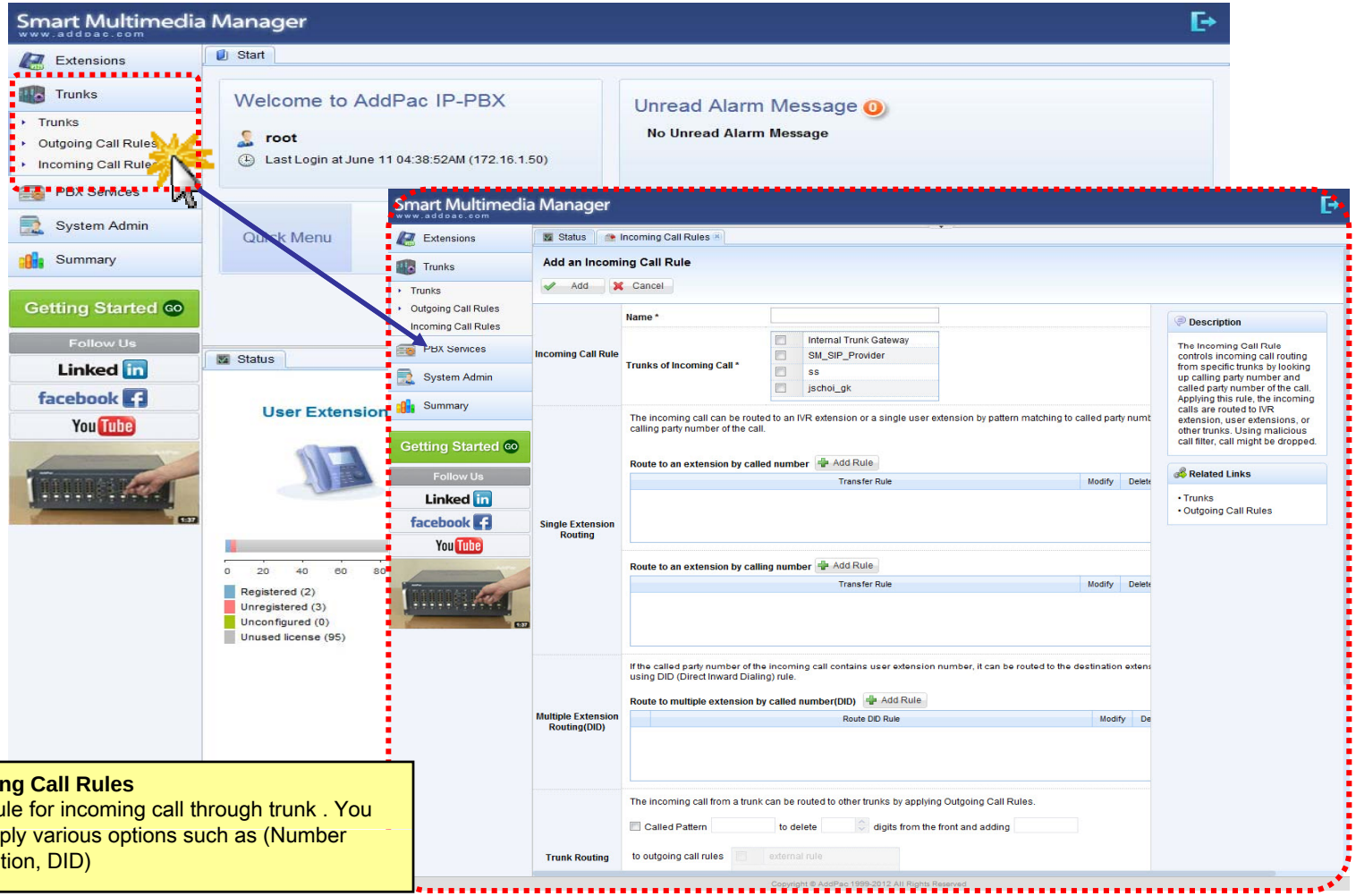

**Trunk**
A trunk setup for IP-PBX in order to make a call. You may setup VoIP Trunk, SIP Proxy Server, and H.323 Gatekeeper as well as to check the register status in accordance with Trunk types.

# Trunk - Outgoing Call Rules



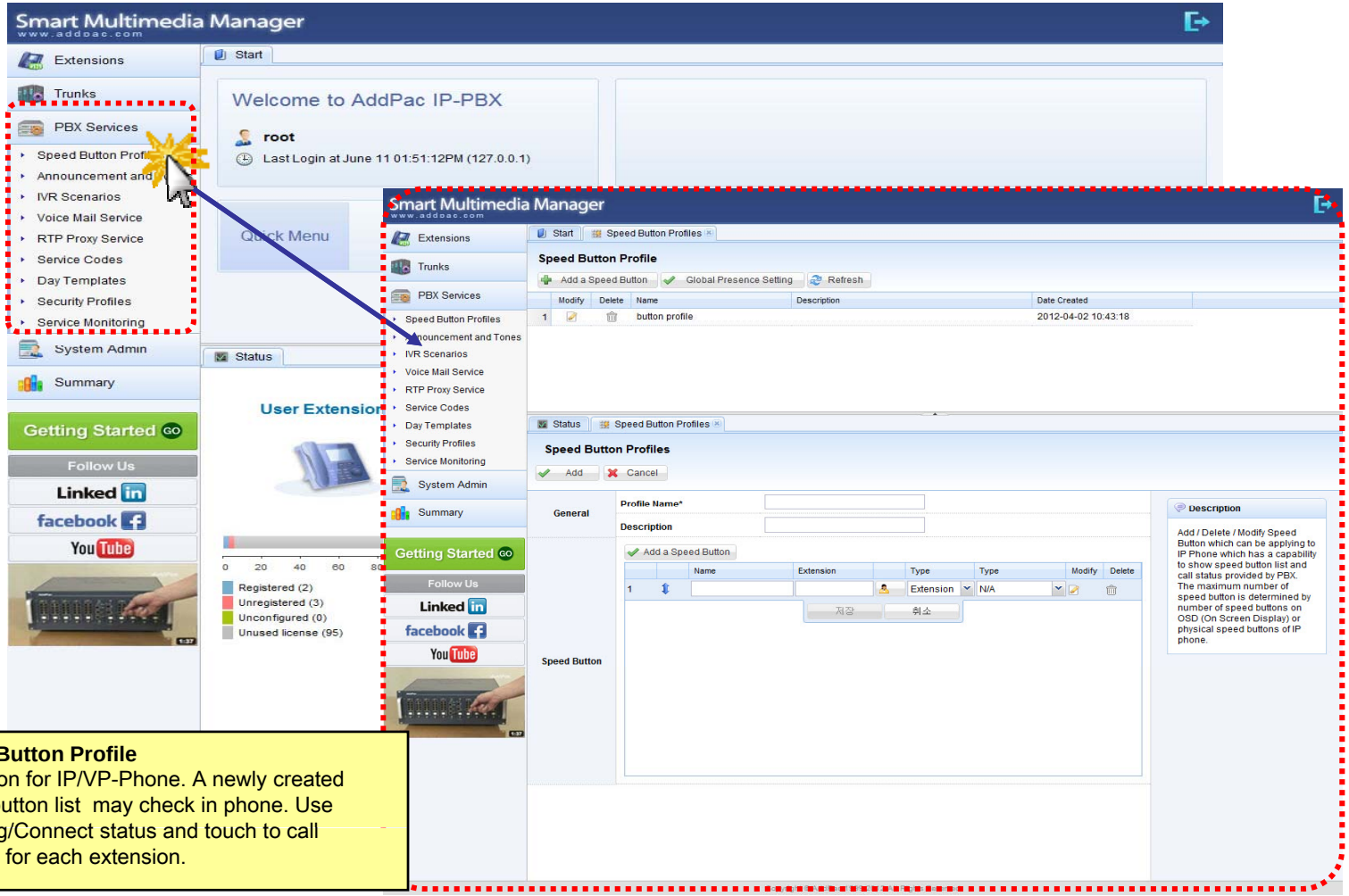

**Outgoing Call Rules**
A call rule for external call routing. You may apply various options such as Outgoing call rule (Number Translation, Routing Mode, Display Name Presentation, P-Asserted Identity Presentation) for outgoing call rule.

# Trunk - Incoming Call Rules



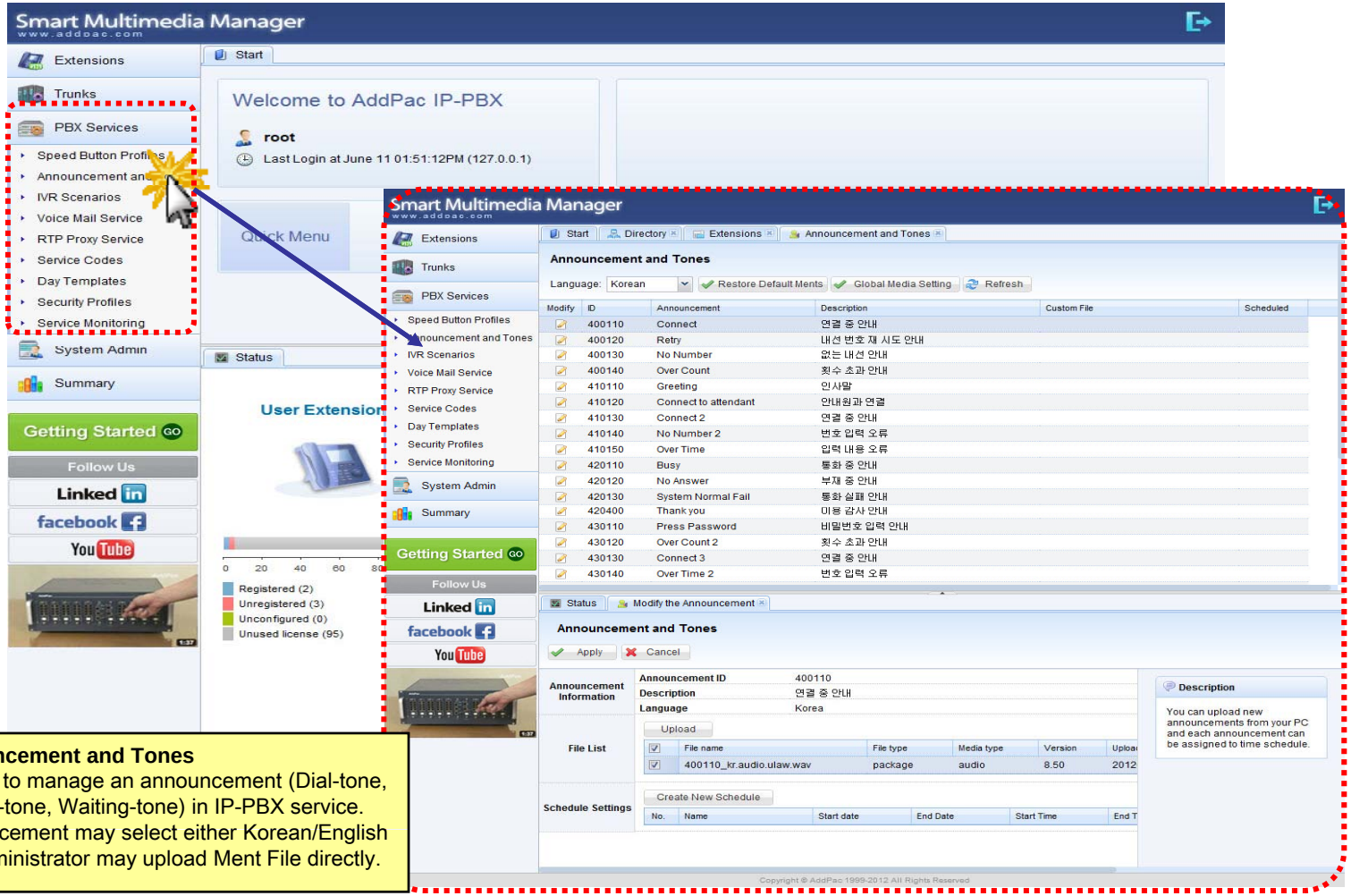

**Incoming Call Rules**
A call rule for incoming call through trunk . You may apply various options such as (Number Translation, DID)

# PBX Service - Speed Button Profiles



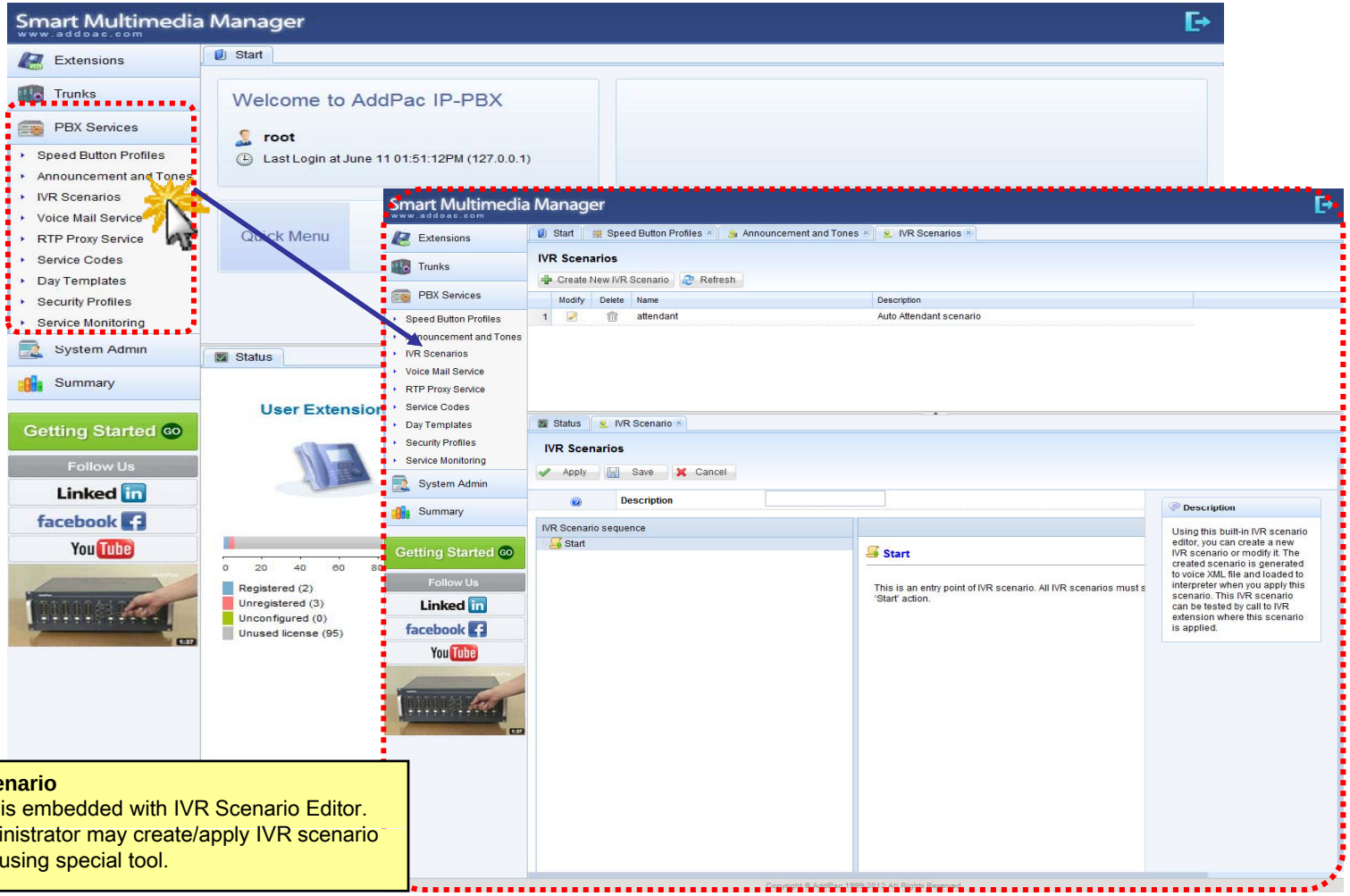

**Speed Button Profile**
A function for IP/VP-Phone. A newly created speed button list may check in phone. Use idle/Ring/Connect status and touch to call function for each extension.

# PBX Service - Announcement and Tones



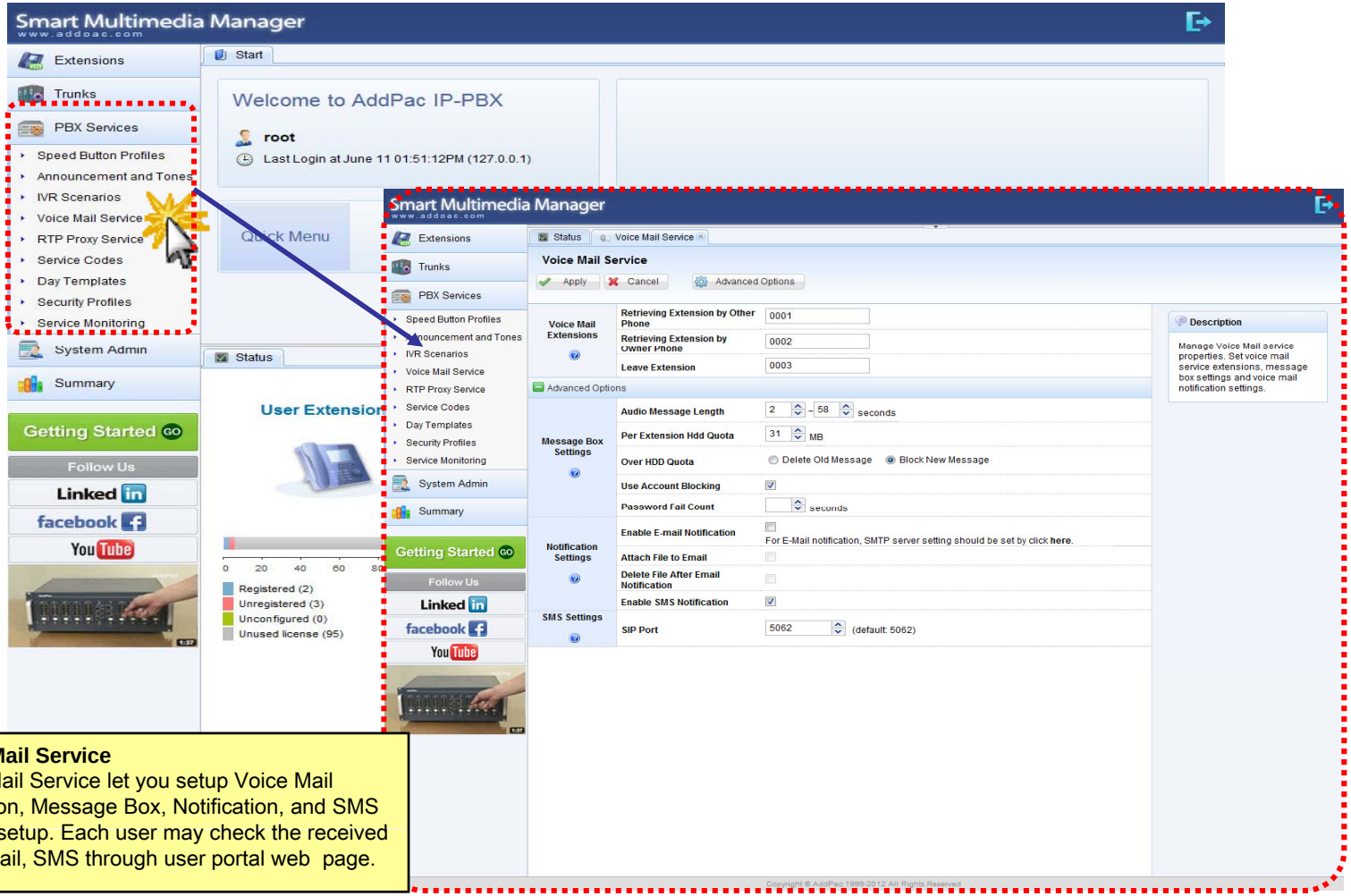

**Announcement and Tones**
A setup to manage an announcement (Dial-tone, Consult-tone, Waiting-tone) in IP-PBX service. Announcement may select either Korean/English and administrator may upload Ment File directly.

# PBX Service - IVR Scenarios



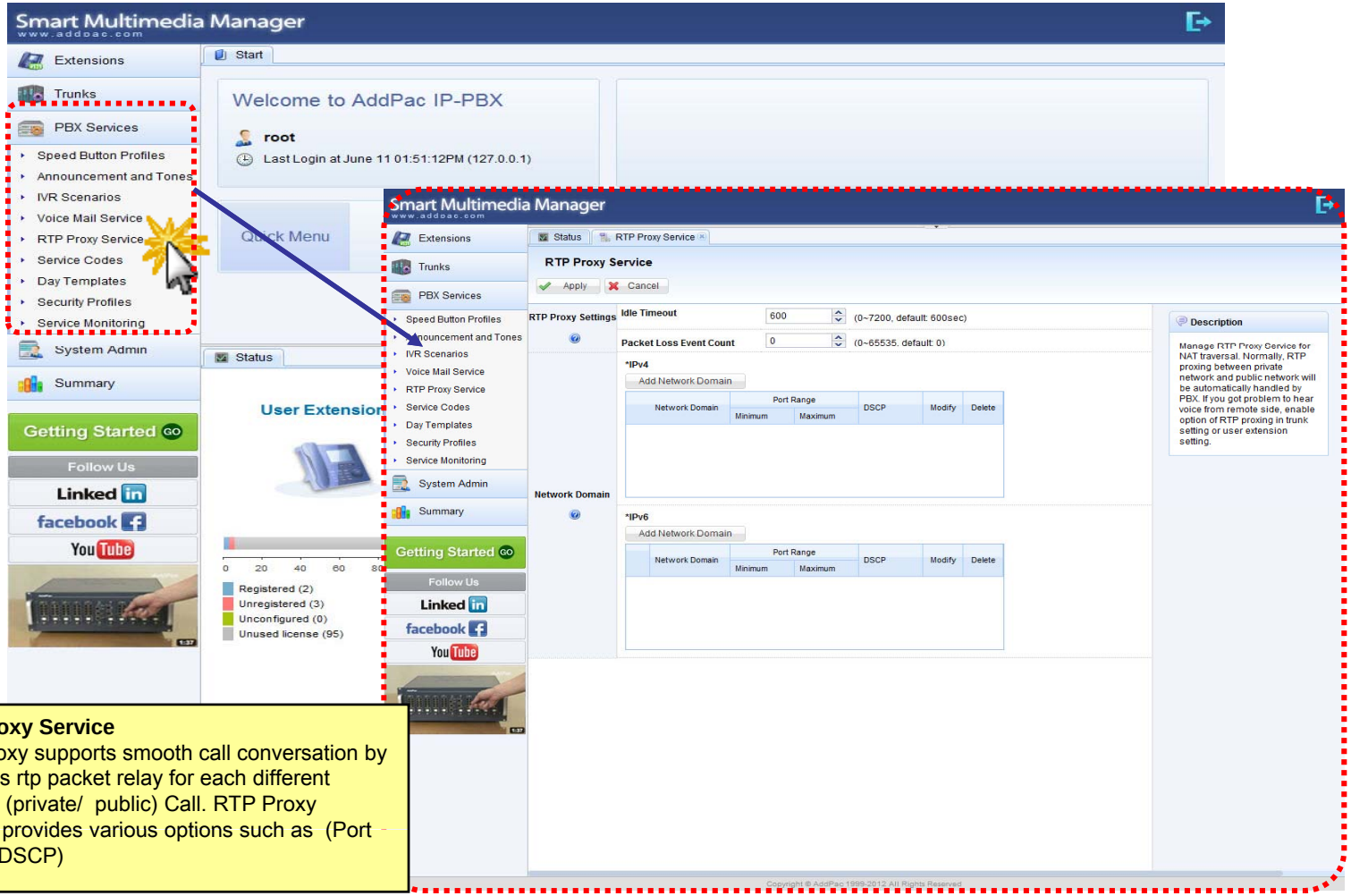

**IVR Scenario**
WSMM is embedded with IVR Scenario Editor. An administrator may create/apply IVR scenario without using special tool.

# PBX Service - Voice Mail Services



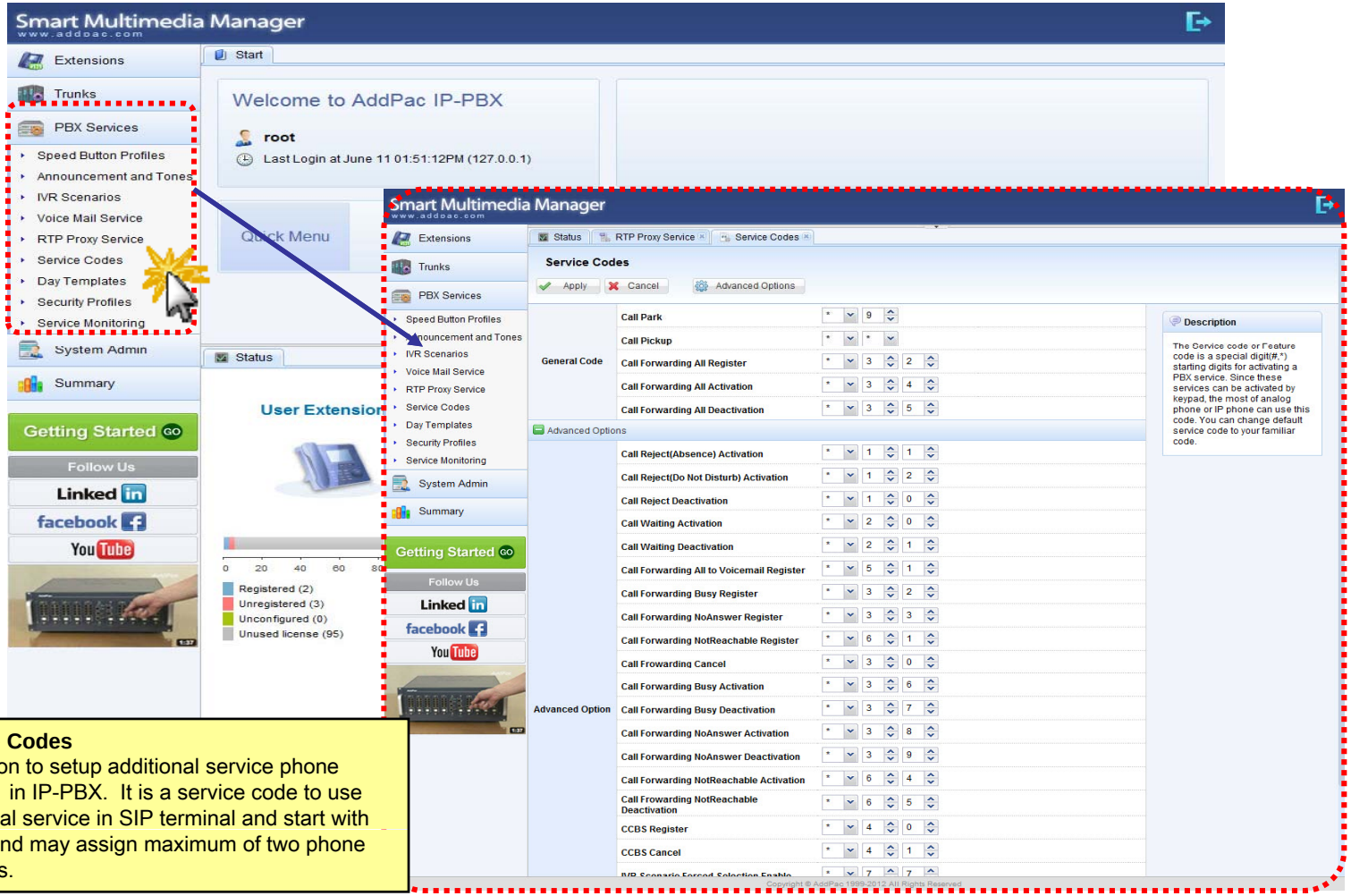

**Voice Mail Service**
Voice Mail Service let you setup Voice Mail Extension, Message Box, Notification, and SMS related setup. Each user may check the received voice-mail, SMS through user portal web page.

# PBX Service - RTP Proxy Service



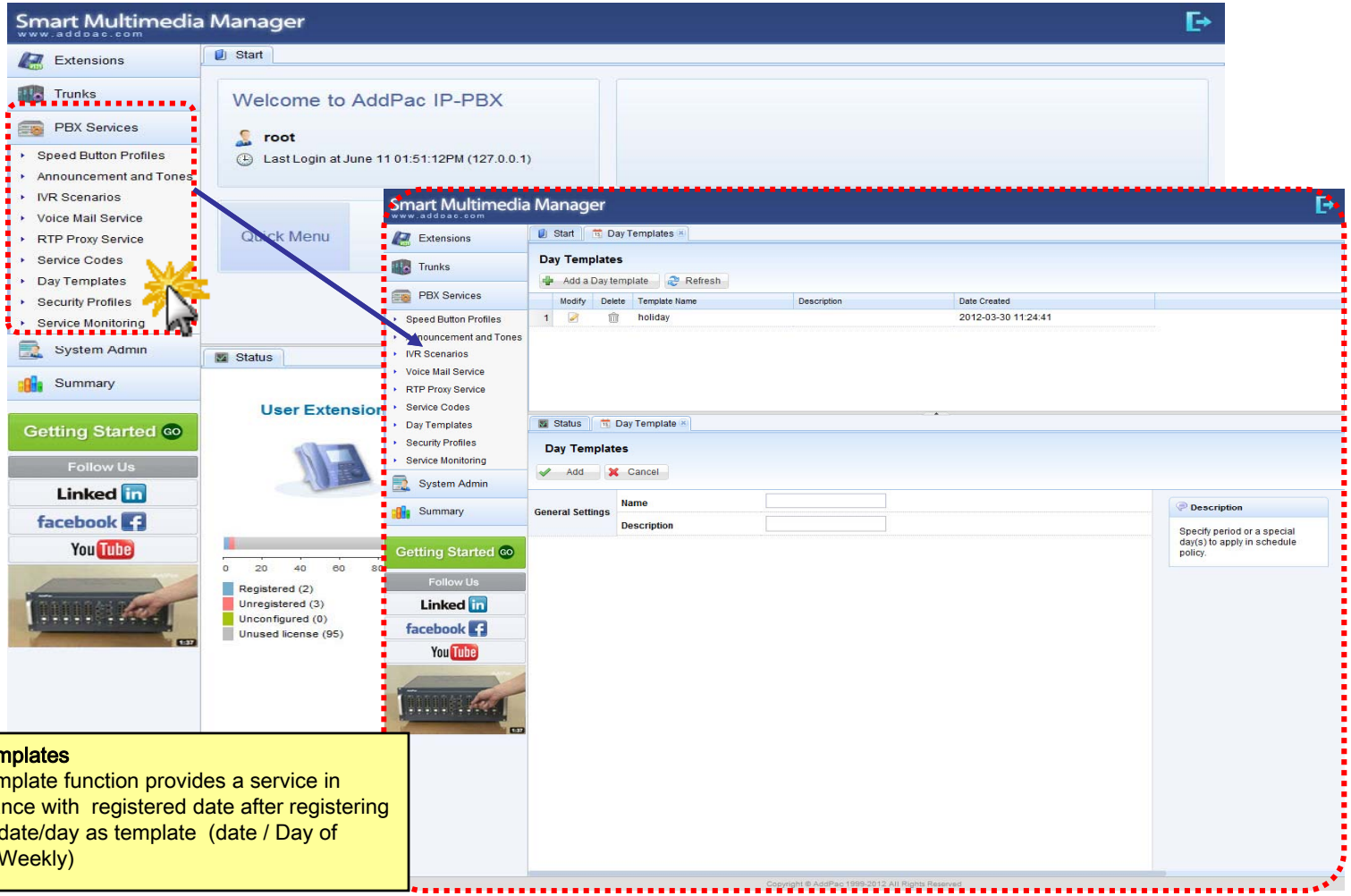

**RTP Proxy Service**
RTP Proxy supports smooth call conversation by acting as rtp packet relay for each different network (private/ public) Call. RTP Proxy Service provides various options such as (Port range / DSCP)

# PBX Service - Service Codes



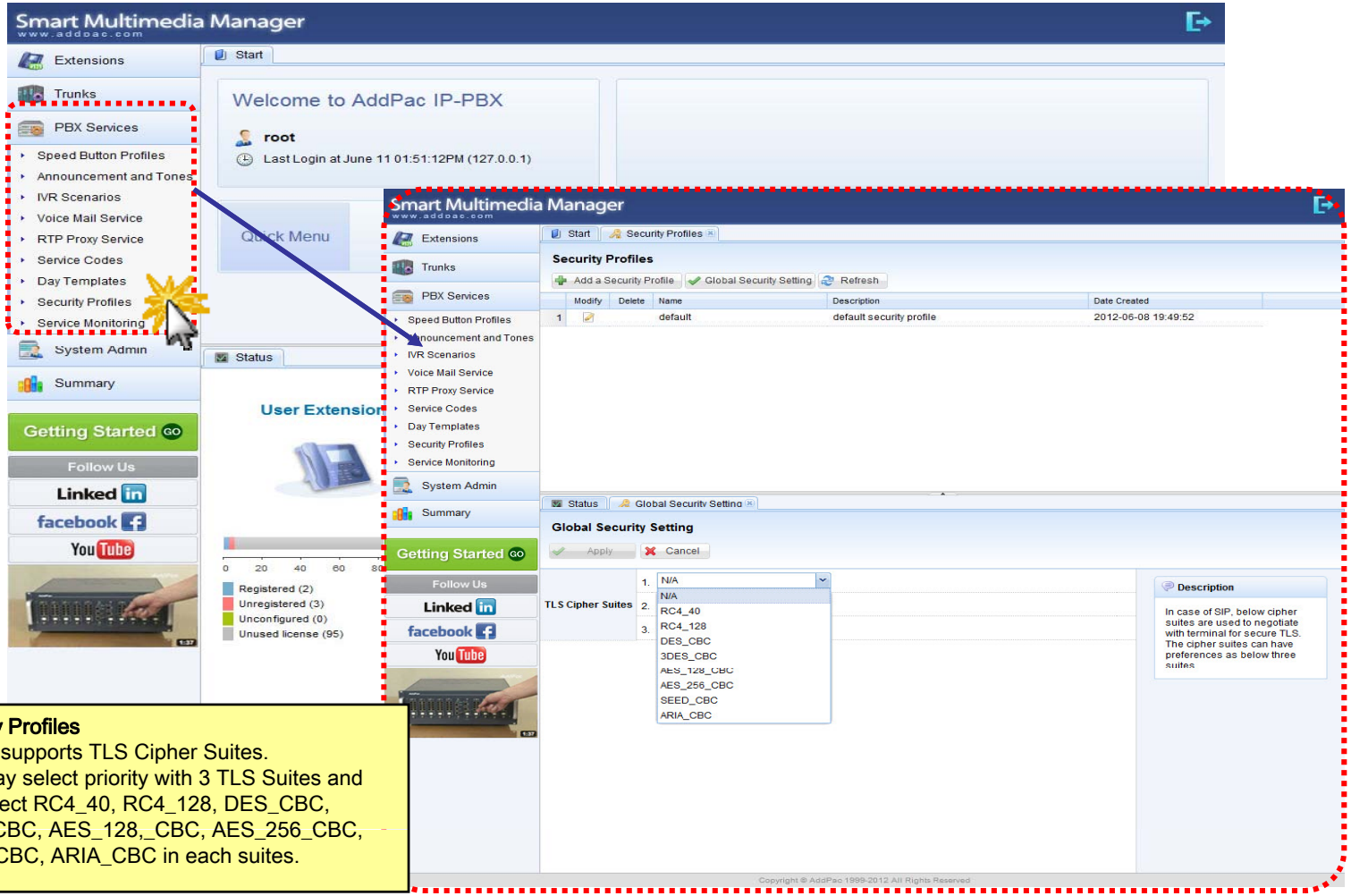

**Service Codes**
A function to setup additional service phone number in IP-PBX. It is a service code to use additional service in SIP terminal and start with # or * and may assign maximum of two phone numbers.

# PBX Service - Day Templates



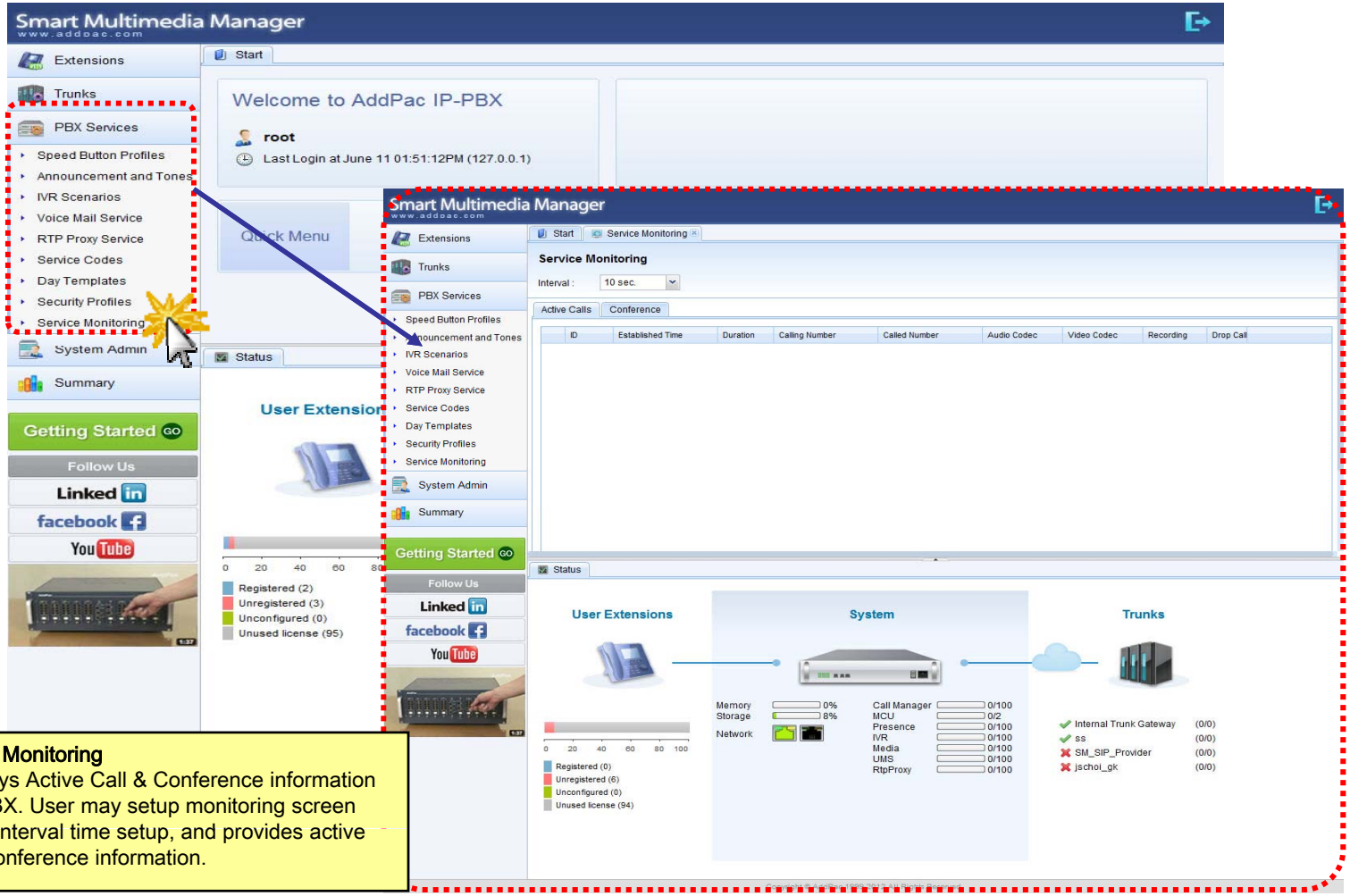

**Day Templates**
Day Template function provides a service in accordance with registered date after registering special date/day as template (date / Day of Week / Weekly)

# PBX Service - Security Profiles



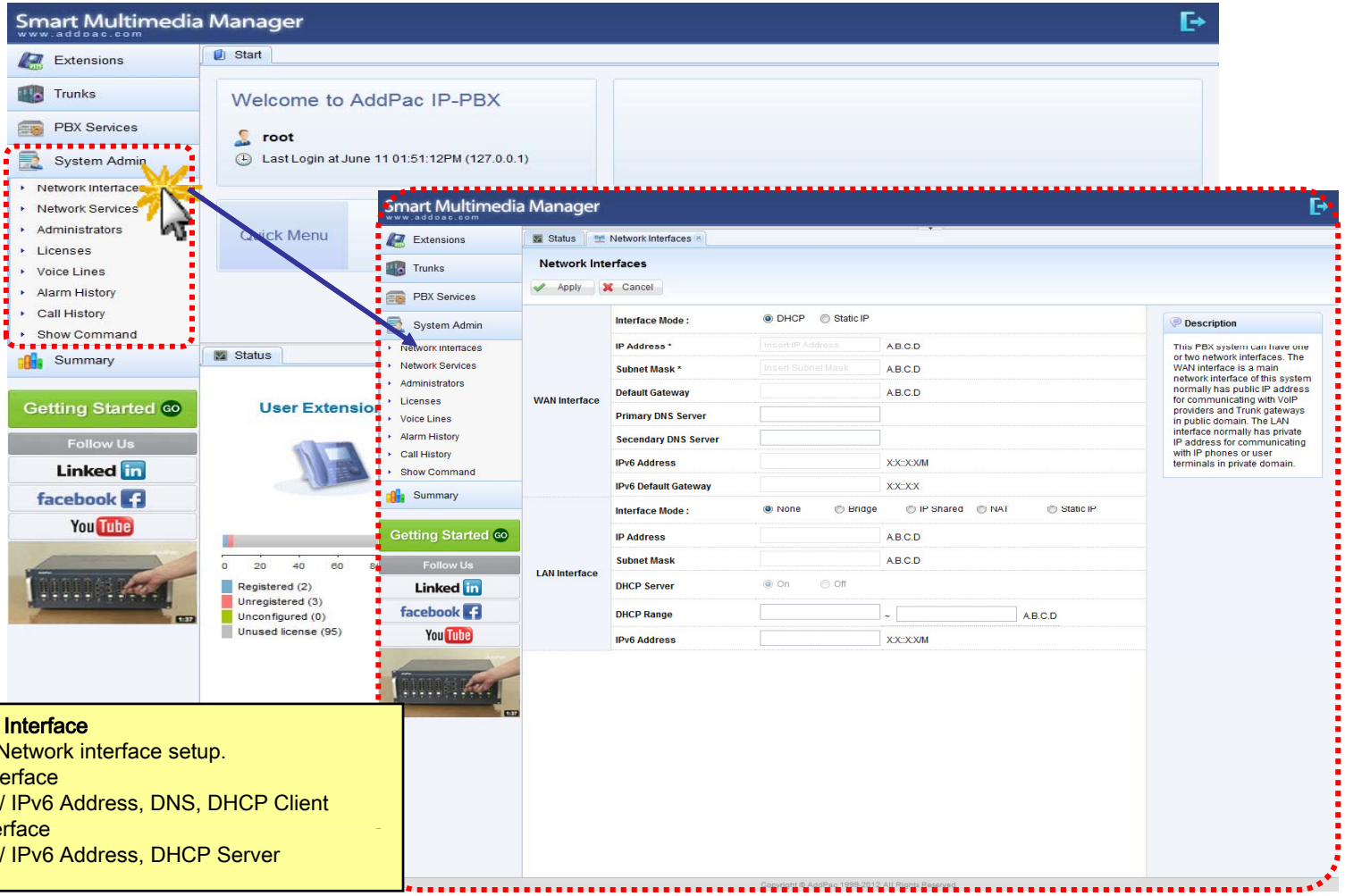

**Security Profiles**
IP-PBX supports TLS Cipher Suites.
User may select priority with 3 TLS Suites and may select RC4_40, RC4_128, DES_CBC, 3DES_CBC, AES_128,_CBC, AES_256_CBC, SEED_CBC, ARIA_CBC in each suites.

# 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.

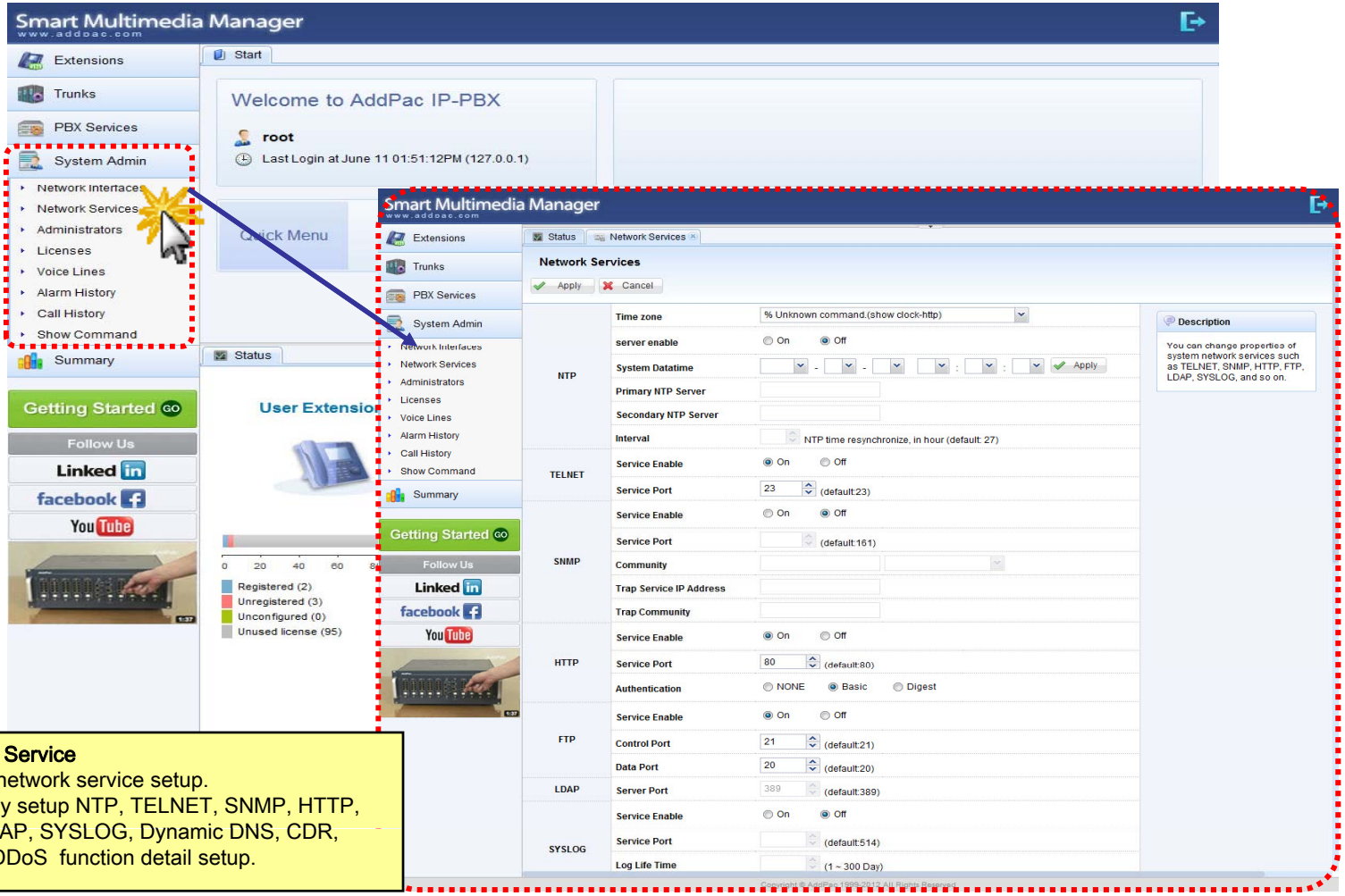
# System Admin - Network Interface




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

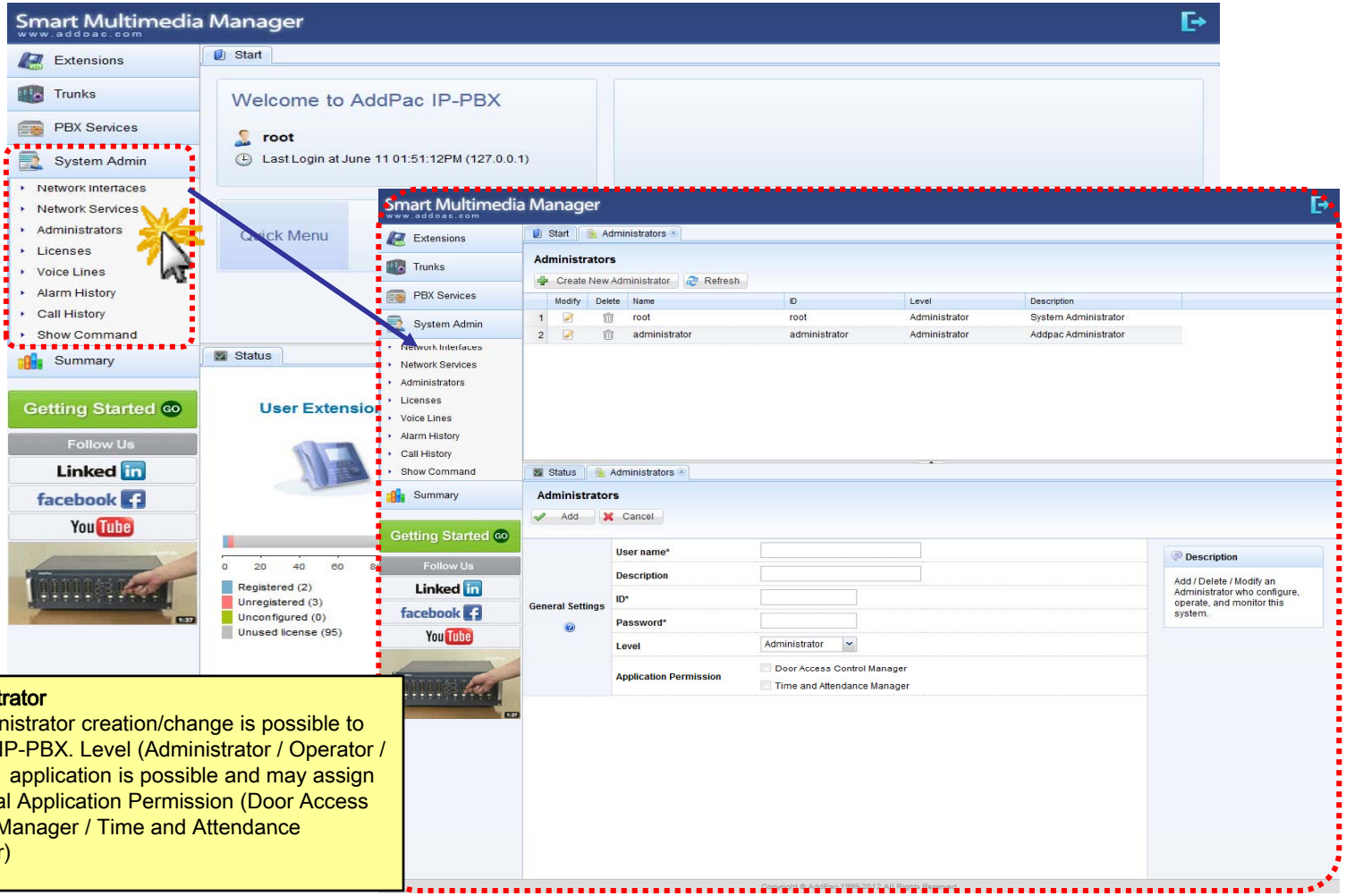
LAN Interface
- IPv4 / IPv6 Address, DHCP Server

# System Admin - Network Services



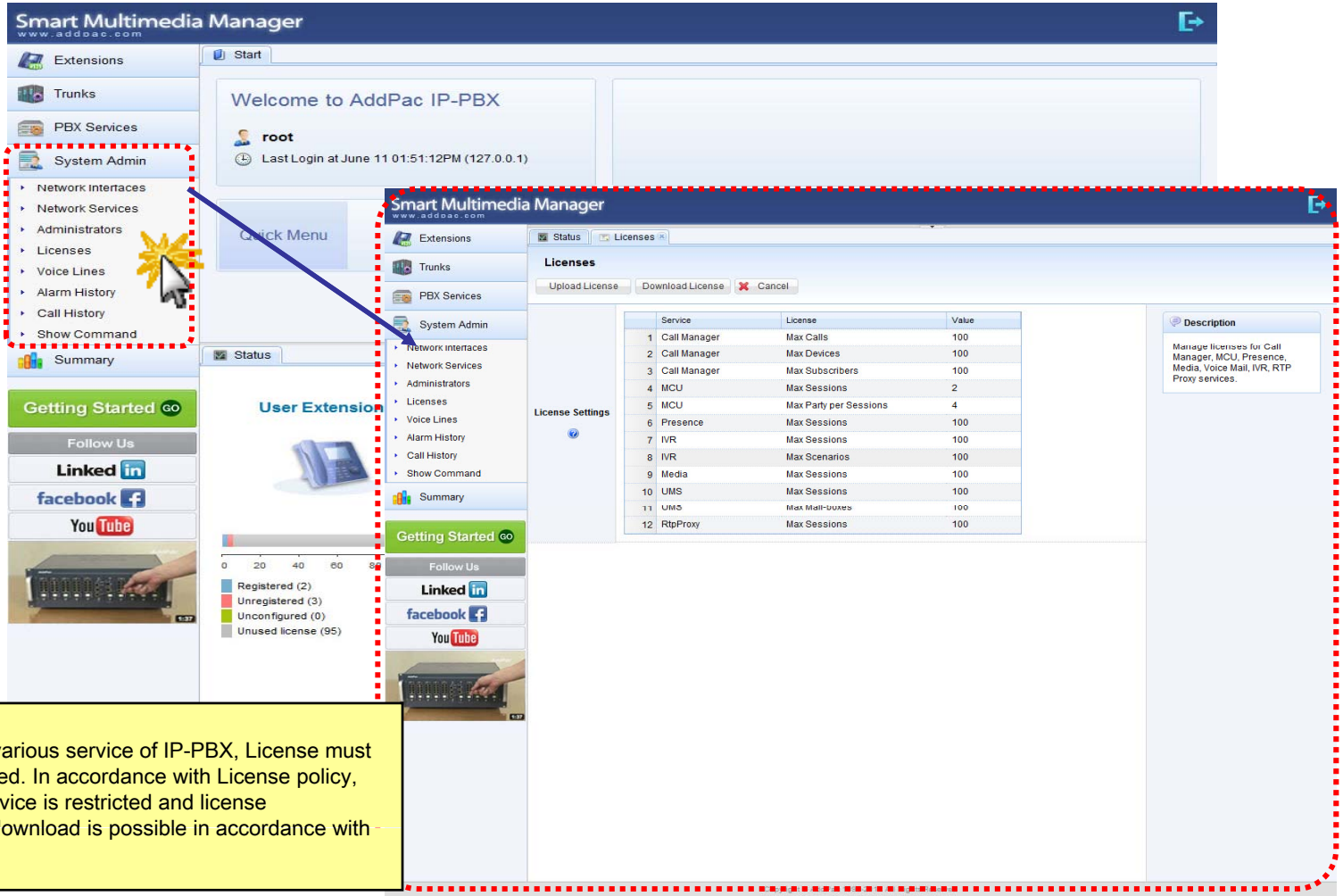

**Network Service**
IP-PBX network service setup.
User may setup NTP, TELNET, SNMP, HTTP, FTP, LDAP, SYSLOG, Dynamic DNS, CDR, SMTP, DDoS function detail setup.

# System Admin - Administrators



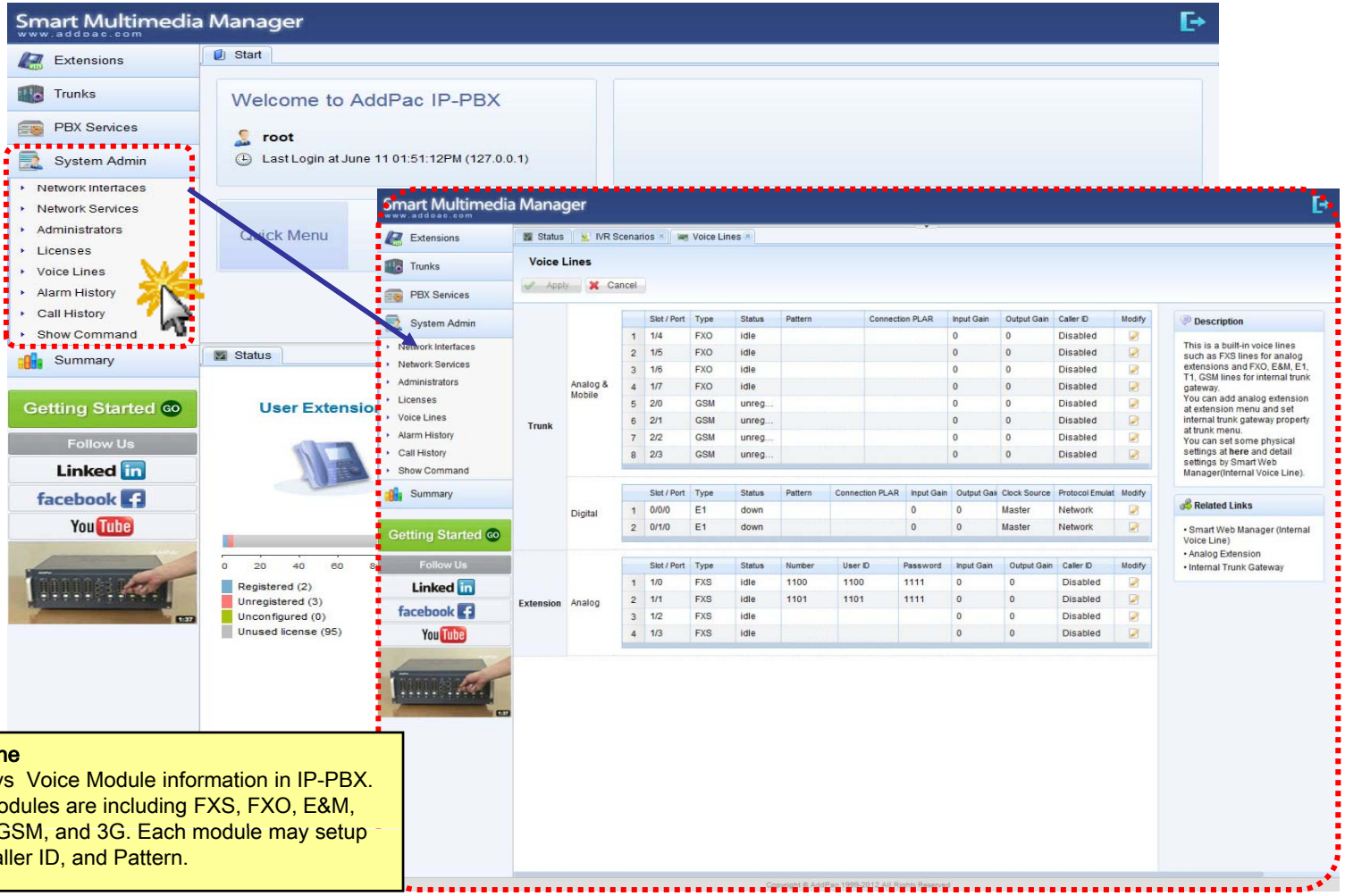

**Administrator**
An administrator creation/change is possible to operate IP-PBX. Level (Administrator / Operator / Monitor) application is possible and may assign additional Application Permission (Door Access Control Manager / Time and Attendance Manager)

# System Admin - Licenses



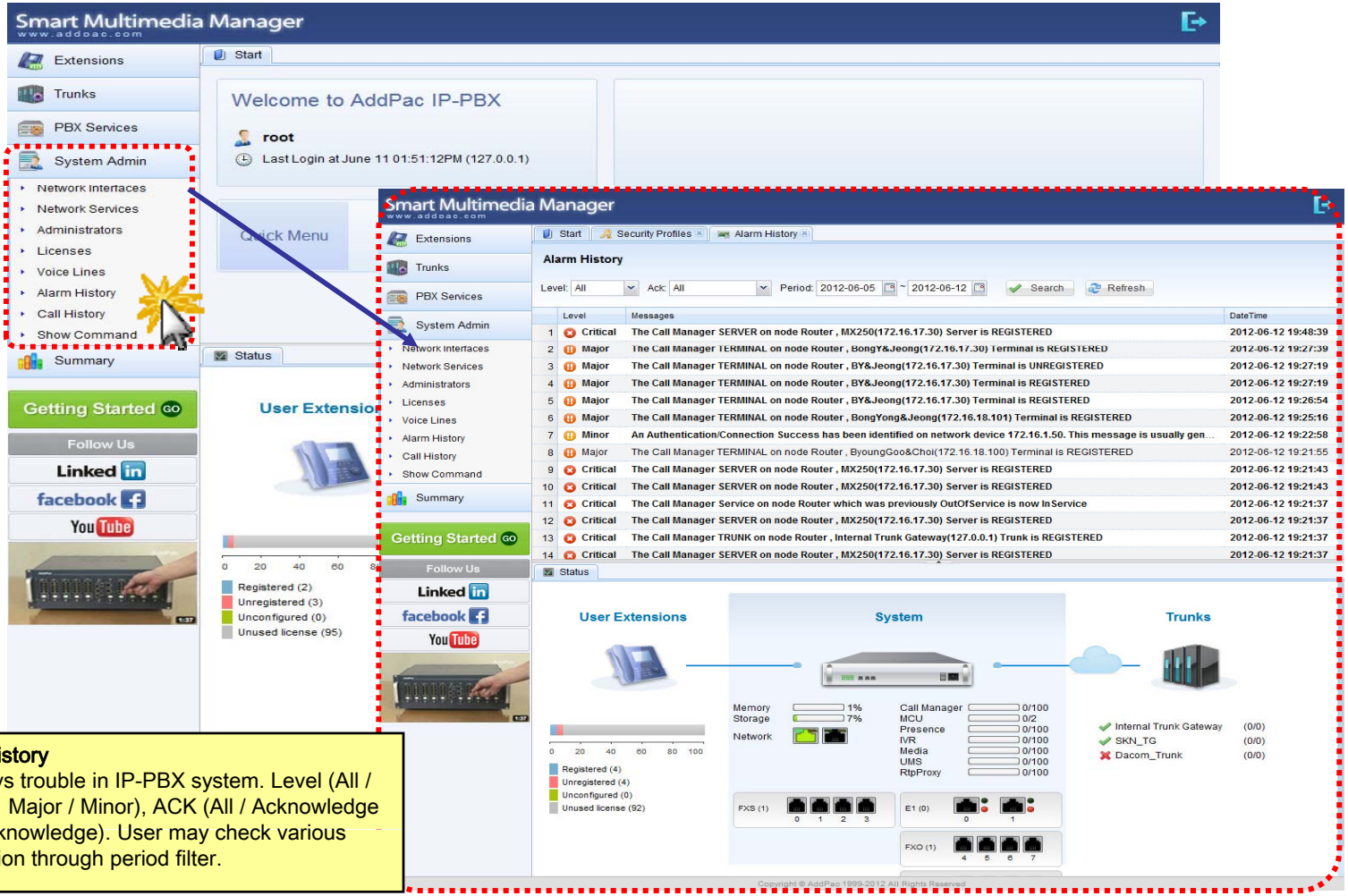

**License**
To use various service of IP-PBX, License must be created. In accordance with License policy, Max Service is restricted and license upload/download is possible in accordance with policy.

# System Admin - Voice Lines



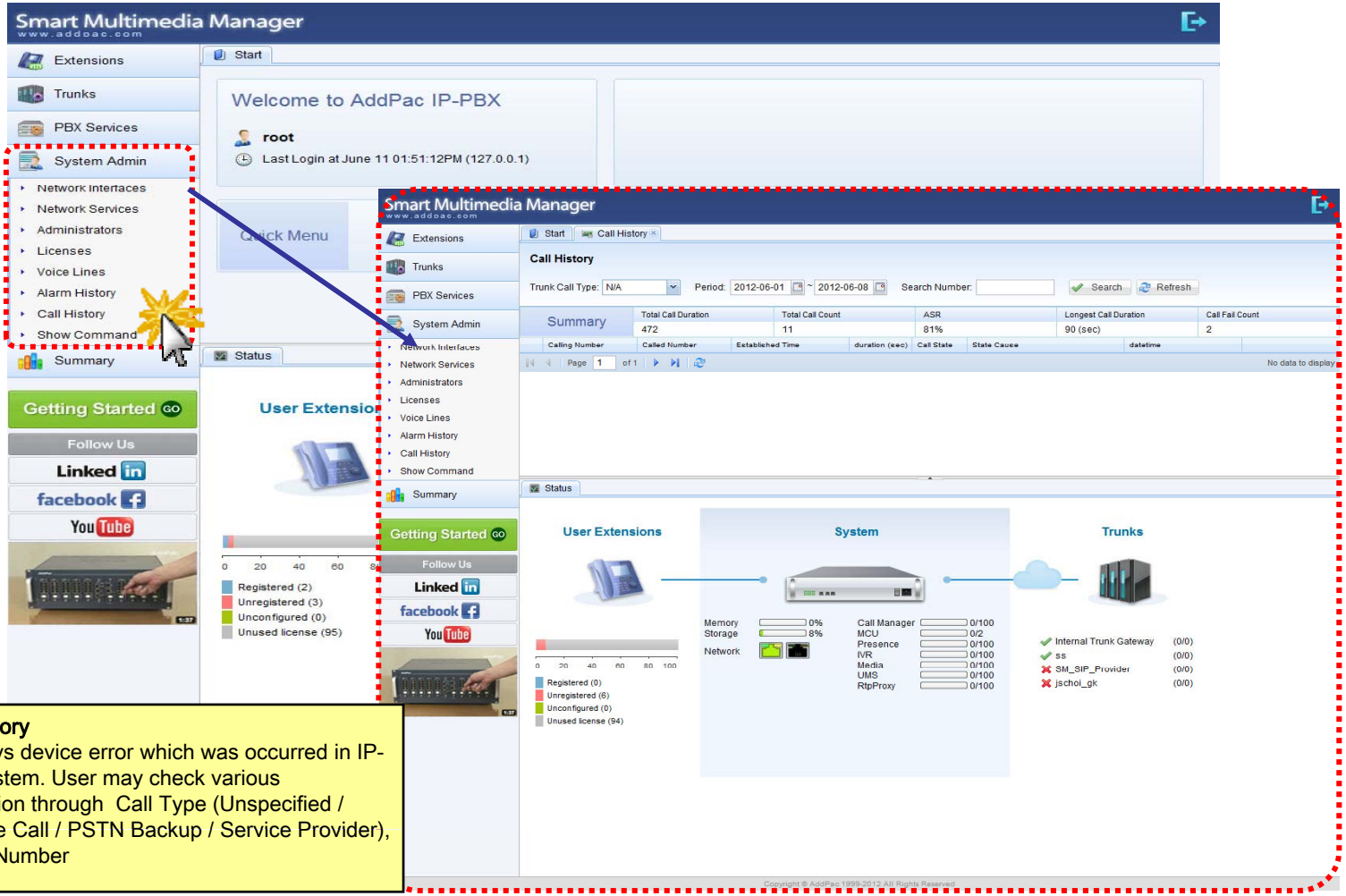

**Voice Line**
It displays Voice Module information in IP-PBX. Voice modules are including FXS, FXO, E&M, E1, T1, GSM, and 3G. Each module may setup Gain, Caller ID, and Pattern.

# System Admin - Alarm History



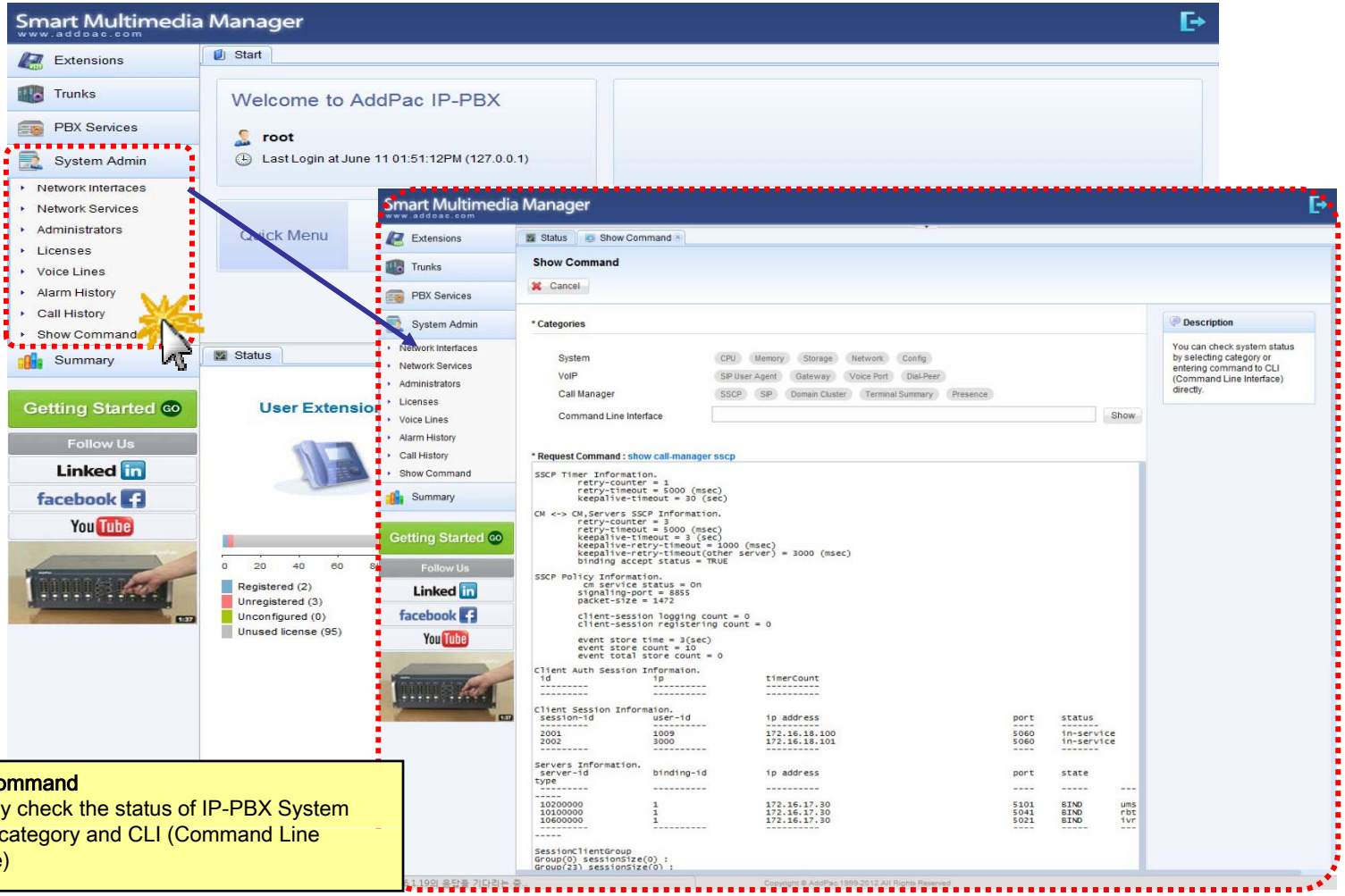

**Alarm History**
It displays trouble in IP-PBX system. Level (All / Critical / Major / Minor), ACK (All / Acknowledge / Not Acknowledge). User may check various information through period filter.

# 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!

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