IP Telephony Solution
(GSM + SMS)

AddPac Technology

Sales and Marketing

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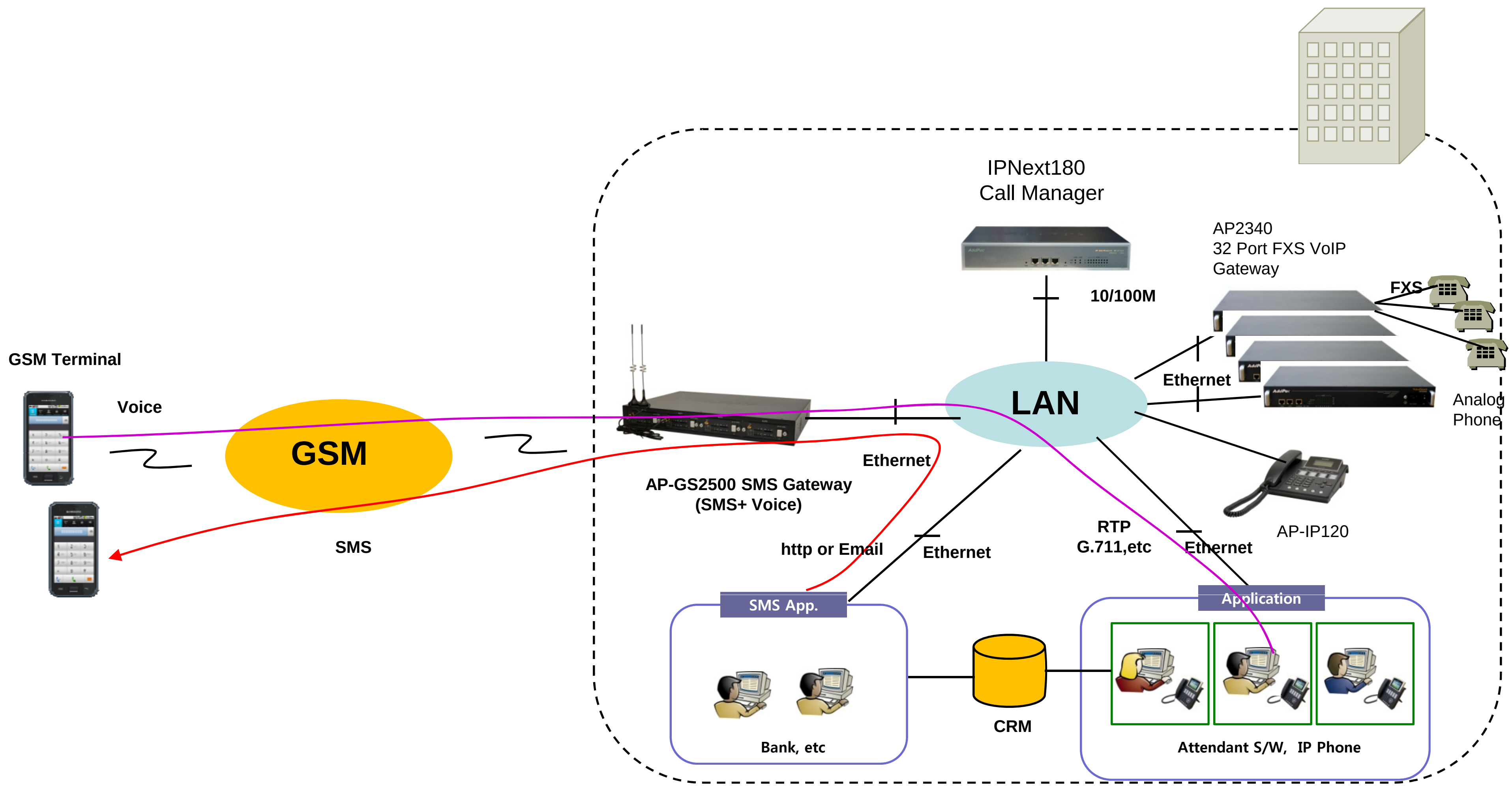
    (32Port x 3 + 8Port x1 = 104 Port)
  - IP Phones (AP-IP120, AP-IP90)
  - AP-GS2500 16-Port GSM SMS Gateway (4Port Module x 2)
  - Smart Billing Software for Enterprise
  - WSMM : Web based Smart Multimedia Manager for IP-PBX

IP Telephony Network Diagram



IP Telephony Solution Components

IPNext180
IP-PBX

Product Overview

IPNext180 Next Generation Hybrid IP-PBX System

- IP based Advanced Hybrid IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Small & Medium Office
- SIP Call Manager Service
- 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

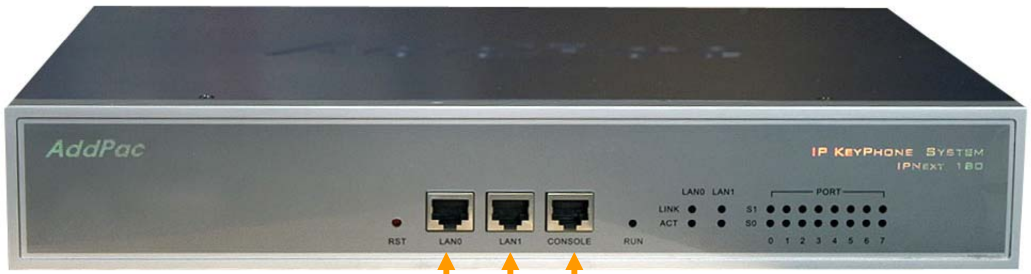
IPNext180 Next Generation Hybrid IP-PBX System

- 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 Next Generation Hybrid IP-PBX System

Front Side



LAN1 (10/100Mbps)

LAN0(10/100Mbps)

Console Port

Hardware Specification

IPNext180 Next Generation Hybrid IP-PBX System

Back Side

PSTN Interface Module

Power Inlet

Power Switch

Hot-Swap Switch & LAMP Indication

Hardware Specification

IPNext180 Next Generation Hybrid IP-PBX System

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)

AP2340 32 Port FXS VoIP Gateway

Product Overview

AP2340 32 Port Analog VoIP Gateway

- H.323/SIP/MGCP Triple Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Performance LAN-to-LAN Routing Capability
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Compact Design with Internal Power Supply
- Four(4) VoIP Module Slot : 8-Port VoIP Module

Hardware Specification

AP2340 32 Port Analog VoIP Gateway

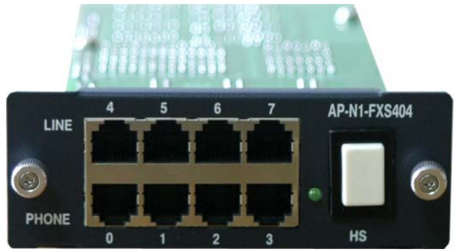
RISC CPU

High-end DSP

- RISC Microprocessor Computing Power
- Up to 32 Port Analog VoIP Gateway
- Four(4) VoIP Module Slots (Hot-Swap)

  - 8-Port FXS Card, 8-Port FXO Card, 4-Port FXS 4-Port FXO Card
- Network Interface

  -Two(2) 10/100Mbps Fast Ethernet (RJ45)
- RS232C Console Interface
- Run LED, LAN LED, Port LEDs
- Internal Power Supply



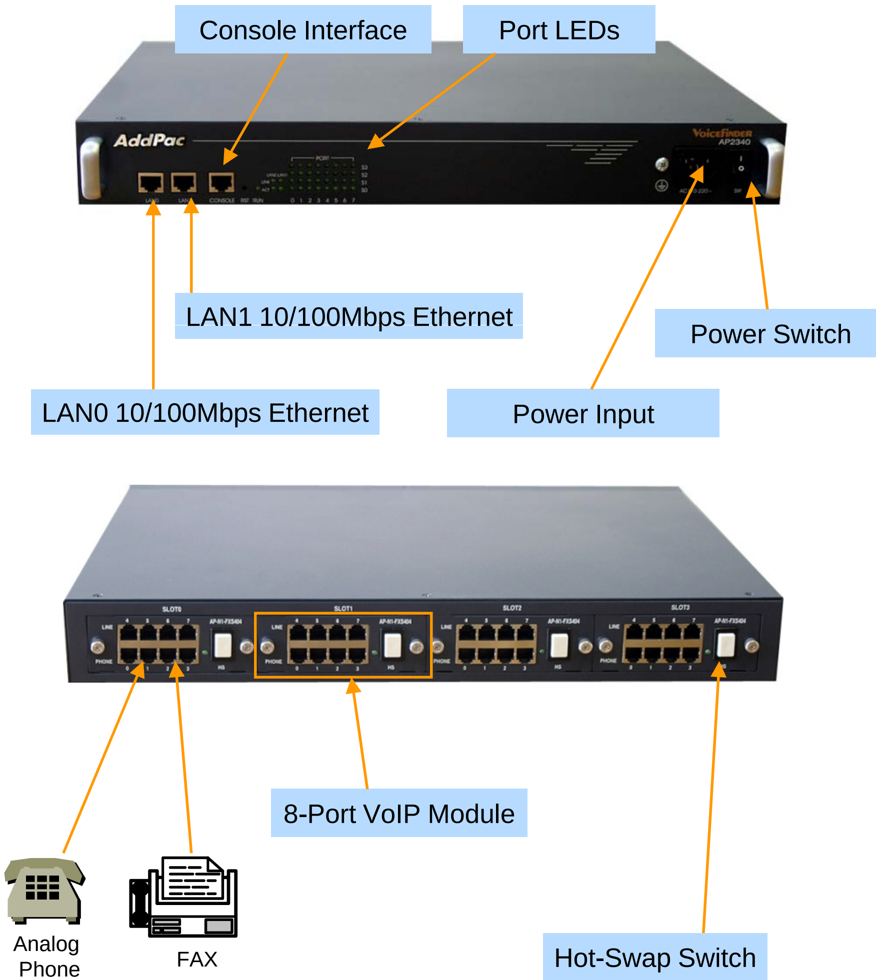
Hardware Specification

AP2340 32Port Analog VoIP Gateway

Hardware Specifications

AP2340 VoIP Gateway	Basic Specifications
Voice Interface	Four(4) VoIP Module Slots
	AP-N1-FXS8, AP-N1-FXO8, AP-N1-FXS4O4
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	32 Mbyte High-speed SDRAM
Power Requirement	Power Supply / VAC 110~220V, 50/60Hz, 5V 15A, 12V 8A
Operating Temperature	0ºC ~ 45ºC (32 ºF ~ 122ºF)
Storage Temperature	-40ºC ~ 85ºC (-40ºC ~ 185ºF)
Relative Humidity	5% ~ 95% (Non-condensing)
Dimensions	W×D×H (440×318×56mm)

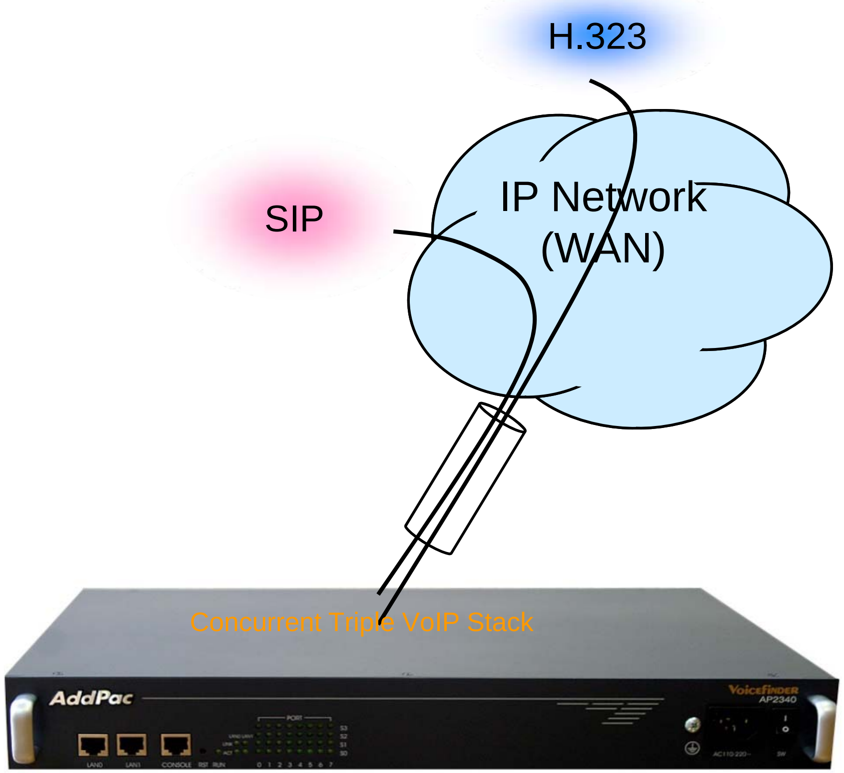
Network interface Configurations

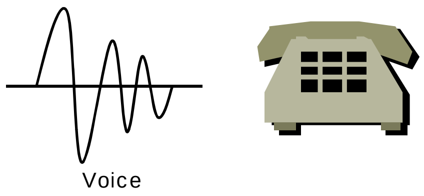


VoIP (Voice over IP) Service

AP2340 32 Port Analog VoIP Gateway

- H.323, SIP Dual VoIP Signaling Stack
- H.323
  - ITU-T Standard H.323 v3 Support
  - Support H.245 Tunneling
  - Including H.235 Security Features
- SIP
  - IETF RFC3261 or RFC2543 SIP Standard





VoIP (Voice over IP) Service

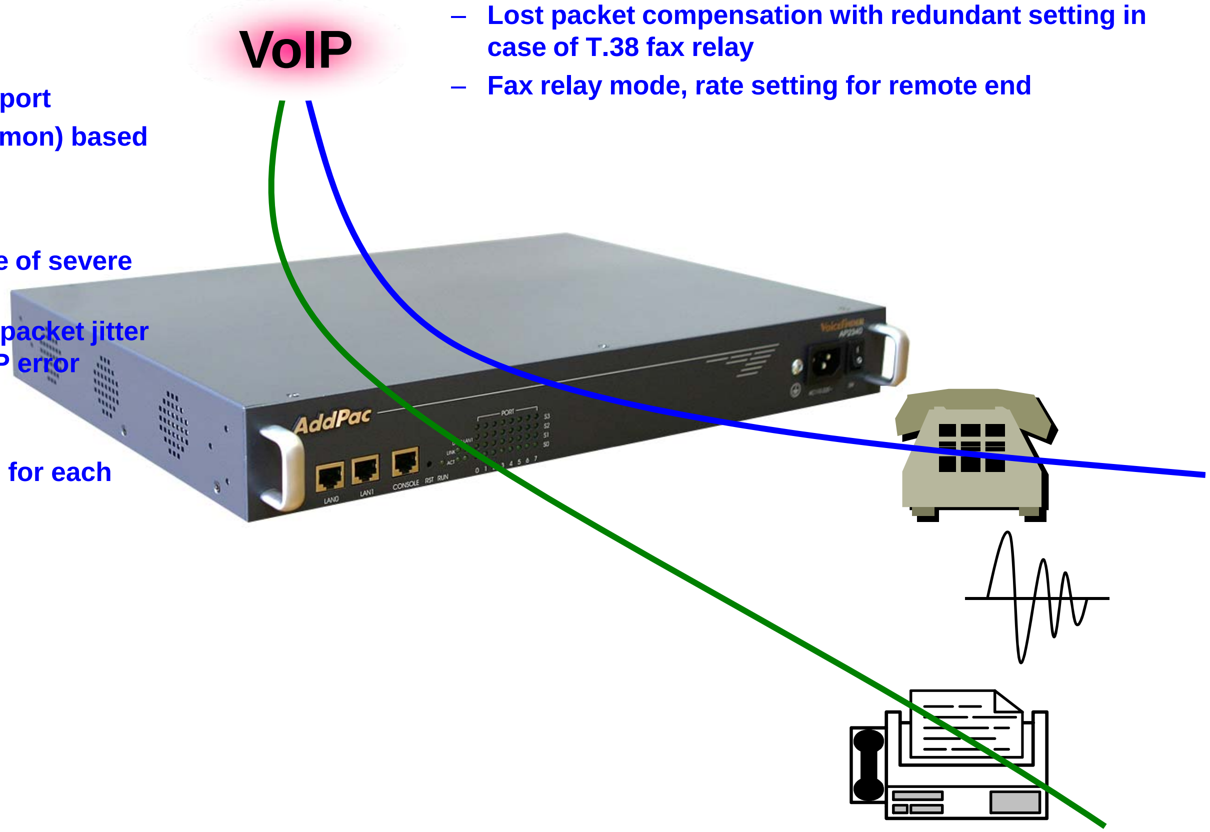
AP2340 32 Port Analog VoIP Gateway

- H.323
  - Fast connect, normal connect support
  - H.245 tunneling support
  - Q.931 response message setting for inbound VoIP calls
  - H.245 logical channel open timing selection function
  - Start H.245 procedure support
  - DTMF / Hook flash relay with H.245 alphanumeric / signal
  - Secondary gatekeeper support
  - Gatekeeper assignment according to the domain name
  - Gatekeeper discovery with multicast
  - Lightweight RRQ support
  - Signaling TCP port assignment
  - Resource threshold setting with RAI
  - H.235 clear-token, crypto-token support
  - canMapAlias support
  - Technical prefix (supported prefix) support
  - Public IP assignment in NAT environment
- SIP
  - Gateway-based / Endpoint-based registration support
  - Secondary proxy-server assignment function
  - SIP signaling port change function
  - SIP proxy server assignment according to the domain name
  - T.38 real-time fax relay support
  - DTMF relay support with RFC2833 / OPTION message
  - Re-INVITE support

VoIP (Voice over IP) Service

AP2340 32 Port Analog VoIP Gateway

- **Voice Codec**
  - G.711 A-Law, G.711 U-Law
  - G.726 r16, G.726 r32
  - G.729A
  - G.723.1 r63, G.723.1 r53
  - VAD (Voice Activity Detection) function support
  - DTMF relay support (H.323, SIP, MGCP common) based on RFC2833
- **RTP**
  - Redundant RTP packet transmission in case of severe packet loss
  - Dynamic jitter buffer management and RPT packet jitter and loss compensation with heuristic & DSP error concealment
  - Static jitter buffer setting support
  - Voice frame per RTP packet number control for each codec
  - In-band ring-back tone support
  - Virtual ring-back tone support
  - Tone parameter change support
- **FAX**
  - Fax relay mode supporting T.38, inband-T.38, bypass mode
  - Lost packet compensation with redundant setting in case of T.38 fax relay
  - Fax relay mode, rate setting for remote end

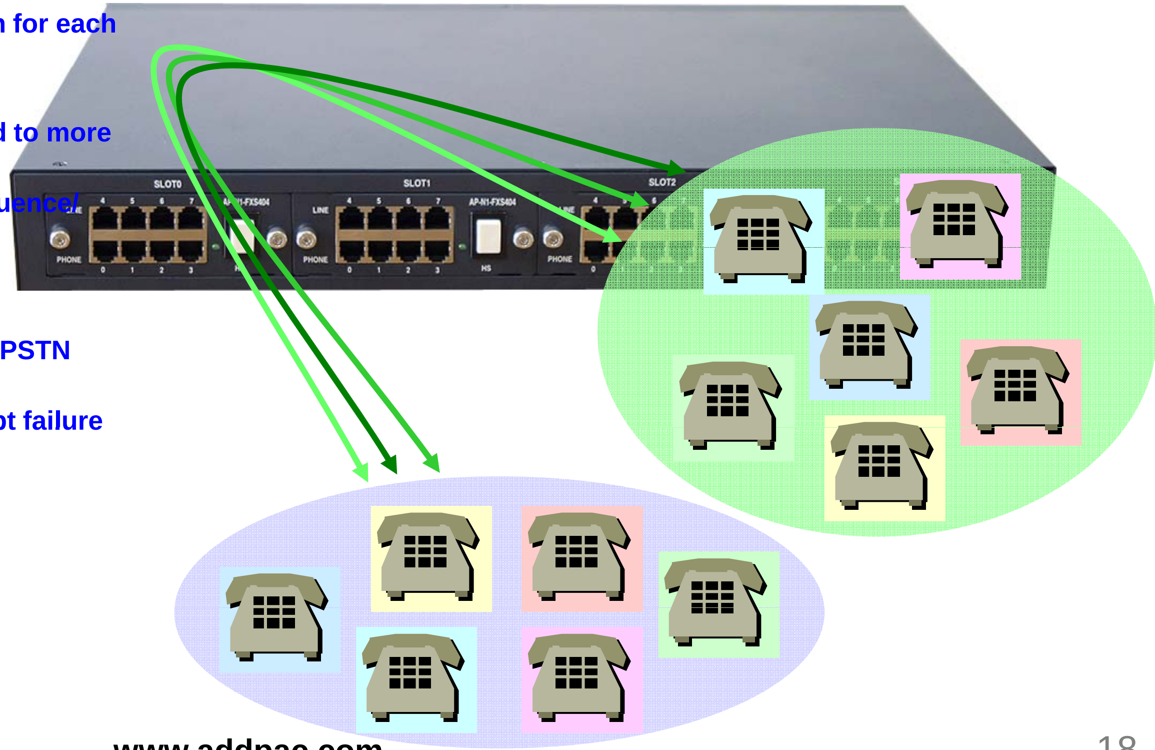


VoIP (Voice over IP) Service

AP2340 32 Port Analog VoIP Gateway

- **VoIP Call Controls**
  - **Hot line connection function with PLAR (Private Line Auto Ring Down)**
  - **Leased line emulation function**
  - **Connection monitoring function**
  - **Fault tolerant with Redundancy and Call Distribution among Gateways for load balancing**
  - **Call attempt with IP address**
  - **H.323, SIP, MGCP inbound call connection for each voice port**
  - **Multiple E.164 setting for one voice port**
  - **One E.164 or digit pattern can be assigned to more than one voice port**
  - **Hunting with Longest match/ priority/ sequence/ random**
  - **One stage call setup by Digit forwarding**
  - **Call barring with specific digit patterns**
  - **Calling and called number conversion for PSTN outbound calls**
  - **PSTN rerouting in case of VoIP call attempt failure**

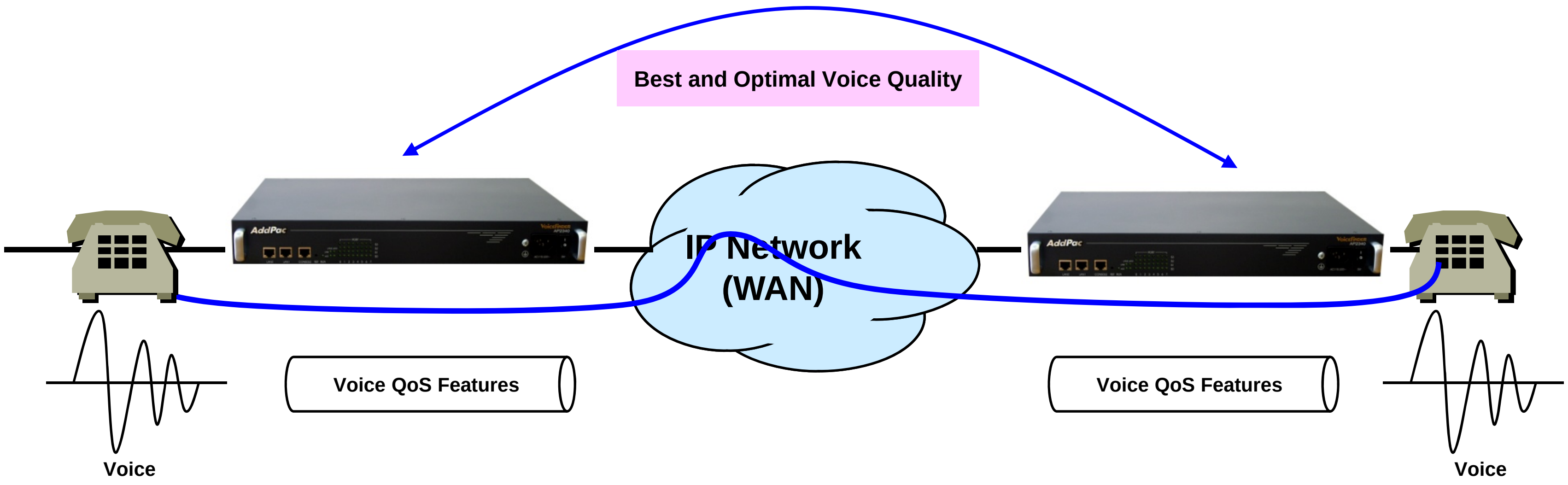
- **VoIP Call Controls (cont.)**
  - **Call transfer for internal calls**
  - **Call pickup for internal calls**
  - **Calling and called number conversion for VoIP outbound calls**
  - **Calling and called number conversion for VoIP inbound calls**
  - **Fax broadcasting call control**



Advanced QoS Features

AP2340 32 Port Analog VoIP Gateway

- **Enhances Transmit Voice QoS Features**
  - Voice Traffic Priority Queuing
  - QoS Service Profiling
  - Providing Virtual Network Transmit Algorithm
  - Real-time Voice Traffic QoS Support
  - RTP Packet Transmit Interval Control
  - Supporting RTP Packet Redundancy Scheme
  - IP Header Control such as ToS, Diffserv
- **Enhances Receive Voice QoS Features**
  - Dynamic Jitter Buffer Management
  - Error Concealment
  - Support T.38 FAX Data Error Recovery Scheme



Network Protocols

AP2340 32Port Analog VoIP Gateway

Basic Network Protocols

- ARP, IPv4, TCP, UDP, ICMP, SCTP, IGMP, MLD

Routing Protocol

- IPv4 : Static

Service Protocol

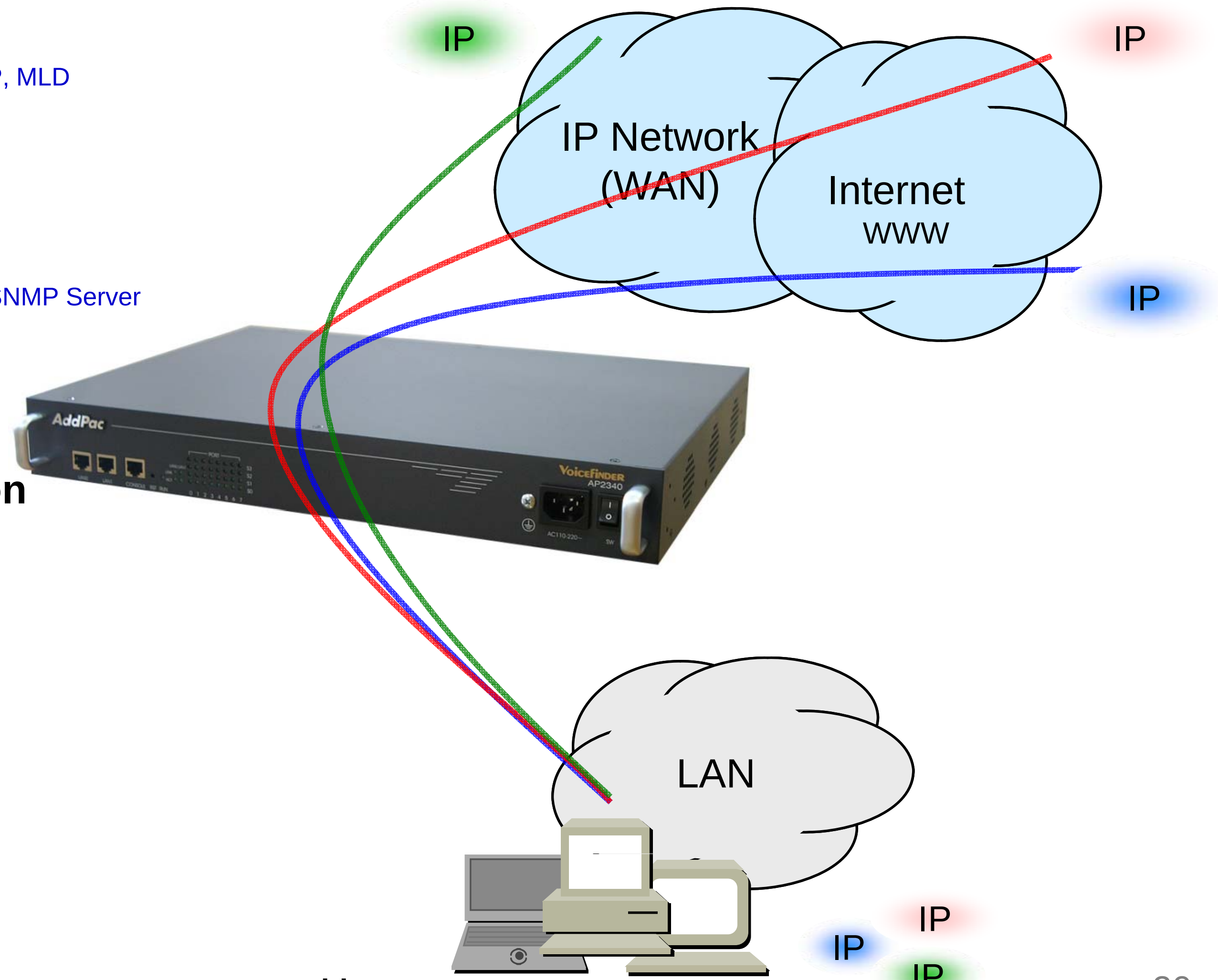
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

IPv4 Address Configuration

- Fixed (Static)
- DHCP
- PPPoE

Miscellaneous

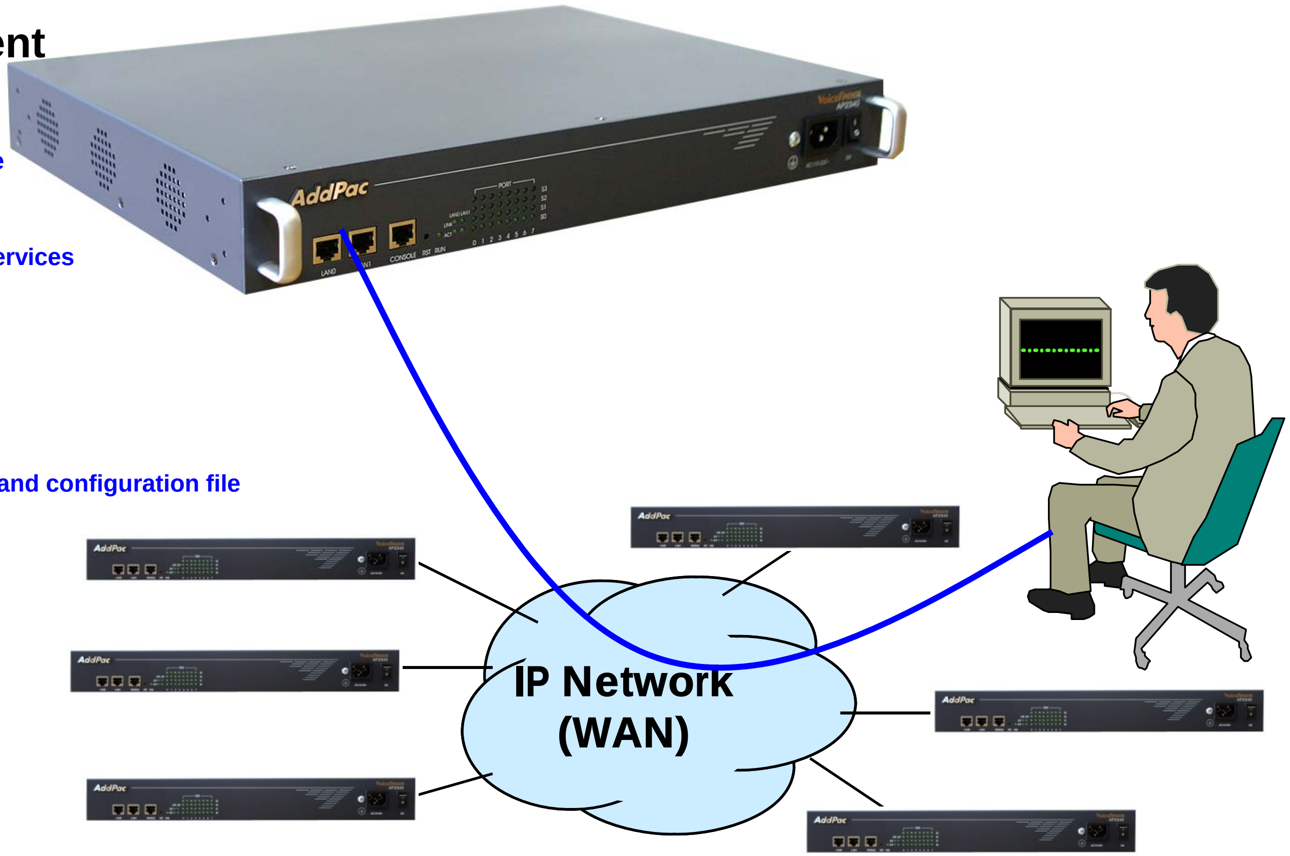
- Cisco Style CLI
- Standard & Extended IPv4 Access List
- Multi-level User Account Management
- IP accounting
- STUN Client



Network Management

AP2340 32Port Analog VoIP Gateway

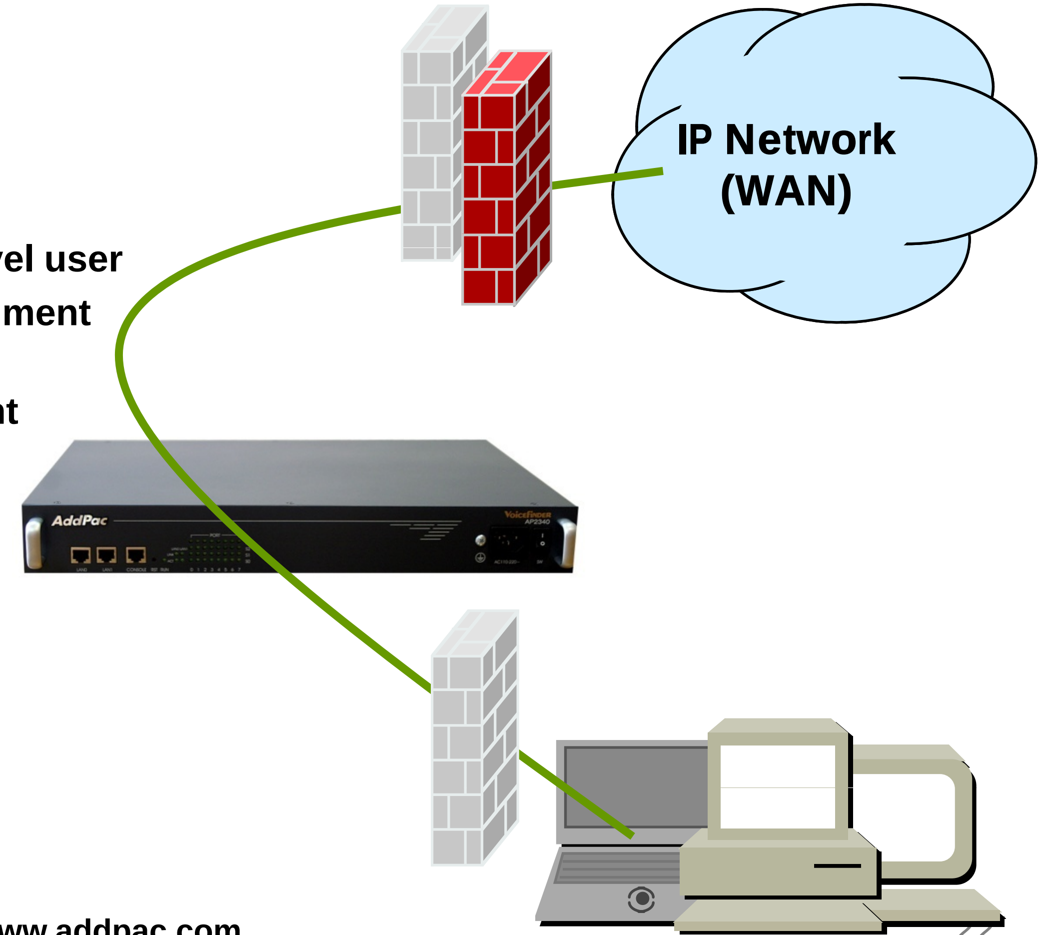
- **SNMP**
  - Standard Simple Network Management Protocol(SNMP) Agent support
  - MIB v1 and v2 Support
- **Web-based Management**
  - Smart Easy Setup
  - Standard Voice Interface
  - Standard PSTN Back-up Interface
- **Watch-dog Function**
  - Hardware, Software watch-dog services
- **Remote Management**
  - Telnet
  - Rlogin
- **Auto Upgrade Service**
  - HTTP server based APOS image and configuration file auto-upgrade support
- **Batch Job Function**
  - Text based script downloading
- **Interoperable with AP-VPMS Service**
  - AddPac VoIP Plug & Play Management System (AP-VPMS)



Security Management

AP2340 32 Port Analog VoIP Gateway

- IP packet filtering
- IP access list
- User authentication function
  - Password Authentication Protocol (PAP)
  - Challenge Handshake Authentication Protocol (CHAP)
- Enable/Disable specific protocols
- Auto-square connect of Telnet session
- Account Management function for multi-level user
- SNMP/TELNET/FTP/HTTP/TFTP port assignment function
- SNMP/TELNET/FTP access list management
- Boot mode security checking function



IP Phone

IP Phone Comparison Table

	AP-IP300 	AP-IP230 	AP-IP160 	AP-IP120 	AP-IP90 
LCD Size	4.3 Inch Color LCD	5 Inch Color LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD	4 Text Line Graphic LCD
Touch Screen	N/A	Support	N/A	N/A	N/A
Speed-Dial Keys	25 Key with Presence LED	Touch Screen based 25 Keys	16 Key with Presence LED	12 Key with Presence LED	N/A
Voice Codec	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723	G.711/G.726/ G.729/G.723
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
3-Party Conversation	Support	Support	Support	Support	Support
LAN Port	2	2	2	2	2
PoE(Option)	Support	Support	Support	Support	Support
FXO(Option)	Support	Support	Support	Support	Support

AP-IP120
IP Phone

Product Overview

AP-IP120 IP Phone

- IP Phone Solution
- 12 Speed-Dial Key with Presence Indication Lamp
- Audio Broadcasting Solution
- SIP and H.323 VoIP Signaling Support
- G.711,G.729, G.723.1 Various Voice Codec Support
- Speaker Phone Service (Acoustic Echo Canceller)
- PoE, FXO Support (Option)
- High-performance Audio, and Voice Service
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection

Hardware Specification

AP-IP120 IP 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

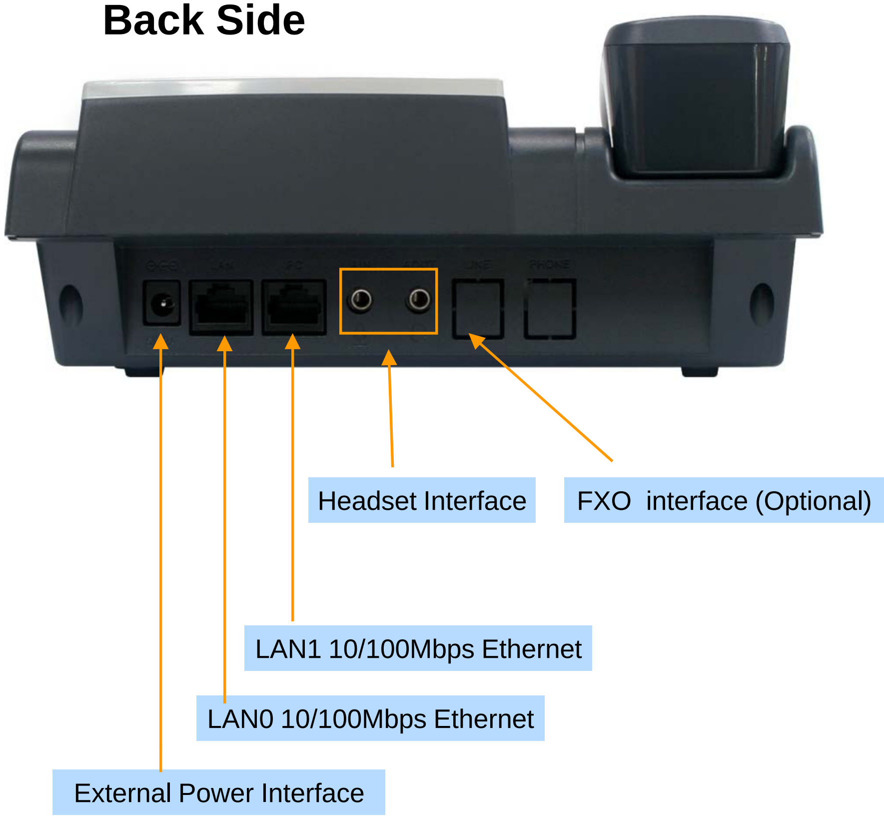
AP-IP120 IP Phone

Hardware Specifications

AP-IP160 IP 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-IP90
IP Phone

Product Overview

AP-IP90 IP Phone

- IP Phone Solution
- SIP/H.323 Dual VoIP Signaling Support
- G.711,G.729, G.723.1 Various Voice Codec Support
- Speaker Phone Service (Acoustic Echo Canceller)
- FXO Analog Interface Support :Option
- Headset Interface Support
- Two(2) LAN Interface Support
- Various VoIP Voice Codec Support
- High-performance Audio, and Voice Service
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection
- PoE (Power over Ethernet) Function Support : Option

Hardware Specification

AP-IP90 IP Phone

- RISC+DSP Microprocessor Computing Power
  (Dual Processor Architecture)
- Optional PSTN Backup (FXO) Interface : AP-IP90E
- High quality Audio and Voice Interface
  - Stereo Audio Input Connector
  - Stereo Audio Output Connector
- Network Interface
  - Two(2) 10/100Mbps Fast Ethernet
- Graphic LCD Window
- Power Supply
  - External Power Adaptor (5V, 2A)
  - PoE (Power over Ethernet) Support : Option

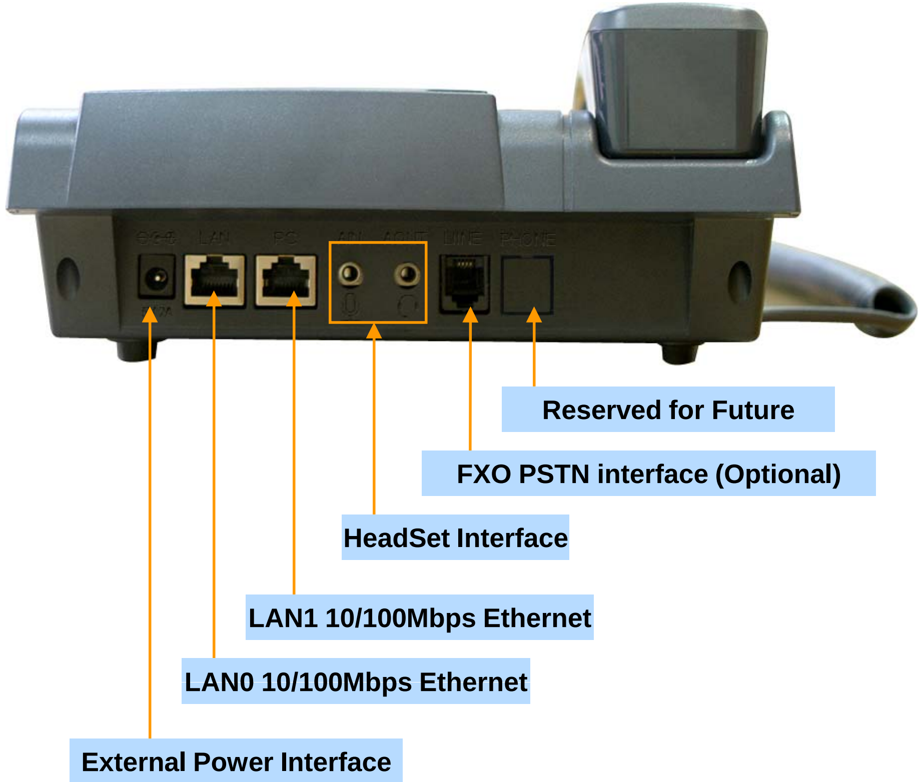
Hardware Specification

AP-IP90 IP Phone

Hardware Specifications

AP-IP90 IP Phone	Basic Specifications
CPU	RISC Microprocessor
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
PSTN Port (Optional)	1-Port FXO PSTN 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
Operating Temperature	0ºC ~ 45ºC (32 ºF ~ 122ºF)
Storage Temperature	-40ºC ~ 85ºC (-40ºC ~ 185ºF)
Relative Humidity	5% ~ 95% (Non-condensing)
Dimensions	W x D x H (200mm x 210mm x 60mm)
Weight (g)	1Kg

Network interface Configurations



AP-GS2500
GSM SMS Gateway

Main Features

AP-GS2500 GSM SMS Gateway

- Four(4) Module Slots for 4-Port GSM Module, 8FXS/8FXO Analog Interface (Up to 12-Port 3G +8FXS/8FXO Module, 16-Port 3G)
- H.323/SIP/MGCP Triple Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Two(2)10/100Mbps Fast Ethernet
- One(1) RS-232C Port for Command Line Interface
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) & NMS for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Powerful Web based Management
- Rack Mountable Chassis with Internal Power Supply

Hardware Specification

AP-GS2500 GSM SMS Gateway

- RISC Microprocessor Computing Power
- Four(4) Module Slot for 3G, GSM, CDMA, Analog/Digital Interface
- 4-Port 3G Module(AP-N1-3G4), Hot-Swap
  - One(1) 3G Antenna Interface (4 Channel Combiner), 4-Port USIM Card Slot
- 4-Port GSM Module(AP-N1-GSM4), Hot-Swap
  - One(1) GSM Antenna Interface (4 Channel Combiner), 4-Port SIM Card Slot
- VoIP Interface Module, Hot-Swap
  - 8-Port FXS Module, Digital E1/T1 Module (AP-N1-FXS8, AP-N1-FXO8, etc)
- Network Interface
  -Two(2) 10/100Mbps Fast Ethernet (RJ45)
- RS232C Console Interface for CLI
- Run LED, LAN LED, Port LEDs
- Internal Power Supply

Hardware Specification

AP-GS2500 GSM SMS Gateway

RISC
CPU

High-end
DSP

Front Side View

Power Switch

Console Interface

Port LEDs

Power Input

LAN0 10/100Mbps Ethernet

LAN1 10/100Mbps Ethernet

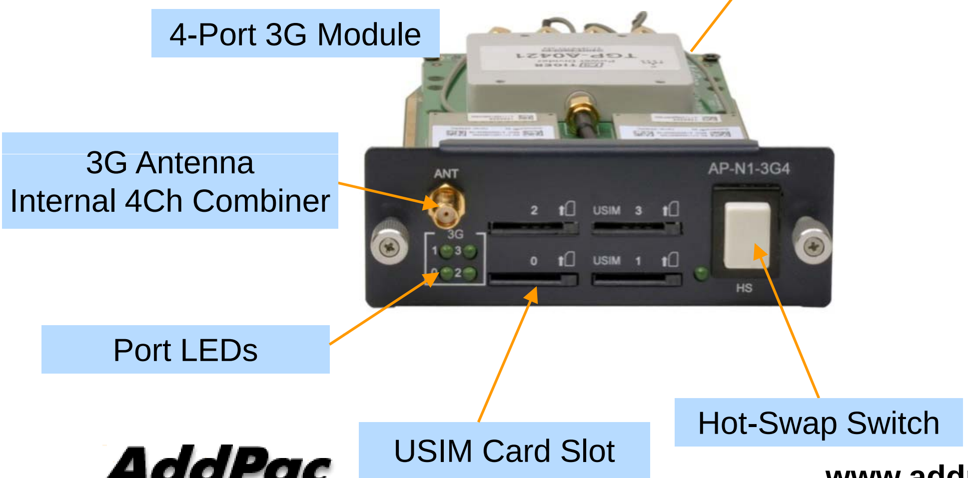
Hardware Specification

AP-GS2500 GSM SMS Gateway

RISC CPU

High-end DSP

Back Side View



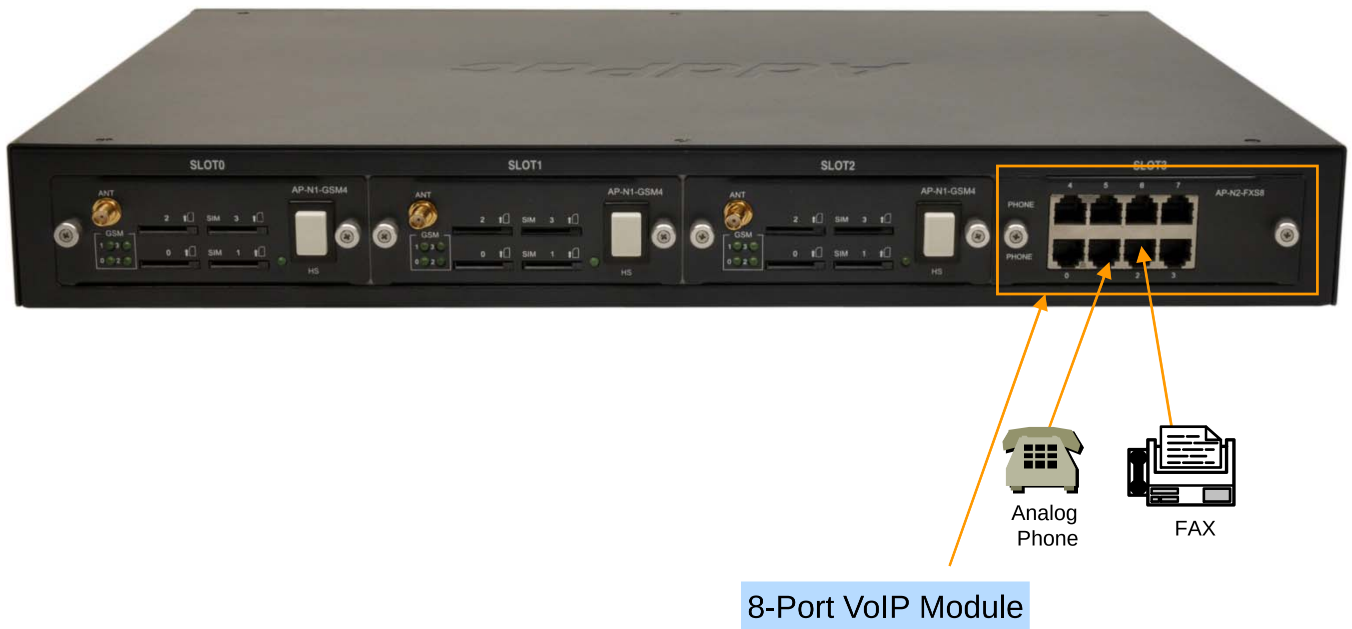
Hardware Specification

AP-GS2500 GSM SMS Gateway

RISC
CPU

High-end
DSP

Back Side View



Hardware Specification

AP-GS2500 GSM SMS Gateway

RISC
CPU

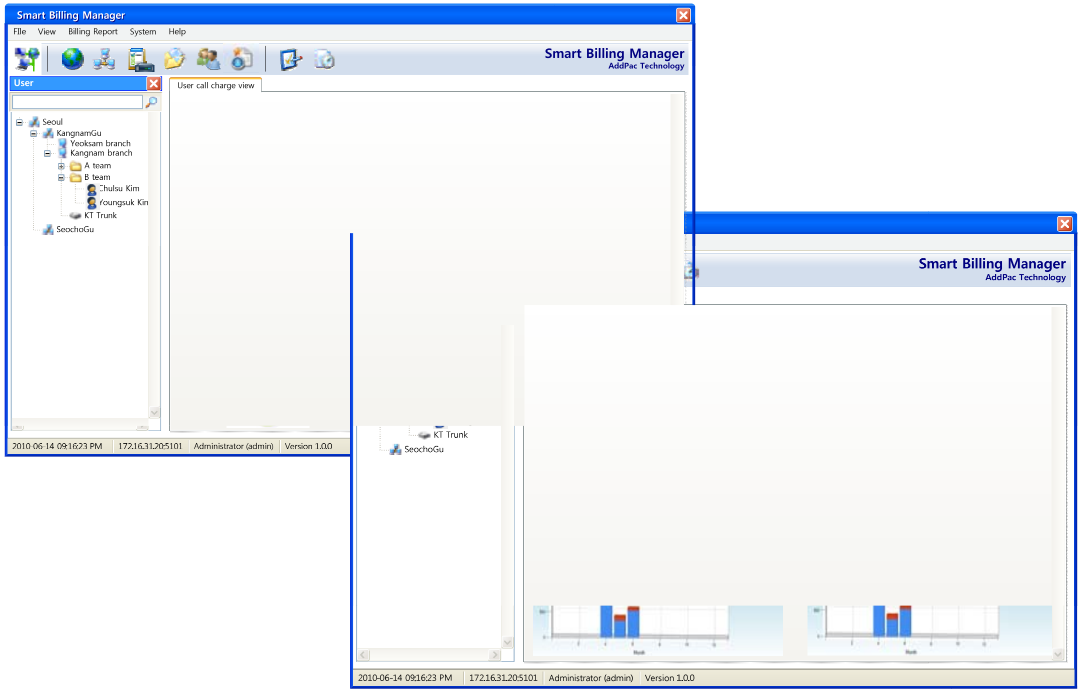
High-end
DSP

Hardware Specification(Voice Module)

AP-GS2500 GSM SMS Gateway

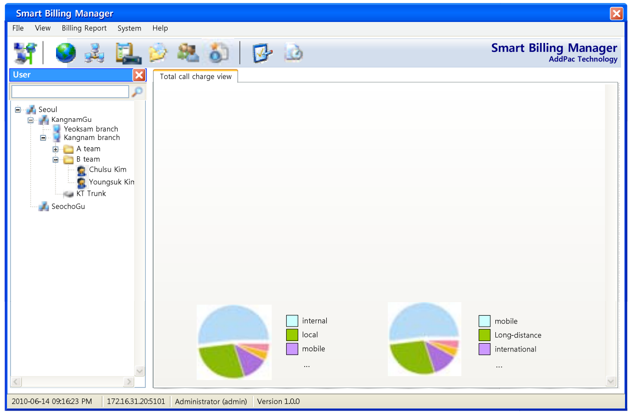
AP-N1-3G4 4-Port 3G Module	
AP-N1-GSM4 4-Port GSM Module	
AP-N1-CDMA4 4-Port CDMA Module	
AP-N1-FXS8 8-Port FXS Module	
AP-N1-FXO8 8-Port FXO Module	
AP-N1-FXS4O4 4-Port FXS&4-Port FXO Module	
AP-N1-E1 1-Port Digital E1/T1 Module	

Smart Billing System



Contents

- System Requirement
- Main Features
- System Configuration
- Site / Phone User / Department / Trunk Management
- Call Report Generation
- Call Report Notification
- System Performance Monitoring
- Access Level Management



System Requirement

Billing Server

- OS : Linux CentOS 5.4 or higher
- CPU : Intel Quad-Core 2.0 GHz / 1333 MHz FSB 2x4 MB cache
- Physical Memory : 8 GB
- HDD : 300G
- JRE (Java Runtime Environment) 1.5.1 or Higher
- Database : PostgreSQL 8.1.11

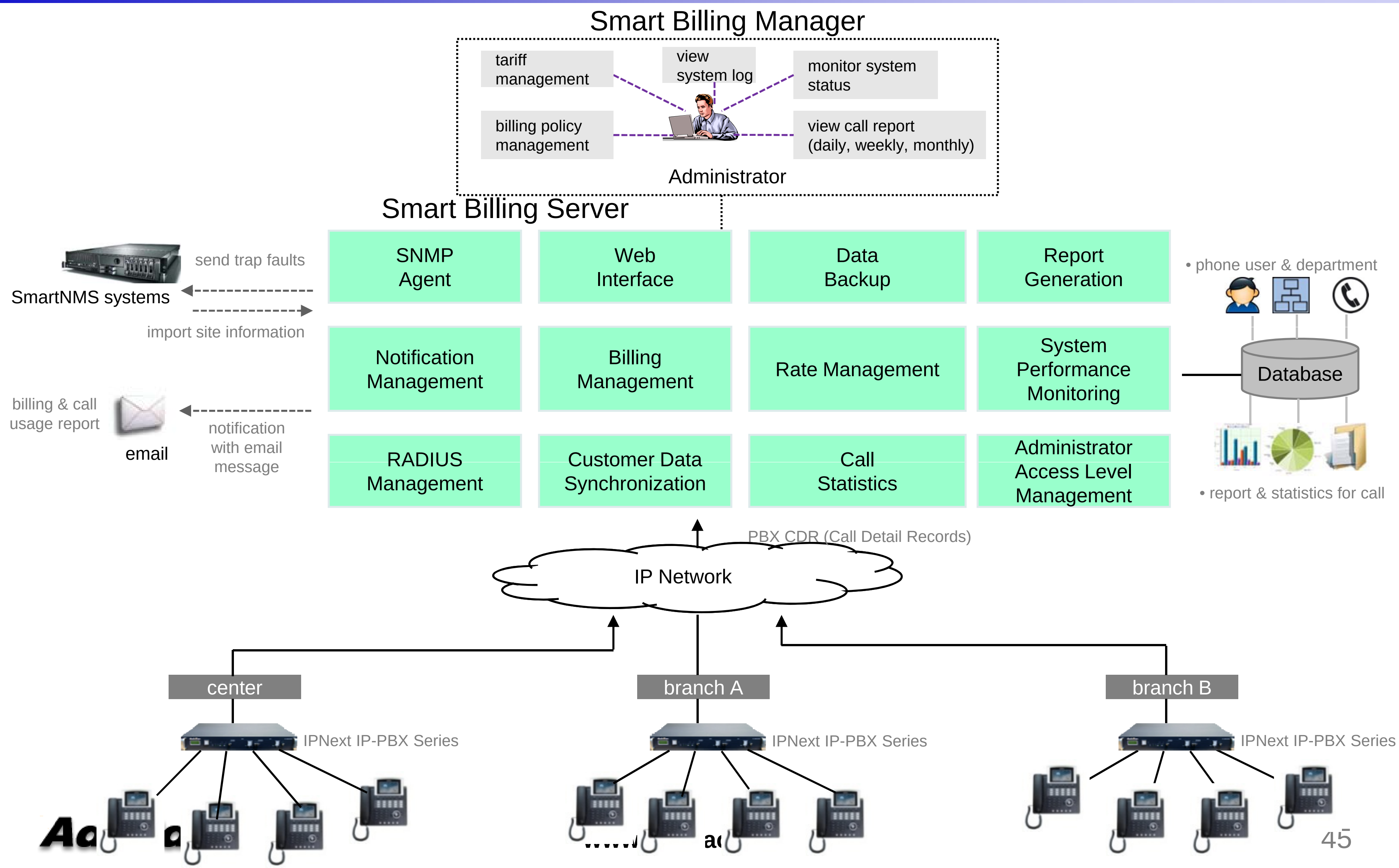
Billing Manager

- Windows XP, Vista, Windows 7, windows Server2000/2003
- Microsoft Internet Explorer 6.0 or higher

Main Features

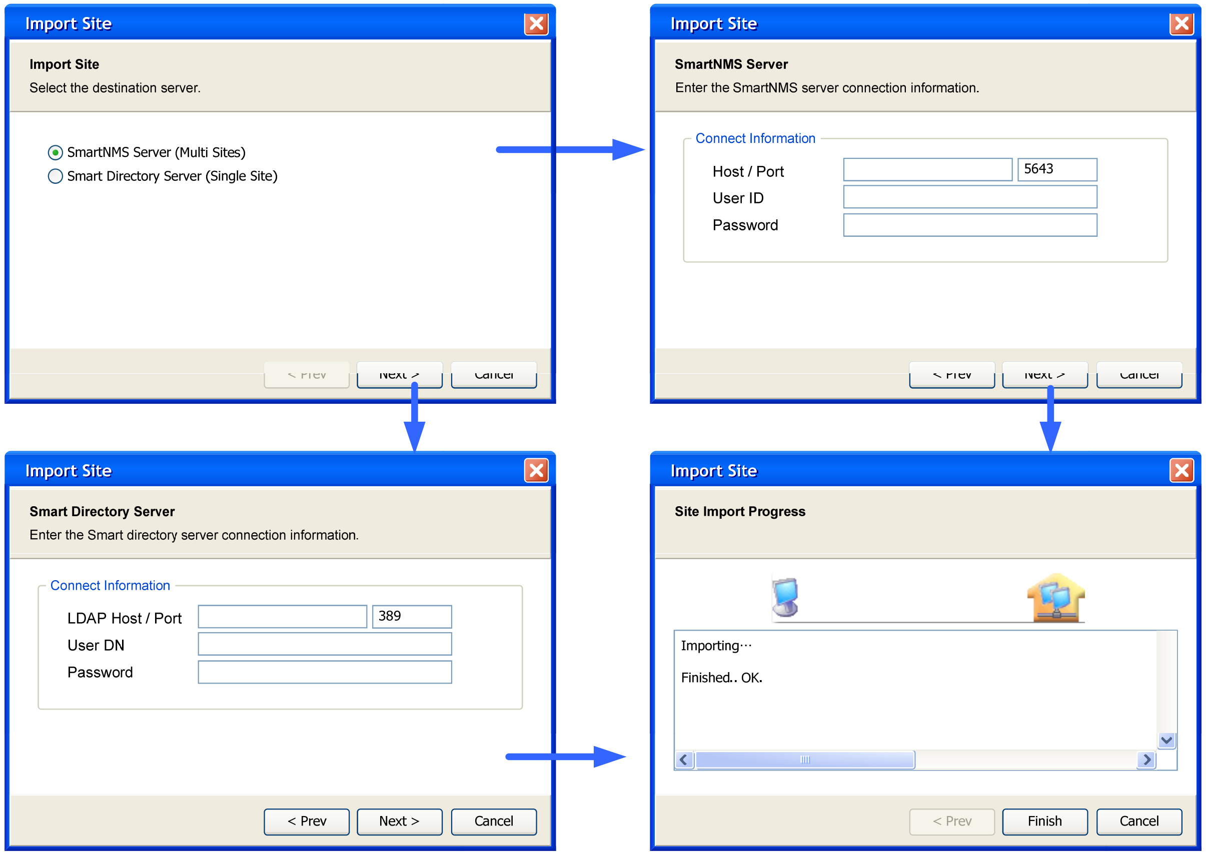
- Generate call reports based on daily / weekly / monthly and call type for site, trunk, department and phone user.
- Notify phone user or department of call report through email attachments with various policy.
- Import site / trunk / department / phone user data from NMS and PBX system.
- Manage different levels of administrator access.
- Monitor system performance for cpu utilization, memory and disk usage.
- Provide billing operating database backup with monthly based scheduling.

System Configuration



Site / Trunk / Department / Phone User Management

- Import sites information from Smart NMS operating data.
- Get trunk, department and phone user from PBX data through import wizard.



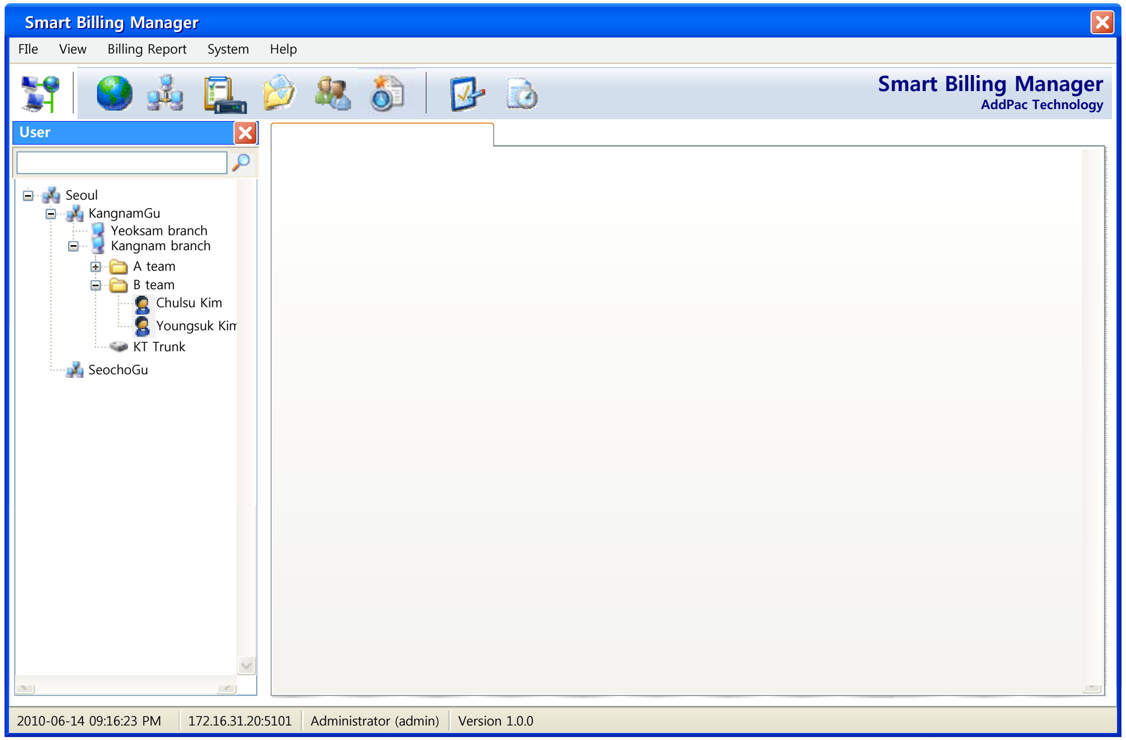
Call Report Generation

Call Report Generation

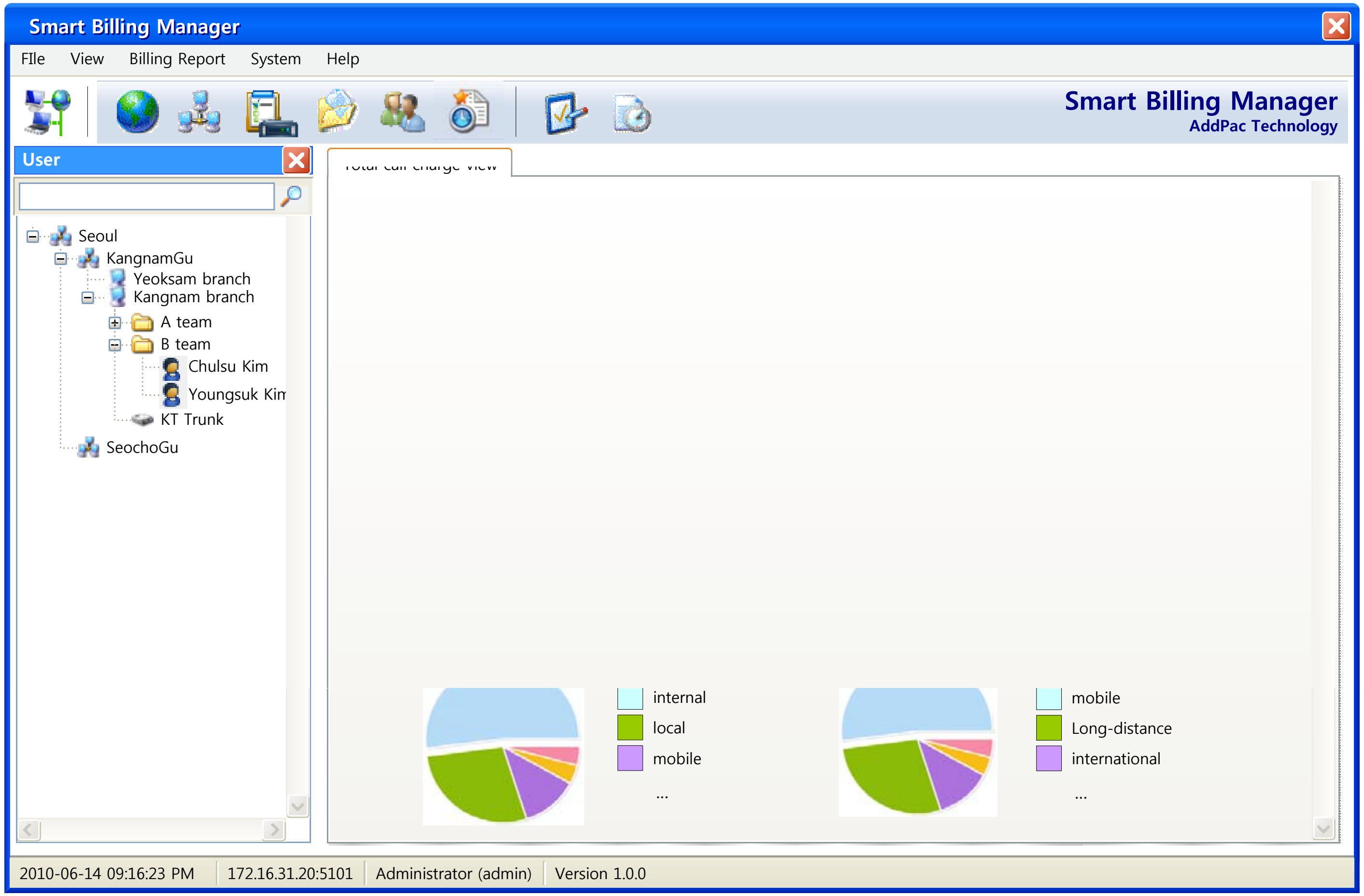
- Total Call Usage Monthly Statistic
- Total Call Charge View
- Site Call Charge View
- Trunk Call Charge View
- Department Call Charge View
- User Call Charge View
- Total Call Charge Monthly Statistic
- Total Number of Call Monthly Statistic
- Total Call Type Monthly Statistic
- Incompletion Calls by reason for Monthly Statistics of total sites
- Call Charge View for each site
- Call Usage Monthly Statistic for each site
- Number of Calls Monthly Statistic for each site
- Call Charge Monthly Statistic for each site

Call Report Generation

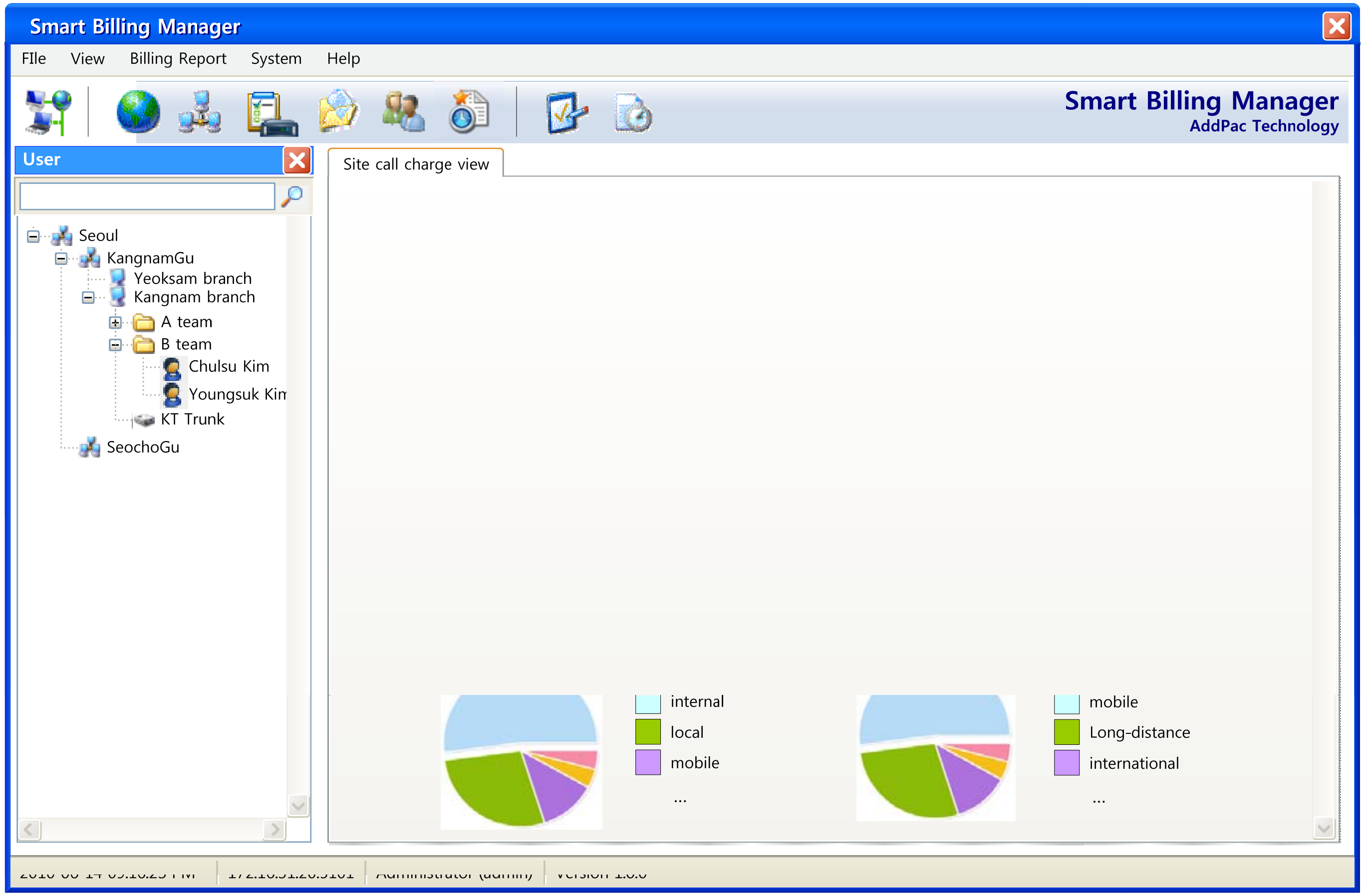
- Generate daily, weekly, monthly call usage report for site, trunk, department and phone users.
- Provide call usage with summary and chart type.



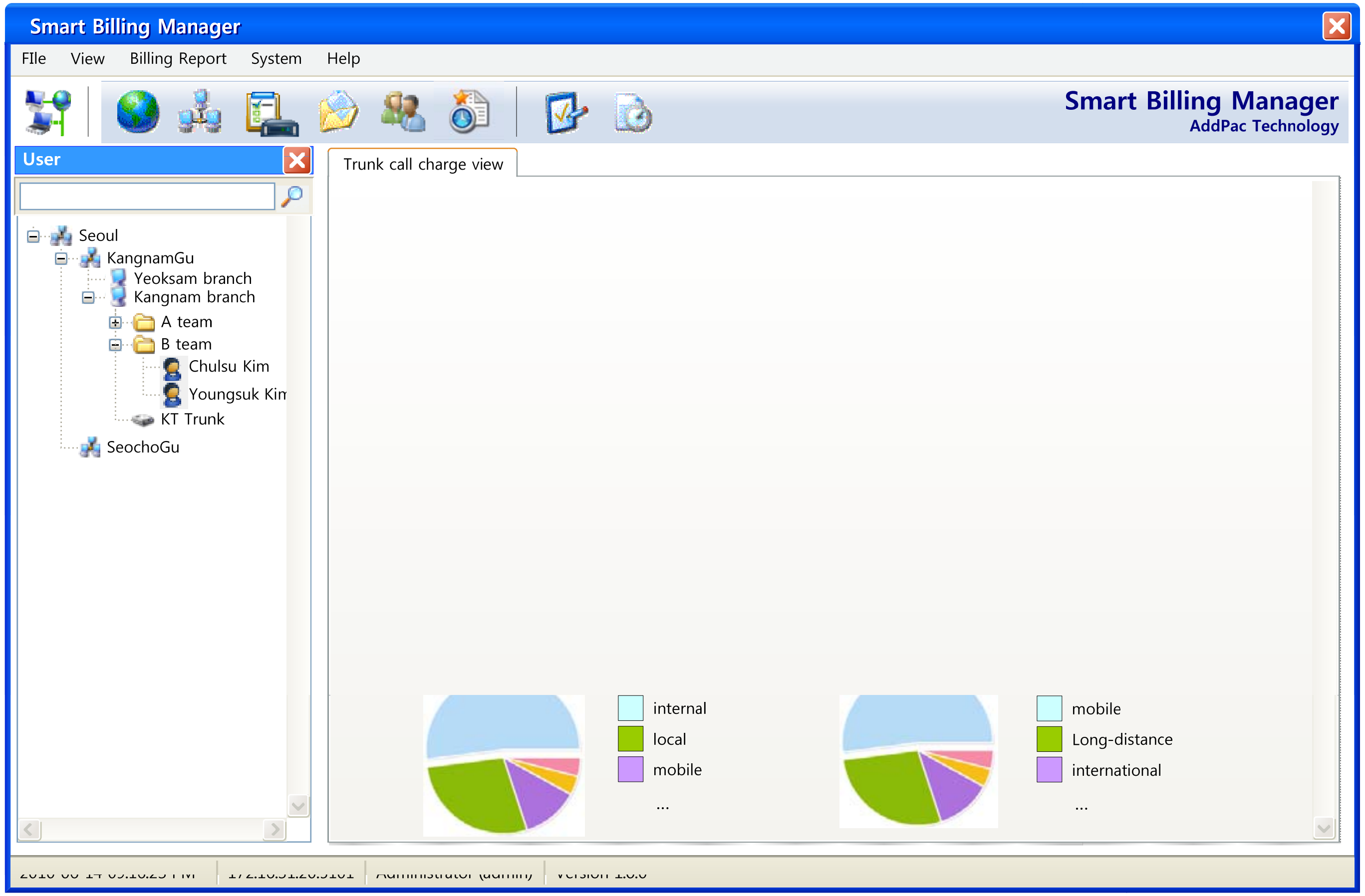
Call Report Generation



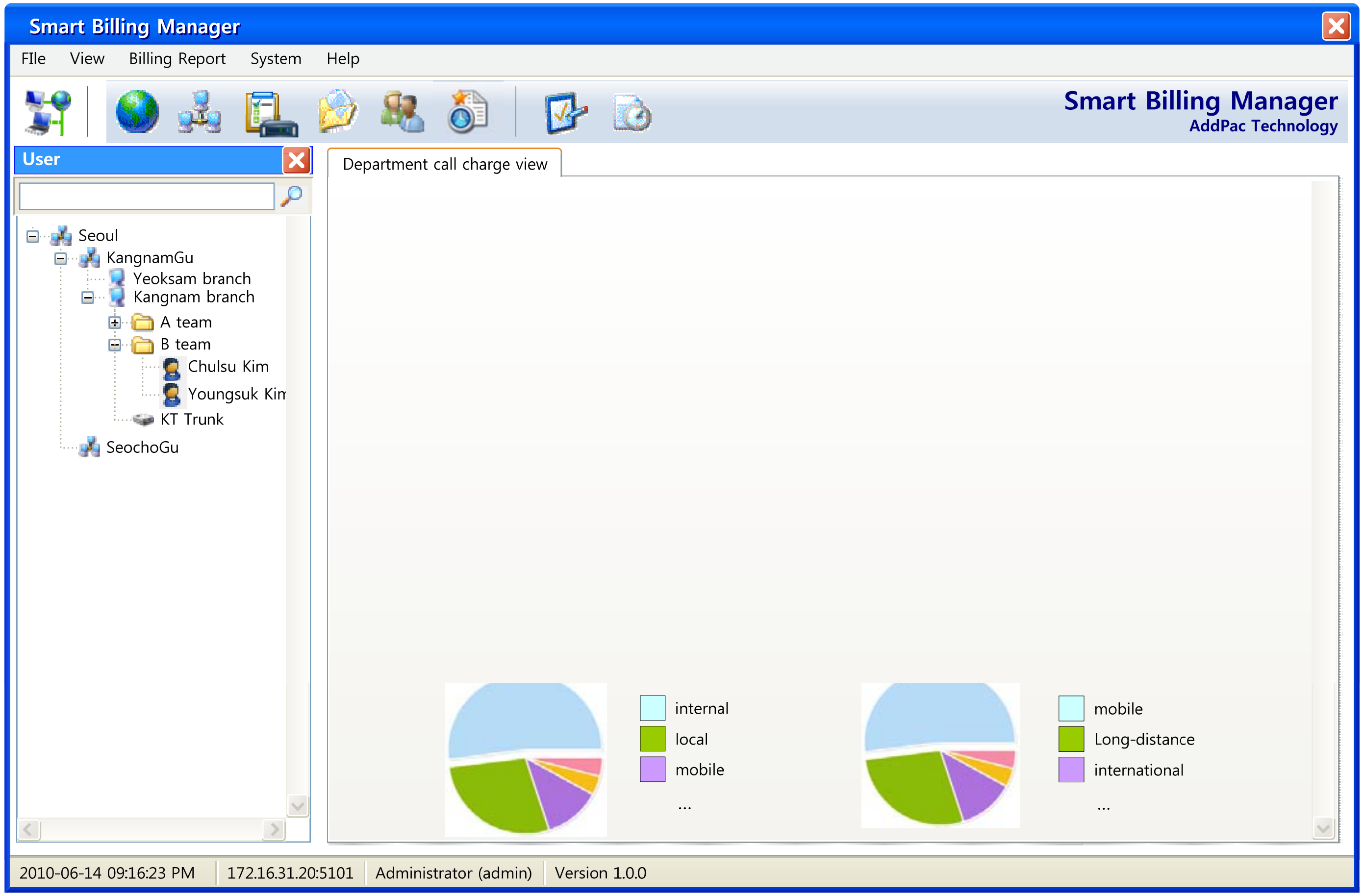
Call Report Generation



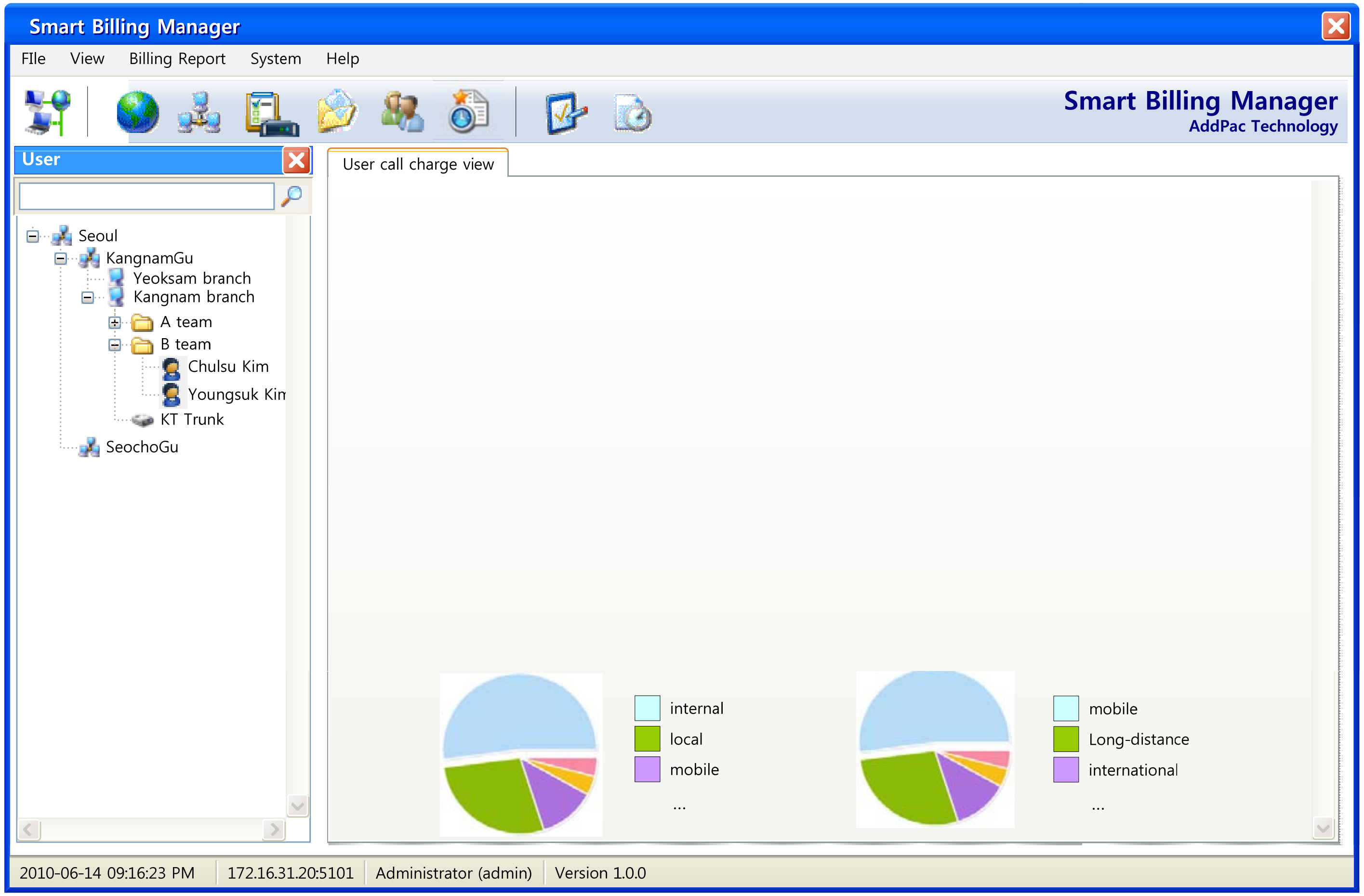
Call Report Generation



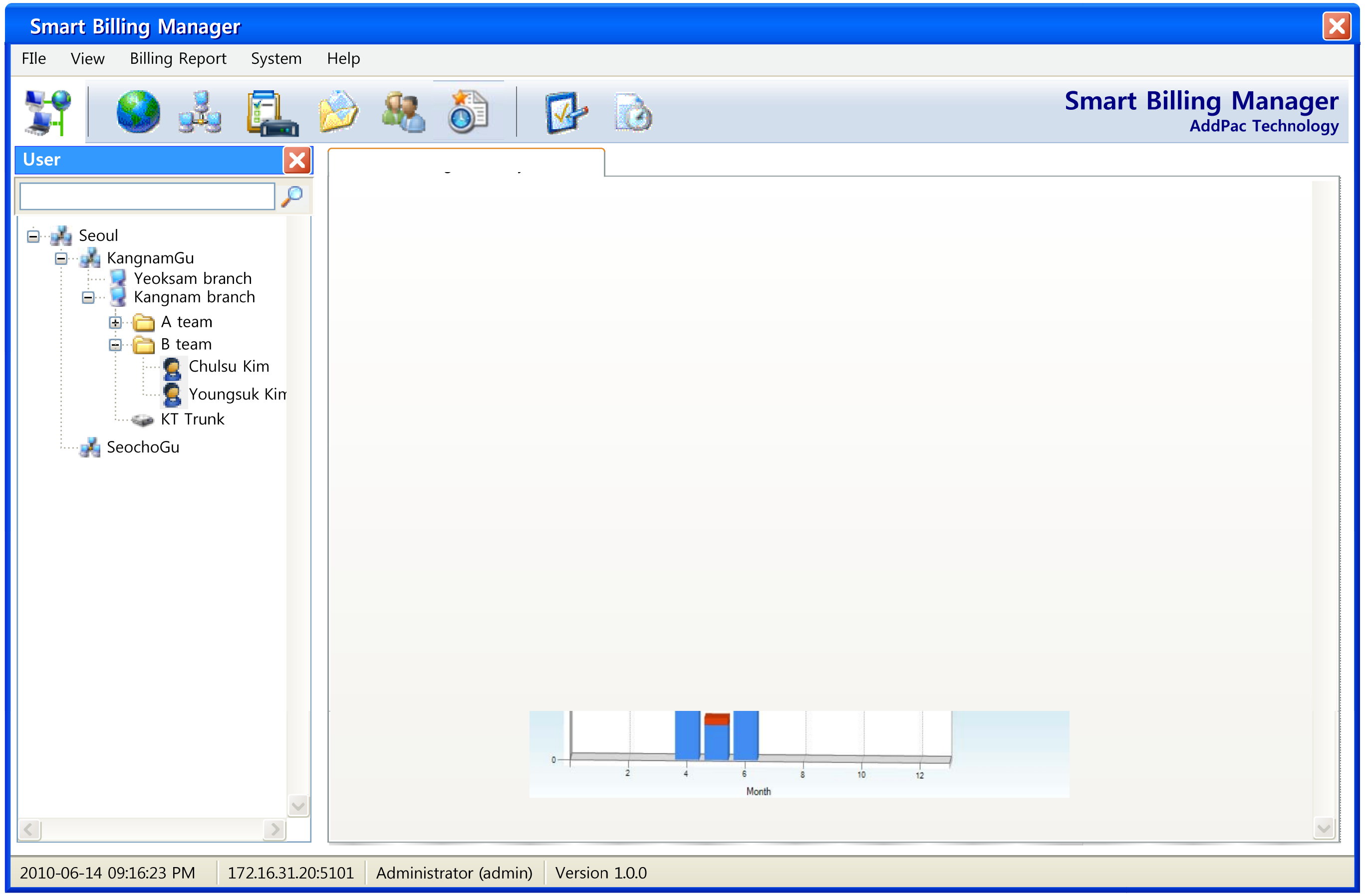
Call Report Generation



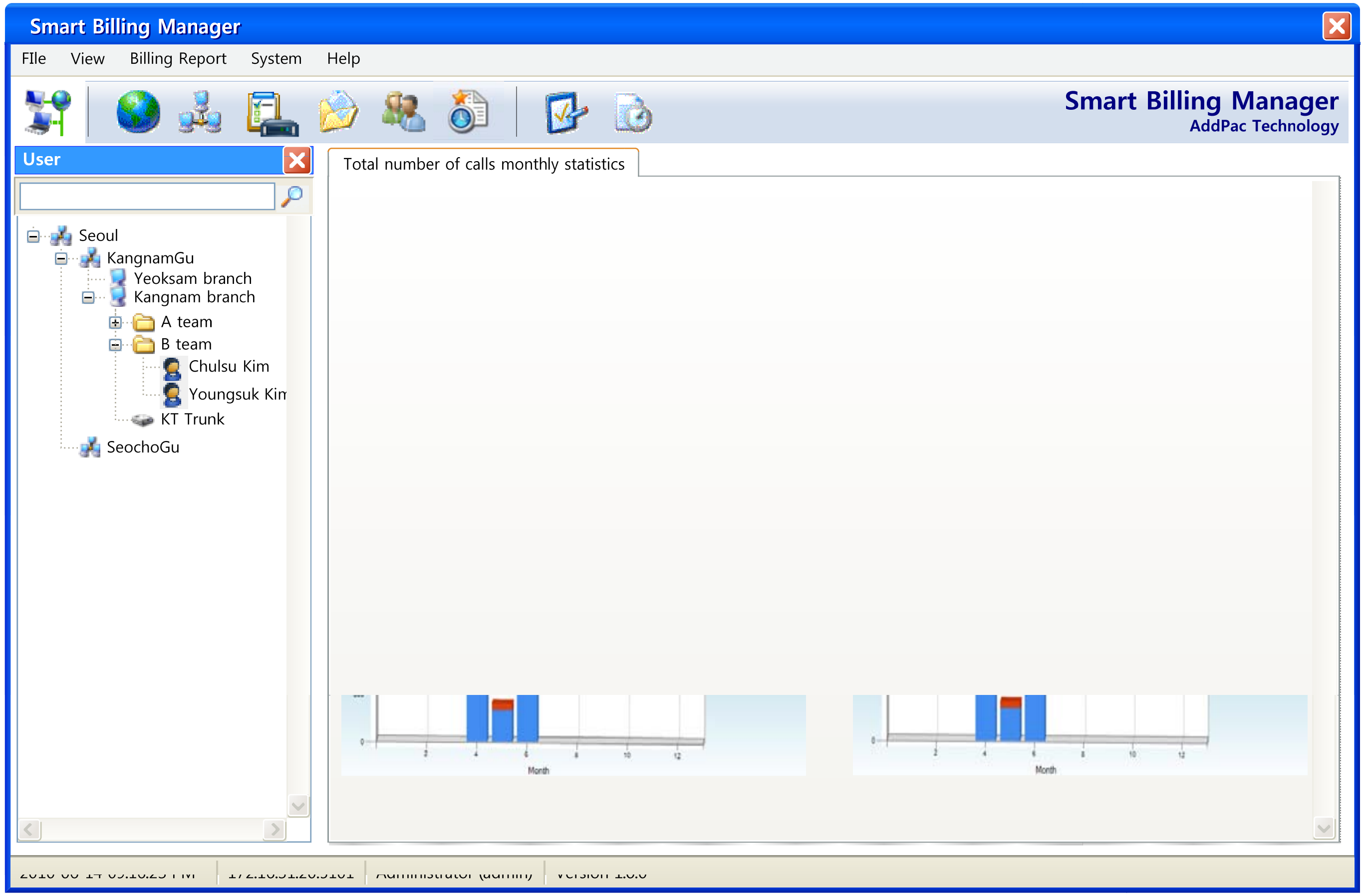
Call Report Generation



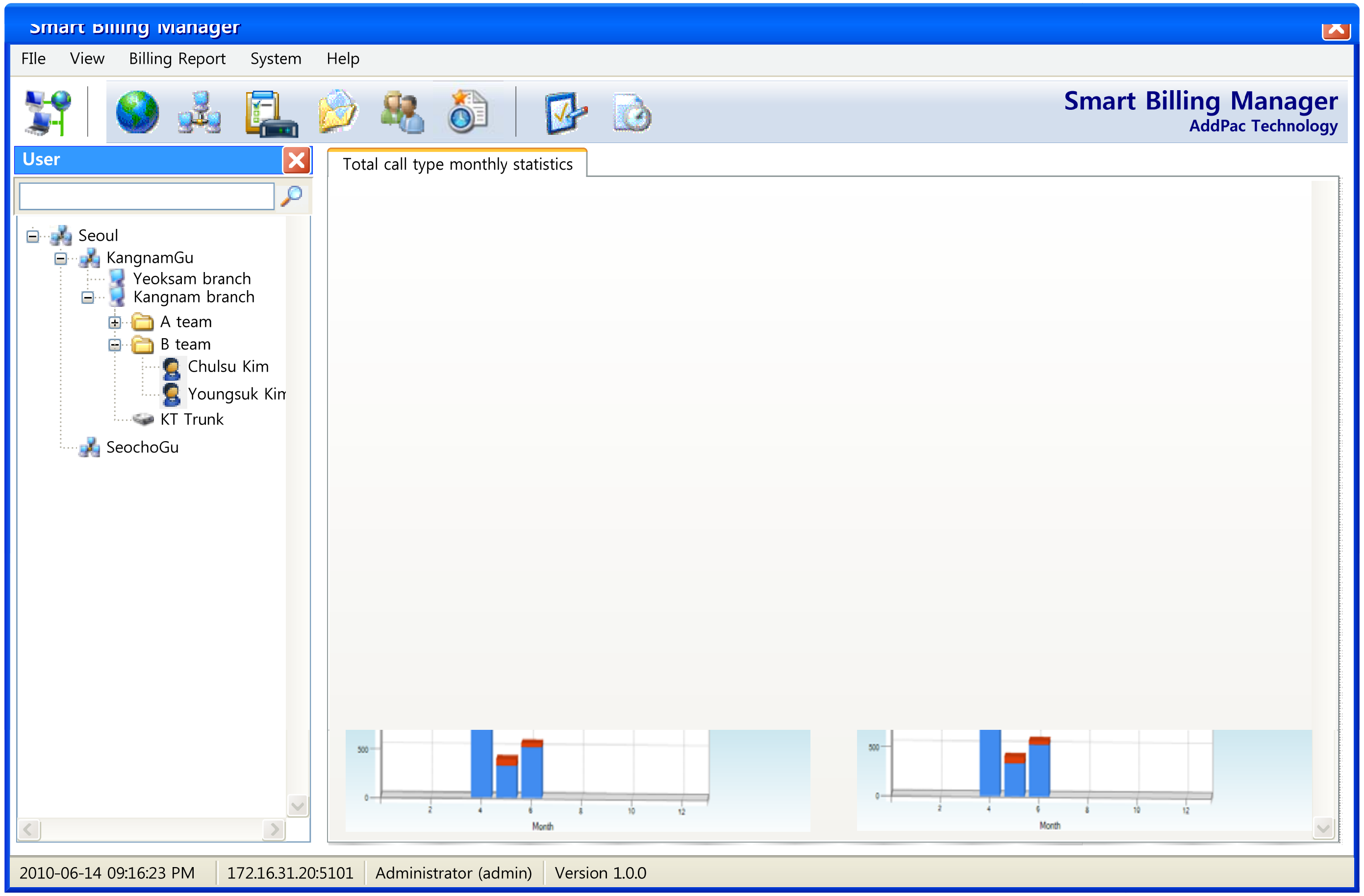
Call Report Generation



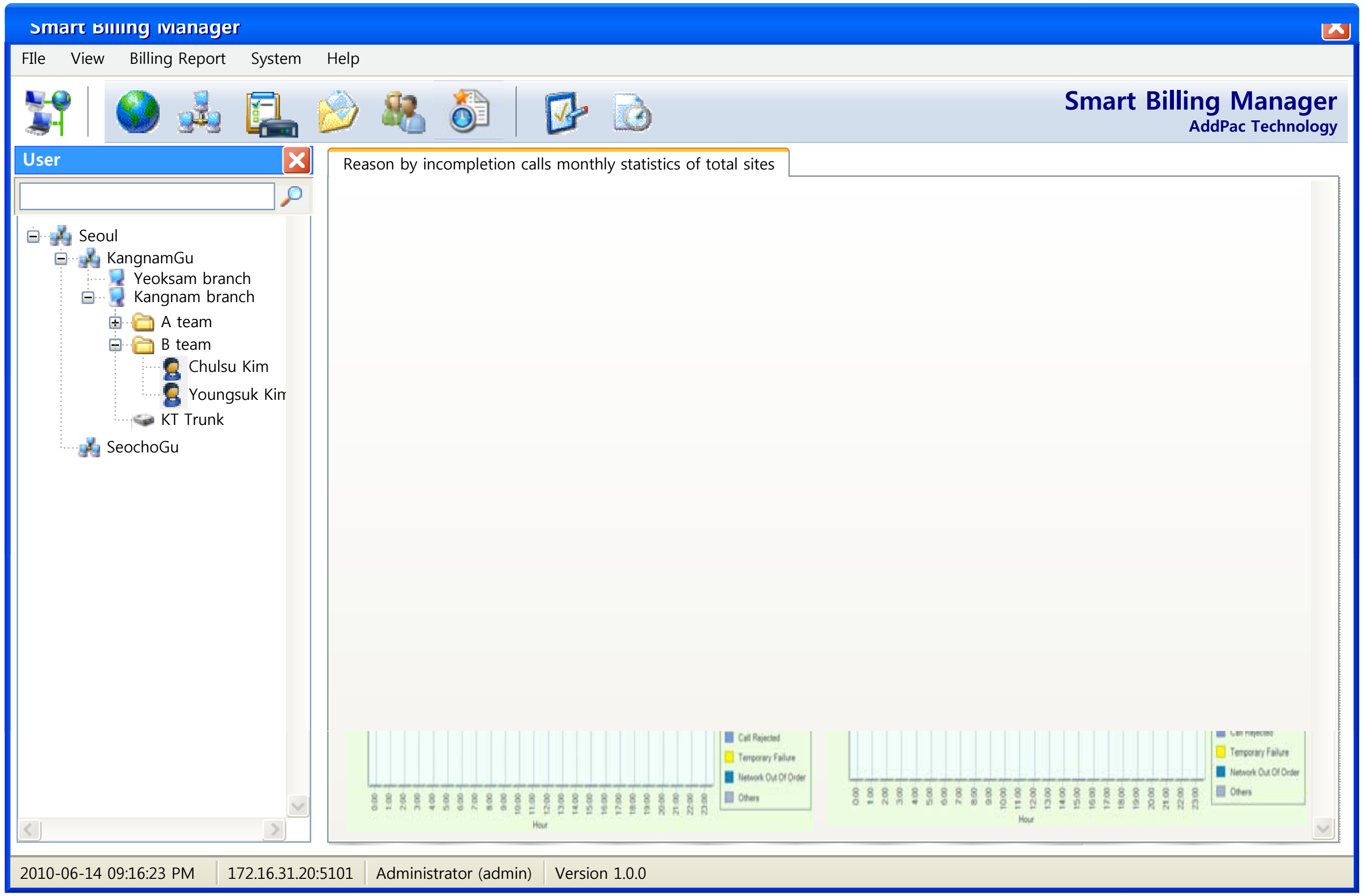
Call Report Generation



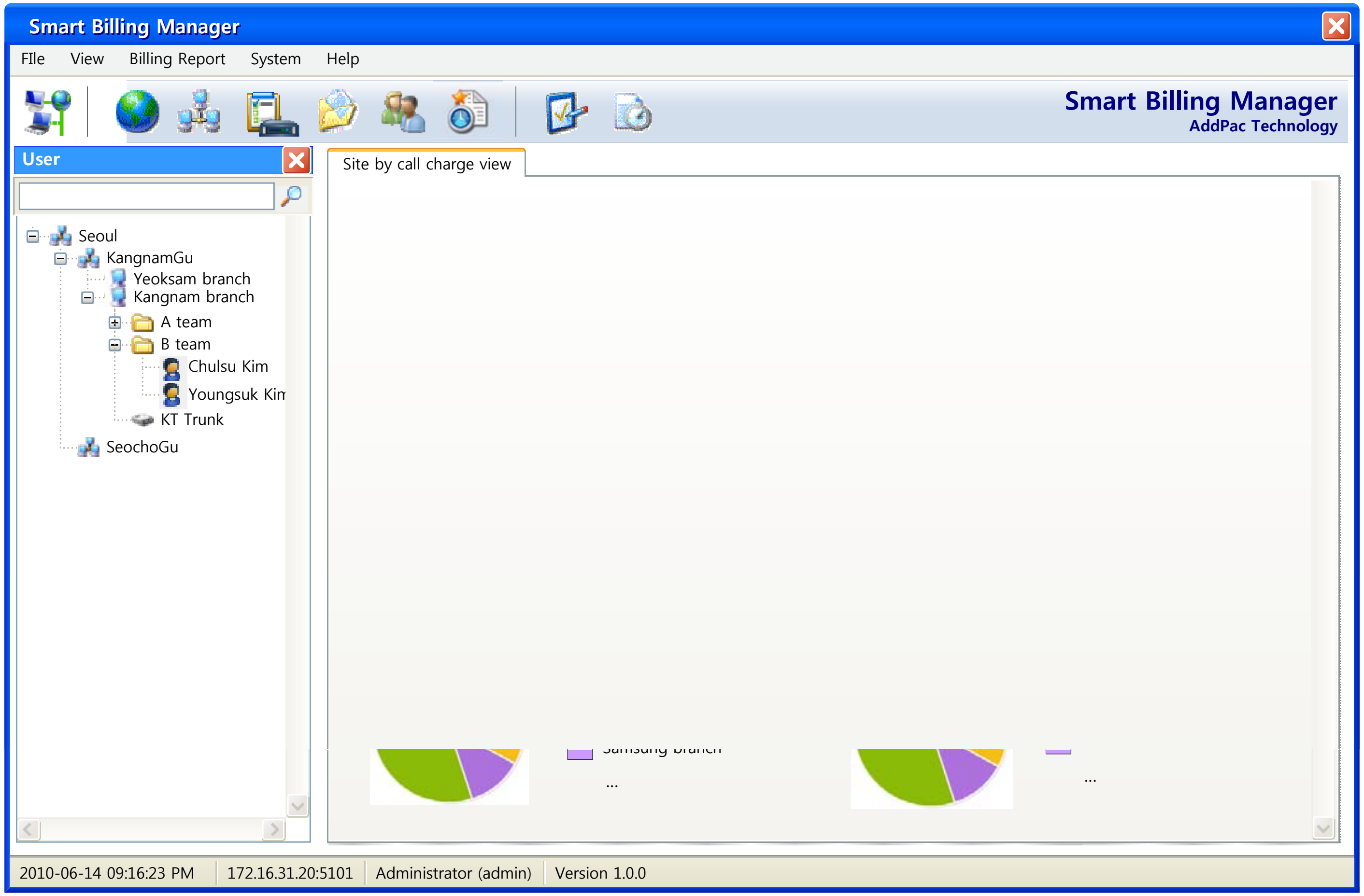
Call Report Generation



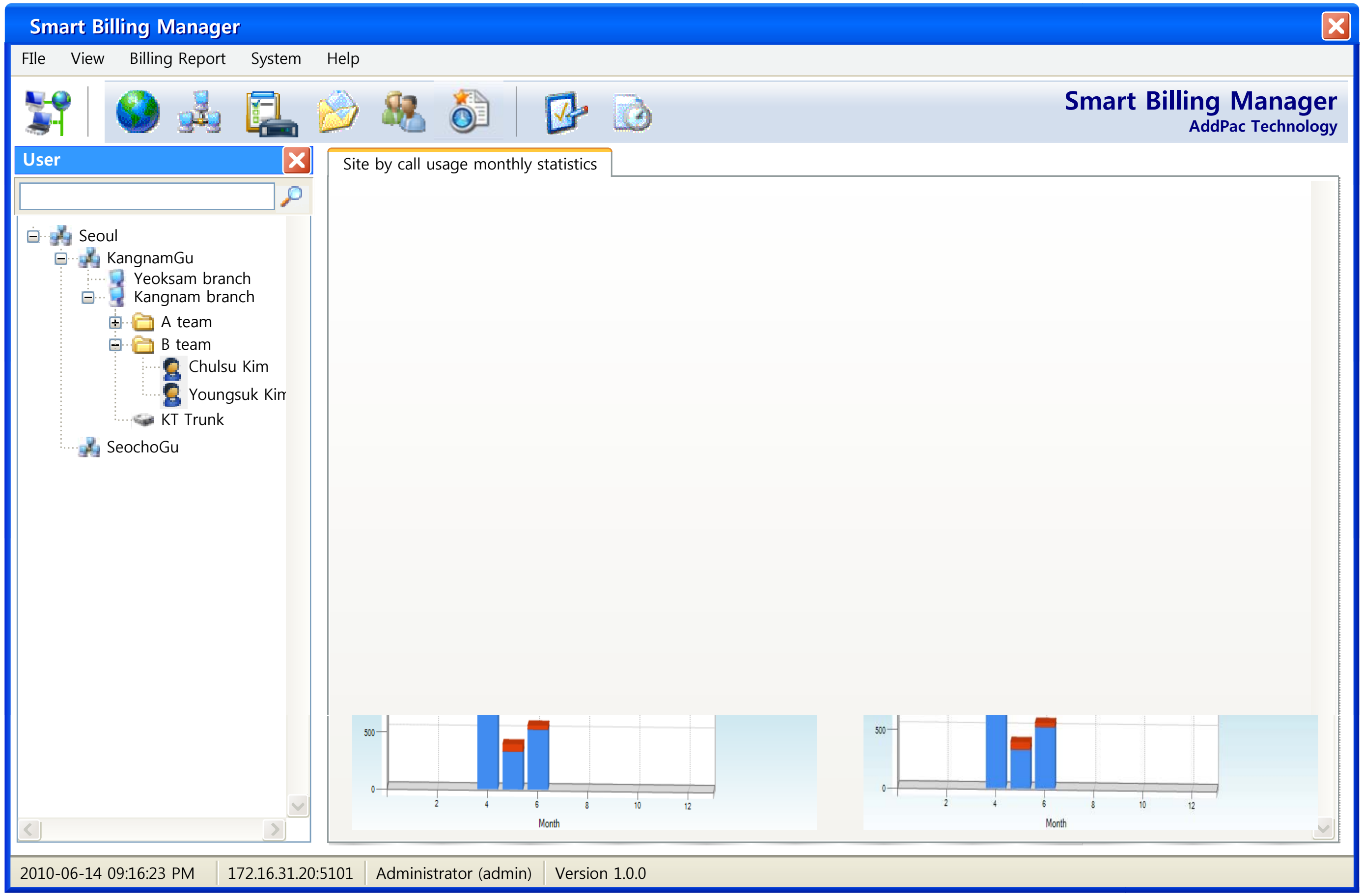
Call Report Generation



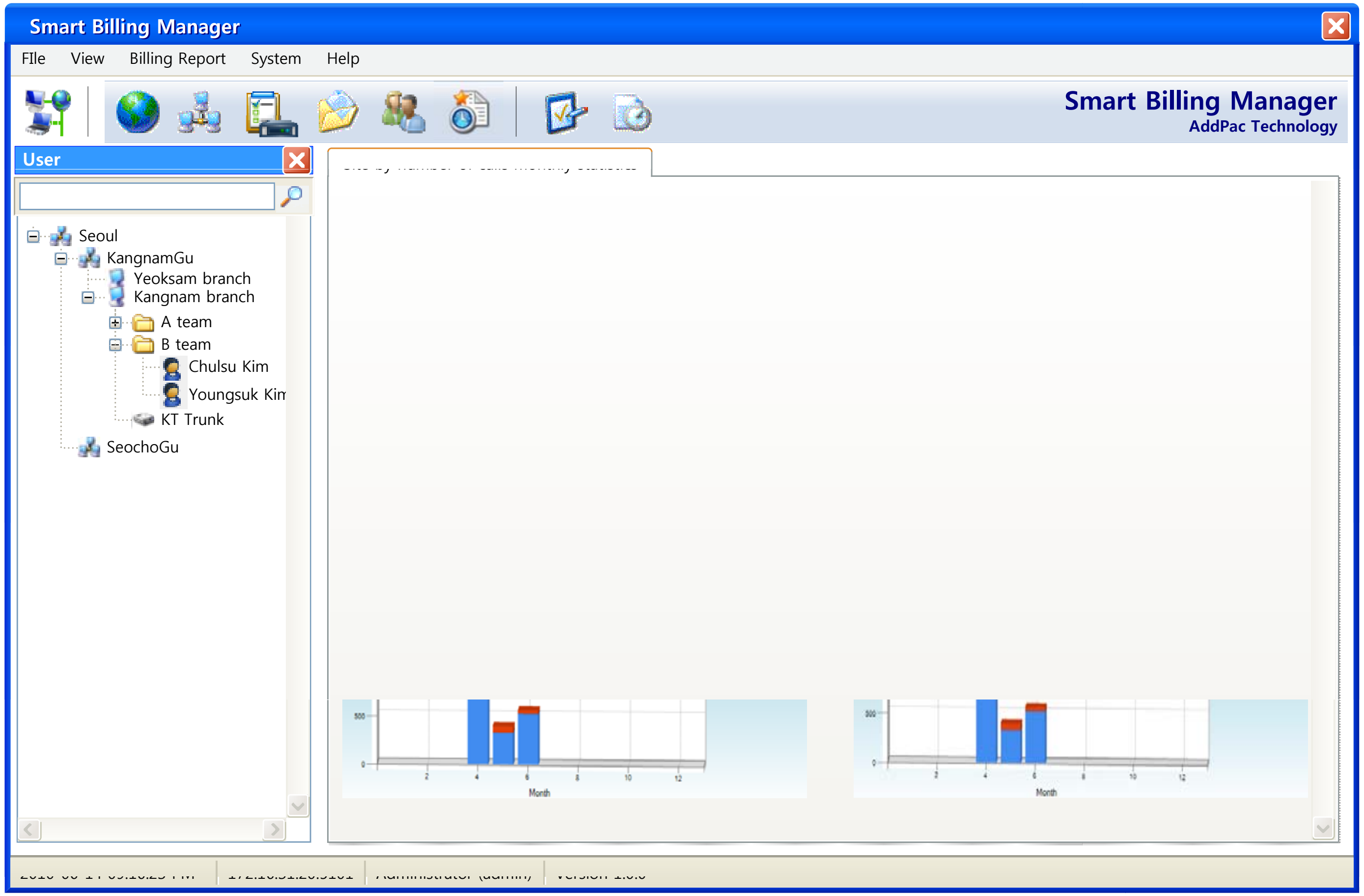
Call Report Generation



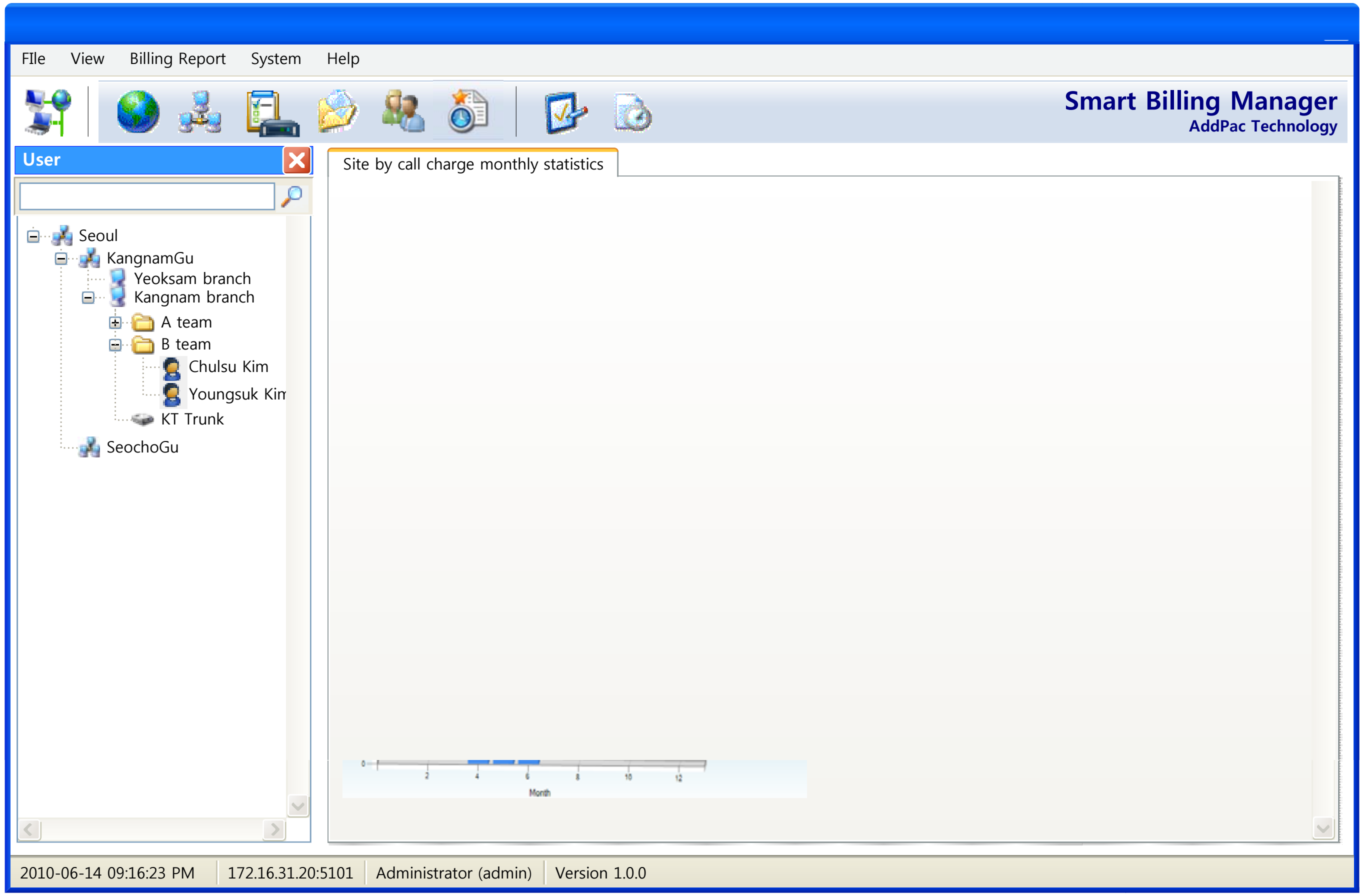
Call Report Generation



Call Report Generation

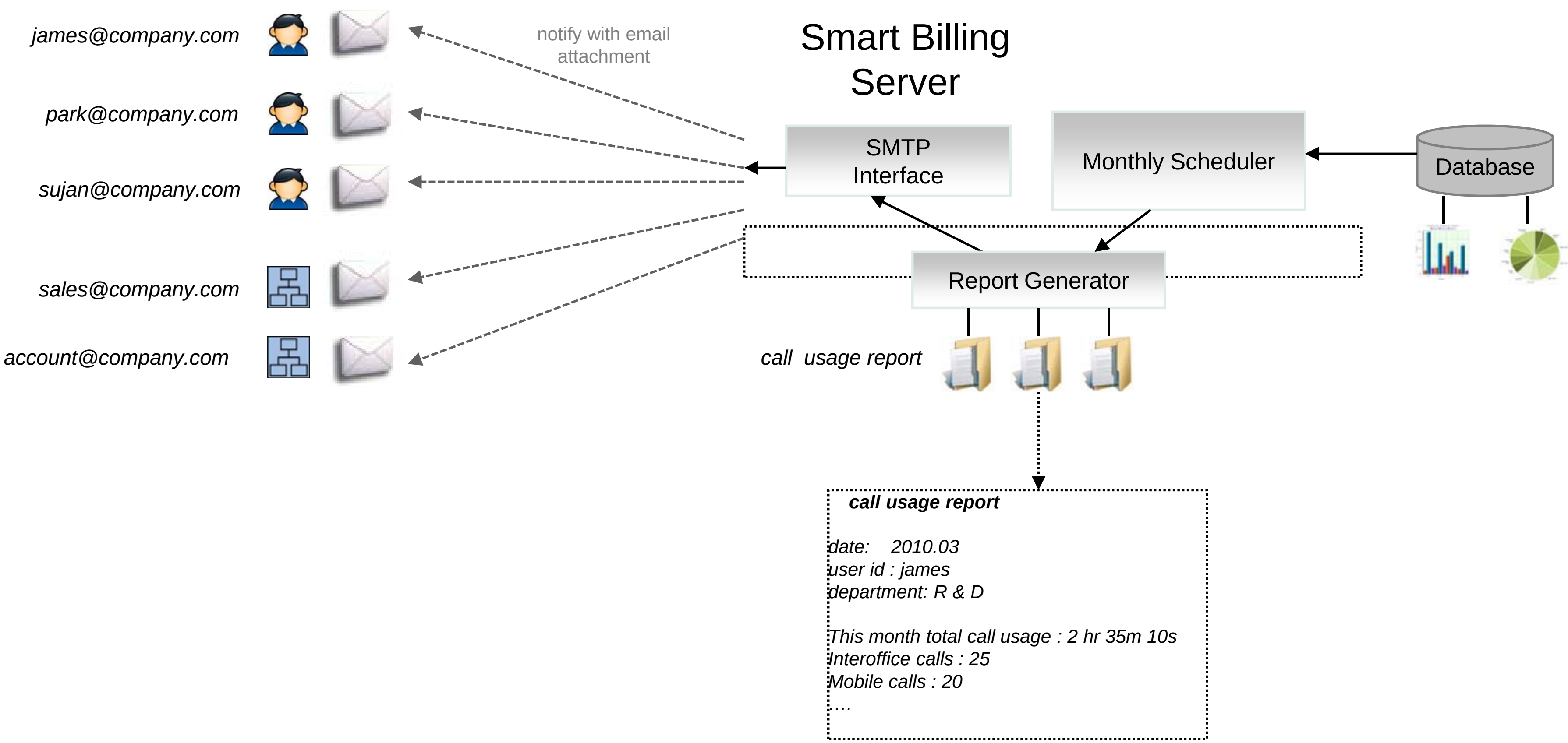


Call Report Generation



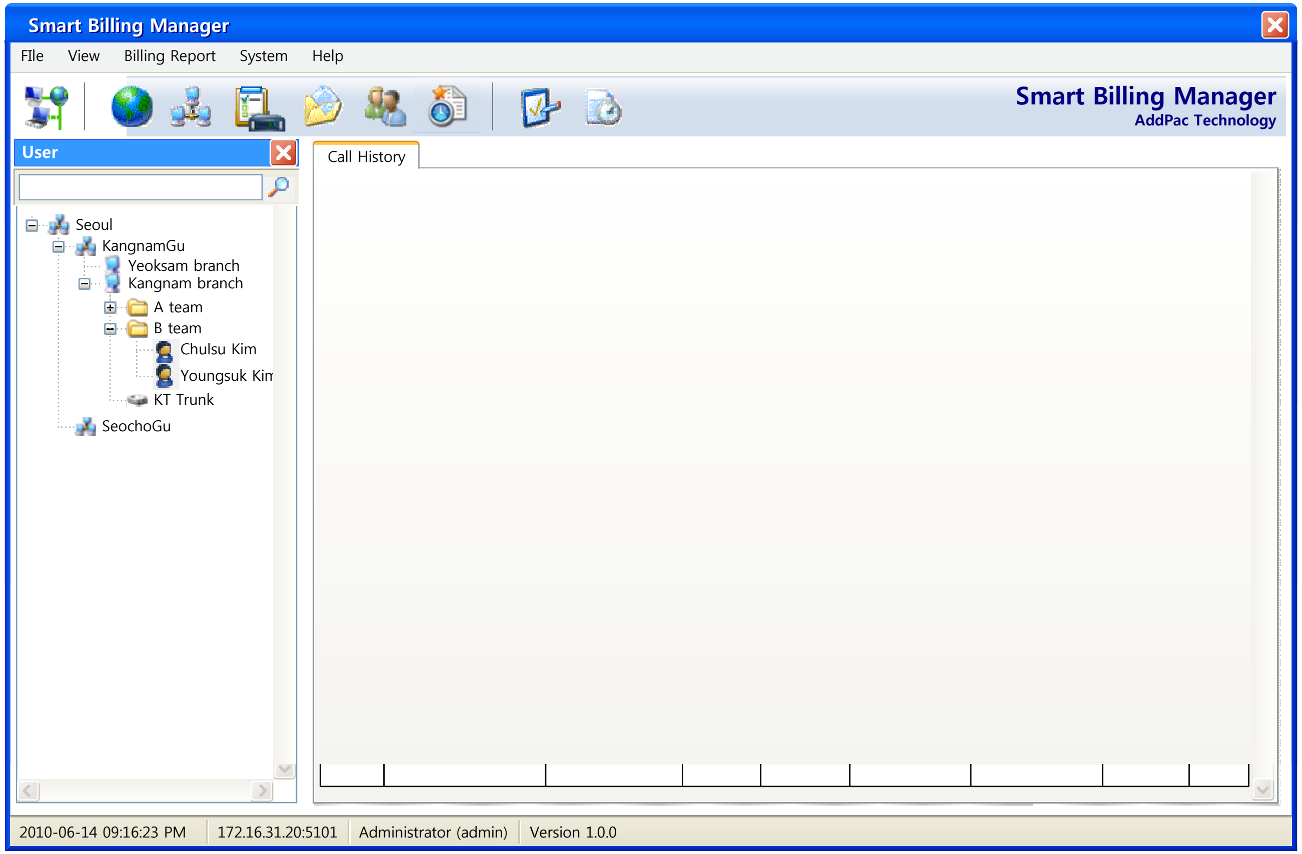
Call Report Notification

- **Notify phone user or department with email attachment for call usage report.**
- **Manage notification list such as all, phone user or department.**



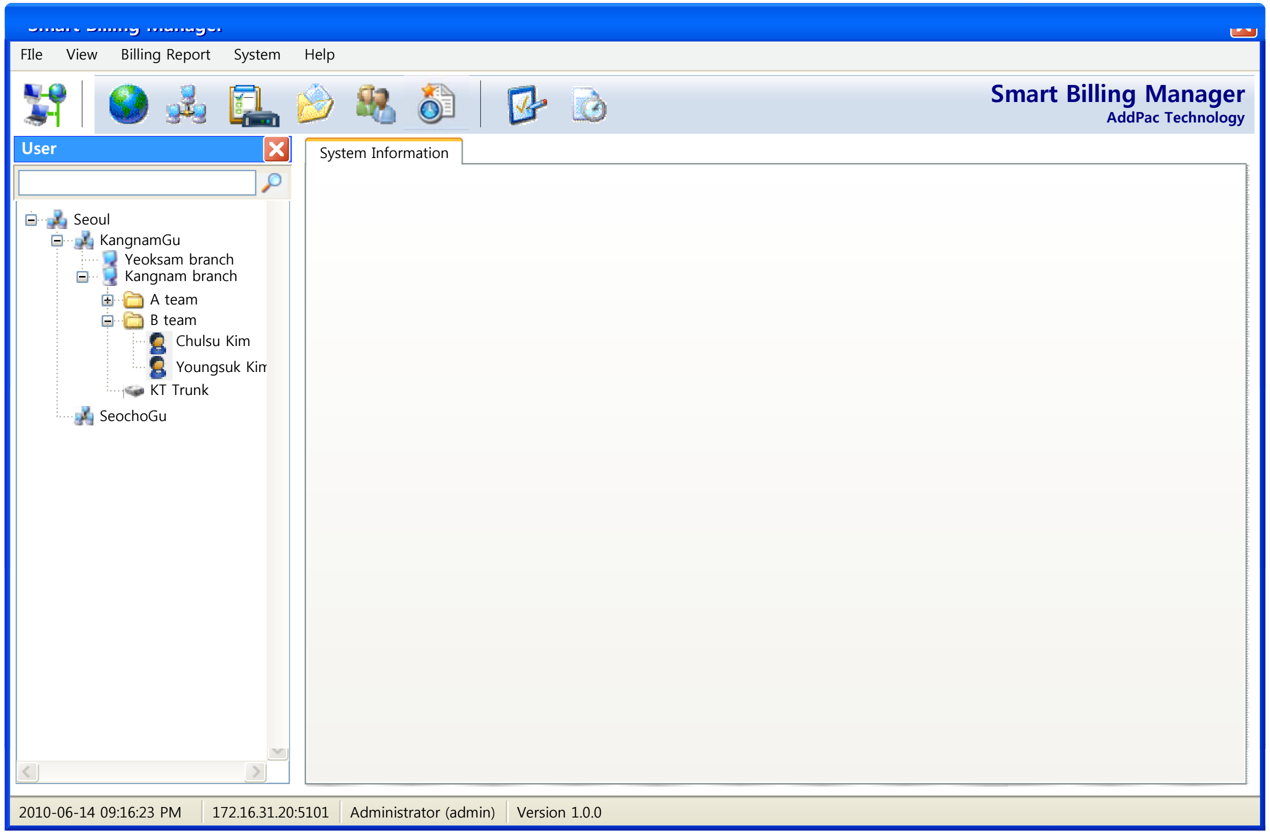
Call History

- View call detail record history with various search conditions.
- Export call history data to MS-Excel, PDF and HTML.



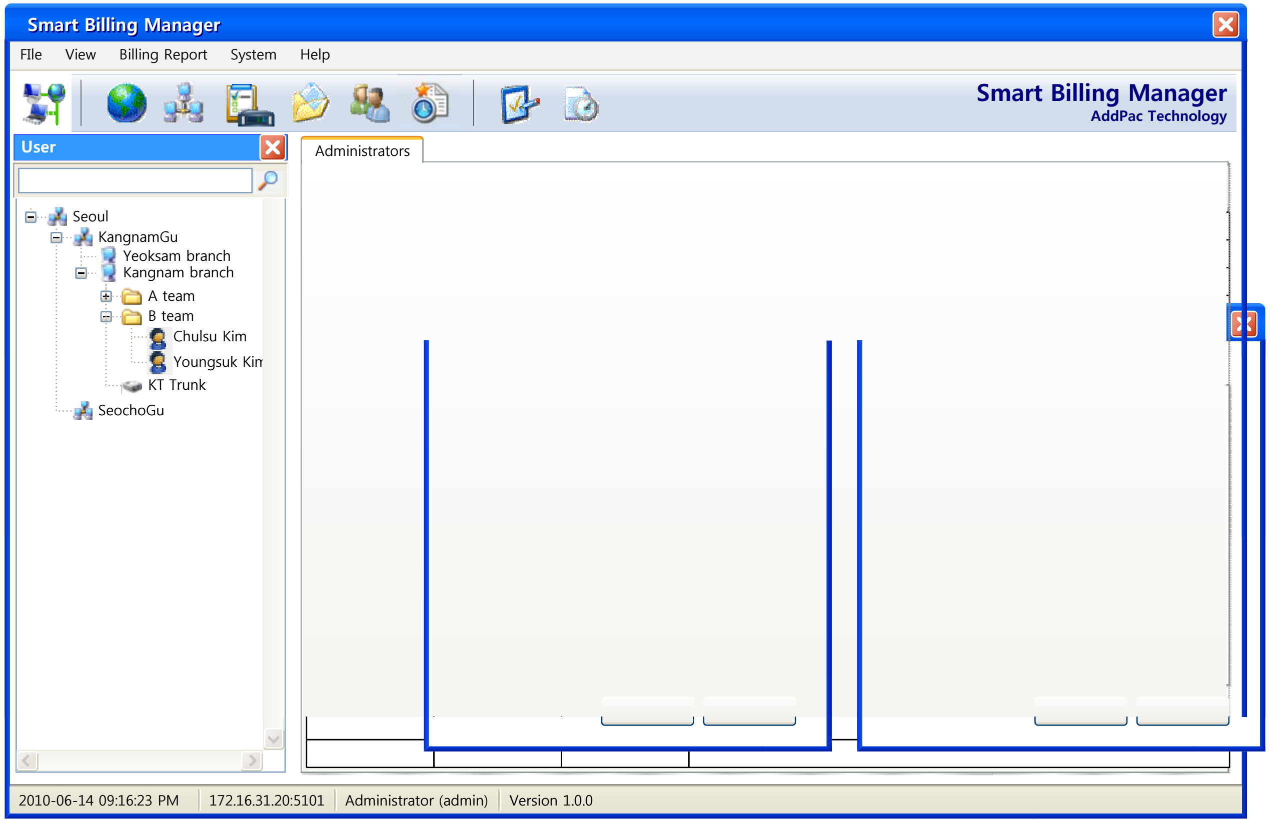
System Performance Monitoring

- **Monitor system performance such as CPU utilization, memory usage and disk space.**



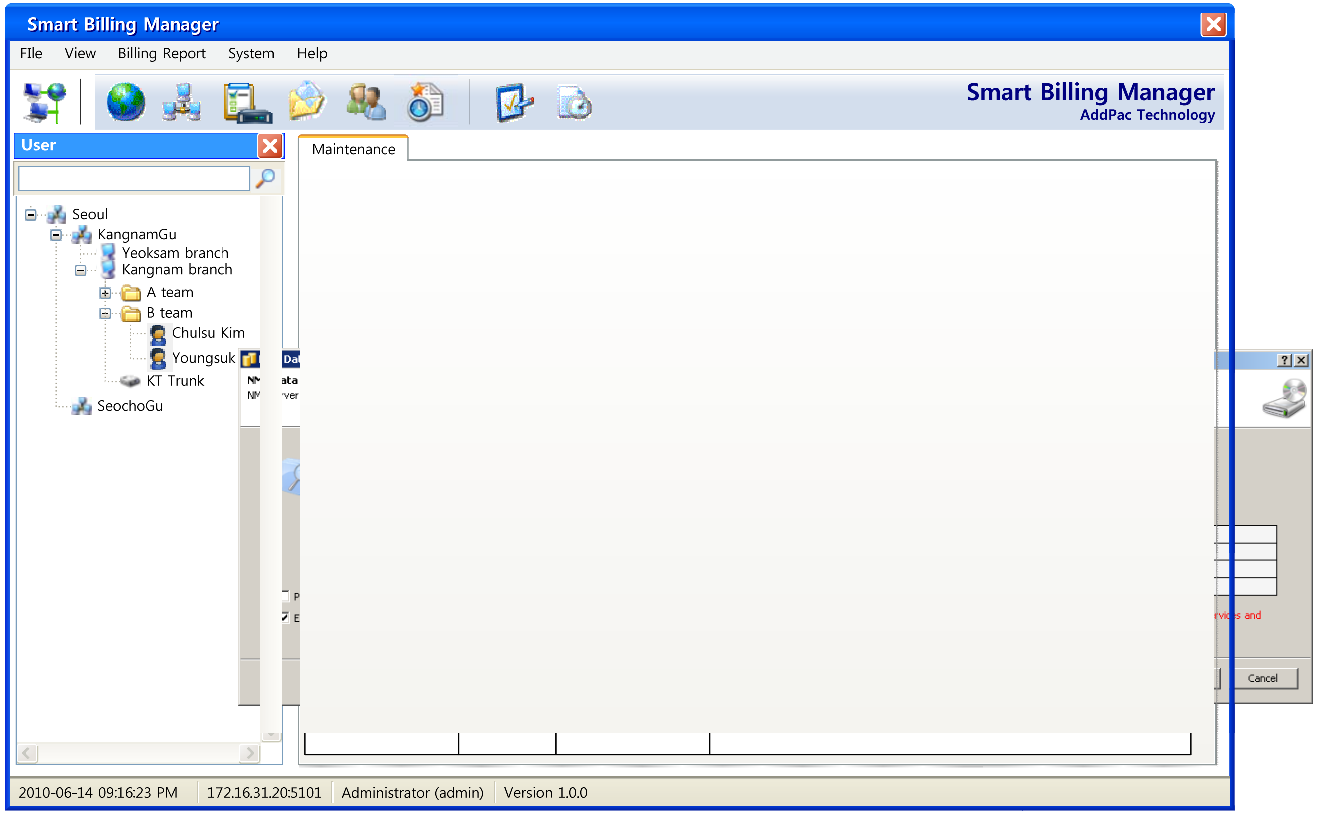
Administrator Access Level Management

- **Provide different levels of administrator access in view and menus.**
- **Manage multiple site and assign resources to administrator.**



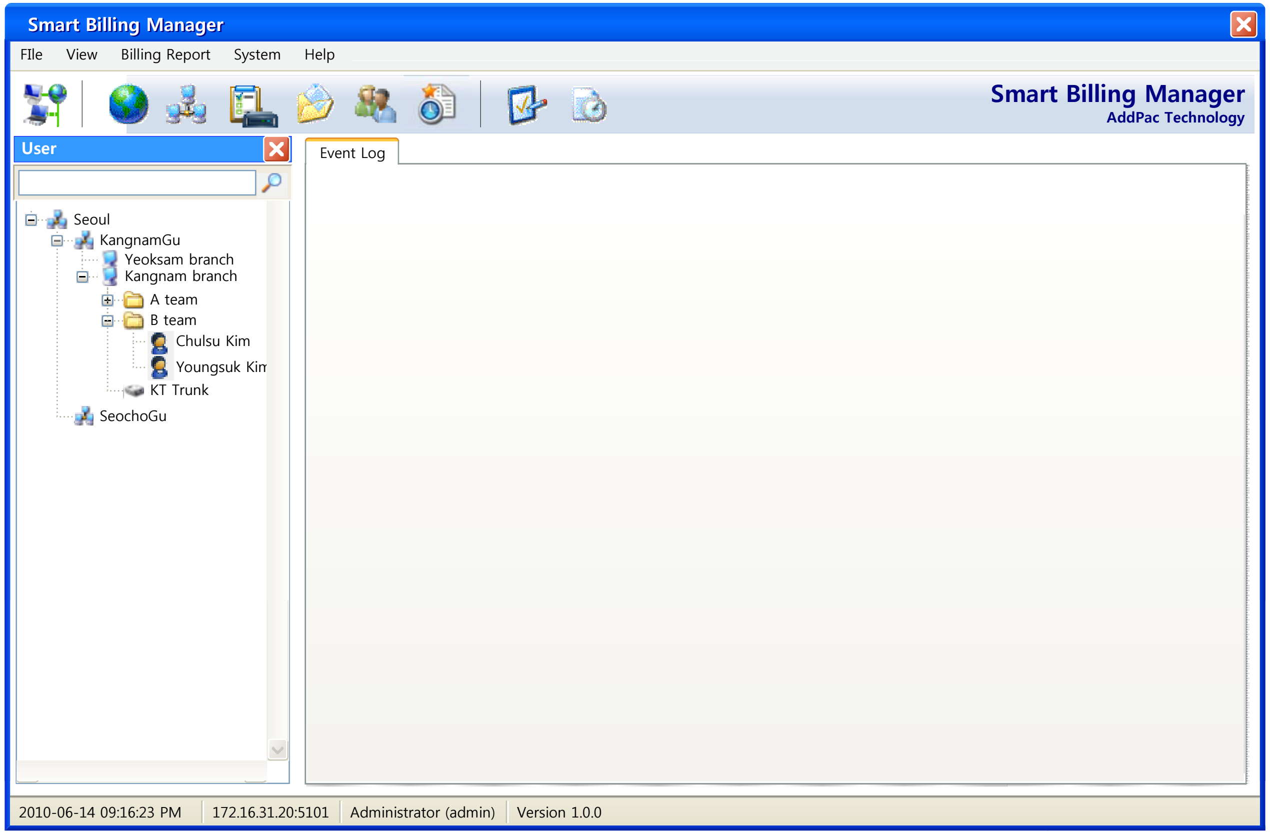
Billing Operating Database Backup

- **Provide schedule-based monthly backup of operating billing data.**
- **Backup manually with wizard style if need arise.**



View System Event Log

- **View system event log with different levels and message.**
- **Search event log with various search conditions.**

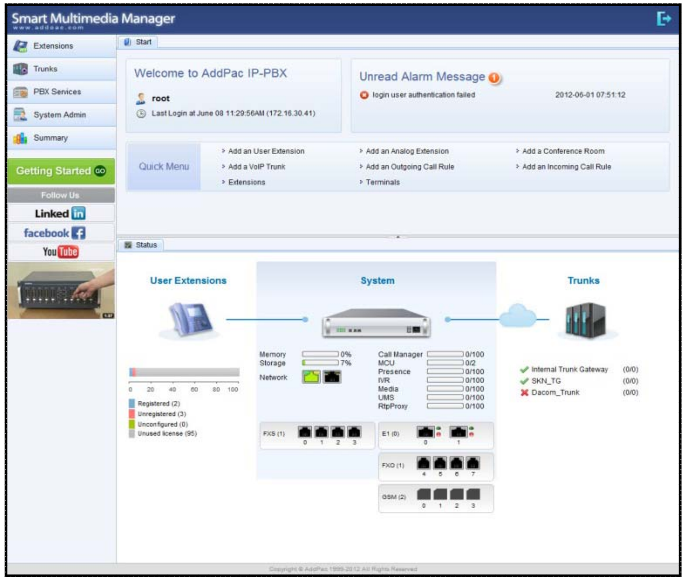


Web based Smart Multimedia Manager (WSMM) for IP-PBX

IP-PBX Series

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

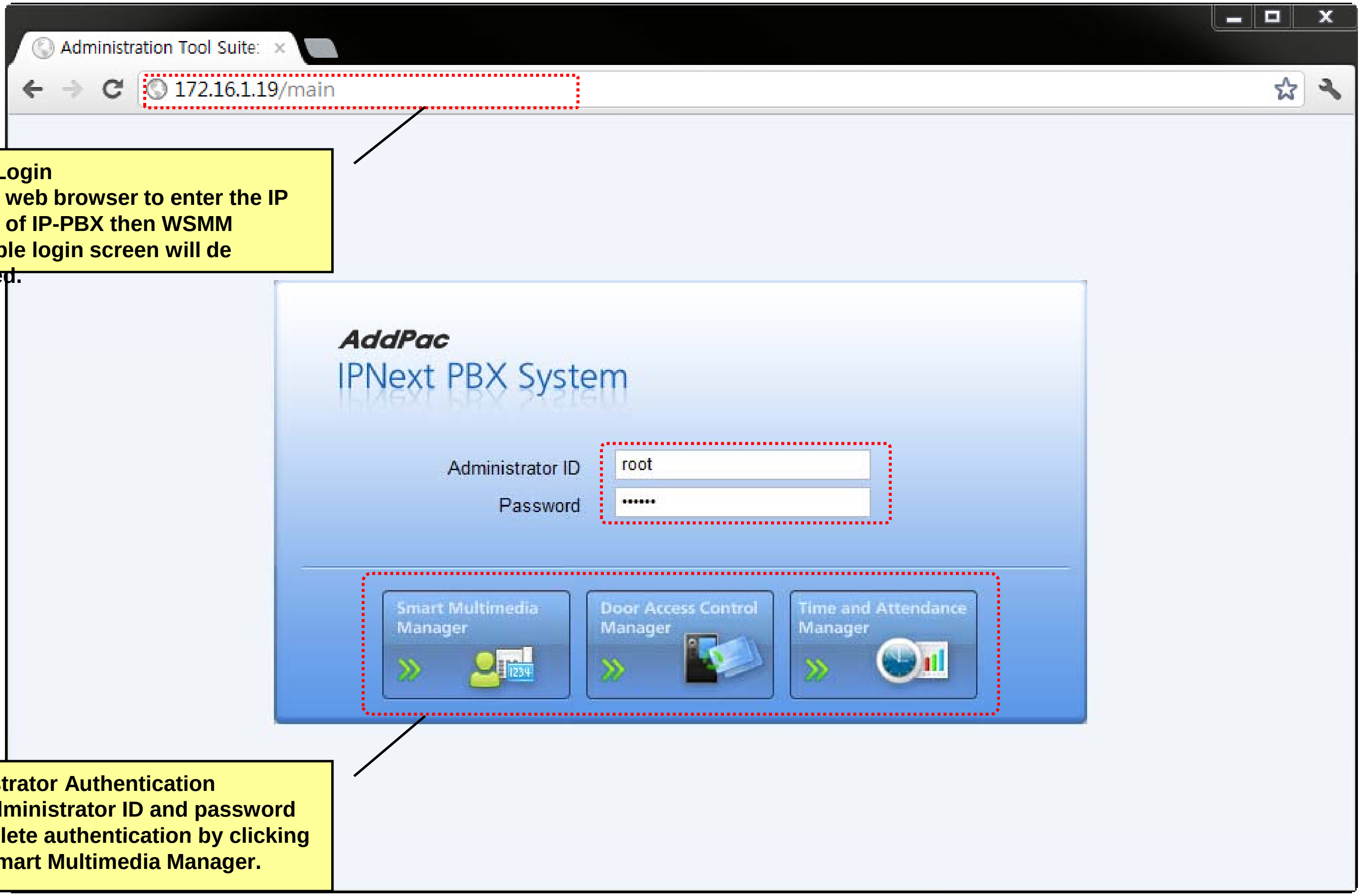
System Requirement

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,...)

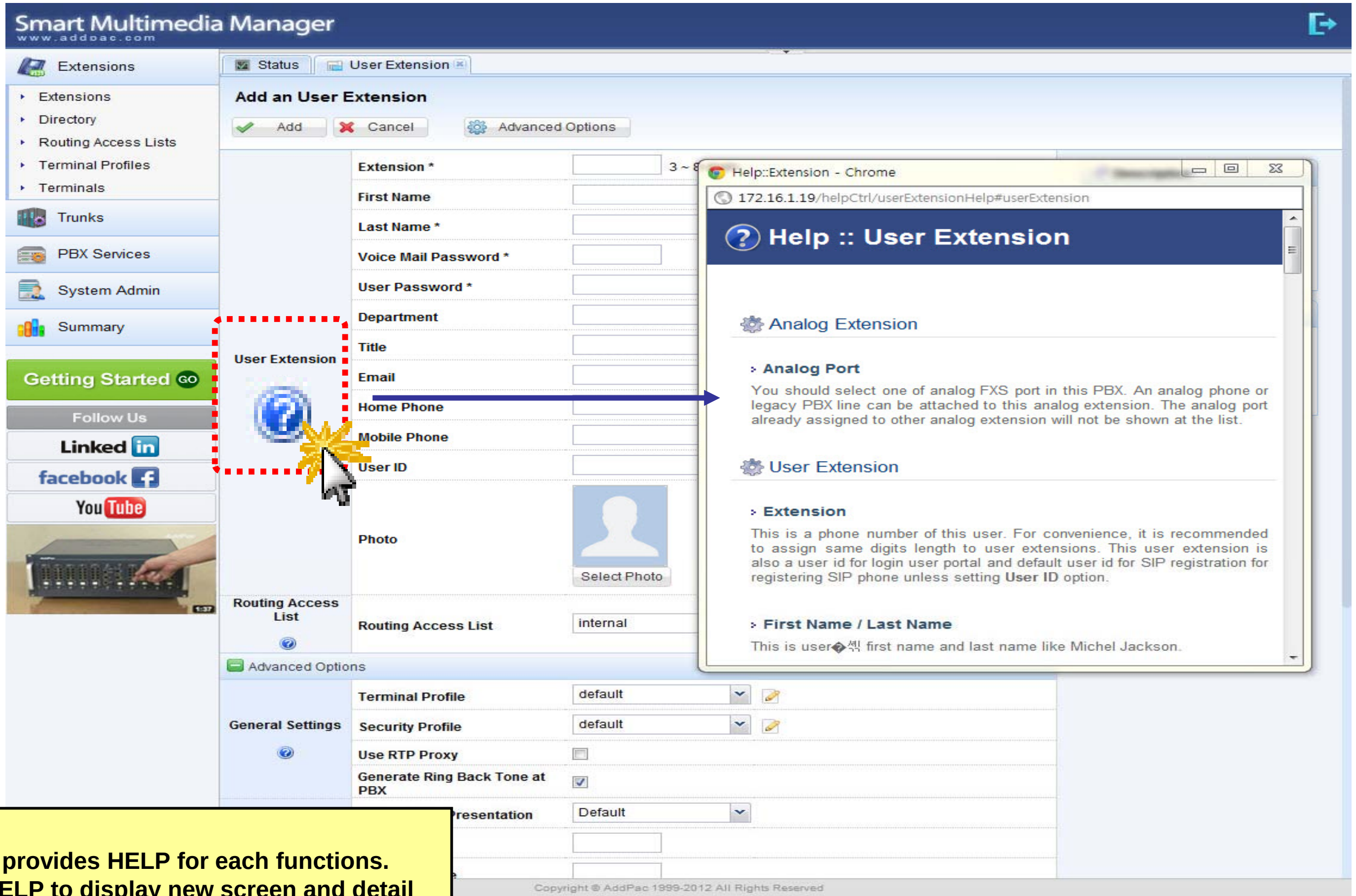
Login

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



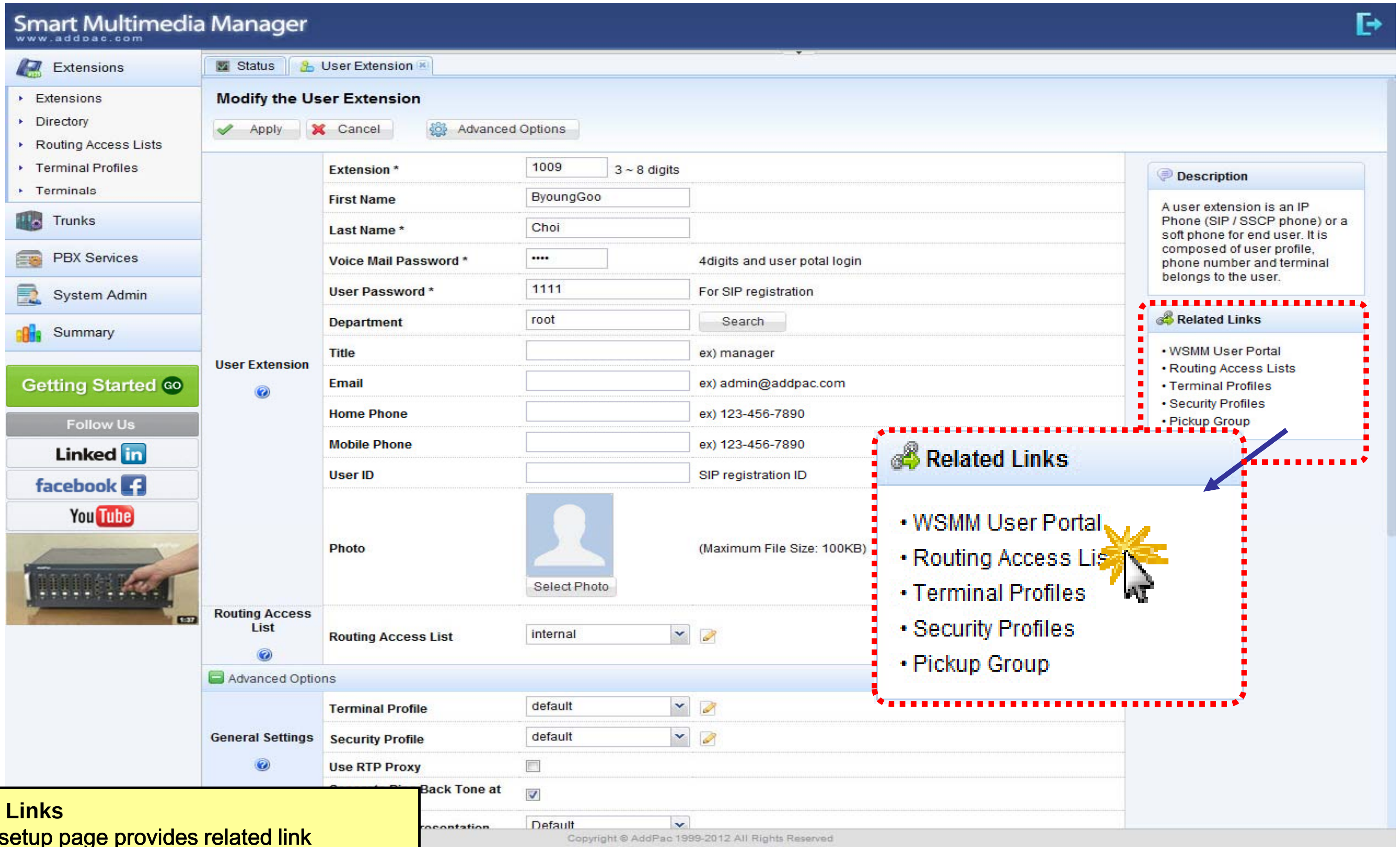
Administrator Authentication
Enter administrator ID and password to complete authentication by clicking login Smart Multimedia Manager.

Help



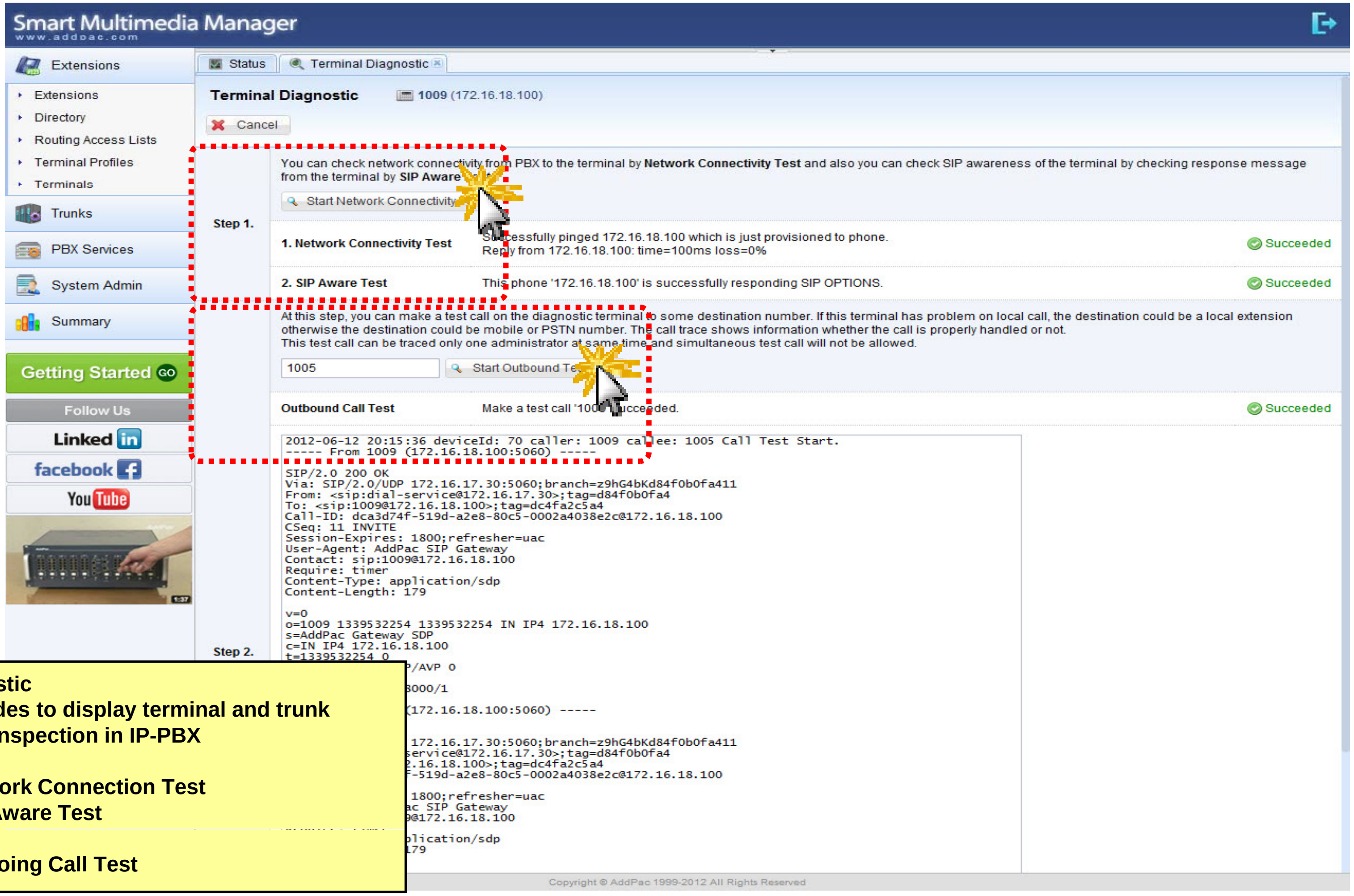
HELP
WSMM provides HELP for each functions.
Click HELP to display new screen and detail description of setup is clearly explained in homepage.

Related Links



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

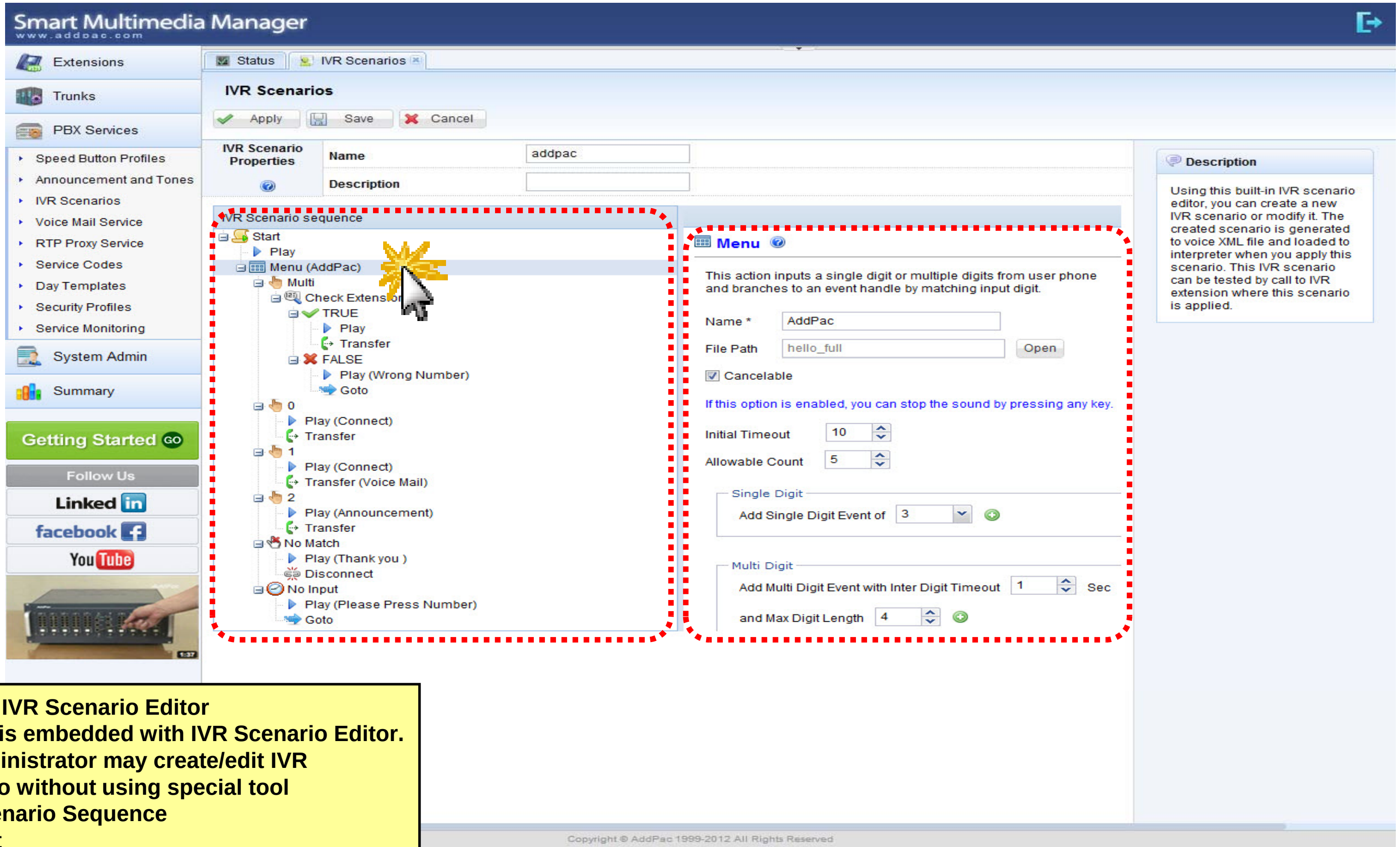
Diagnostic



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

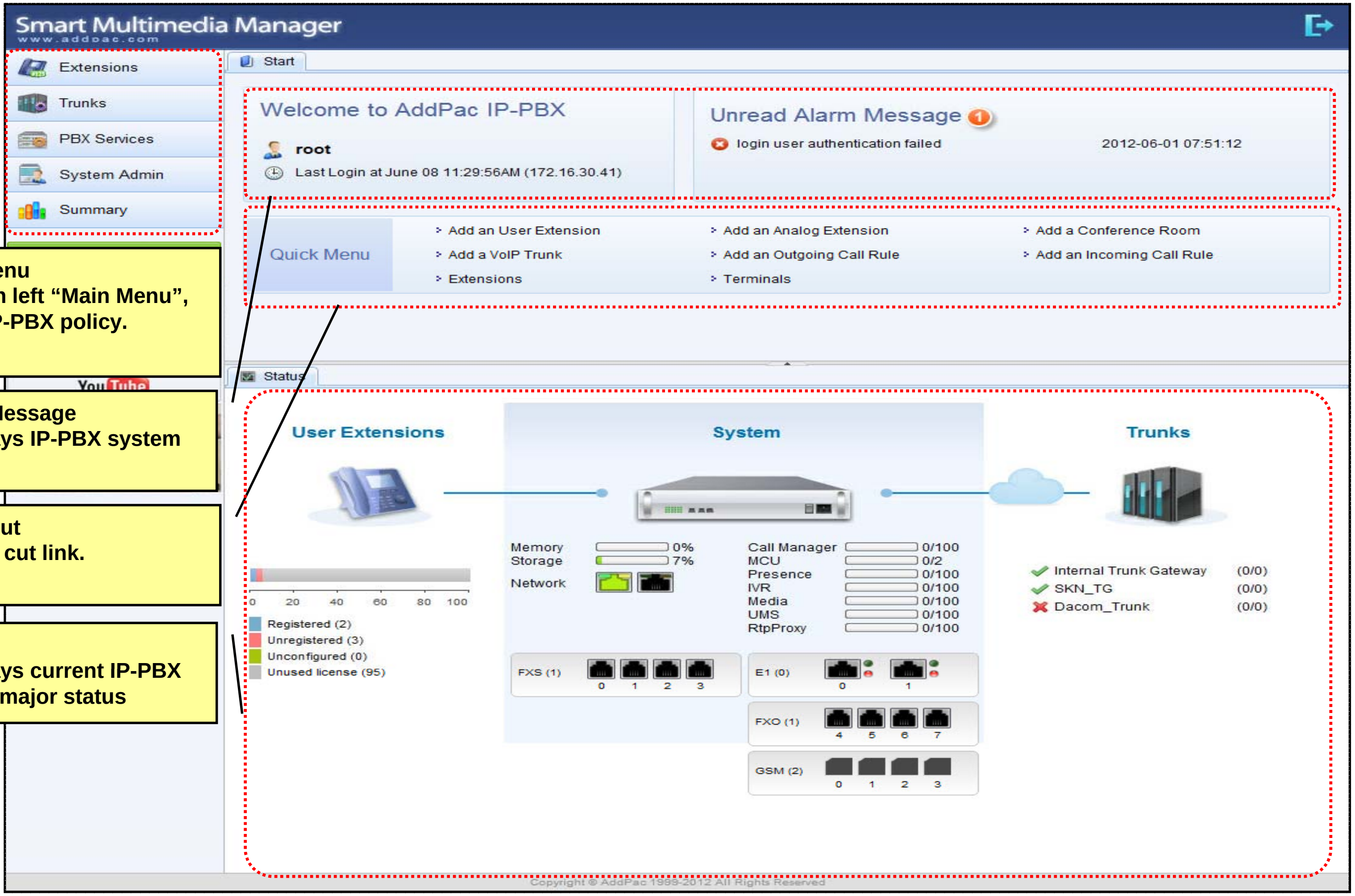
Step 2.
- **Outgoing Call Test**

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



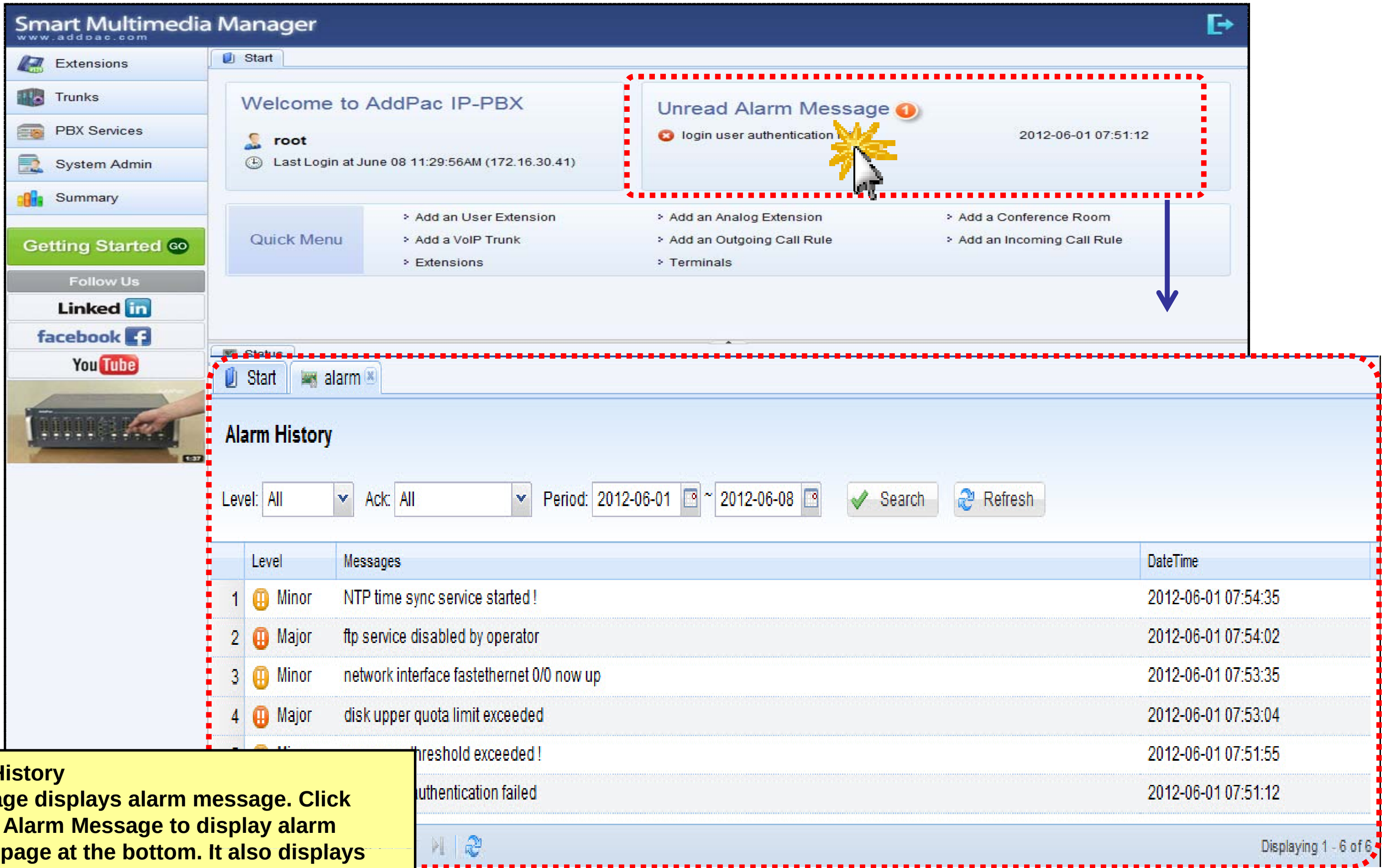
Main Menu
Through left "Main Menu", setup IP-PBX policy.

Alarm Message
It displays IP-PBX system errors

Short Cut
A short cut link.

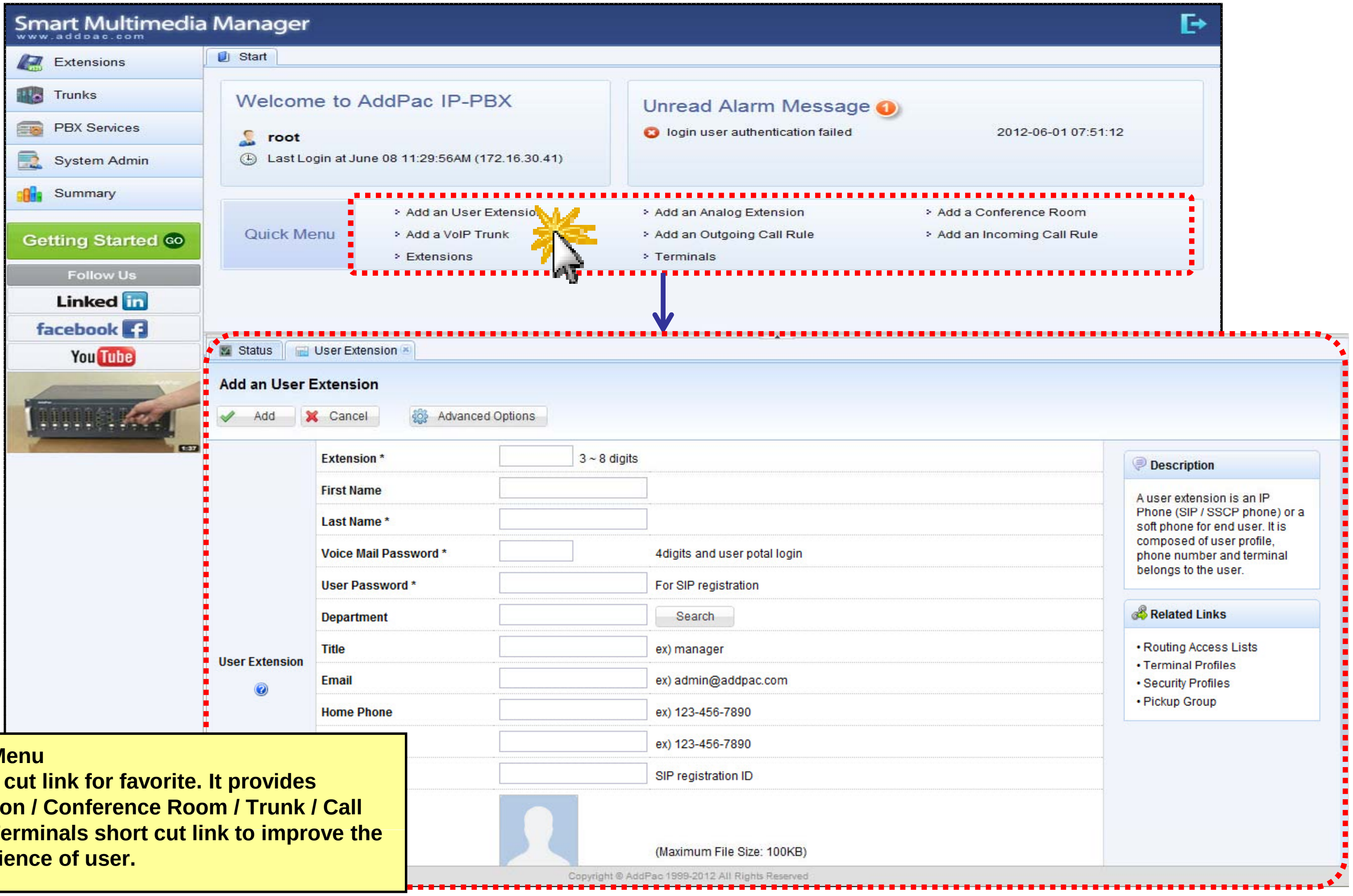
Status
It displays current IP-PBX system major status

Main - Alarm History



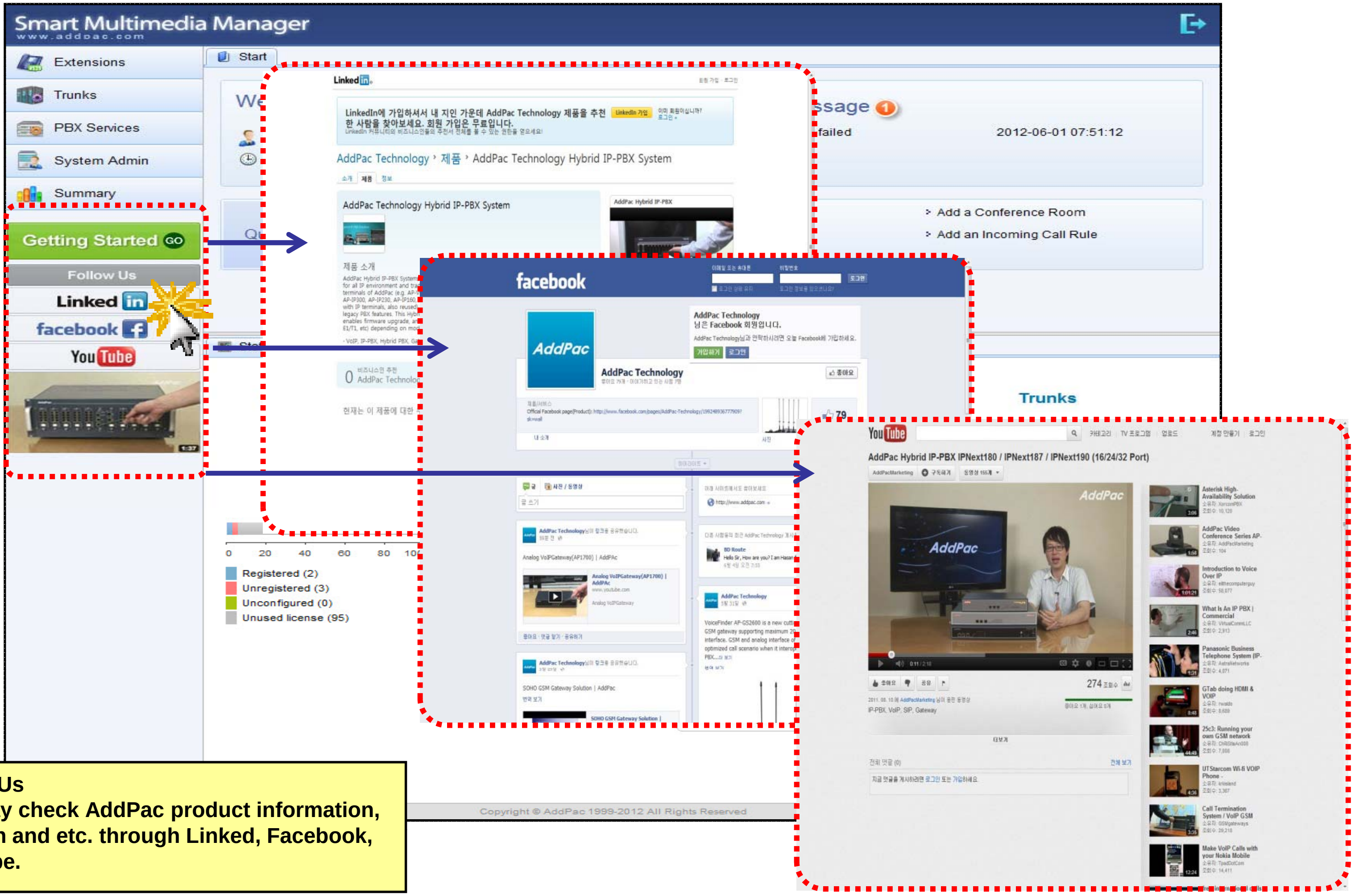
Alarm History
Main page displays alarm message. Click Unread Alarm Message to display alarm history page at the bottom. It also displays IP-PBX system errors.

Main – Quick Menu



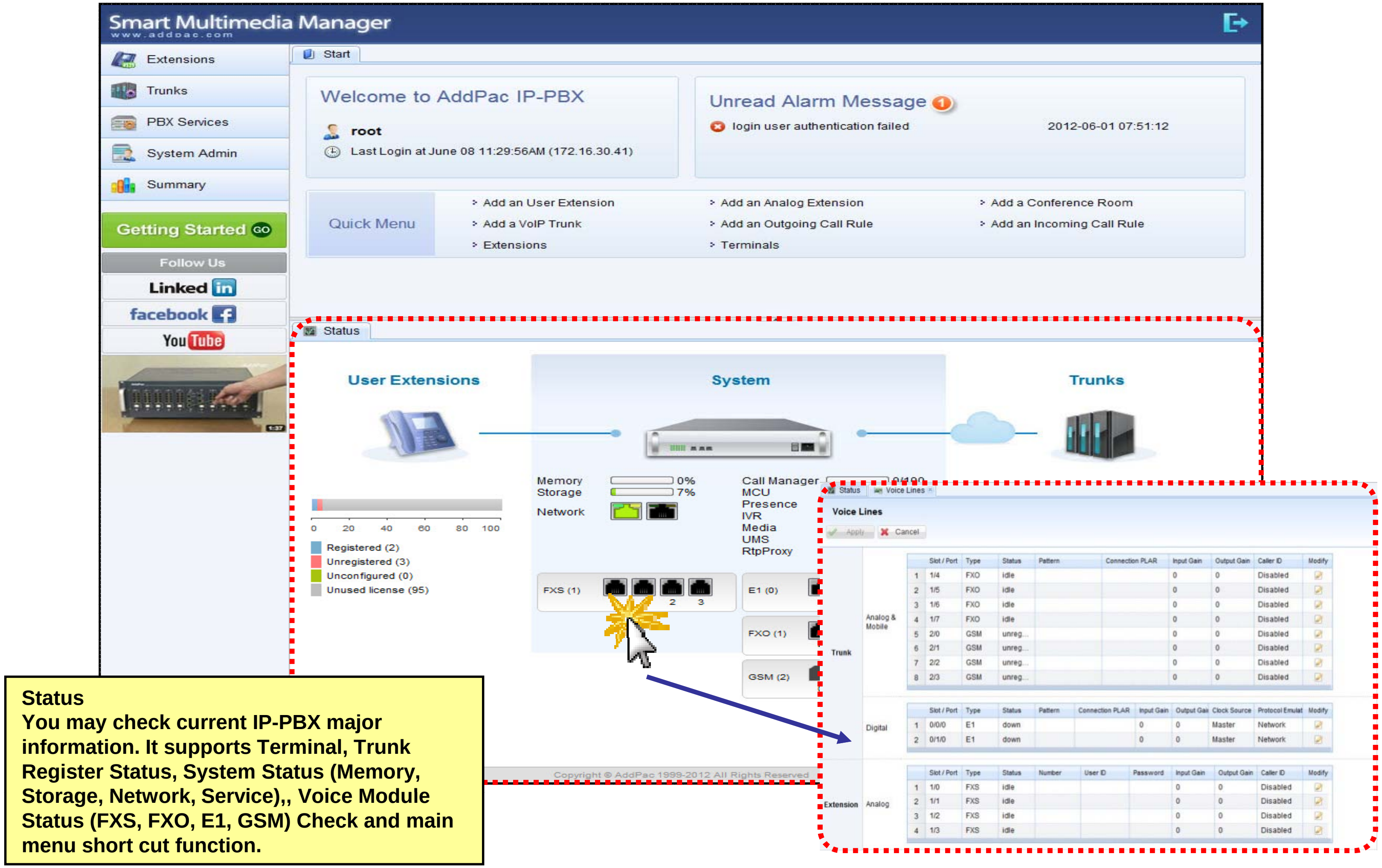
Quick Menu
A short cut link for favorite. It provides Extension / Conference Room / Trunk / Call Rule / Terminals short cut link to improve the convenience of user.

Main – Follow Us



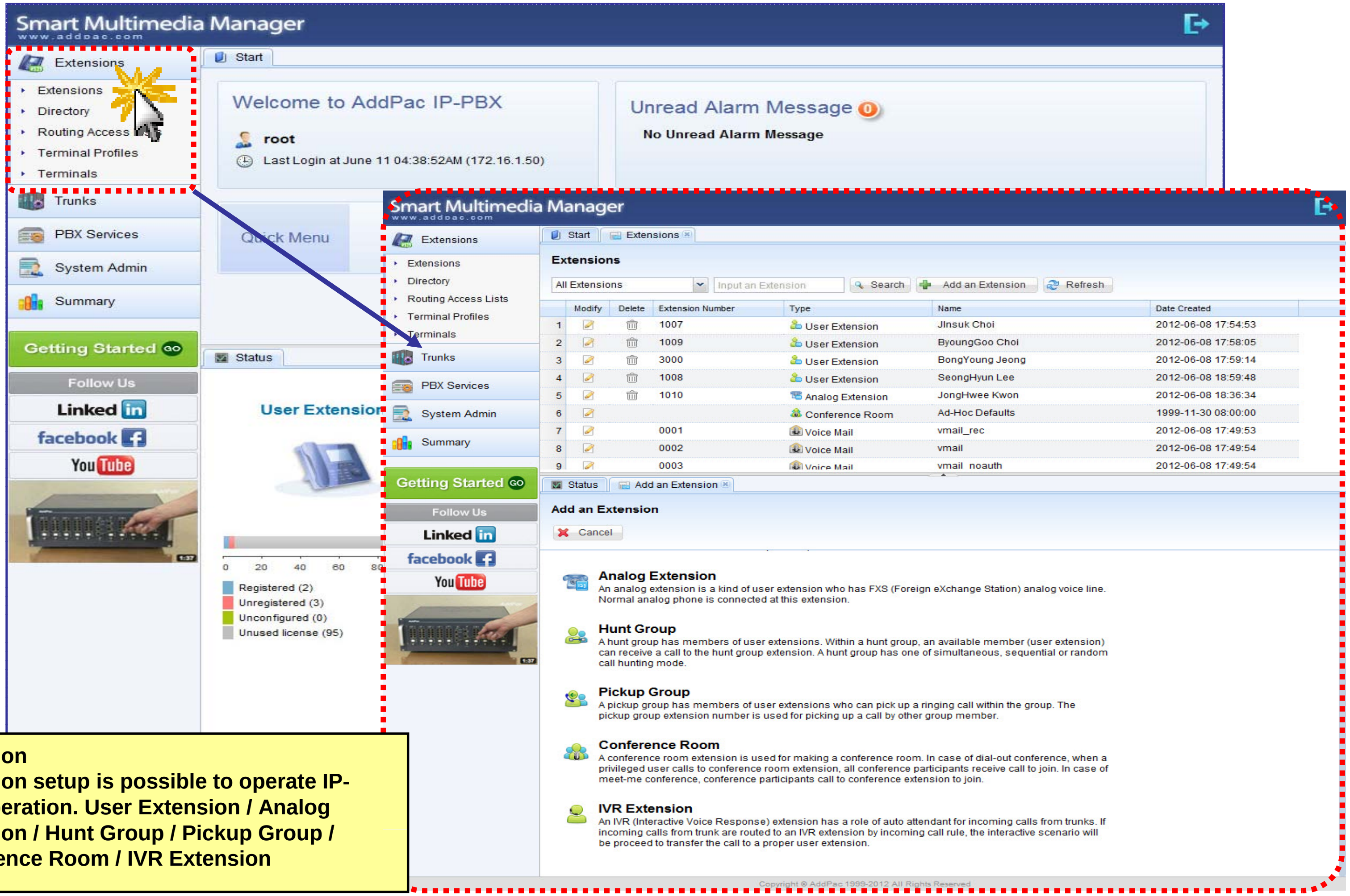
Follow Us
You may check AddPac product information, solution and etc. through Linked, Facebook, YouTube.

Main – Status Monitoring



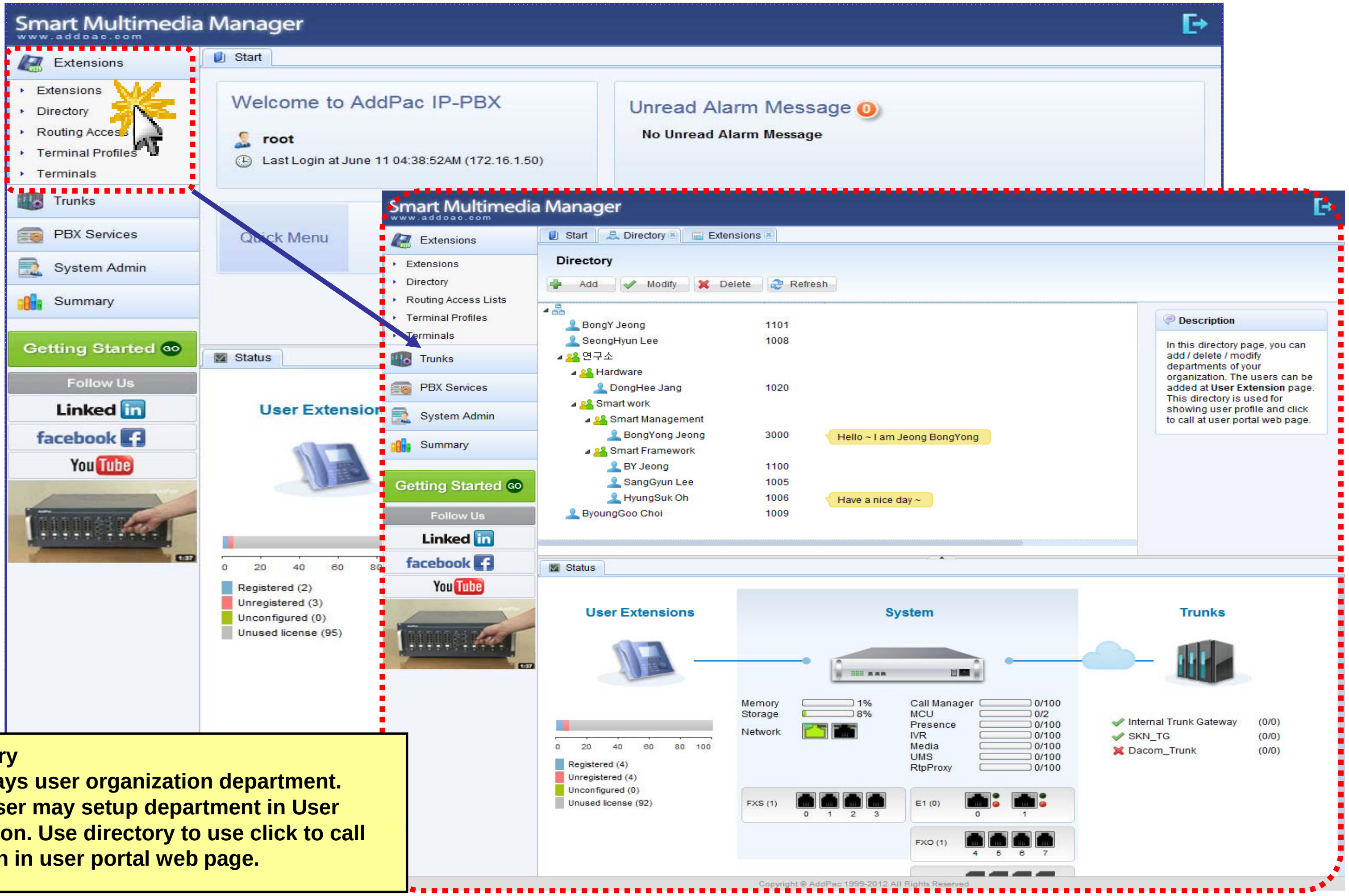
Status
You may check current IP-PBX major information. It supports Terminal, Trunk Register Status, System Status (Memory, Storage, Network, Service),, Voice Module Status (FXS, FXO, E1, GSM) Check and main menu short cut function.

Extension - Extensions



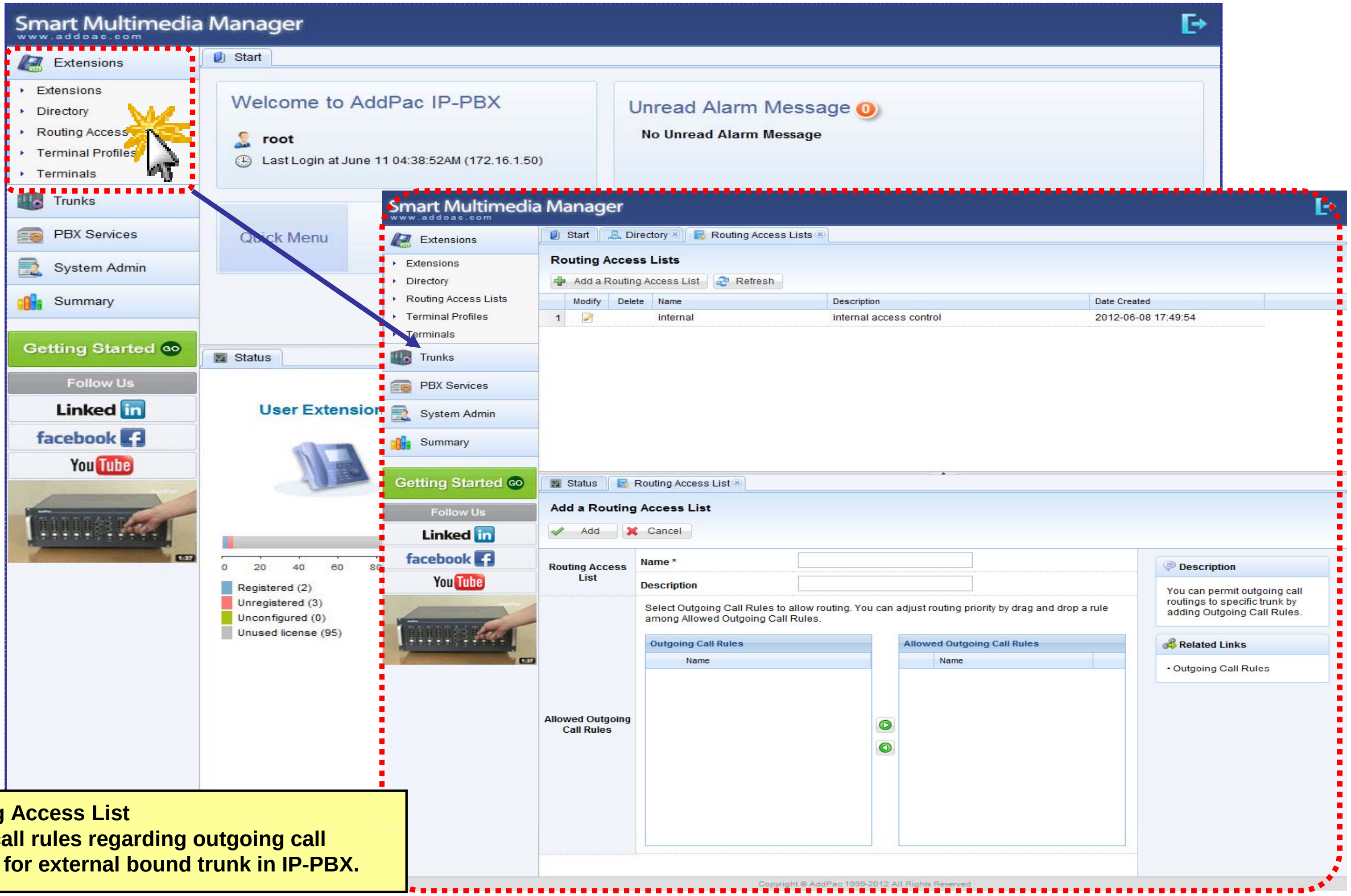
Extension
Extension setup is possible to operate IP-PBX operation. User Extension / Analog Extension / Hunt Group / Pickup Group / Conference Room / IVR Extension

Extension - Directory



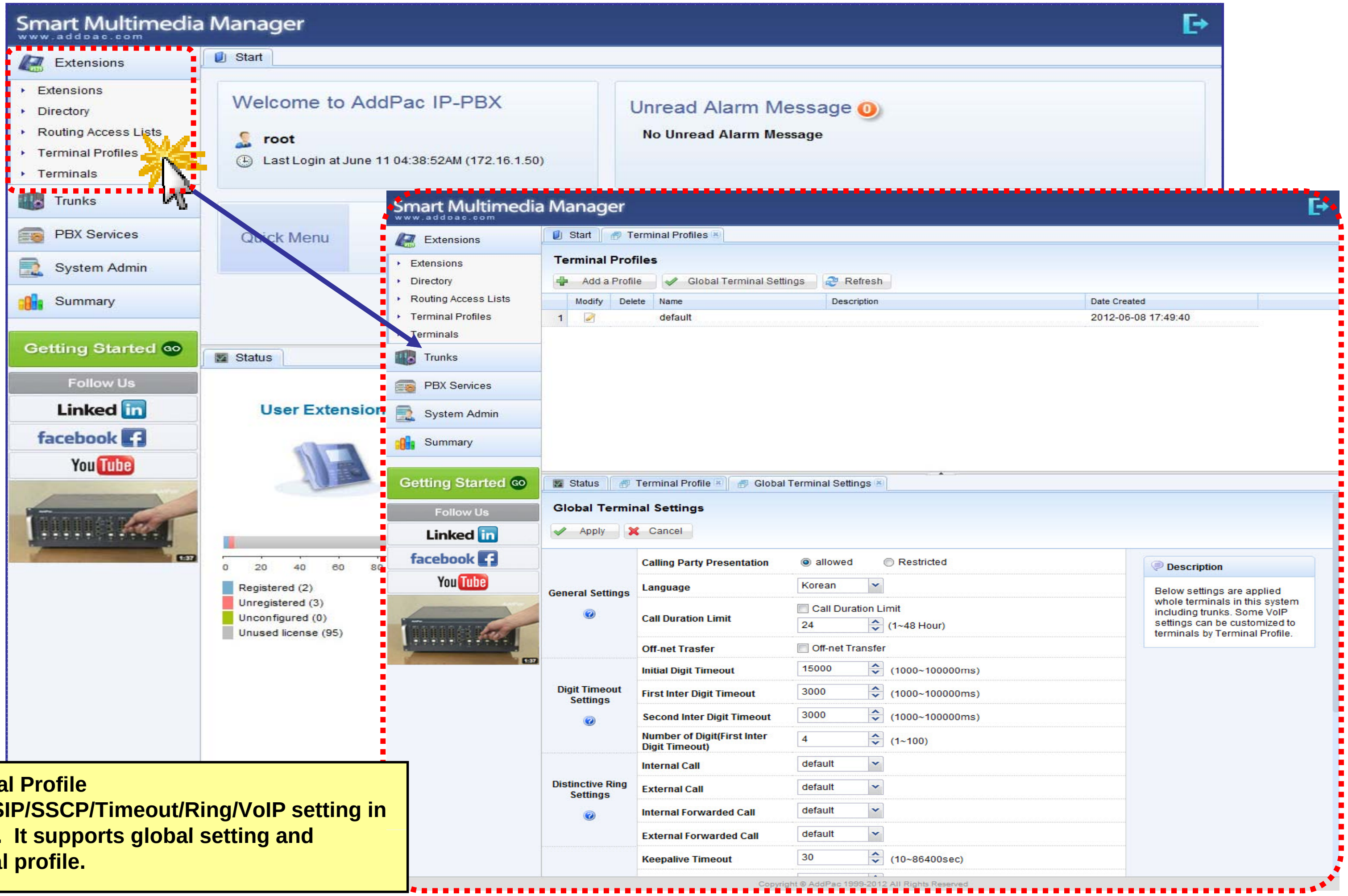
Directory
It displays user organization department.
Each user may setup department in User Extension. Use directory to use click to call function in user portal web page.

Extension - Routing Access List



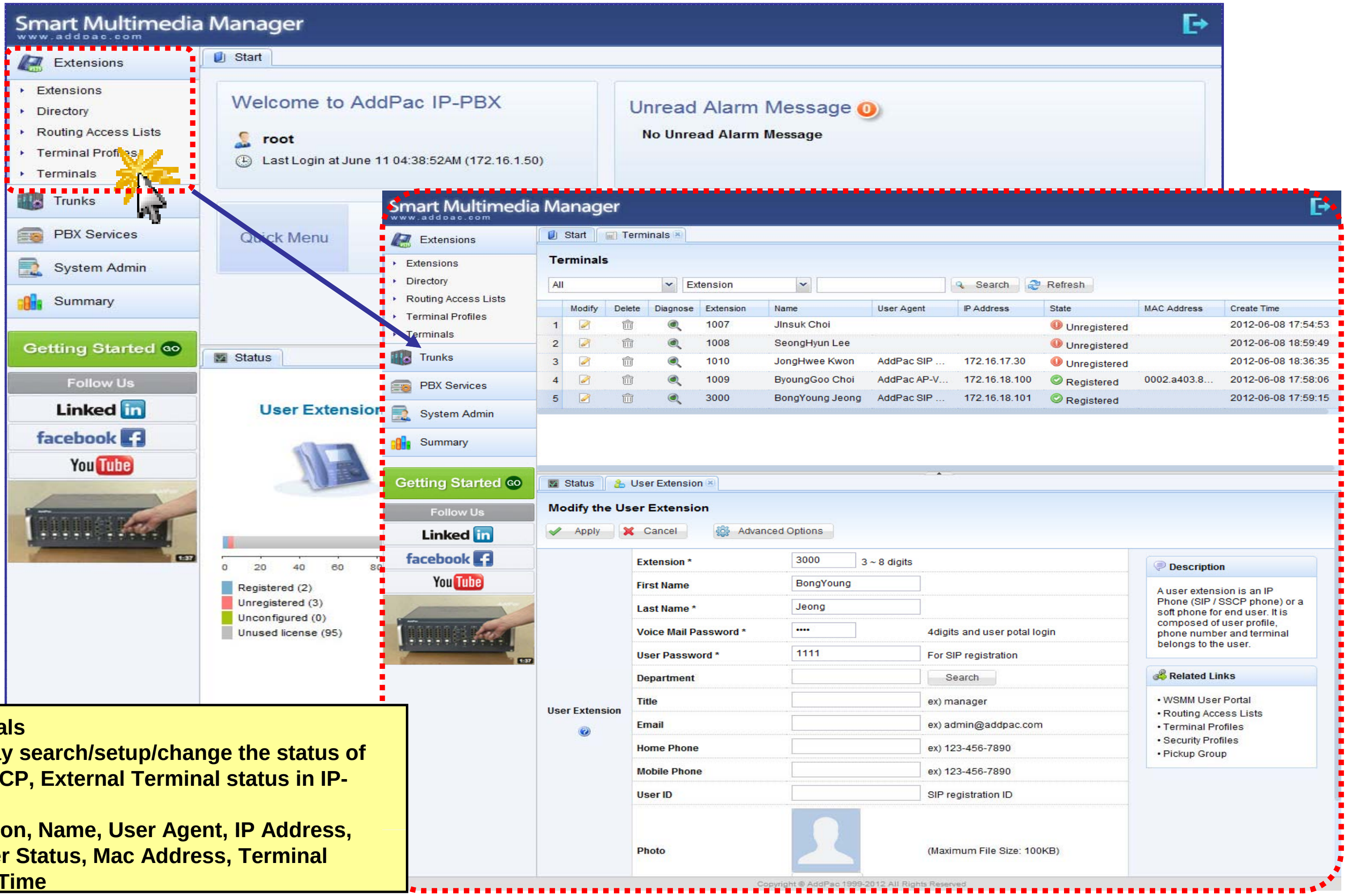
Routing Access List
Apply call rules regarding outgoing call routing for external bound trunk in IP-PBX.

Extension - Terminal Profile



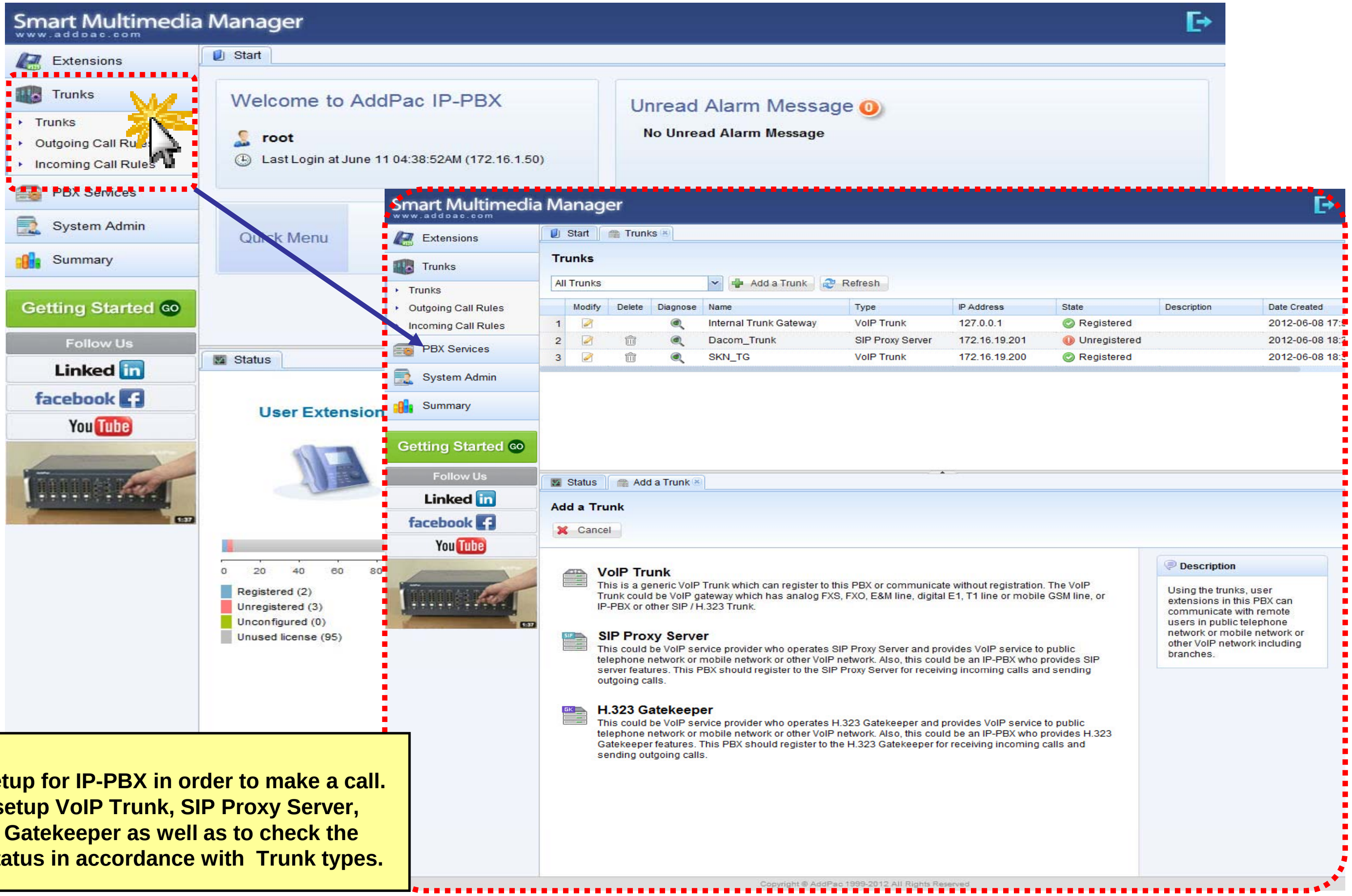
Terminal Profile
Setup SIP/SSCP/Timeout/Ring/VoIP setting in IP-PBX. It supports global setting and terminal profile.

Extension - Terminals



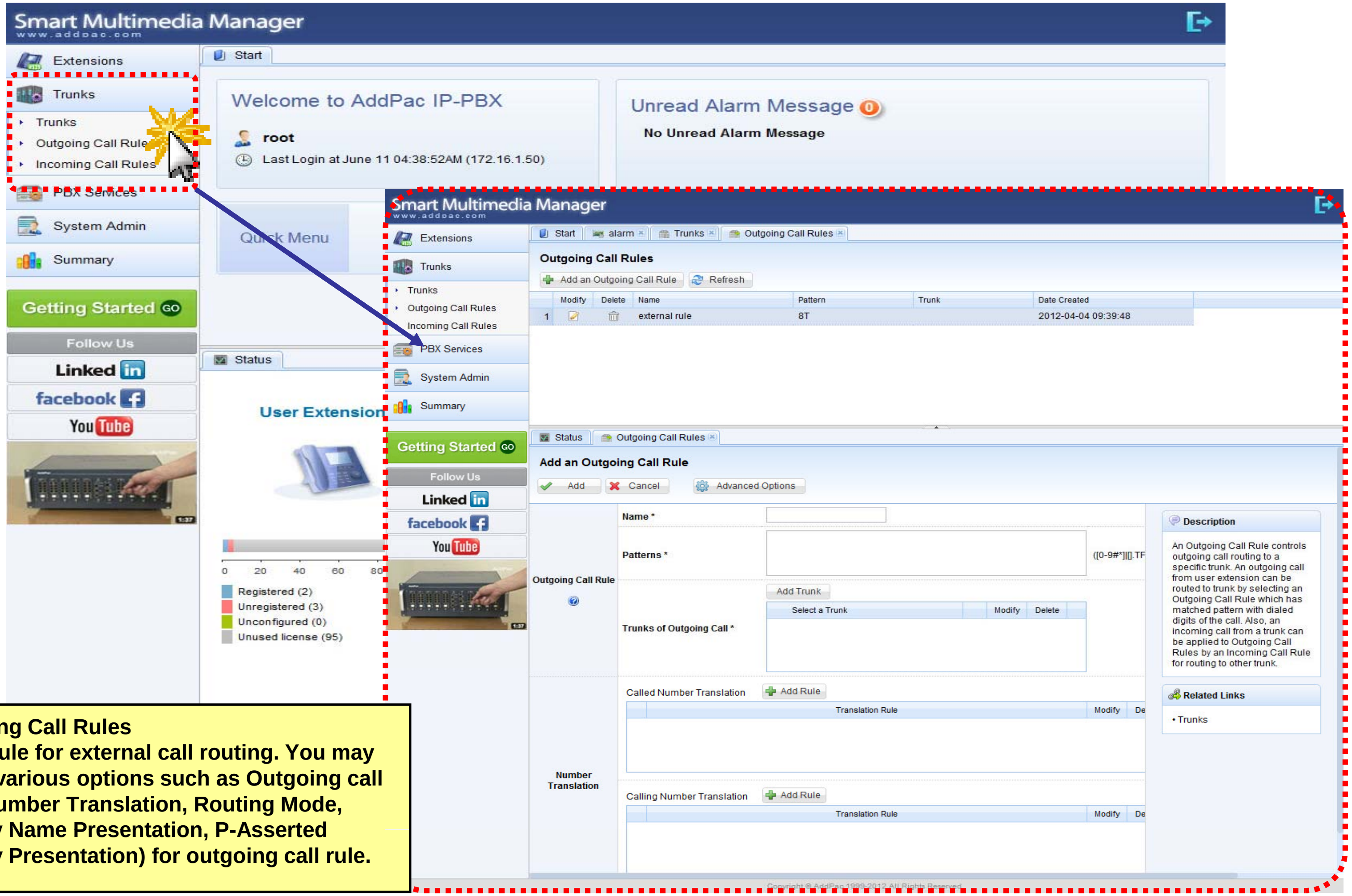
Terminals
You may search/setup/change the status of SIP, SSCP, External Terminal status in IP-PBX.
Extension, Name, User Agent, IP Address, Register Status, Mac Address, Terminal Create Time

Trunk - Trunks



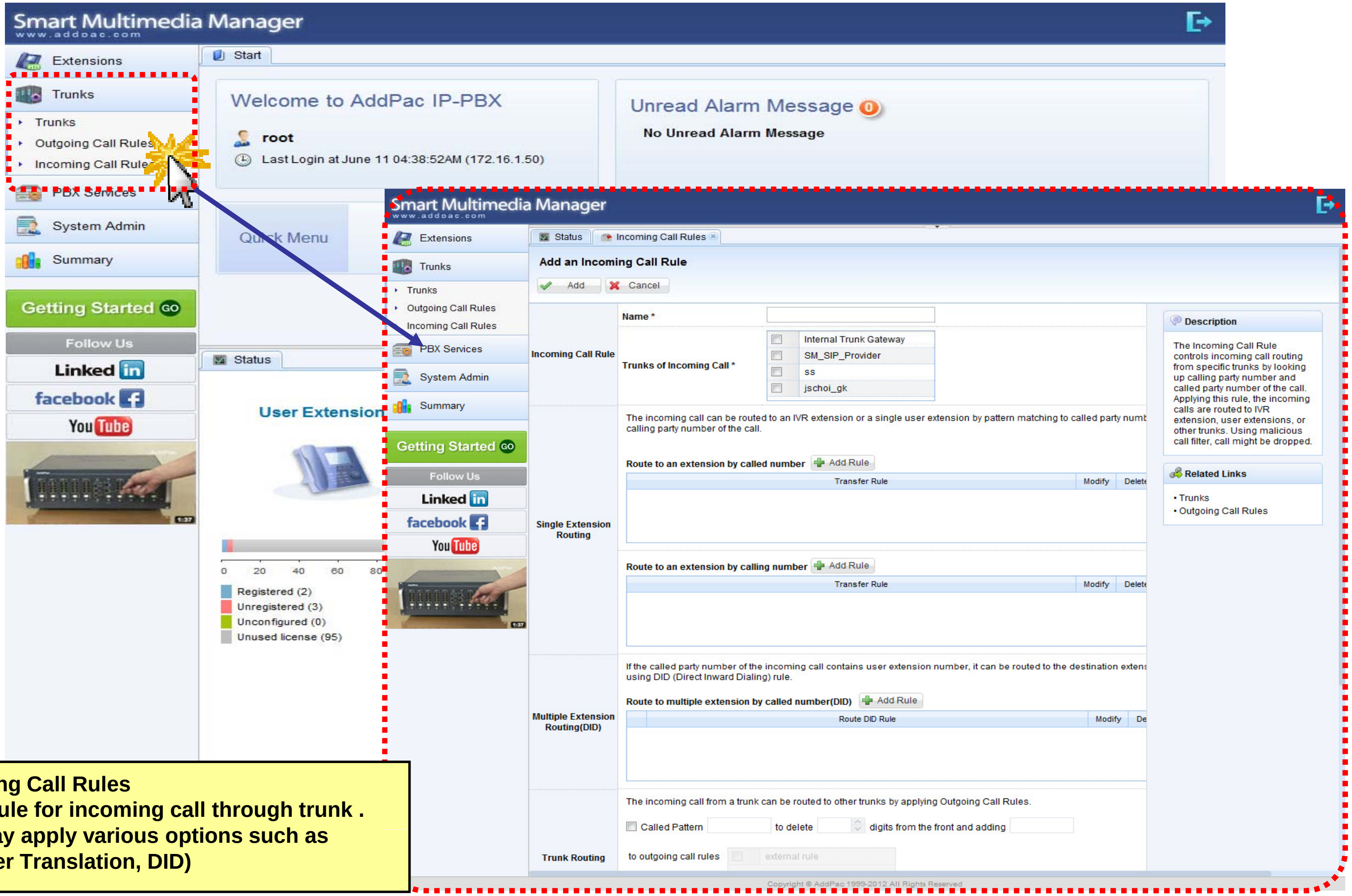
Trunk
A trunk setup for IP-PBX in order to make a call. You may setup VoIP Trunk, SIP Proxy Server, and H.323 Gatekeeper as well as to check the register status in accordance with Trunk types.

Trunk - Outgoing Call Rules



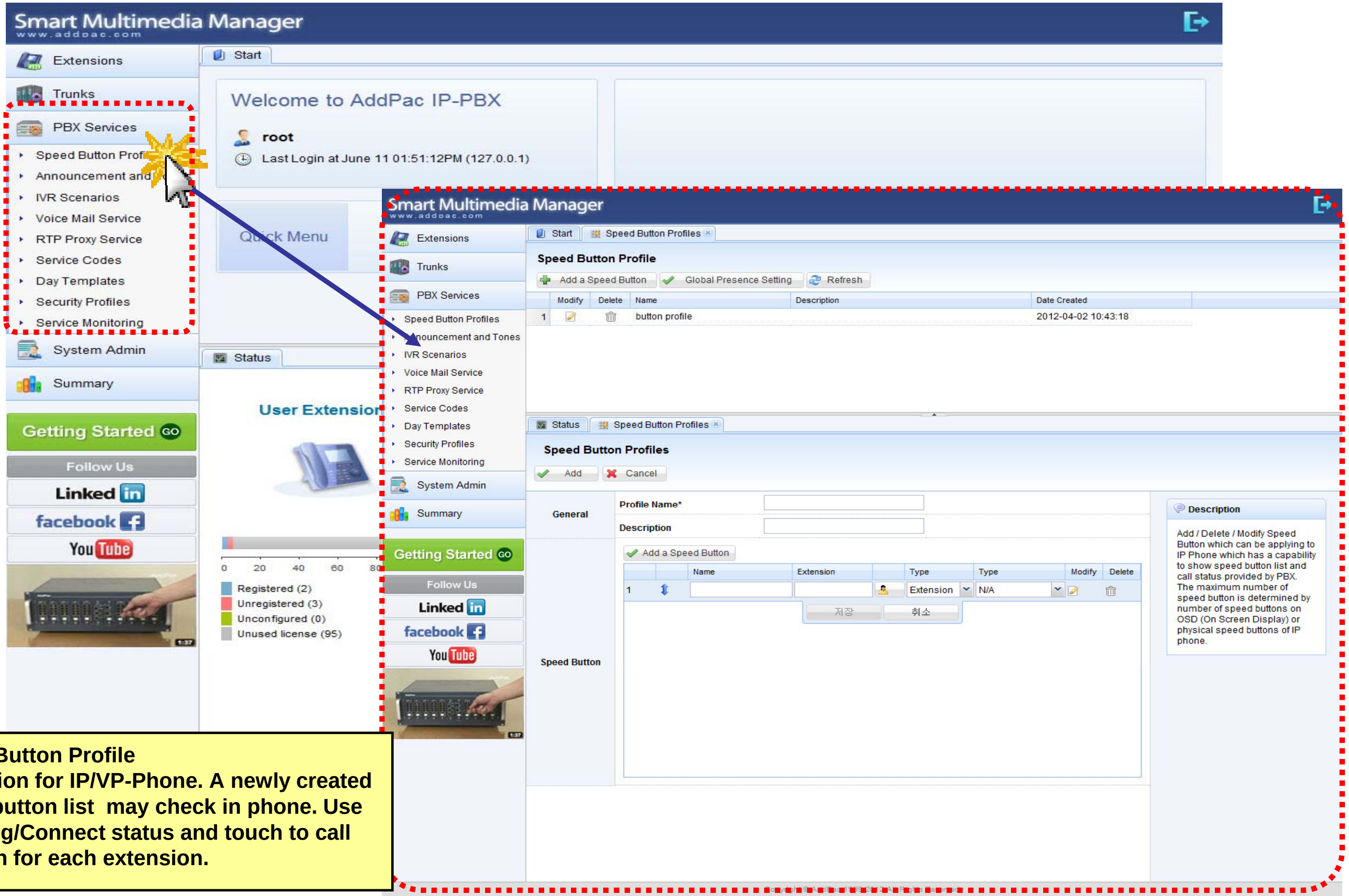
Outgoing Call Rules
A call rule for external call routing. You may apply various options such as Outgoing call rule (Number Translation, Routing Mode, Display Name Presentation, P-Asserted Identity Presentation) for outgoing call rule.

Trunk - Incoming Call Rules



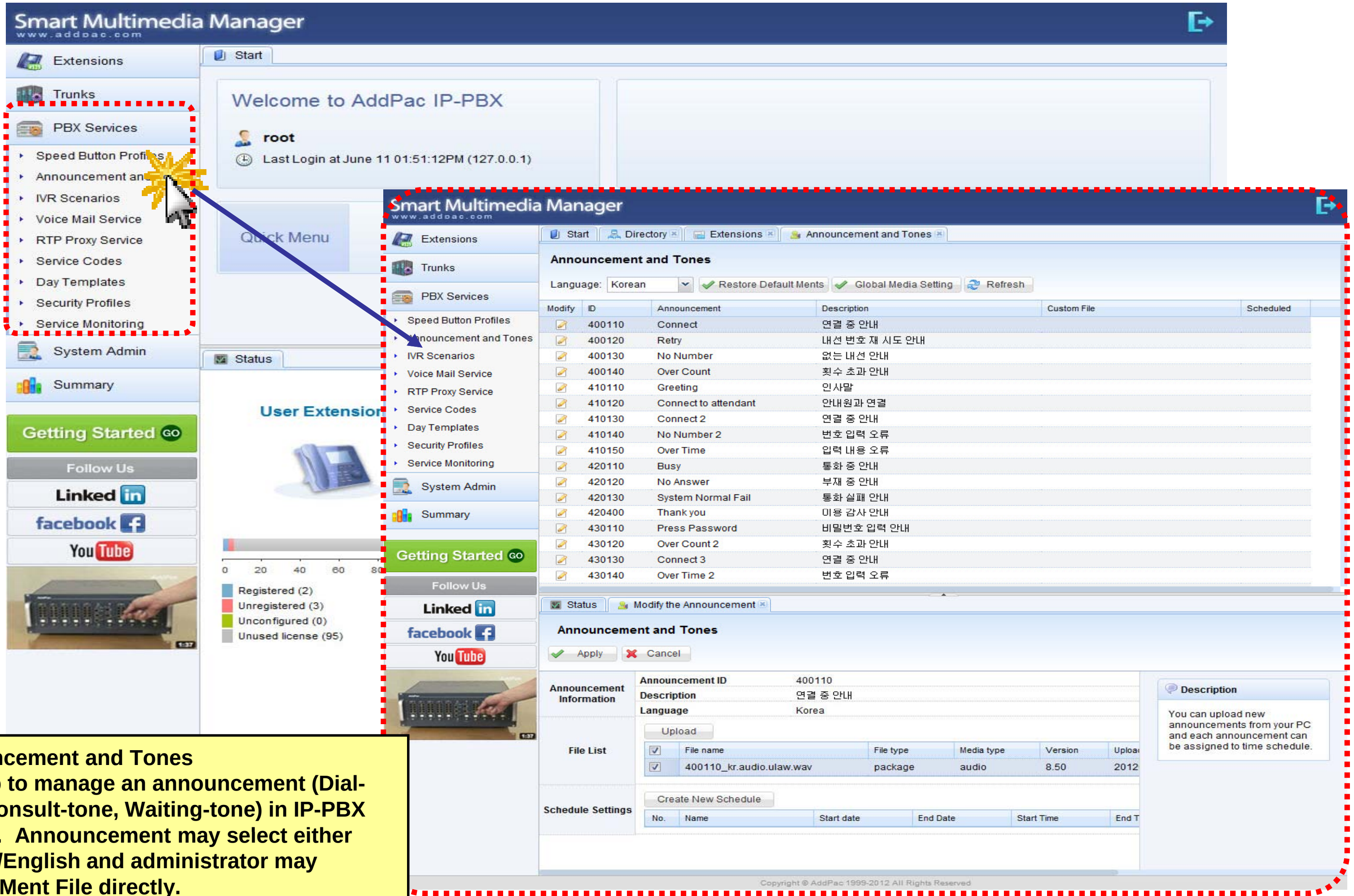
Incoming Call Rules
A call rule for incoming call through trunk .
You may apply various options such as
(Number Translation, DID)

PBX Service - Speed Button Profiles



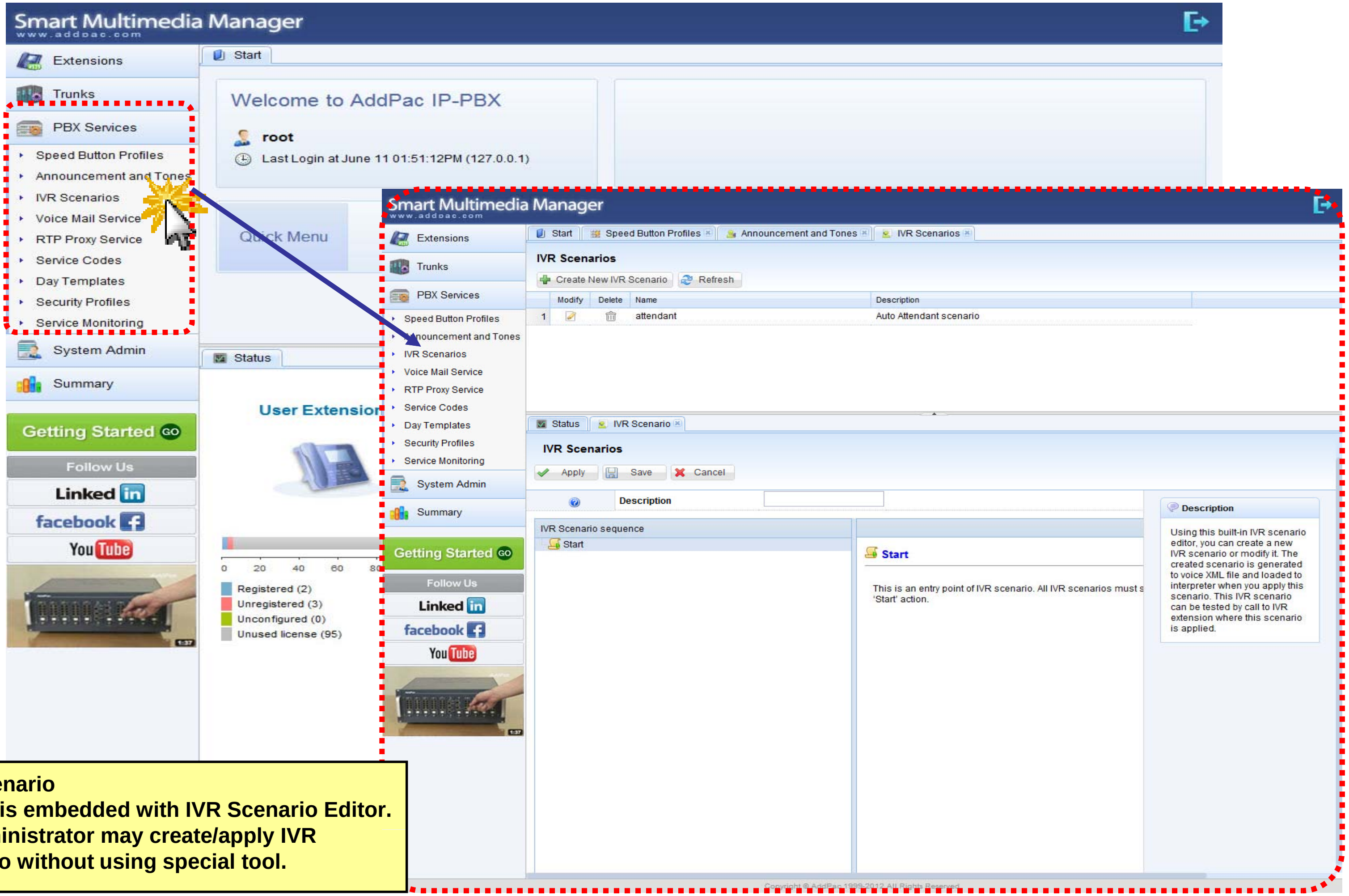
Speed Button Profile
A function for IP/VP-Phone. A newly created speed button list may check in phone. Use idle/Ring/Connect status and touch to call function for each extension.

PBX Service - Announcement and Tones



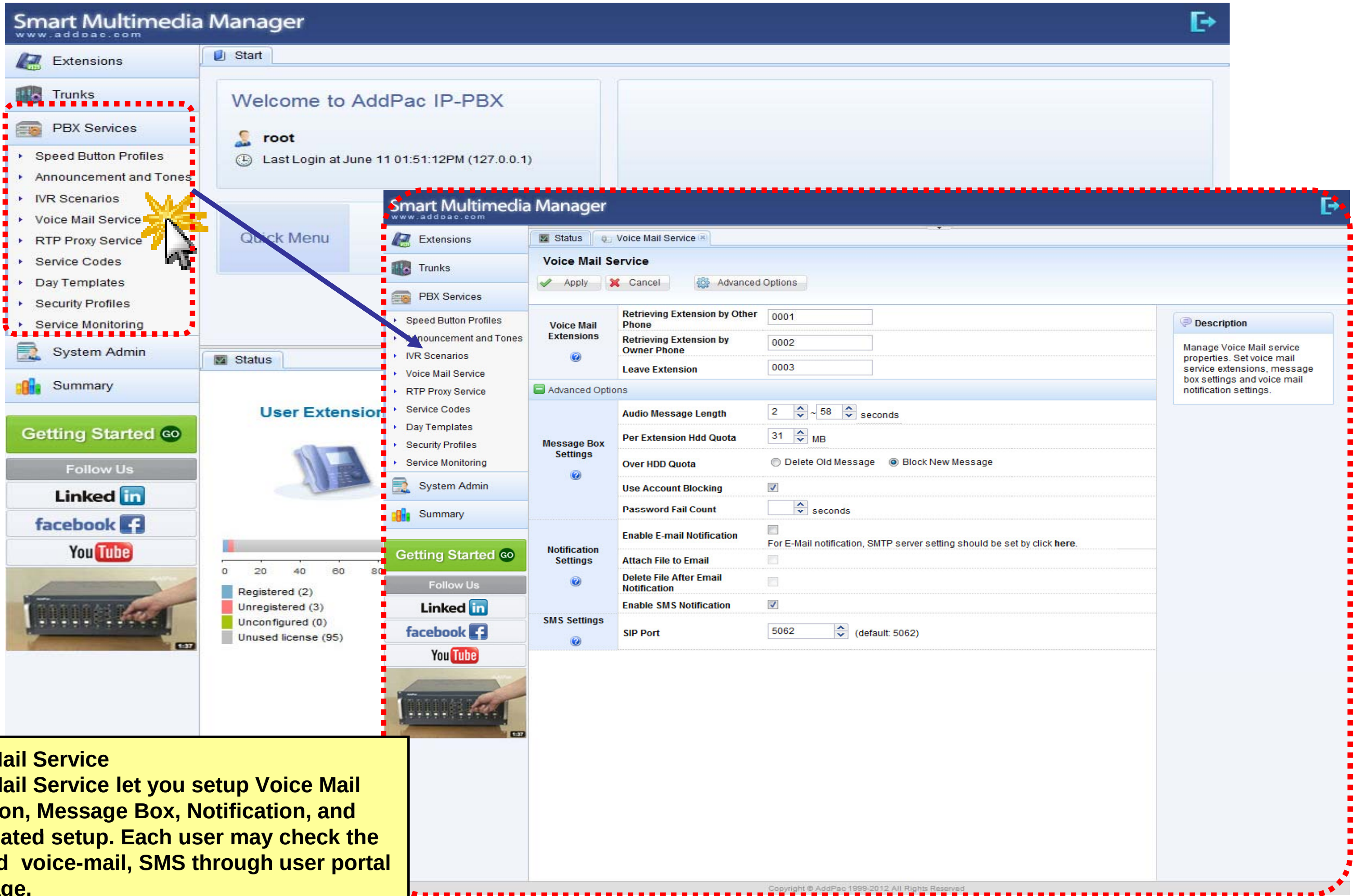
Announcement and Tones
A setup to manage an announcement (Dial-tone, Consult-tone, Waiting-tone) in IP-PBX service. Announcement may select either Korean/English and administrator may upload Ment File directly.

PBX Service - IVR Scenarios



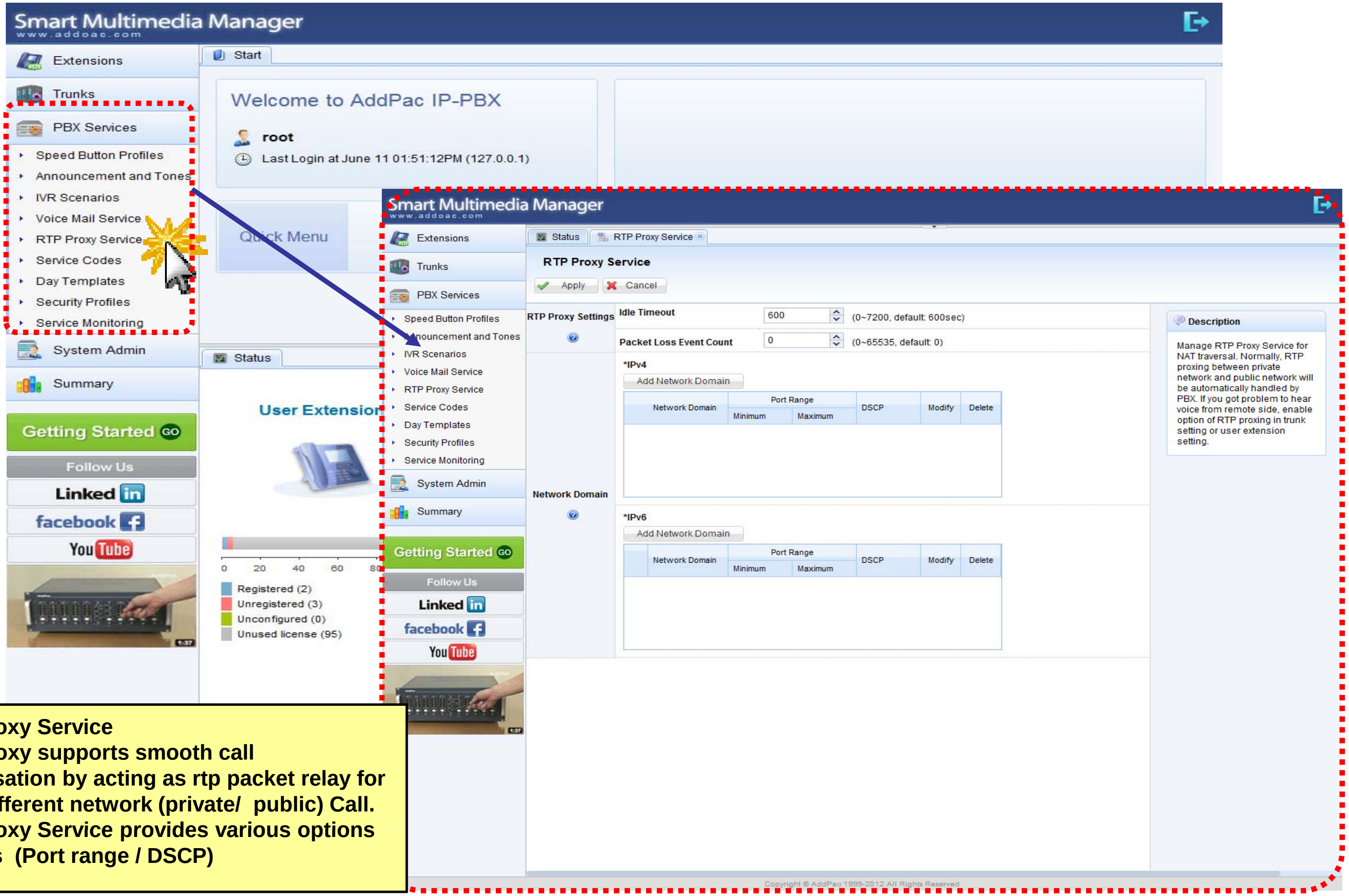
IVR Scenario
WSMM is embedded with IVR Scenario Editor. An administrator may create/apply IVR scenario without using special tool.

PBX Service - Voice Mail Services



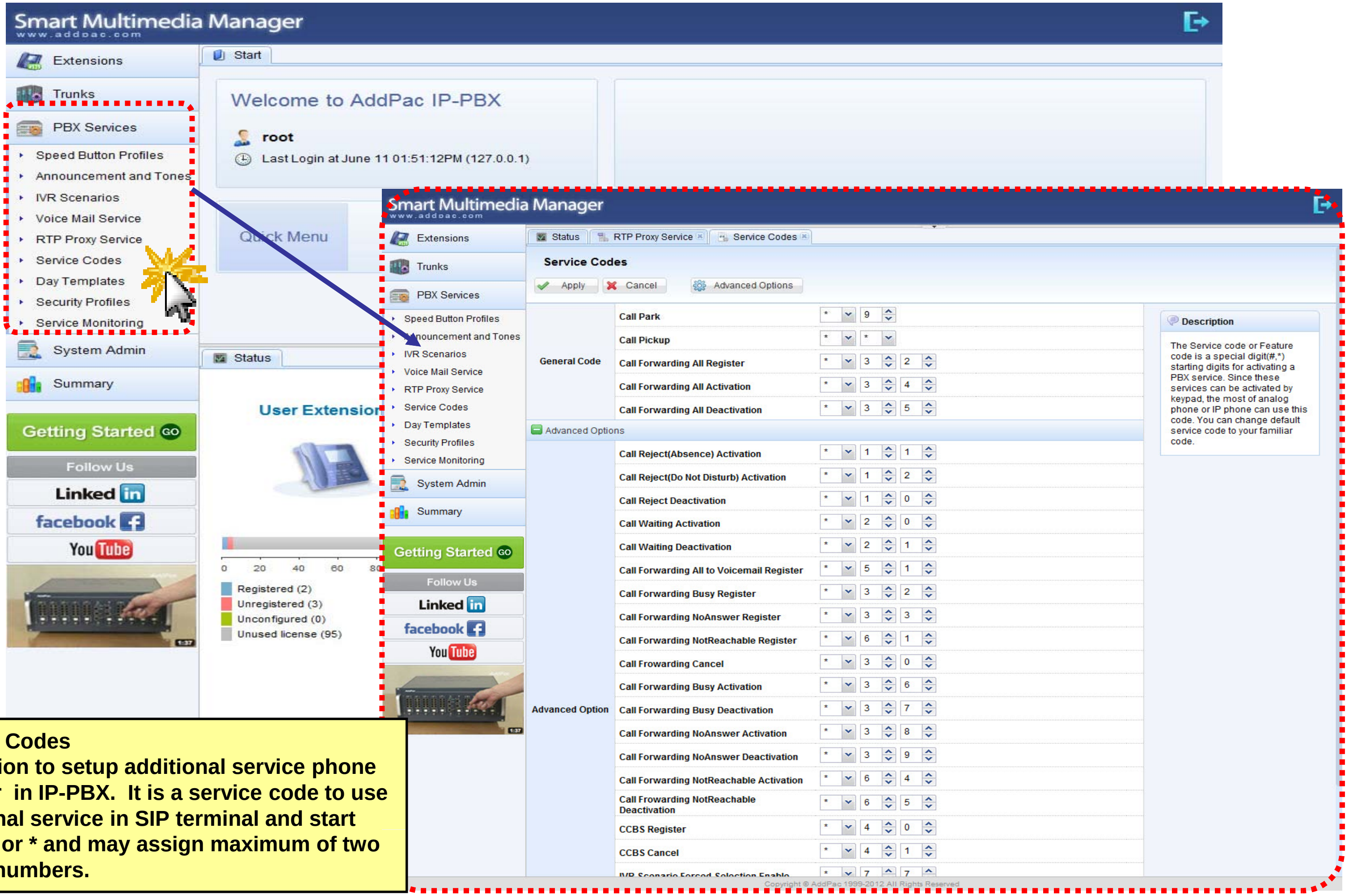
Voice Mail Service
Voice Mail Service let you setup Voice Mail Extension, Message Box, Notification, and SMS related setup. Each user may check the received voice-mail, SMS through user portal web page.

PBX Service - RTP Proxy Service



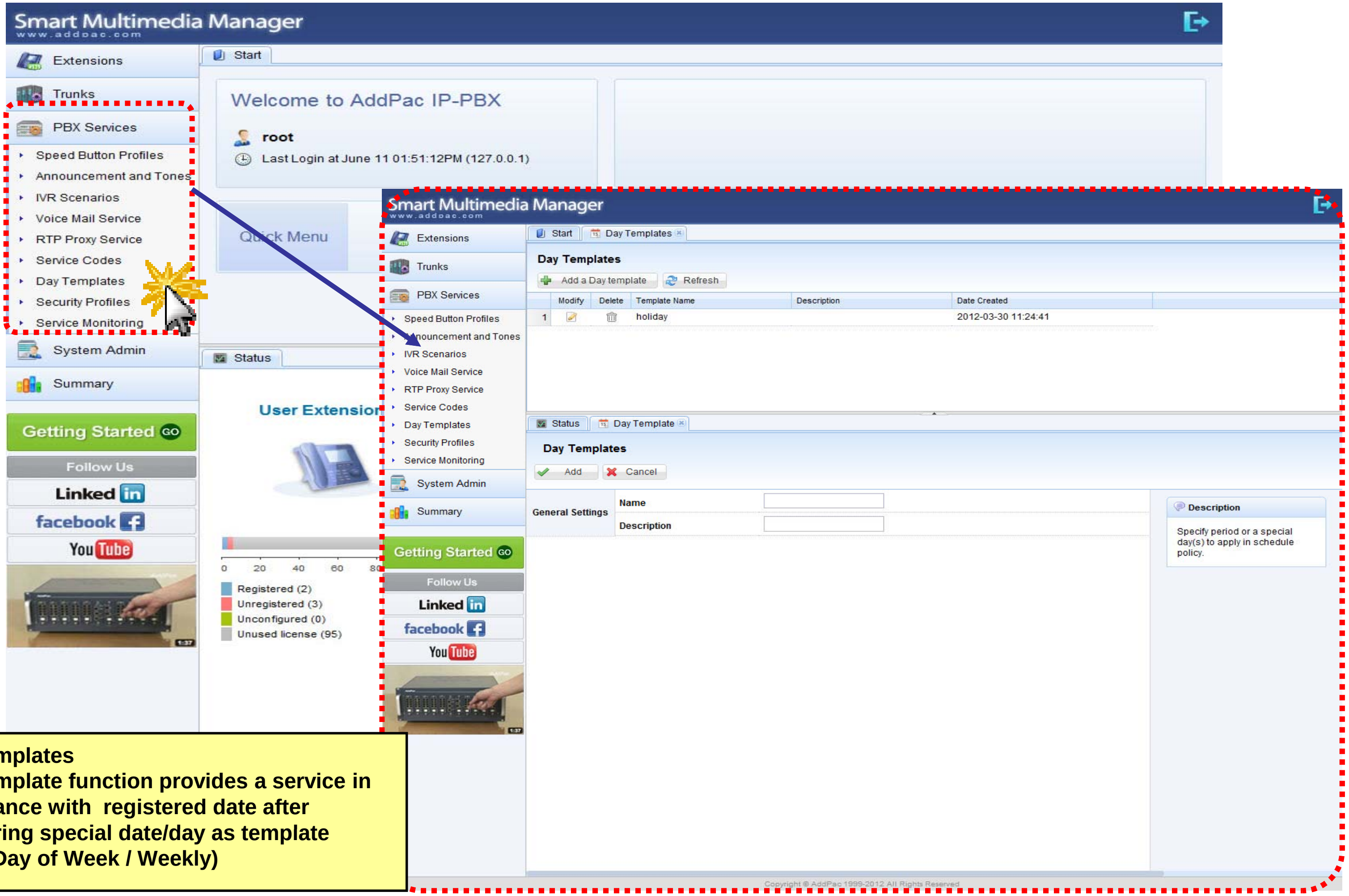
RTP Proxy Service
RTP Proxy supports smooth call conversation by acting as rtp packet relay for each different network (private/ public) Call.
RTP Proxy Service provides various options such as (Port range / DSCP)

PBX Service - Service Codes



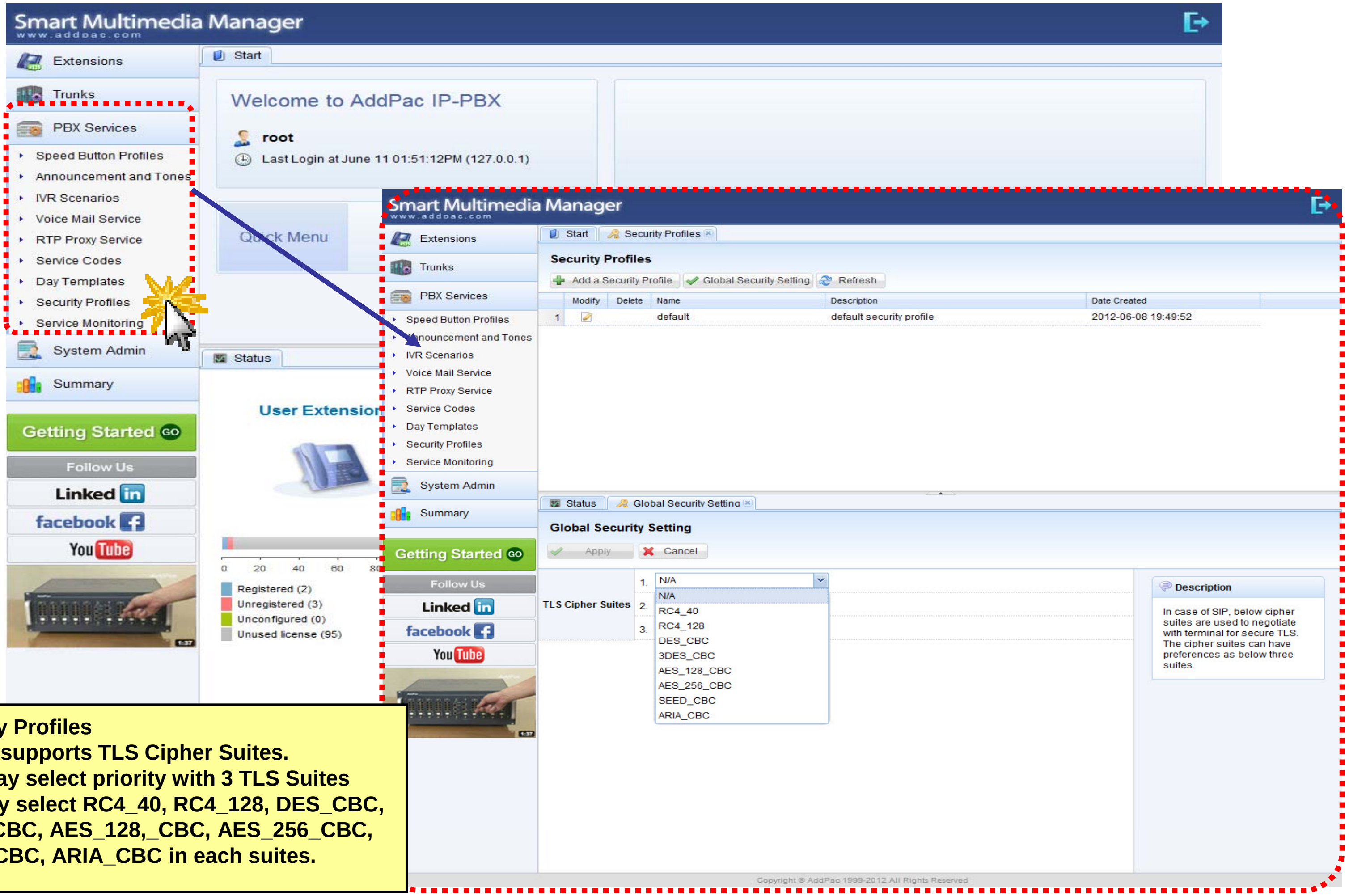
Service Codes
A function to setup additional service phone number in IP-PBX. It is a service code to use additional service in SIP terminal and start with # or * and may assign maximum of two phone numbers.

PBX Service - Day Templates



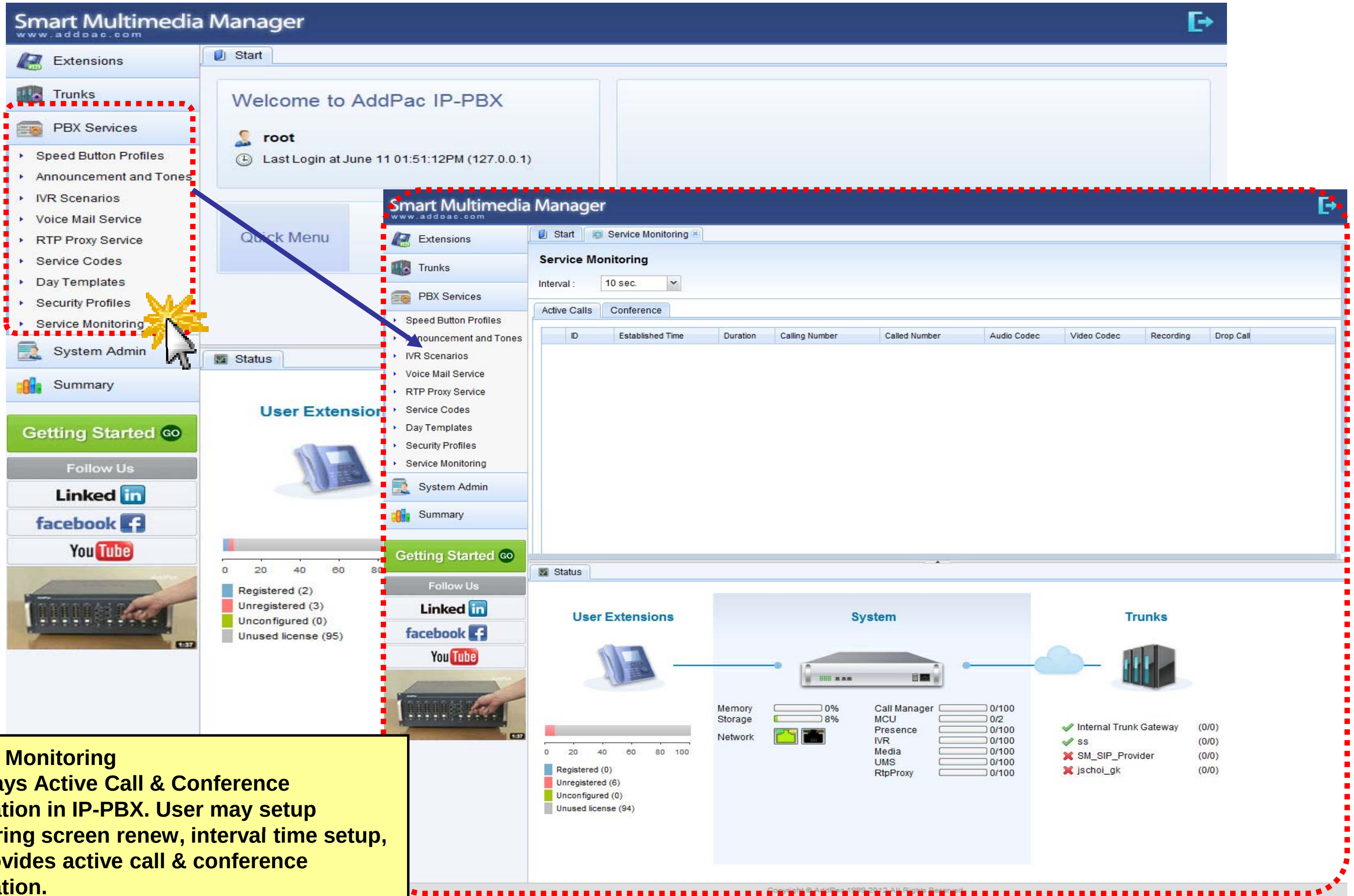
Day Templates
Day Template function provides a service in accordance with registered date after registering special date/day as template (date / Day of Week / Weekly)

PBX Service - Security Profiles



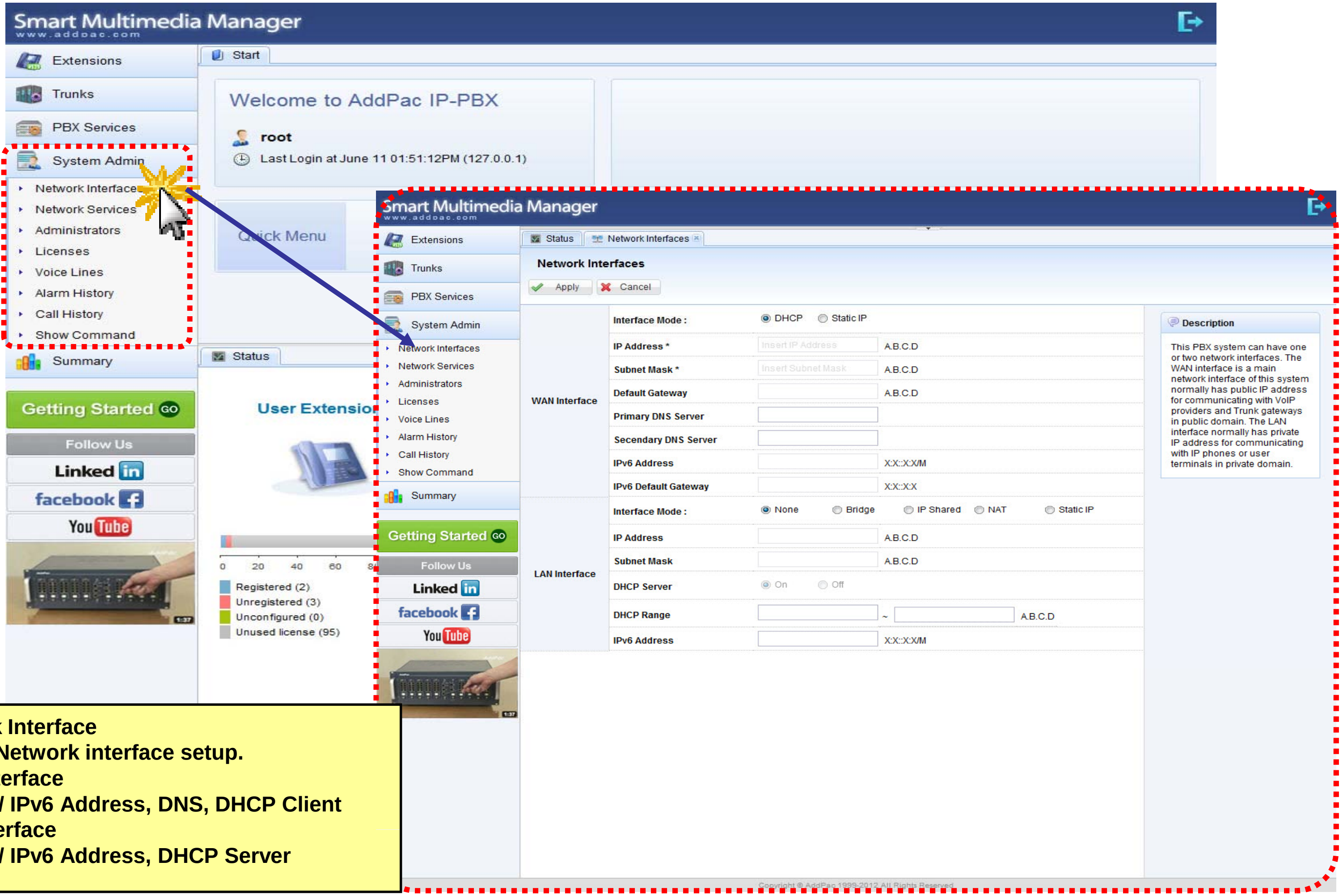
Security Profiles
IP-PBX supports TLS Cipher Suites.
User may select priority with 3 TLS Suites and may select RC4_40, RC4_128, DES_CBC, 3DES_CBC, AES_128,_CBC, AES_256_CBC, SEED_CBC, ARIA_CBC in each suites.

PBX Service - Service Monitoring



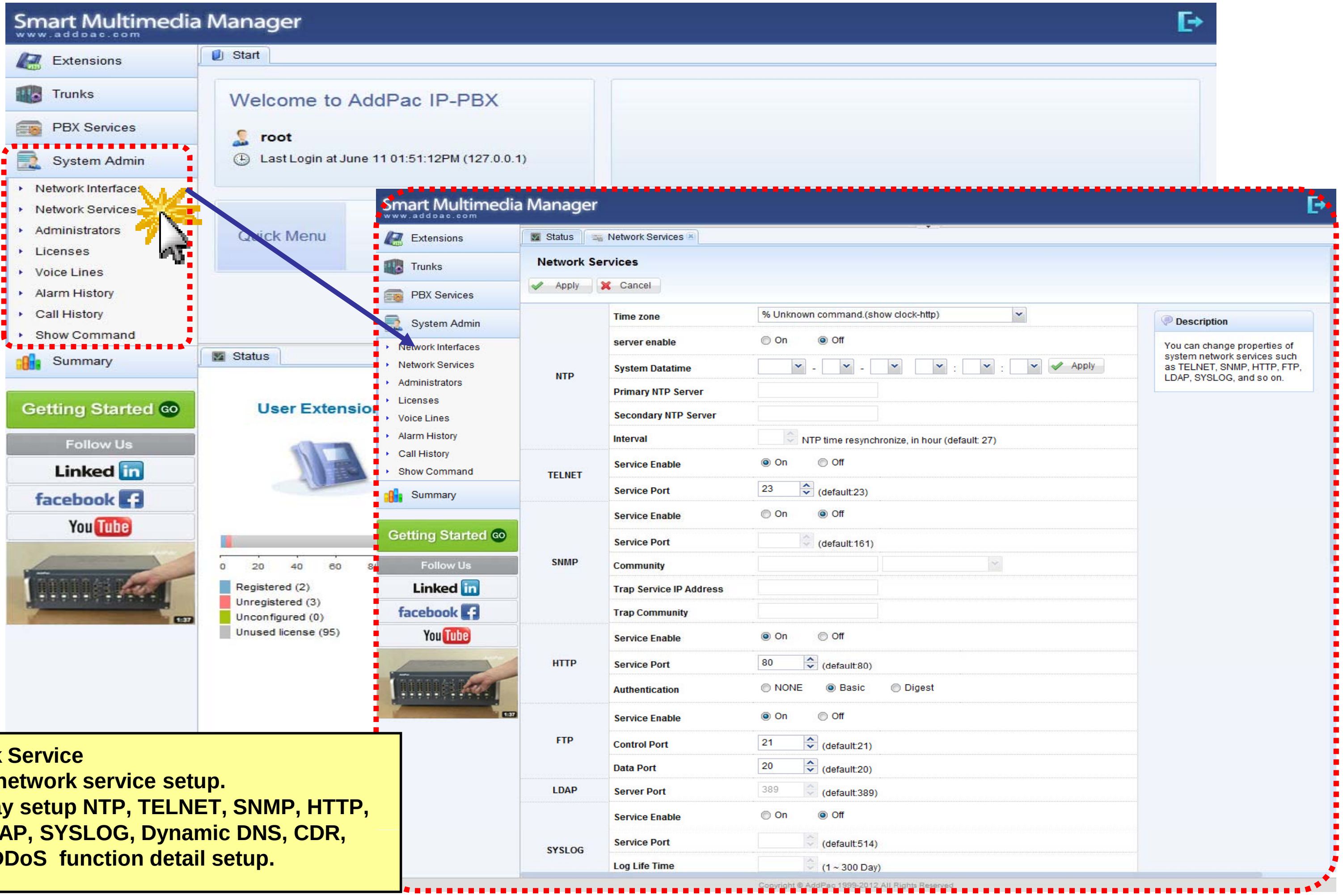
Service Monitoring
It displays Active Call & Conference information in IP-PBX. User may setup monitoring screen renew, interval time setup, and provides active call & conference information.

System Admin - Network Interface



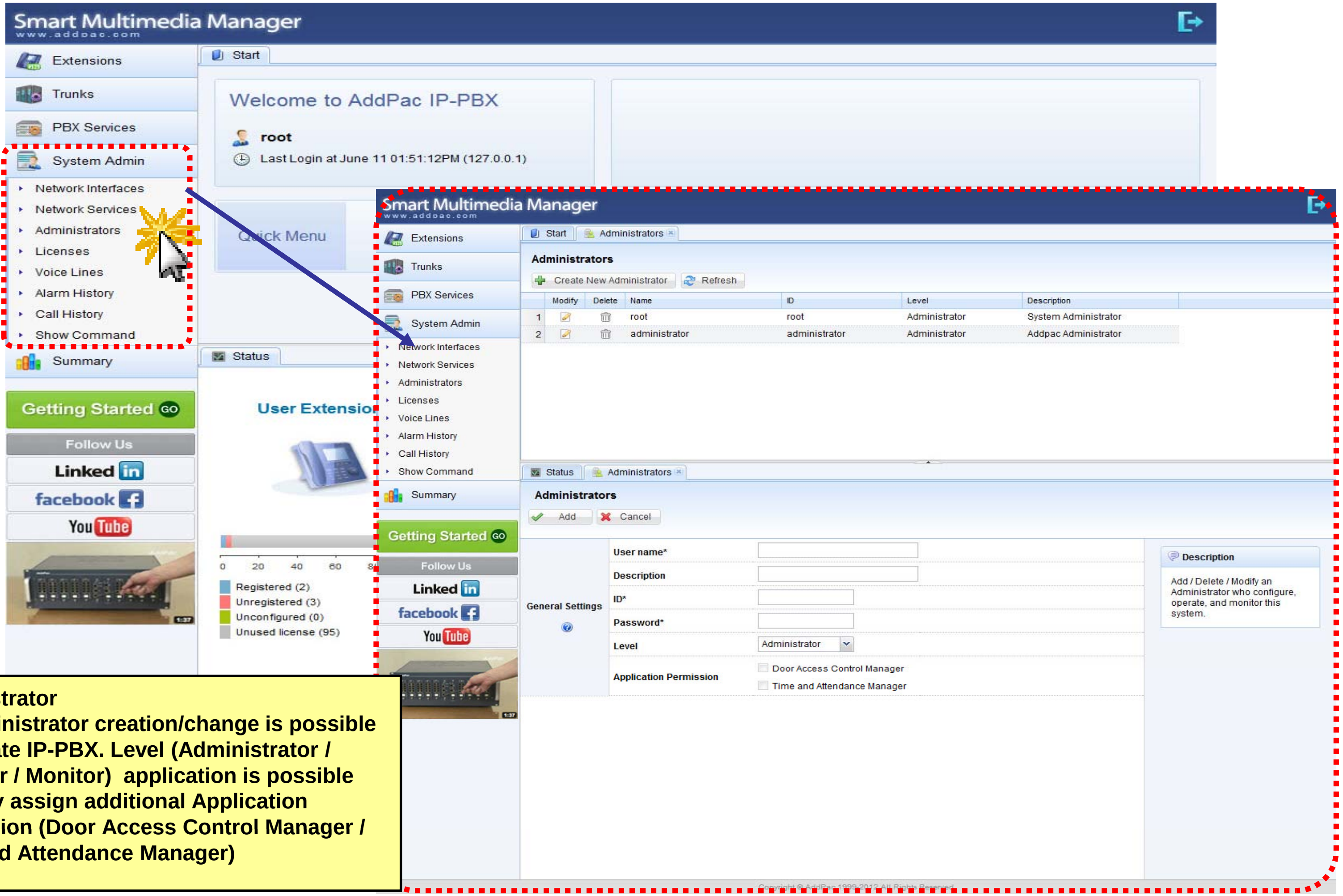
Network Interface
IP-PBX Network interface setup.
WAN Interface
- IPv4 / IPv6 Address, DNS, DHCP Client
LAN Interface
- IPv4 / IPv6 Address, DHCP Server

System Admin - Network Services



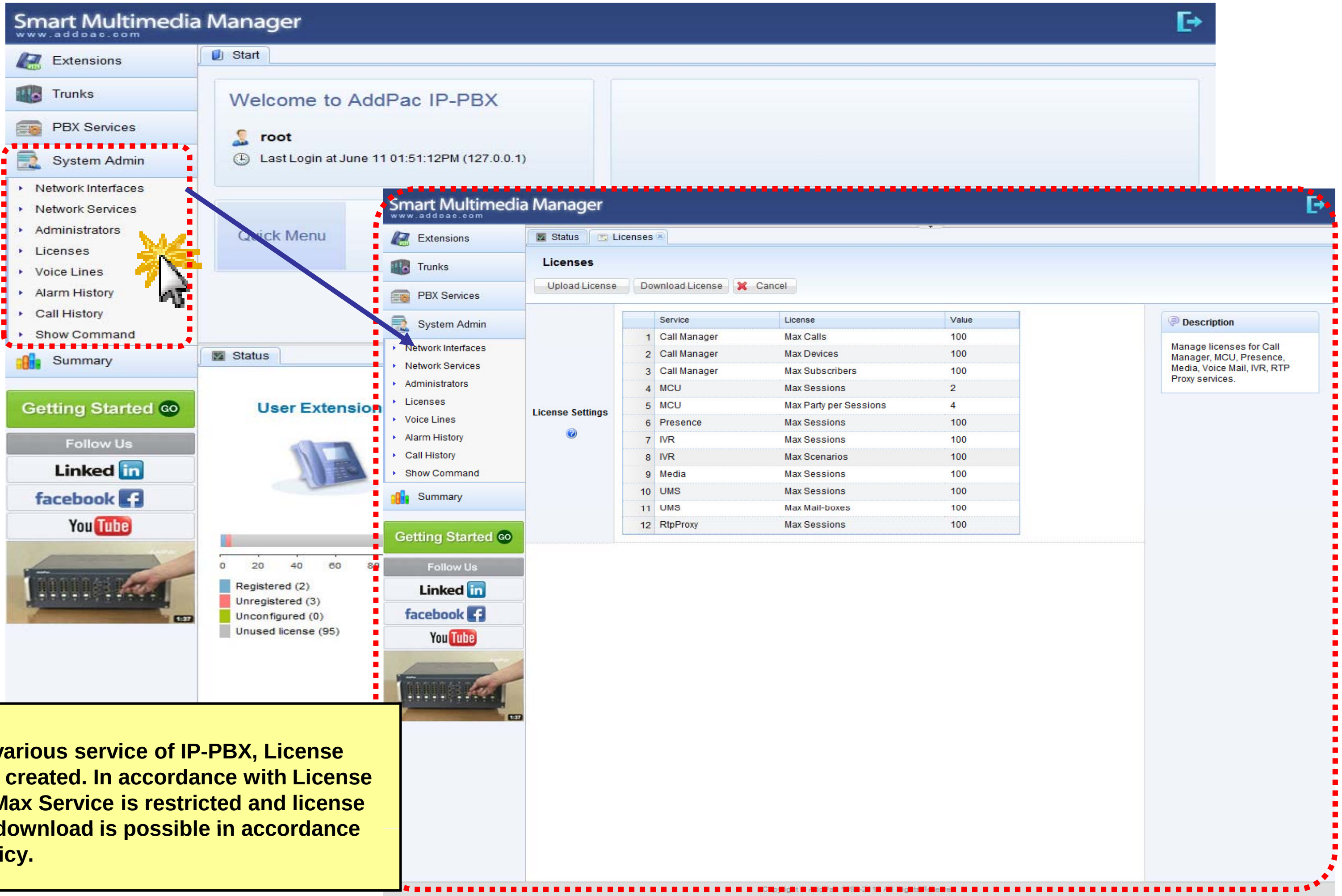
Network Service
IP-PBX network service setup.
User may setup NTP, TELNET, SNMP, HTTP, FTP, LDAP, SYSLOG, Dynamic DNS, CDR, SMTP, DDoS function detail setup.

System Admin - Administrators



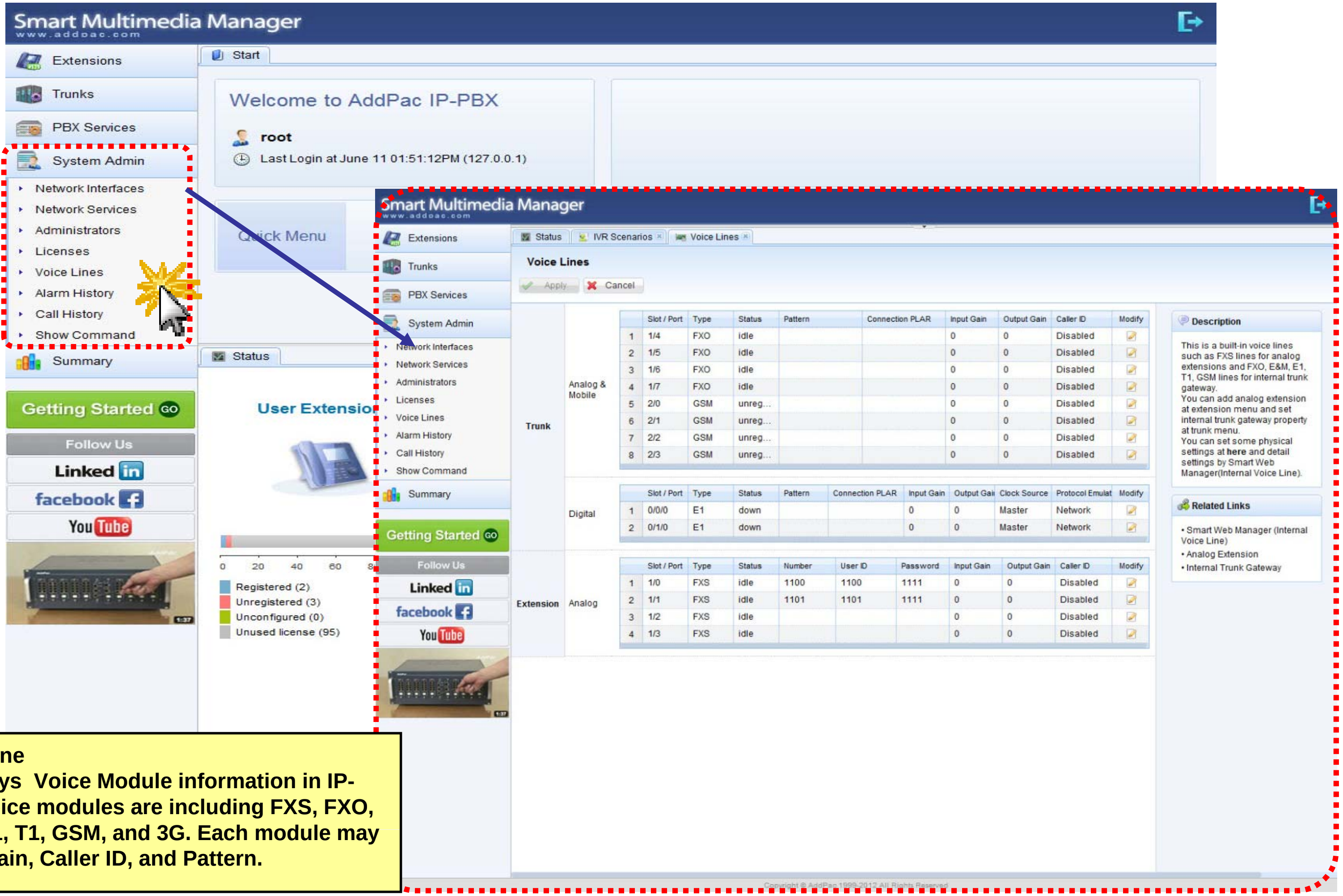
Administrator
An administrator creation/change is possible to operate IP-PBX. Level (Administrator / Operator / Monitor) application is possible and may assign additional Application Permission (Door Access Control Manager / Time and Attendance Manager)

System Admin - Licenses



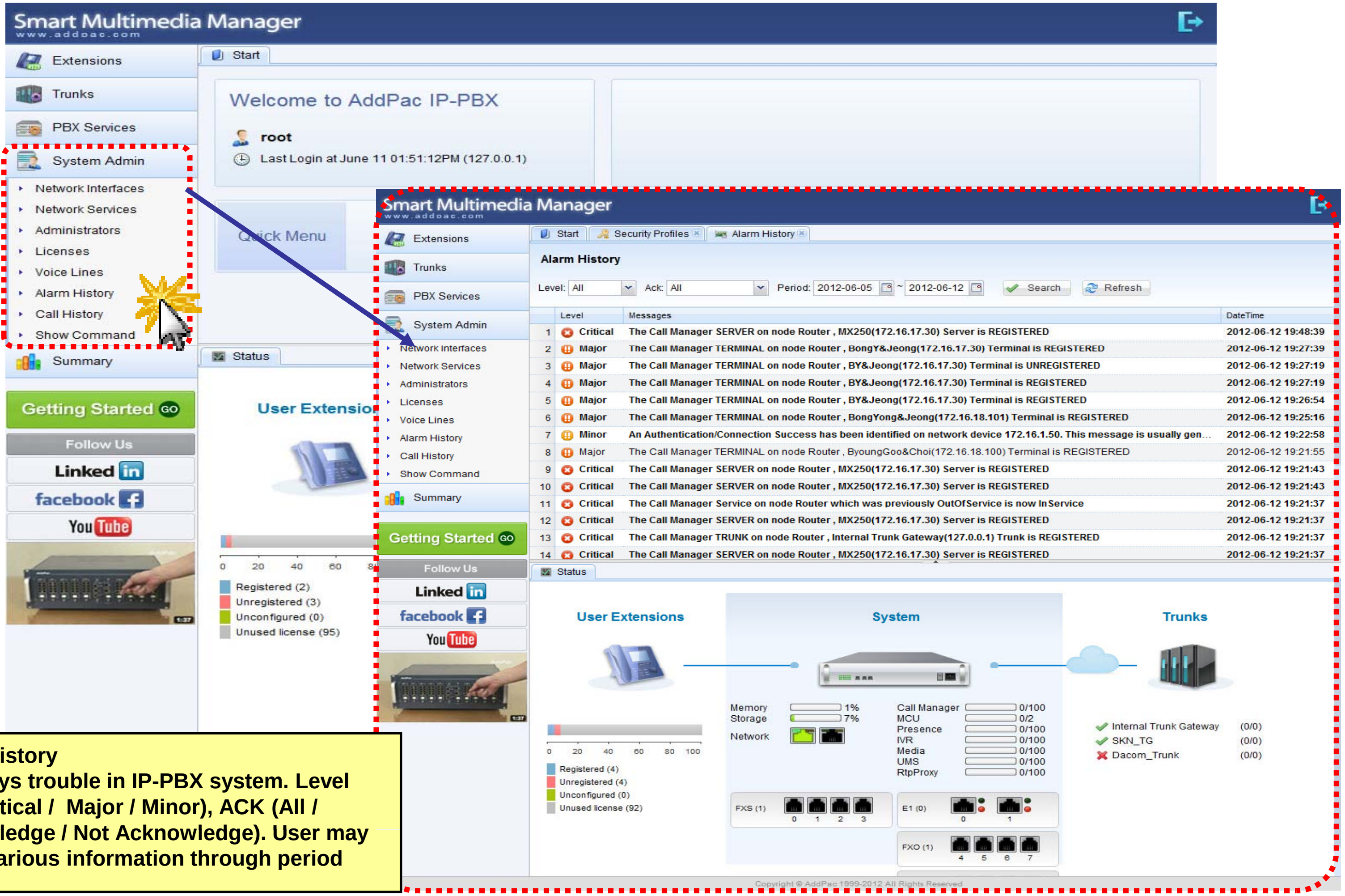
License
To use various service of IP-PBX, License must be created. In accordance with License policy, Max Service is restricted and license upload/download is possible in accordance with policy.

System Admin - Voice Lines



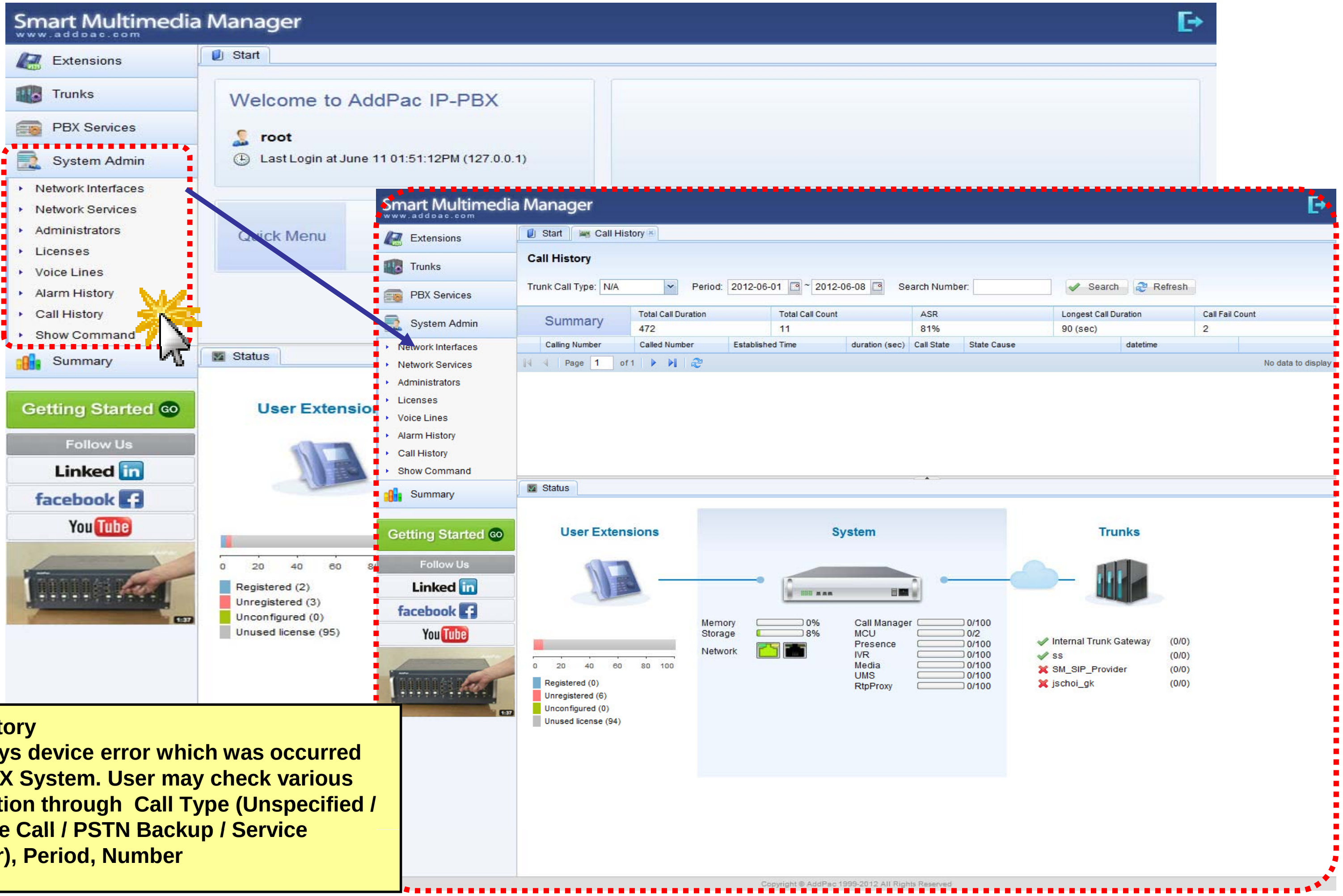
Voice Line
It displays Voice Module information in IP-PBX. Voice modules are including FXS, FXO, E&M, E1, T1, GSM, and 3G. Each module may setup Gain, Caller ID, and Pattern.

System Admin - Alarm History



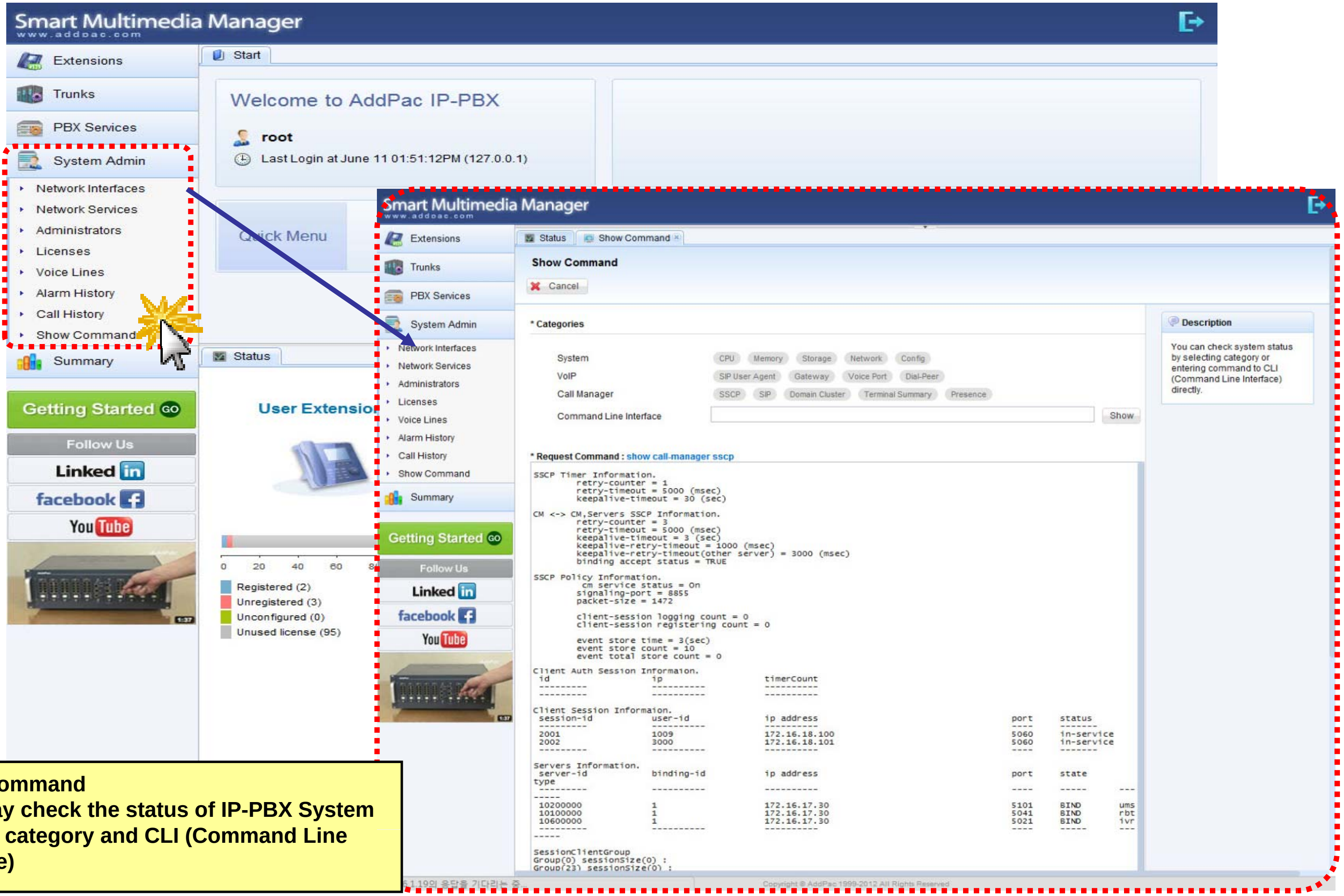
Alarm History
It displays trouble in IP-PBX system. Level (All / Critical / Major / Minor), ACK (All / Acknowledge / Not Acknowledge). User may check various information through period filter.

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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