

# IP Telephony Solution (GSM + SMS)



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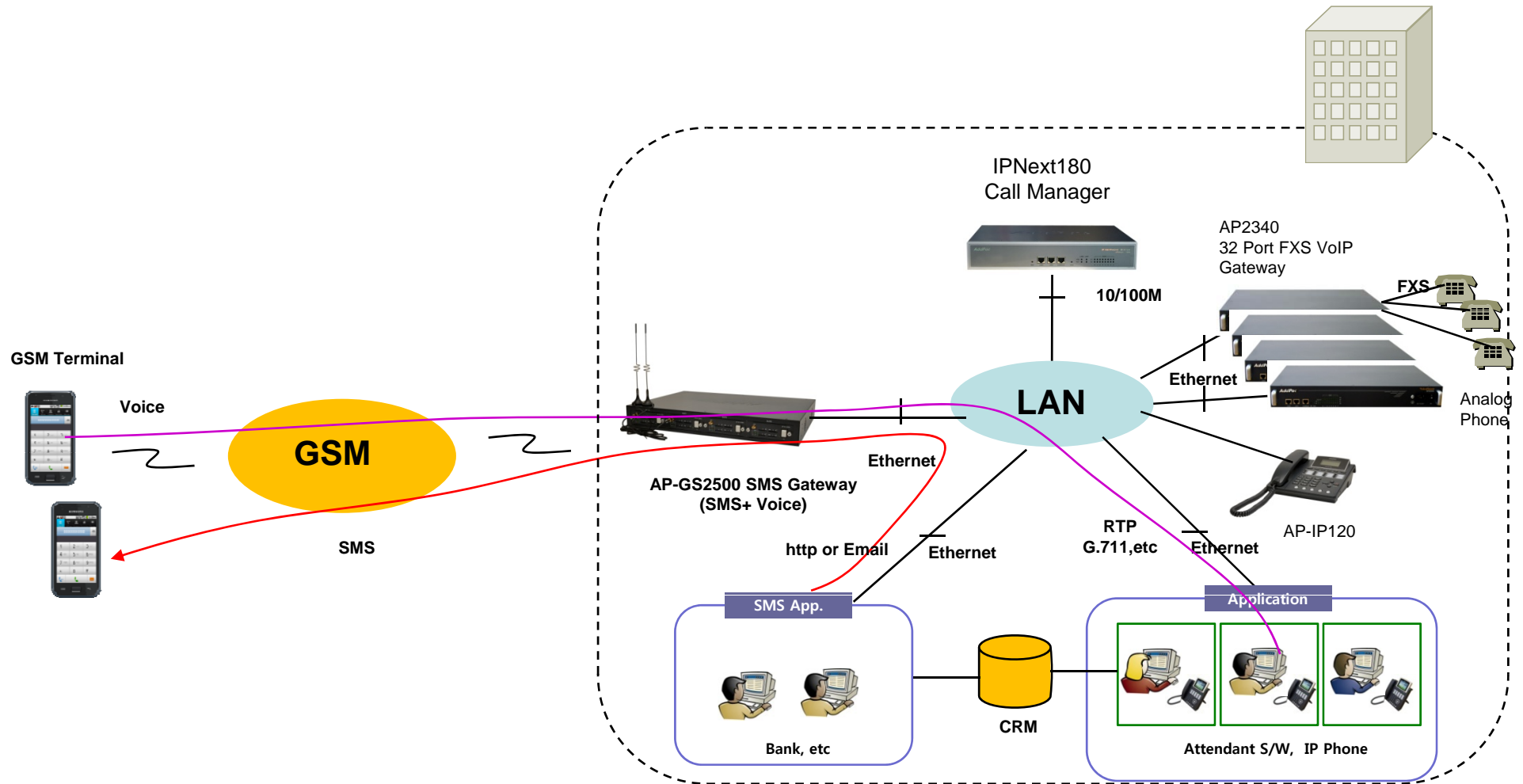
Sales and Marketing

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- IP Telephony Network Diagram
- IP Telephony Solution Components (Example)
  - IPNext180 IP-PBX (150 User License ) with WSMM
  - AP2340 32-Port FXS VoIP Gateway for 100 Analog Line  
(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



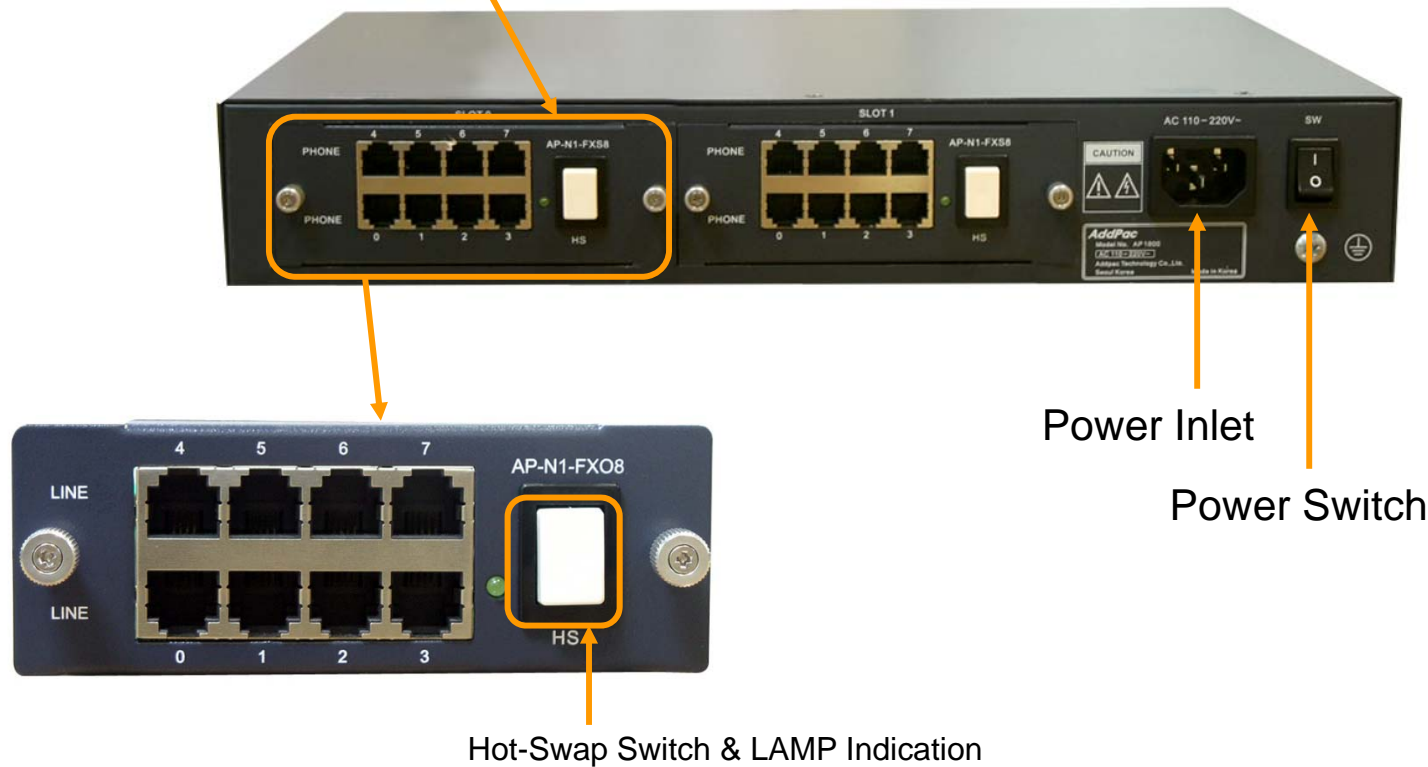
LAN1 (10/100Mbps)  
LAN0(10/100Mbps)  
Console Port

# Hardware Specification

IPNext180 Next Generation Hybrid IP-PBX System

## Back Side

PSTN Interface Module







# Hardware Specification

IPNext180 Next Generation Hybrid IP-PBX System

64bit  
CPU

Audio  
Codec

- **VoIP Interface Module**

|                     |  |   |
|---------------------|--|---|
| <b>AP-N1-FXS8</b>   |  A black VoIP interface module with eight RJ11 ports labeled 0-7 and a central switch labeled 'HS'. The top left has 'LINE' labels and the top right has 'AP-N1-FXS8'.                                   | 8-Port FXS Voice Processing Module<br>(8 x RJ11)                |
| <b>AP-N1-FXO8</b>   |  A black VoIP interface module with eight RJ11 ports labeled 0-7 and a central switch labeled 'HS'. The top left has 'LINE' labels and the top right has 'AP-N1-FXO8'.                                   | 8-Port FXO Voice Processing Module<br>(8 x RJ11)                |
| <b>AP-N1-FXO4S4</b> |  A black VoIP interface module with four RJ11 ports labeled 0-3 on the left and four on the right, and a central switch labeled 'HS'. The top left has 'LINE' labels and the top right has 'AP-N1-FXO8'. | 4-Port FXO and 4-Port FXS Voice<br>Processing Module (8 x RJ11) |
| <b>AP-N1-E1T1</b>   |  A black VoIP interface module with a green PCB on top, a single RJ45 port labeled 'POE1', and a central switch labeled 'HS'. The bottom left has 'ACT' and 'LOS' labels.                              | 1-Port VoIP Digital E1/T1 Interface<br>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



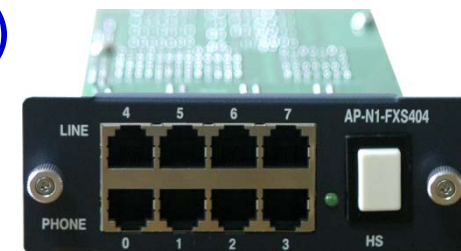
# Hardware Specification

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



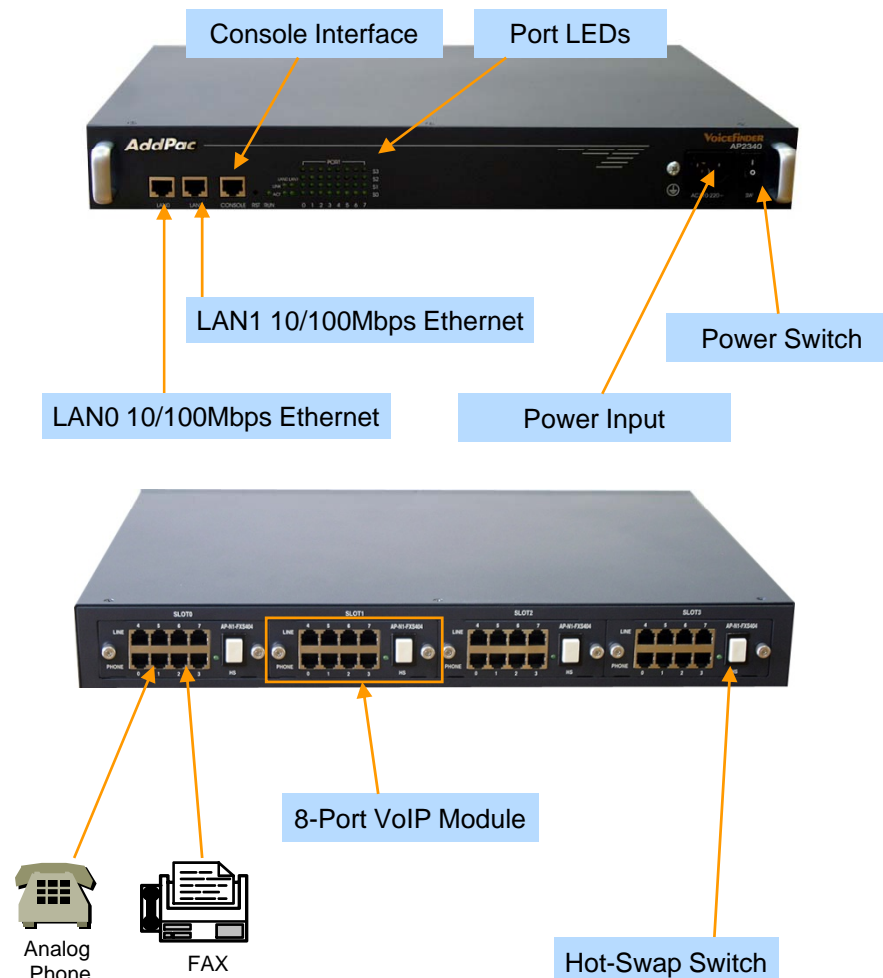
# Hardware Specification

## AP2340 32Port Analog VoIP Gateway

### Hardware Specifications

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

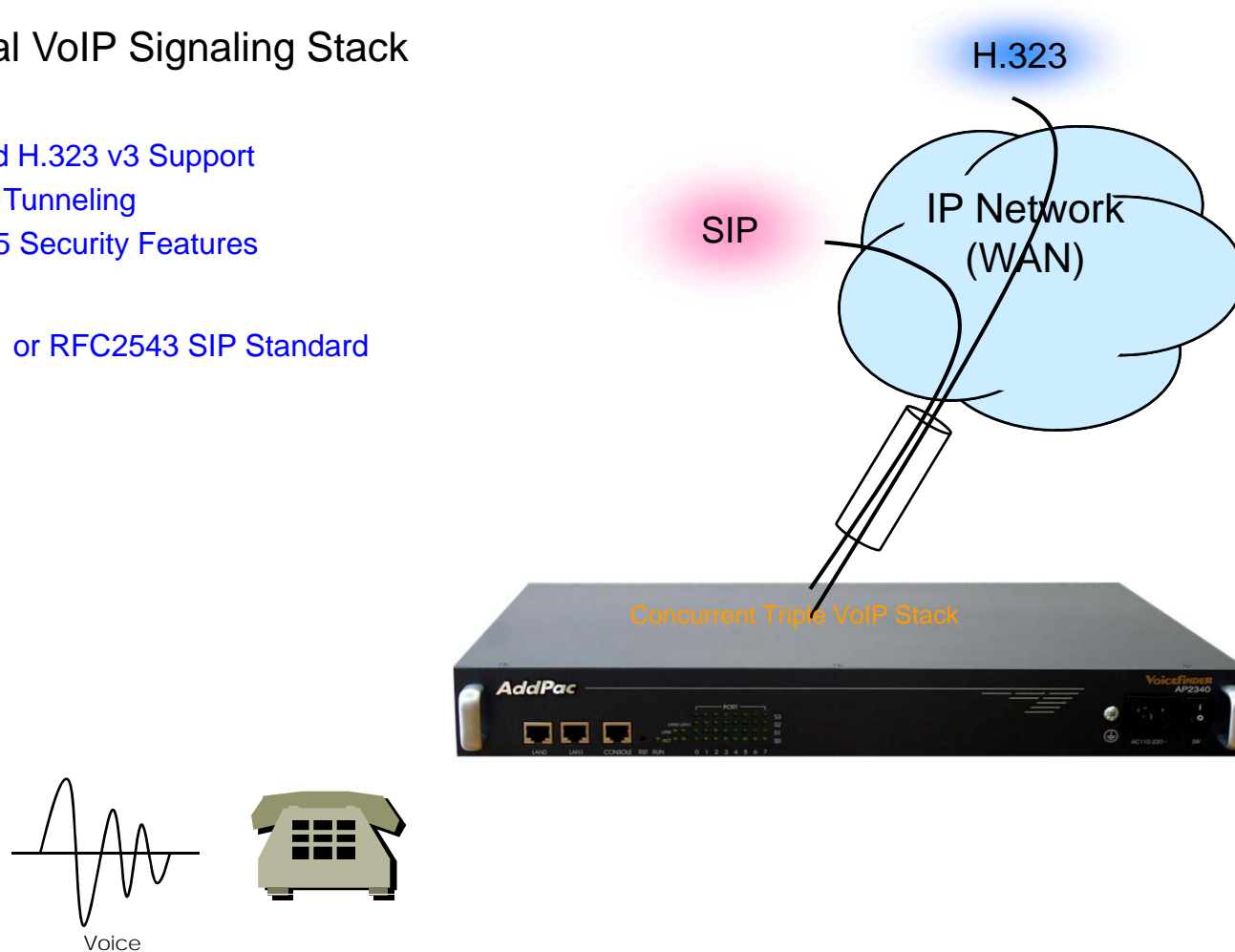
### Network interface Configurations



# 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**

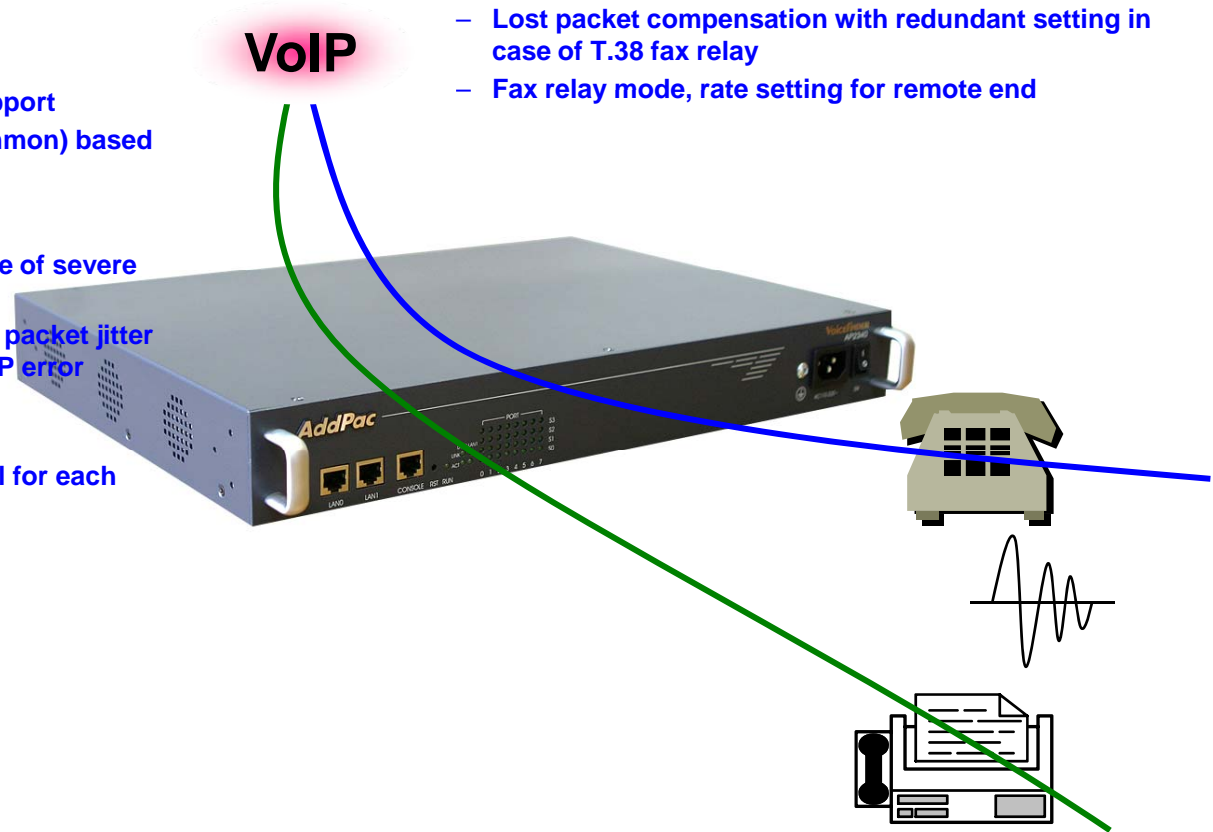
- G.711 A-Law, G.711 U-Law
- G.726 r16, G.726 r32
- G.729A
- G.723.1 r63, G.723.1 r53
- VAD (Voice Activity Detection) function support
- DTMF relay support (H.323, SIP, MGCP common) based on RFC2833

- **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

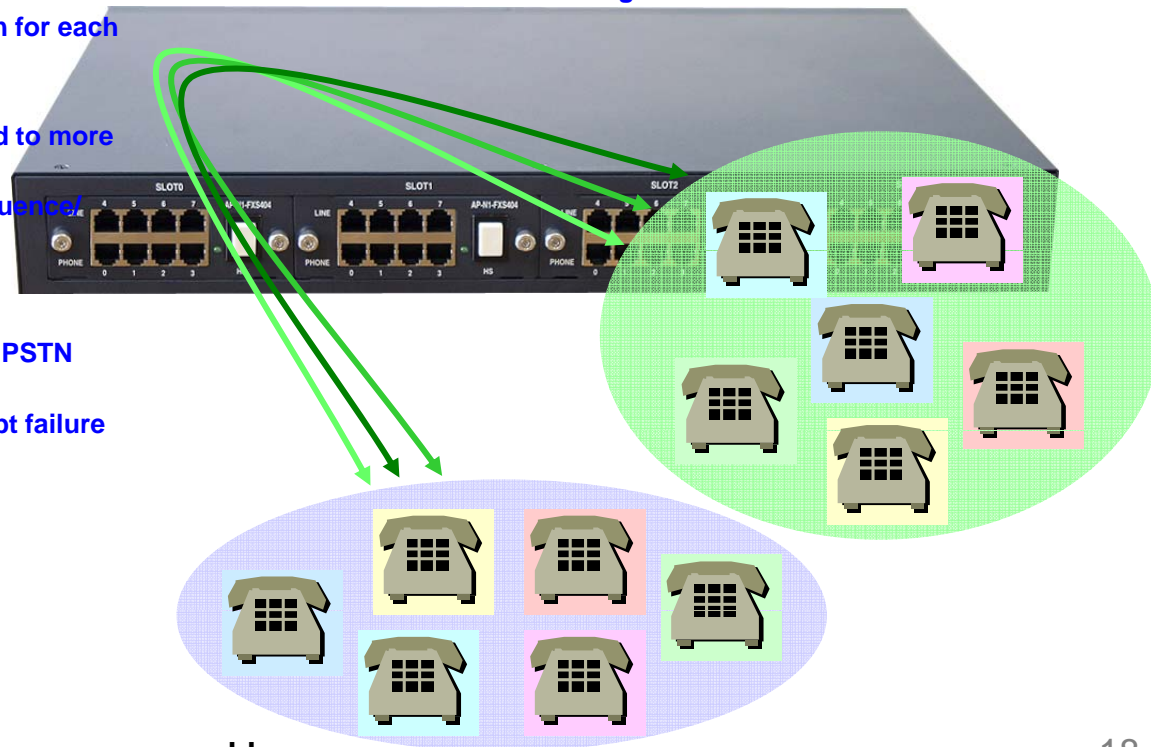
## AP2340 32 Port Analog VoIP Gateway

### • 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.)

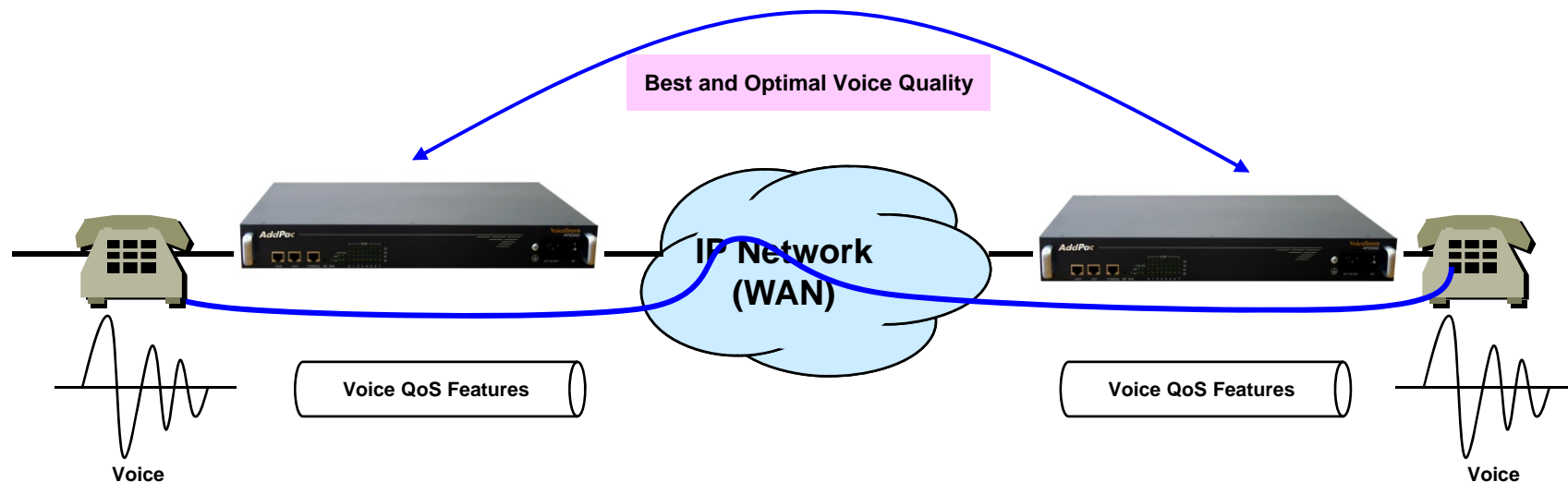
- Call transfer for internal calls
- Call pickup for internal calls
- Calling and called number conversion for VoIP outbound calls
- Calling and called number conversion for VoIP inbound calls
- Fax broadcasting call control



# 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

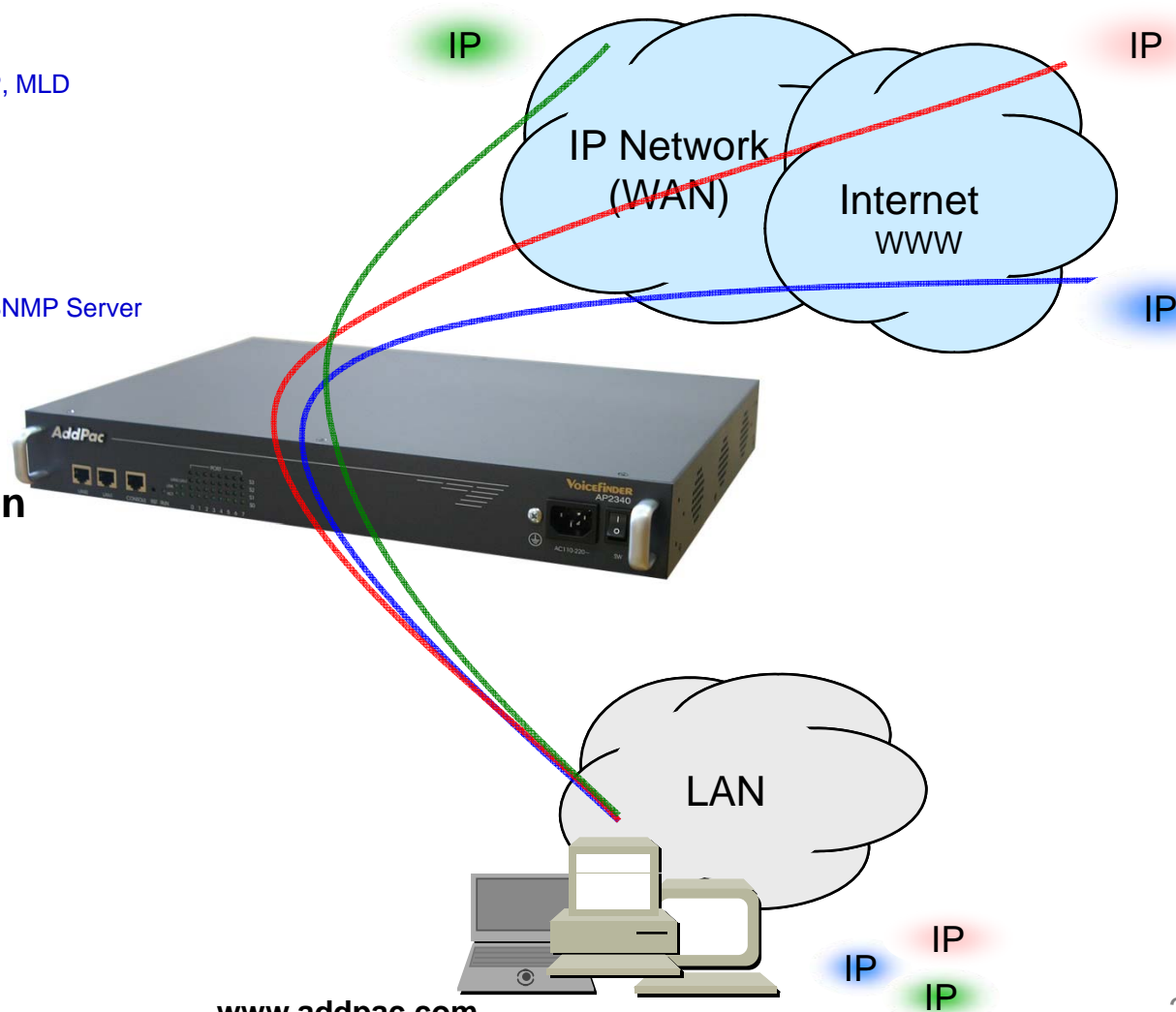
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

### 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**

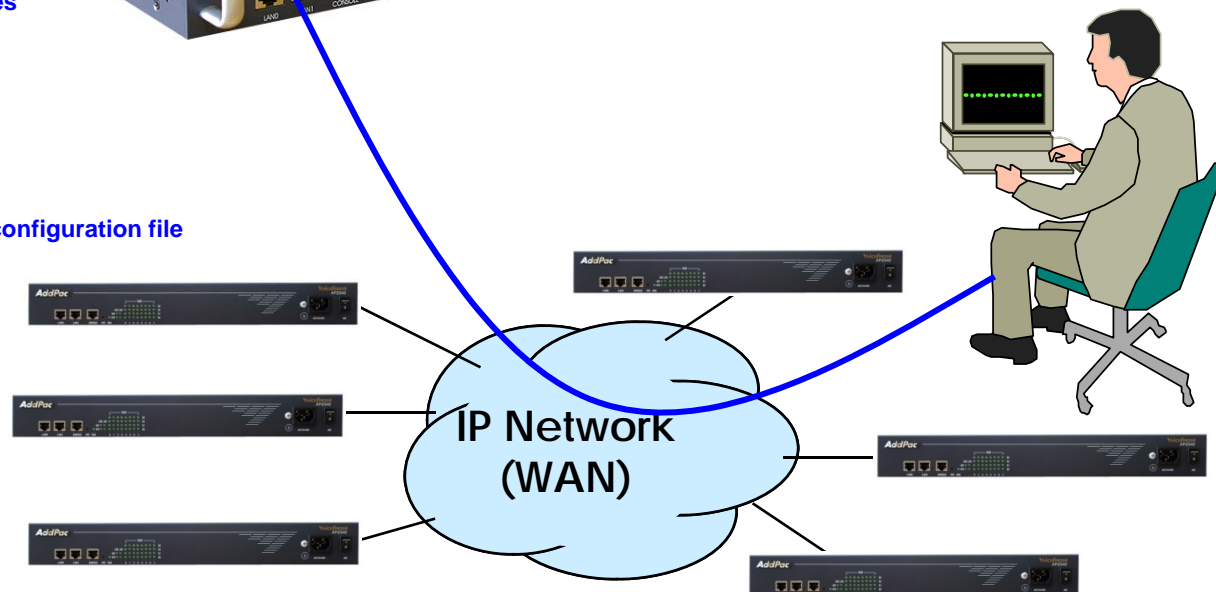
- HTTP server based APOS image and configuration file auto-upgrade support

- **Batch Job Function**

- Text based script downloading

- **Interoperable with AP-VPMS Service**

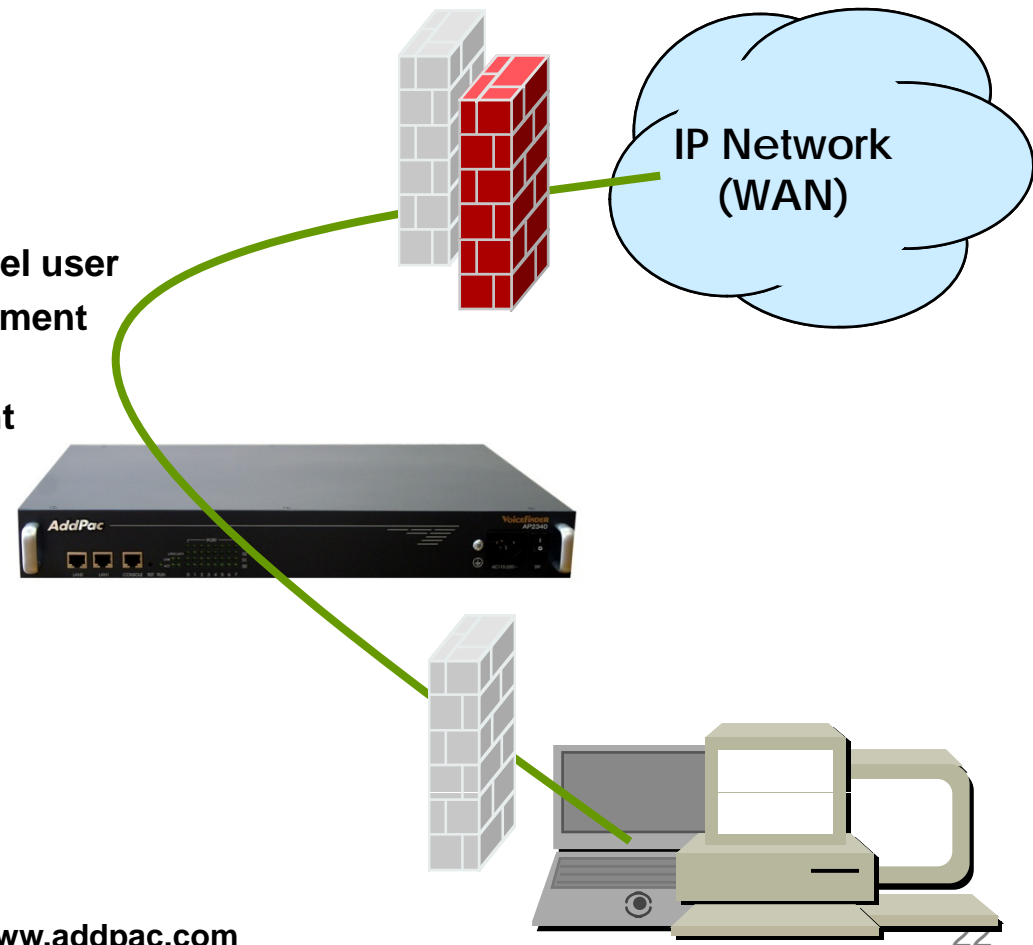
- AddPac VoIP Plug & Play Management System (AP-VPMS)



# 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<br> | AP-IP230<br> | AP-IP160<br> | AP-IP120<br> | AP-IP90<br> |
|----------------------|---|--|---|---|--|
| 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/<br>G.729/G.723   | G.711/G.726/<br>G.729/G.723  | G.711/G.726/<br>G.729/G.723   | G.711/G.726/<br>G.729/G.723   | G.711/G.726/<br>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(Optional)        | Support   | Support  | Support   | Support   | Support  |
| FXO(Optional)        | 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)

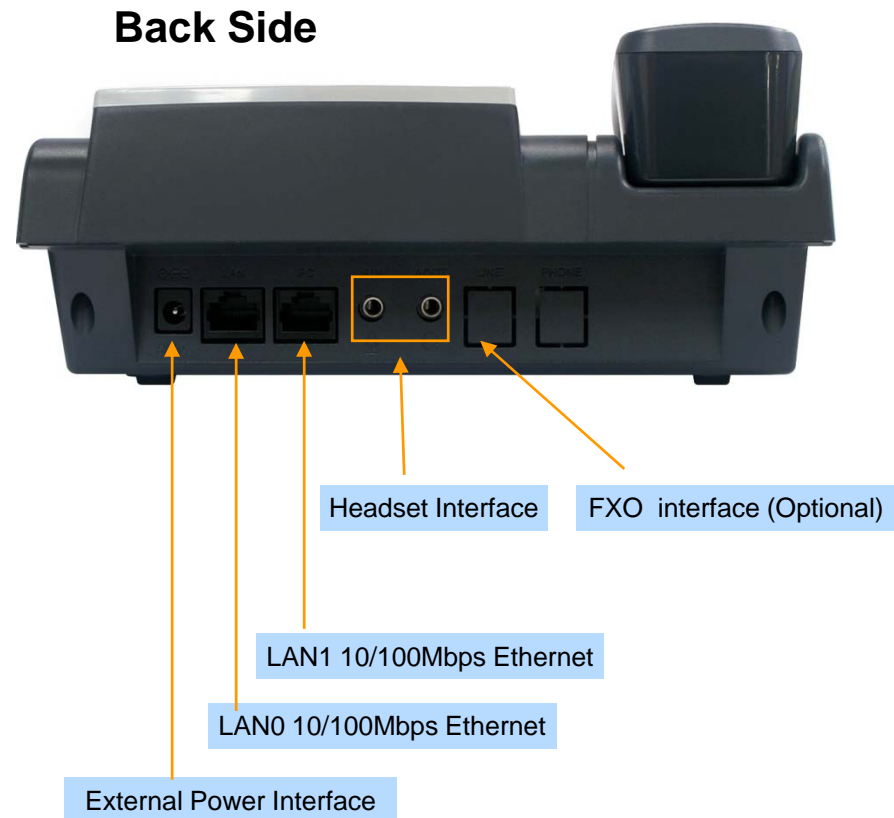
# Hardware Specification

## 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       | 0°C ~ 45°C (32 °F ~ 122°F)   |
| Storage Temperature         | -40°C ~ 85°C (-40°C ~ 185°F)   |
| Relative Humidity           | 5% ~ 95% (Non-condensing)  |
| Dimensions                  | H x W x D ( 70mm x 200mm x 210mm)                                    |
| Weight (g)                  | 1Kg  |

### Network interface Configurations







# AP-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



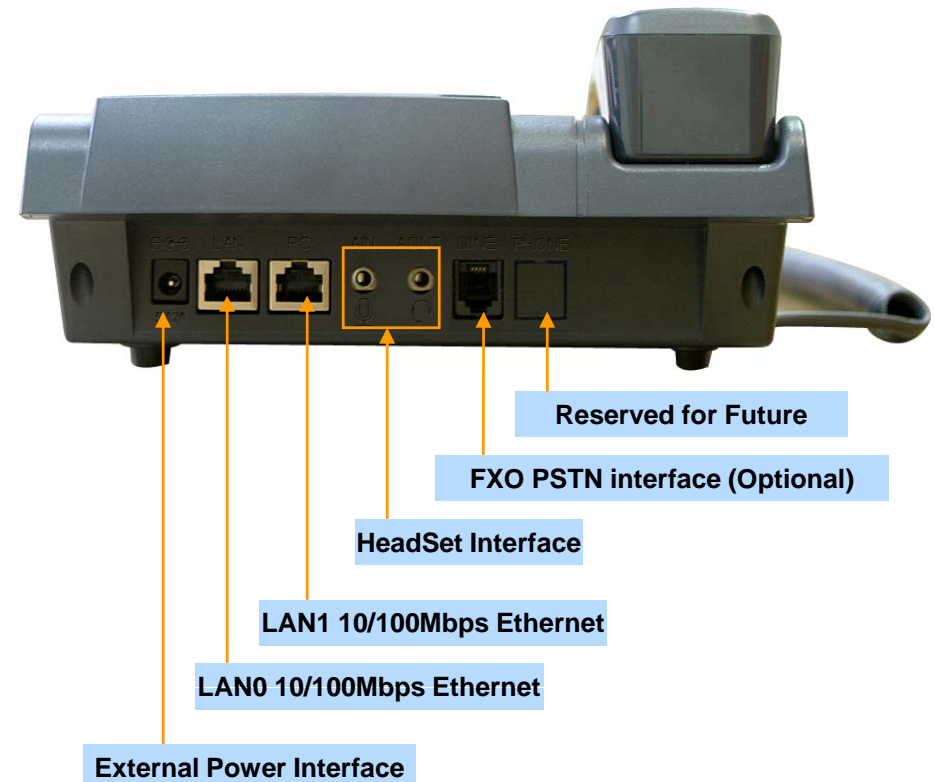
# Hardware Specification


## 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 | 0°C ~ 45°C (32 °F ~ 122°F)                                    |
| Storage Temperature   | -40°C ~ 85°C (-40°C ~ 185°F)                                  |
| Relative Humidity     | 5% ~ 95% (Non-condensing)                                     |
| Dimensions            | 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



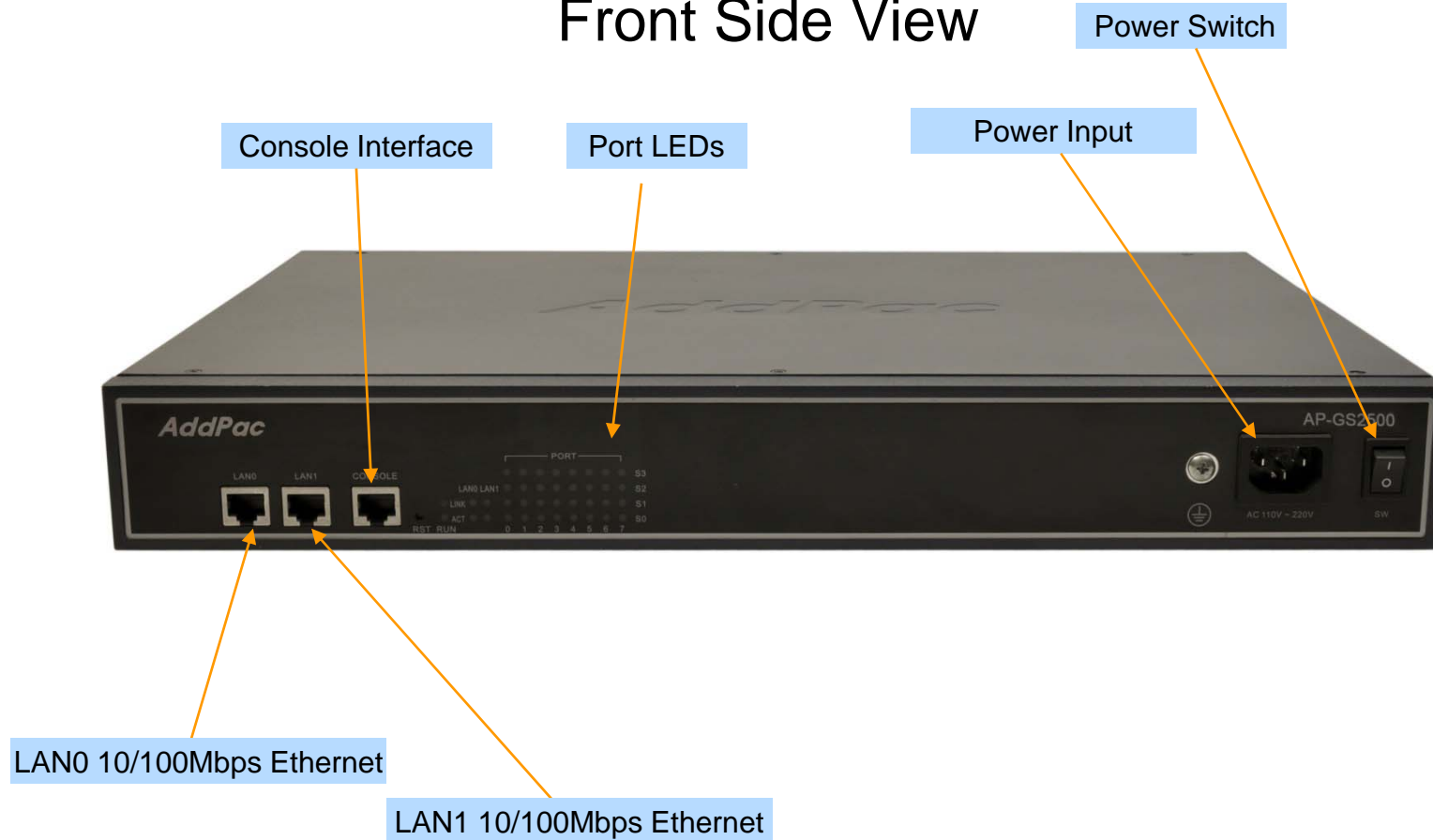
# Hardware Specification

AP-GS2500 GSM SMS Gateway

RISC  
CPU

High-end  
DSP

## Front Side View





# Hardware Specification

AP-GS2500 GSM SMS Gateway

RISC  
CPU

High-end  
DSP

## Back Side View



4-Port 3G Module

3G Antenna  
Internal 4Ch Combiner

Port LEDs

USIM Card Slot

Hot-Swap Switch

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# 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



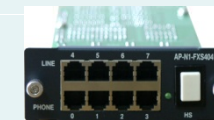
**AP-N1-FXS8** 8-Port FXS Module



**AP-N1-FXO8** 8-Port FXO Module



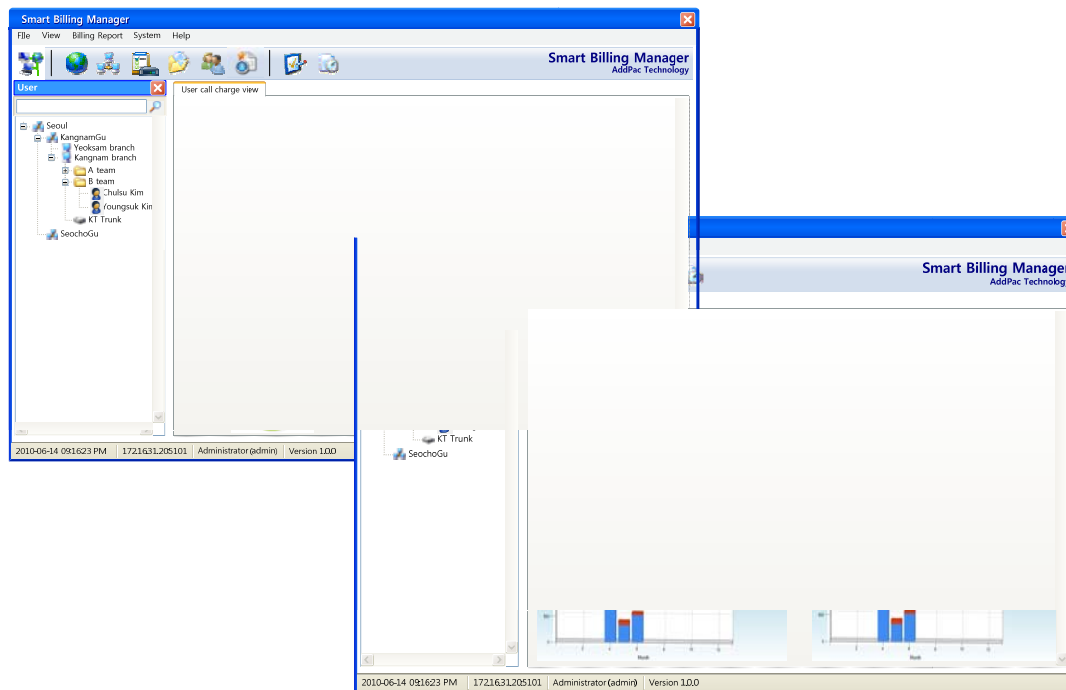
**AP-N1-FXS4O4** 4-Port FXS&4-Port FXO Module



**AP-N1-E1** 1-Port Digital E1/T1 Module

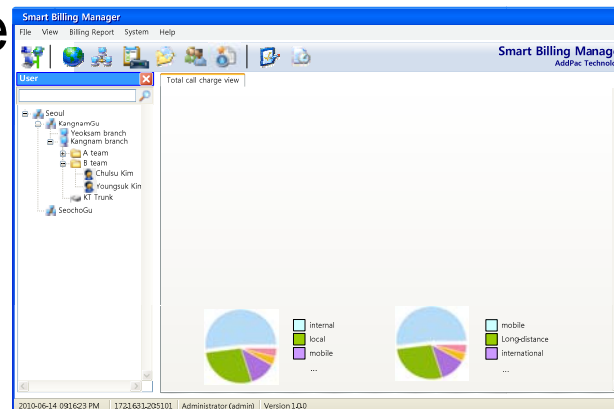


# Smart Billing System



# 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

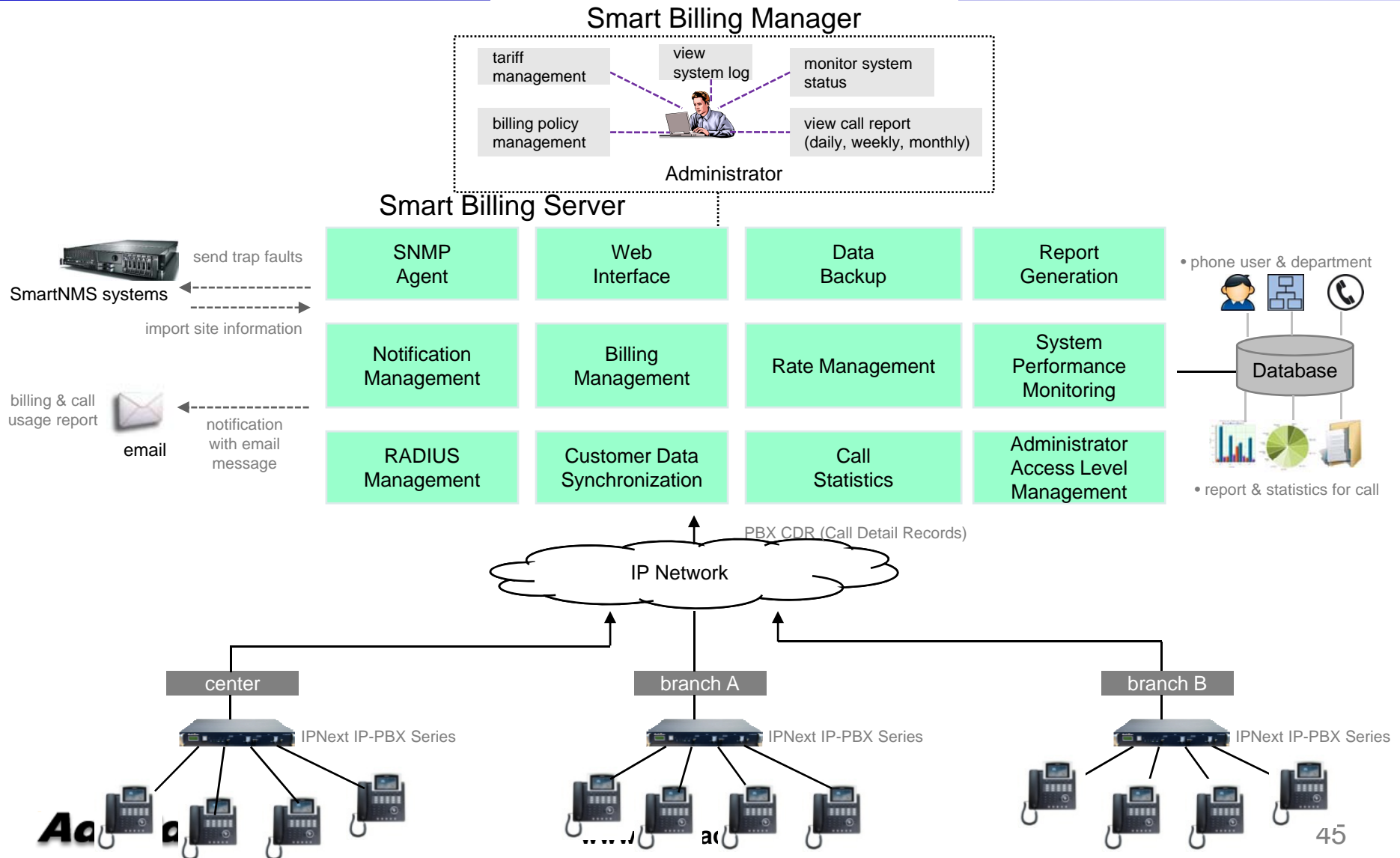
- Windows XP, Vista, Windows 7, windows Server2000/2003
- Microsoft Internet Explorer 6.0 or higher

# 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.

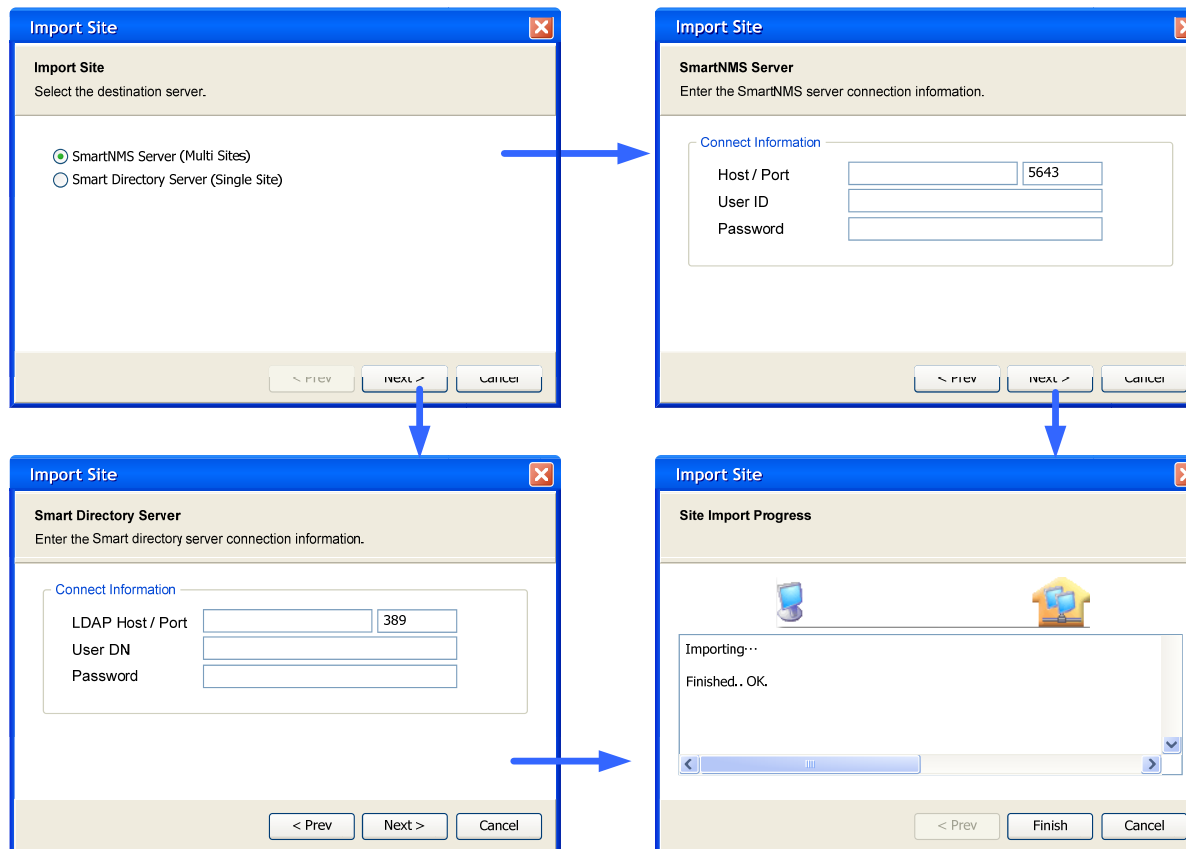


# System Configuration



# 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.





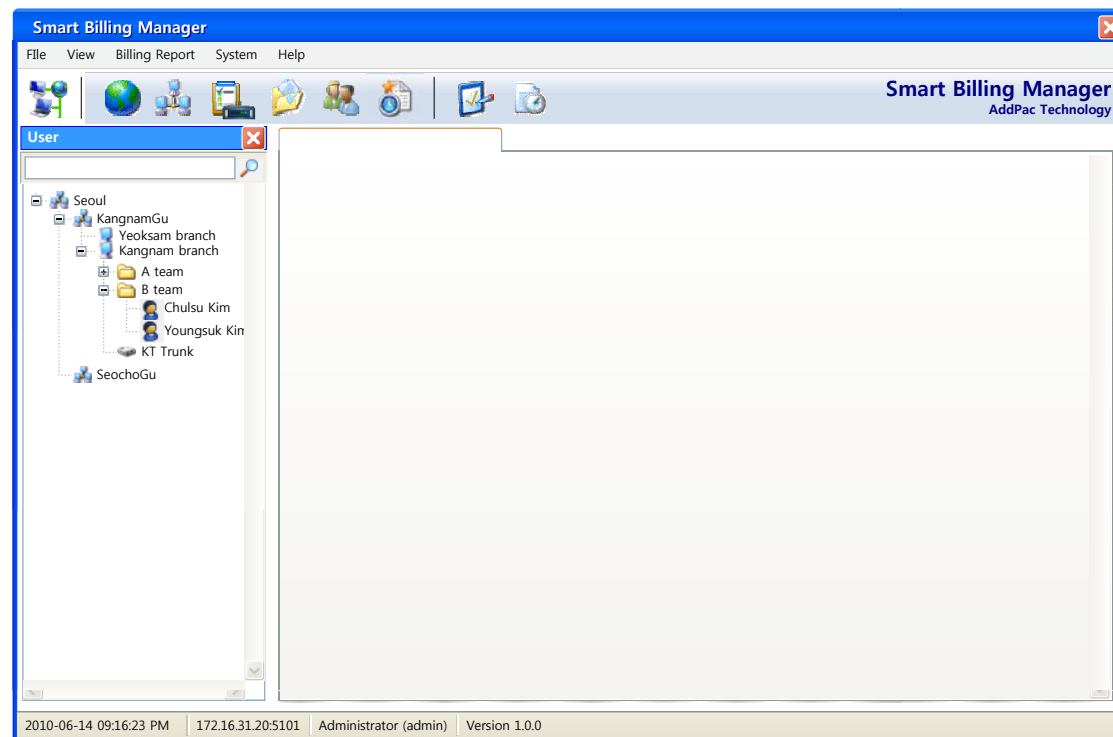
# Call Report Generation

# Call Report Generation

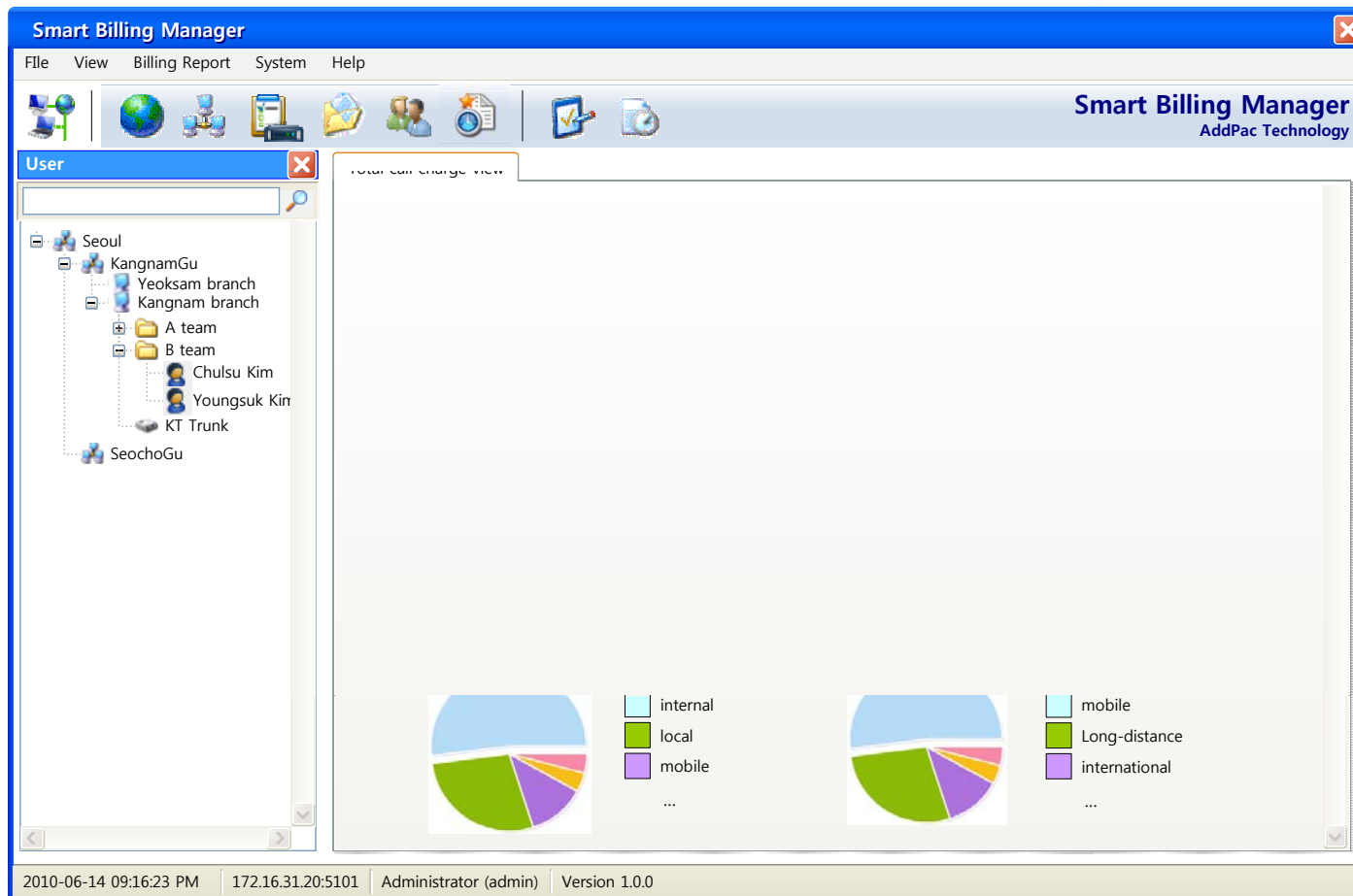
- Total Call Usage Monthly Statistic
- Total Call Charge View
- Site Call Charge View
- Trunk Call Charge View
- Department Call Charge View
- User Call Charge View
- Total Call Charge Monthly Statistic
- Total Number of Call Monthly Statistic
- Total Call Type Monthly Statistic
- Incompletion Calls 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

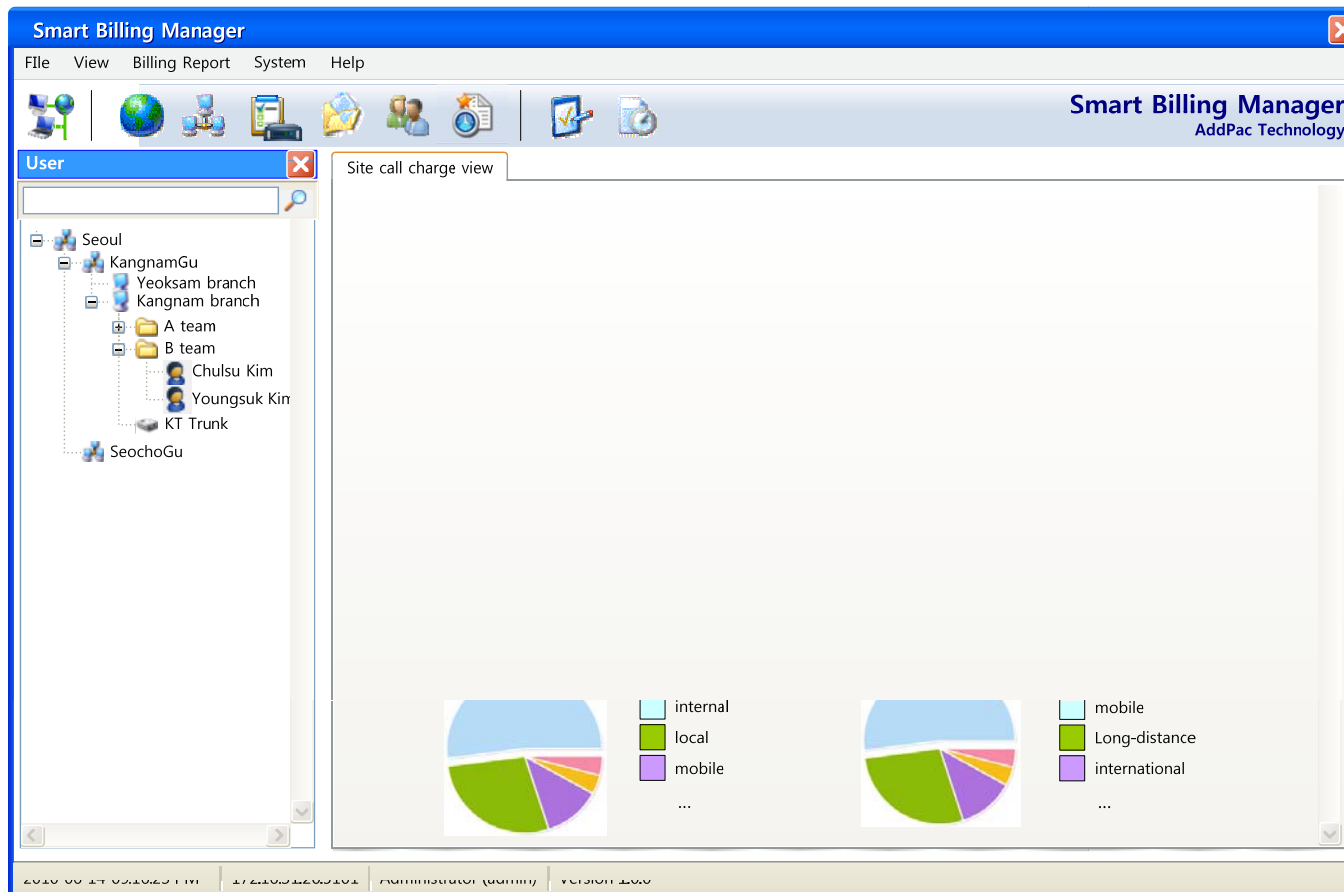
- Generate daily, weekly, monthly call usage report for site, trunk, department and phone users.
- Provide call usage with summary and chart type.



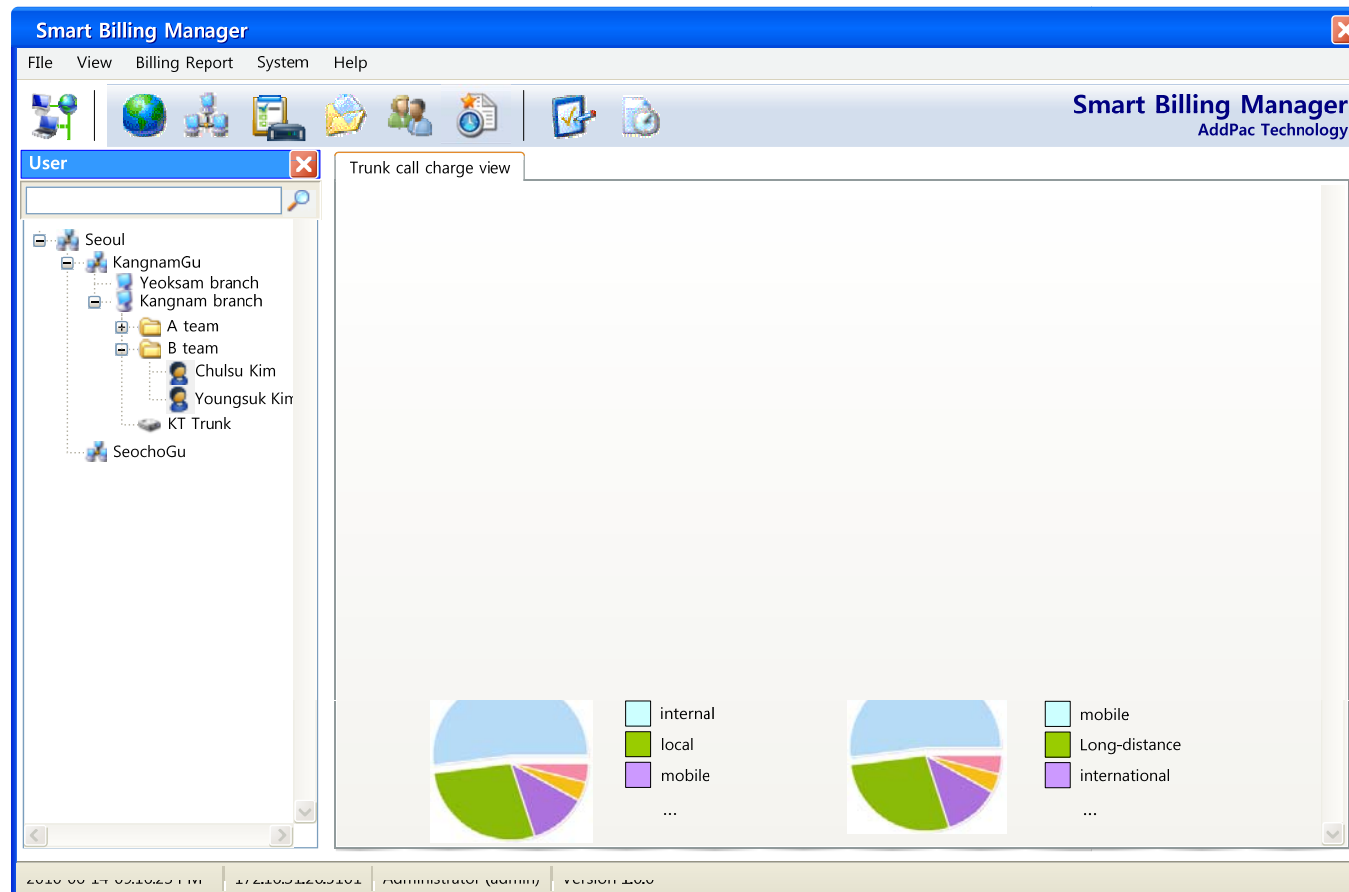
# Call Report Generation



# Call Report Generation

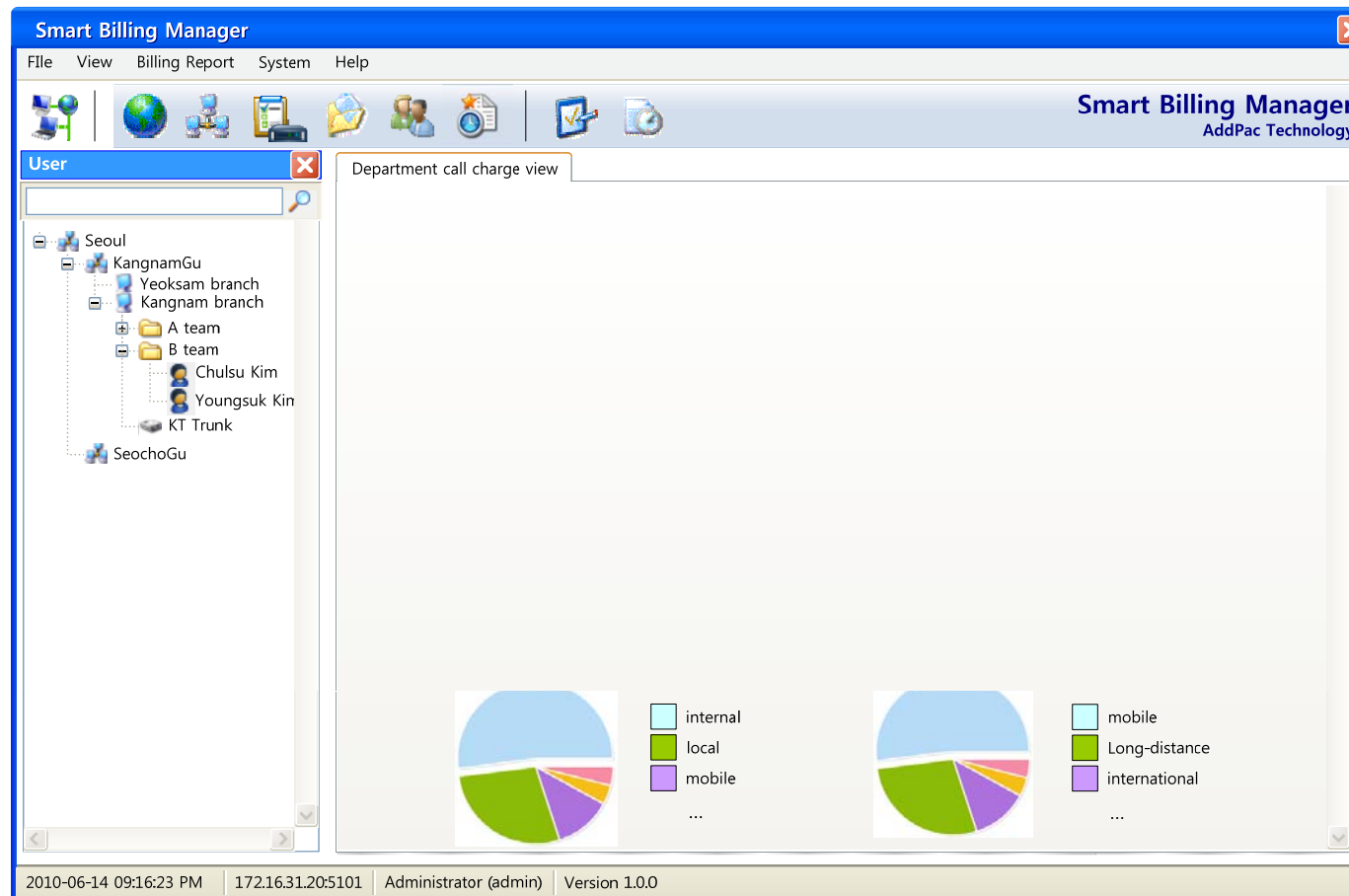


# Call Report Generation

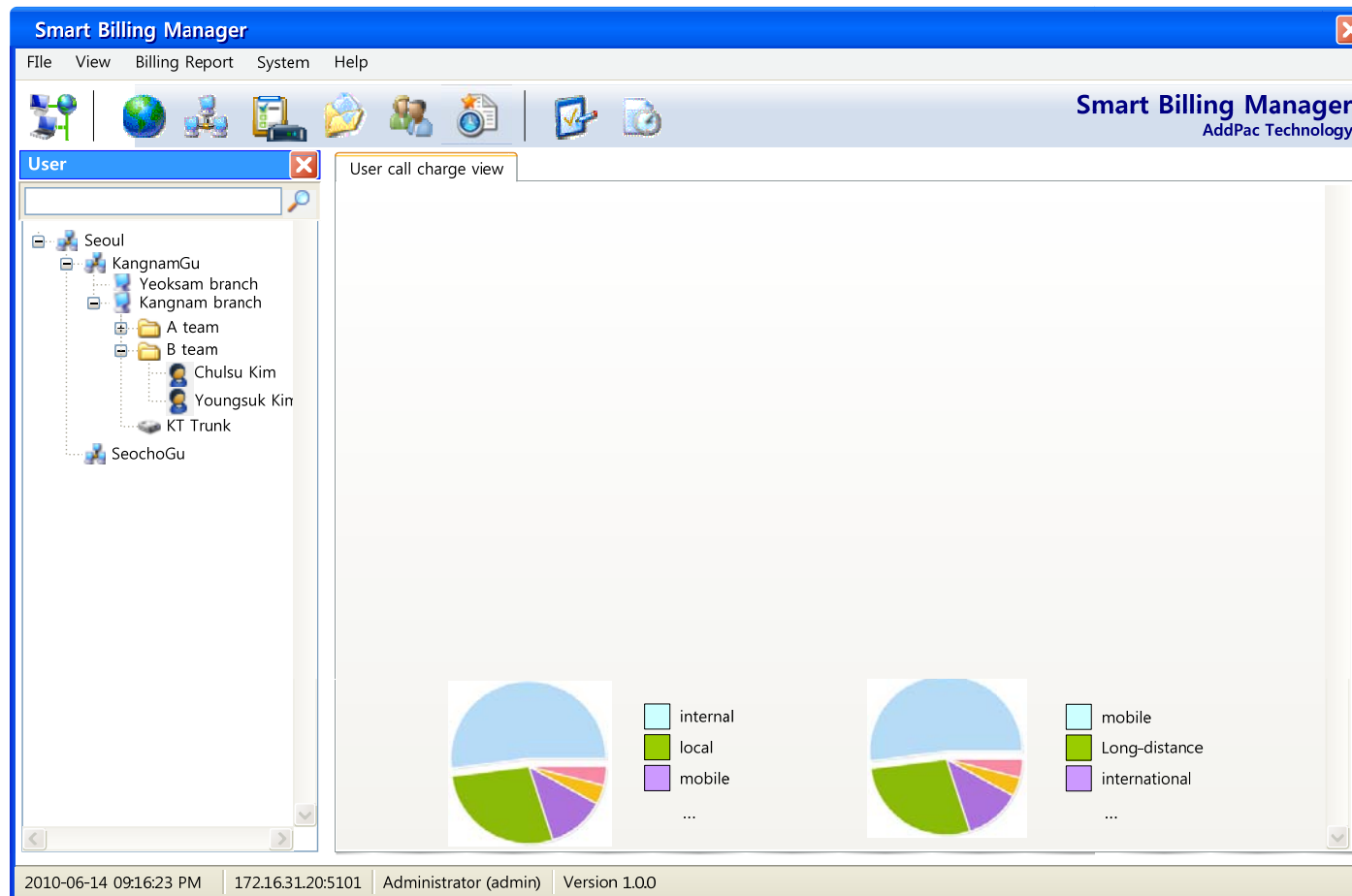




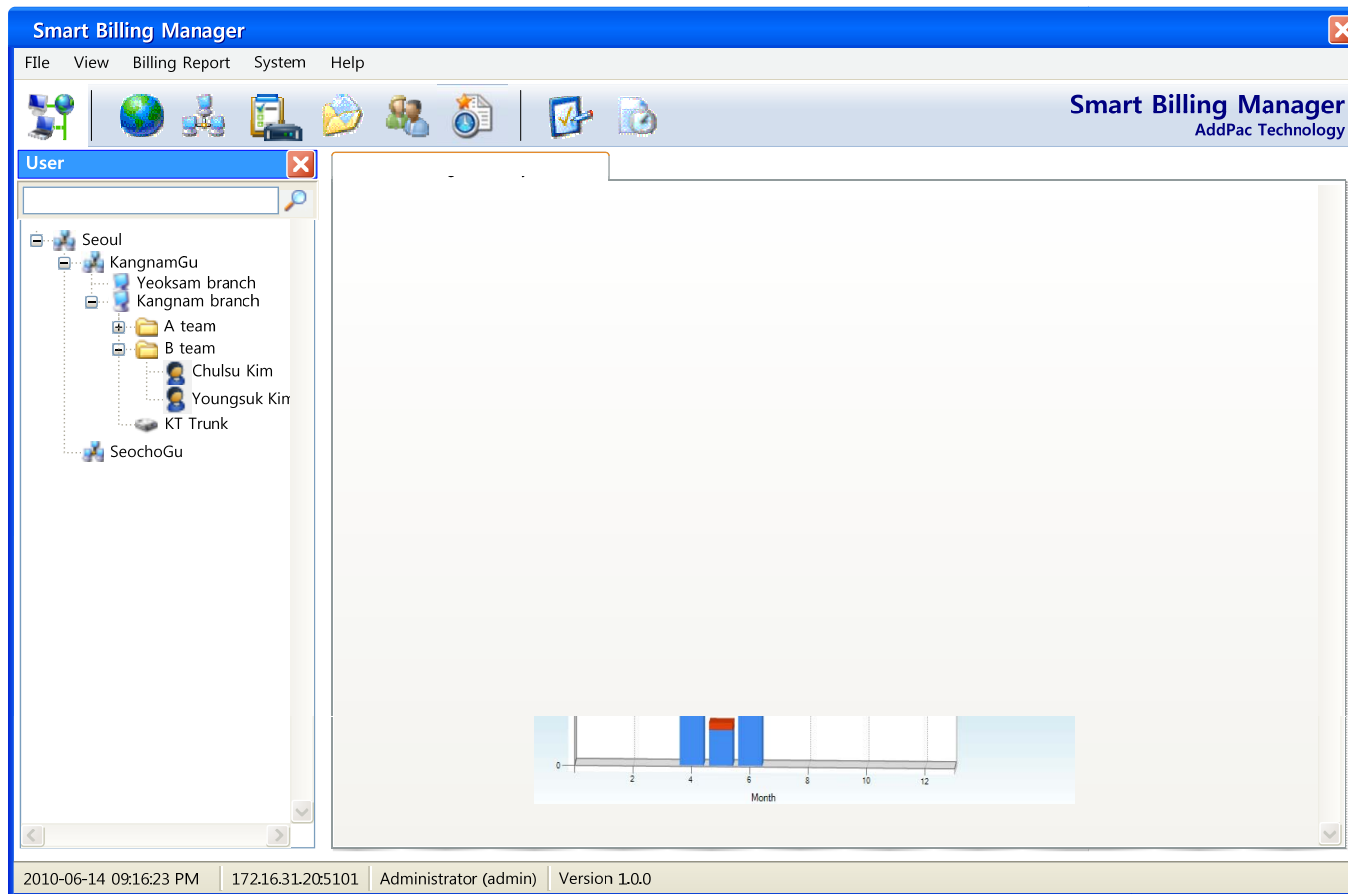
# Call Report Generation



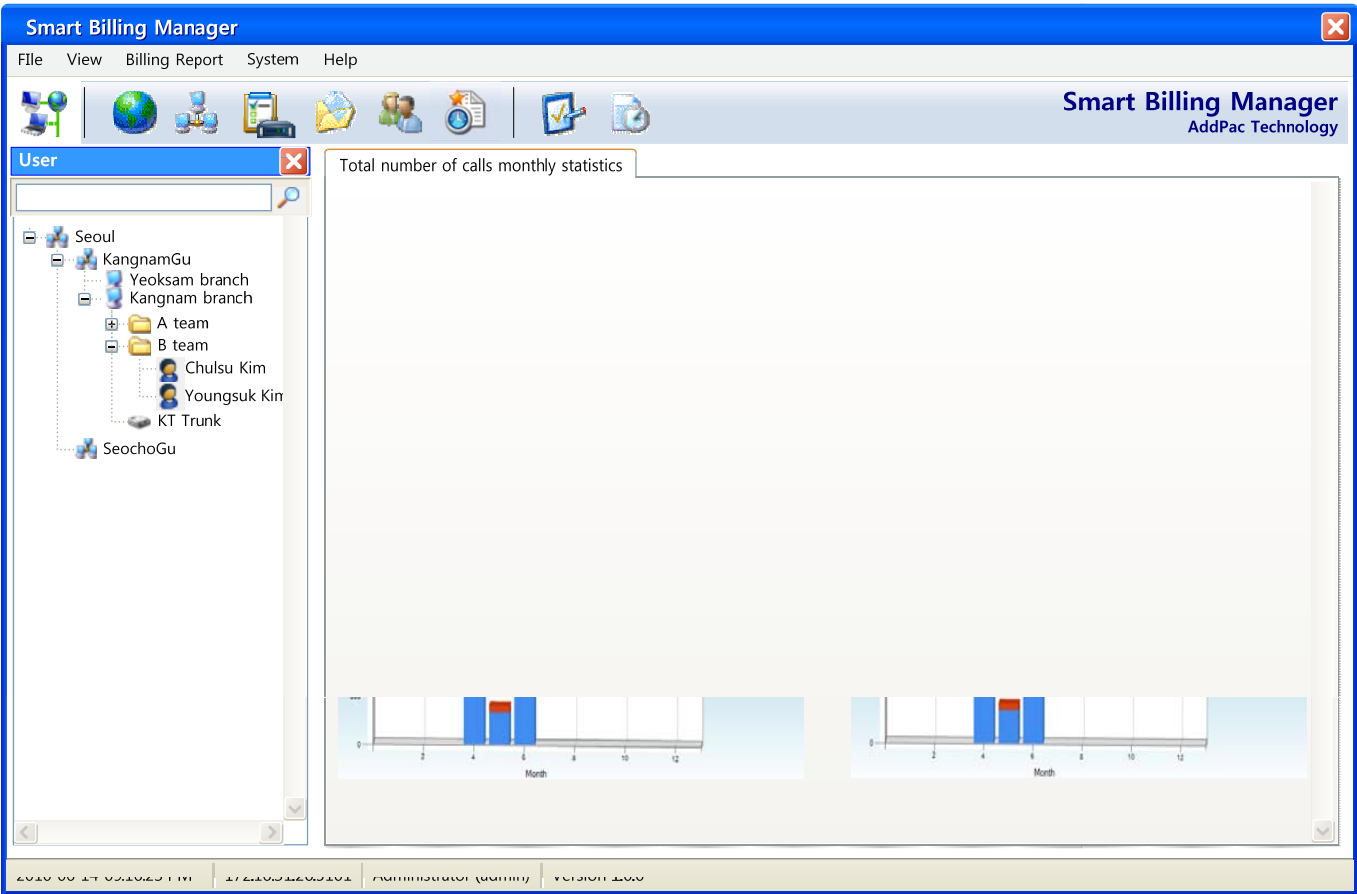
# Call Report Generation



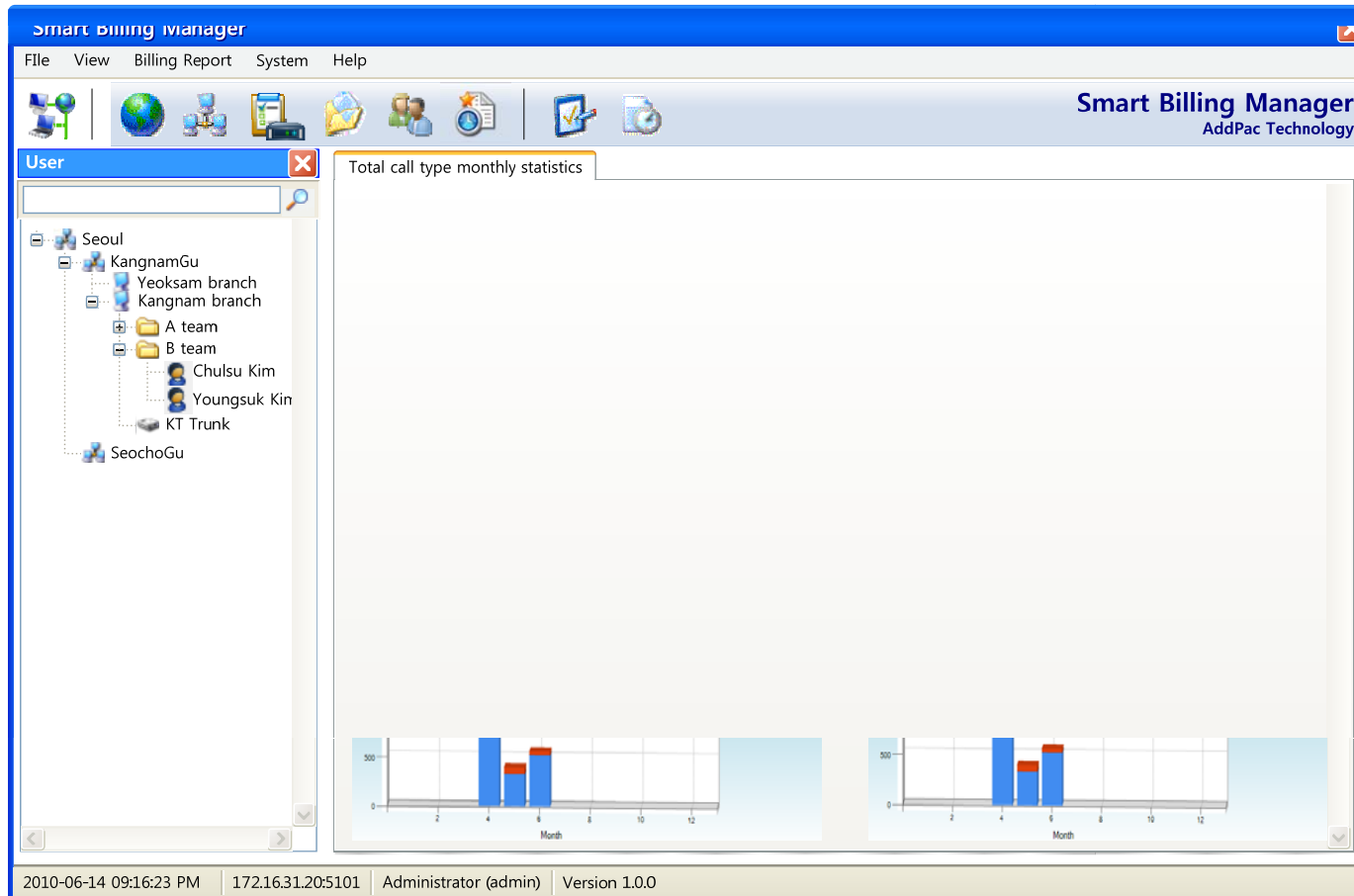
# Call Report Generation



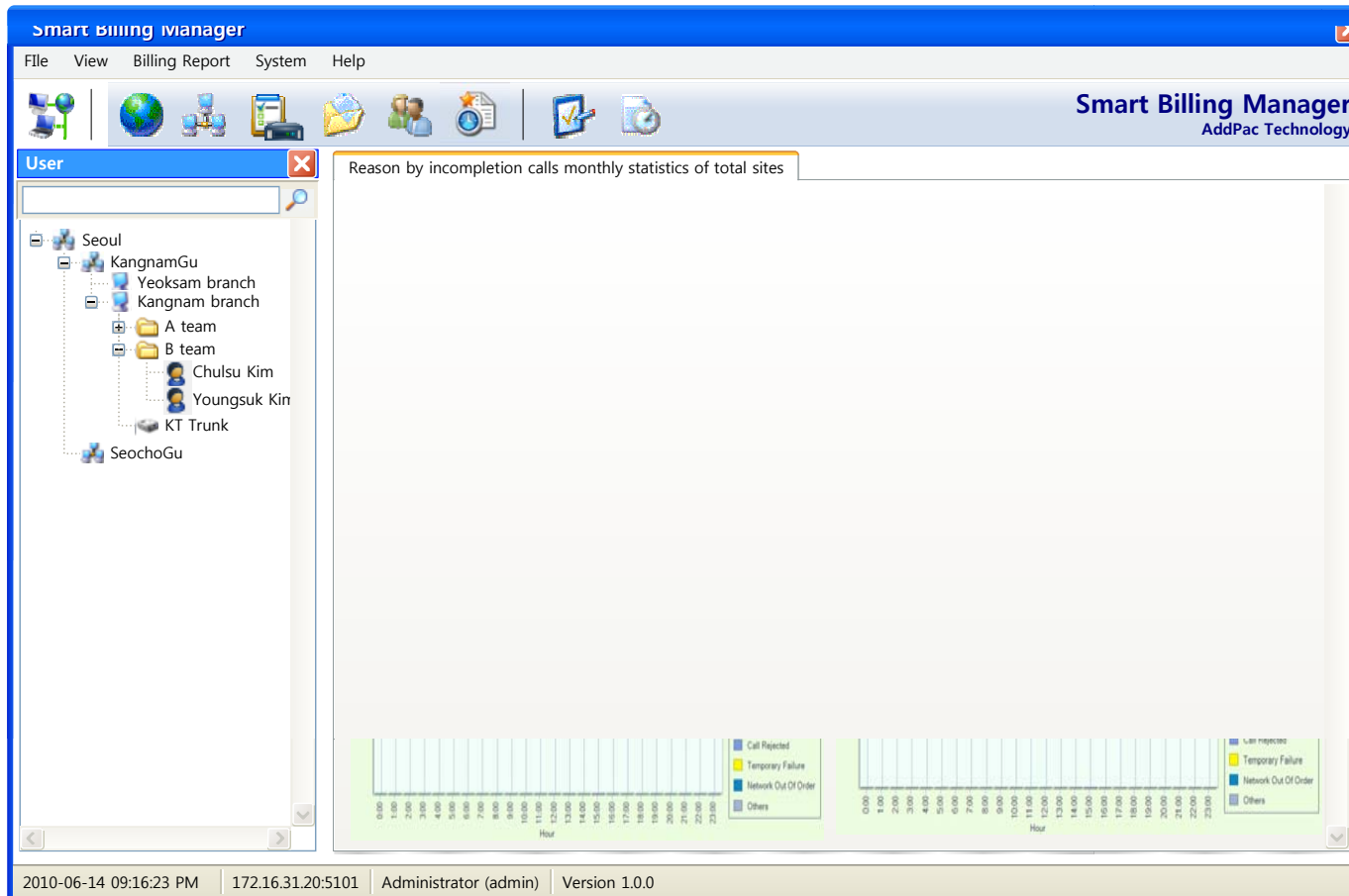
# Call Report Generation



# Call Report Generation



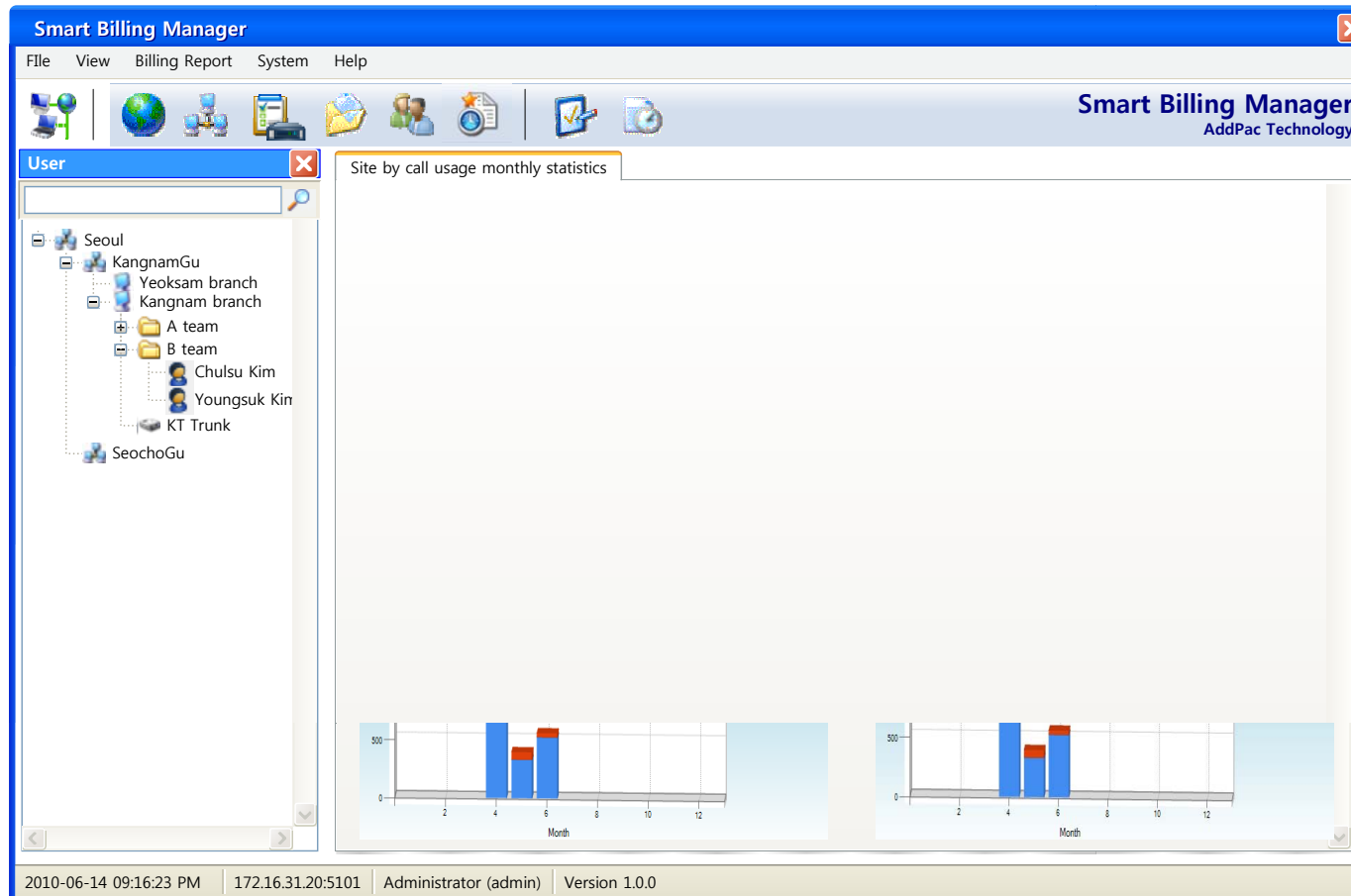
# Call Report Generation



# Call Report Generation

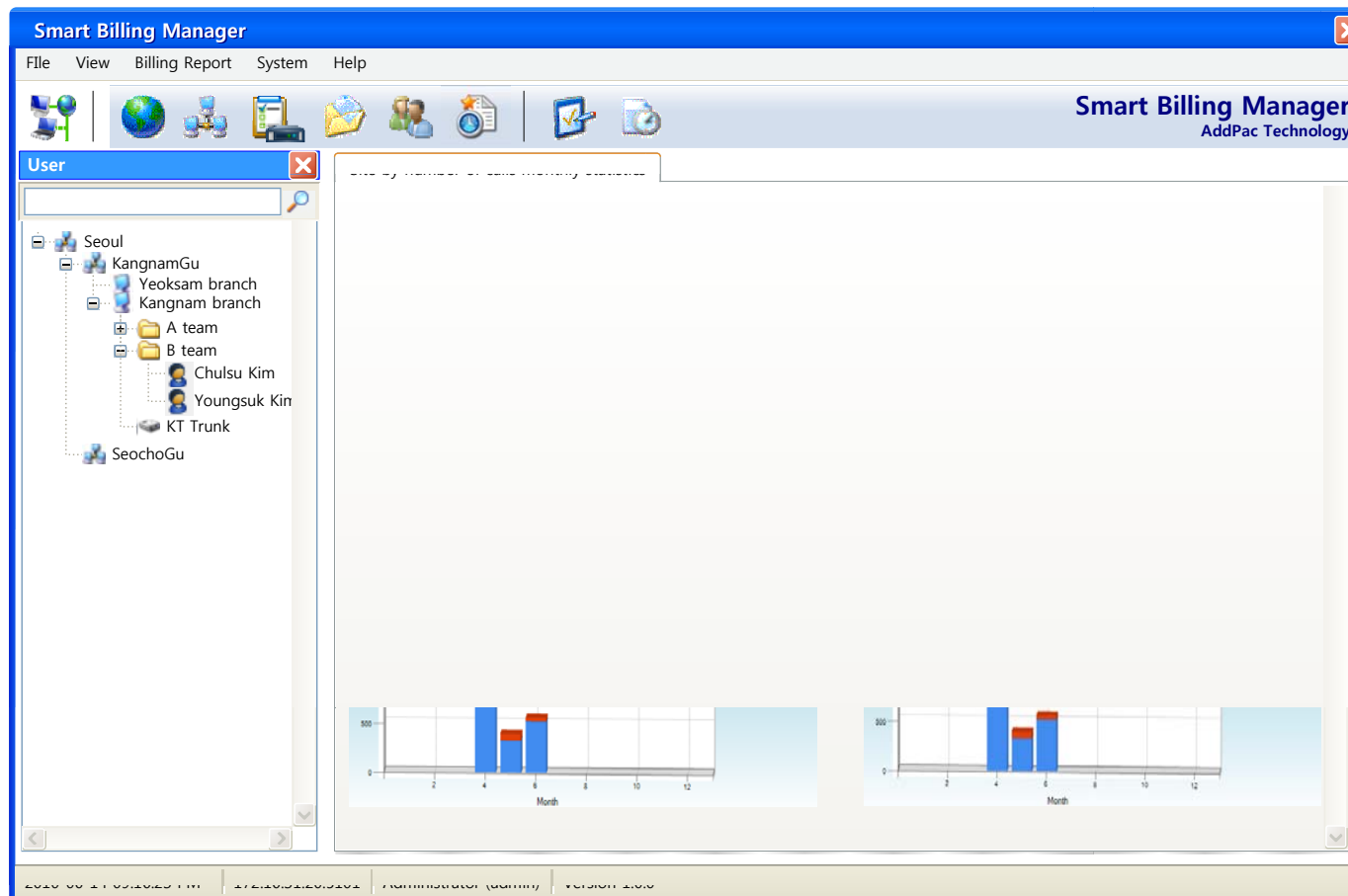
The screenshot displays the 'Smart Billing Manager' application window. The title bar reads 'Smart Billing Manager' with a close button. The menu bar includes 'File', 'View', 'Billing Report', 'System', and 'Help'. The toolbar contains various icons for navigation and reporting. The main interface is divided into two panes. The left pane, titled 'User', shows a hierarchical tree structure of users and locations: Seoul, KangnamGu (with sub-branches Yeoksam branch and Kangnam branch), A team, B team, Chulsu Kim, Youngsuk Kim, KT Trunk, and SeochoGu. The right pane, titled 'Site by call charge view', is currently empty. At the bottom of the window, a status bar shows the date and time '2010-06-14 09:16:23 PM', the IP address '172.16.31.20:5101', the user 'Administrator (admin)', and the version 'Version 1.0.0'. There are also two small pie charts at the bottom of the main content area, one partially visible on the left and one on the right, with a legend below them showing a purple segment labeled 'Samsung branch'.

# Call Report Generation

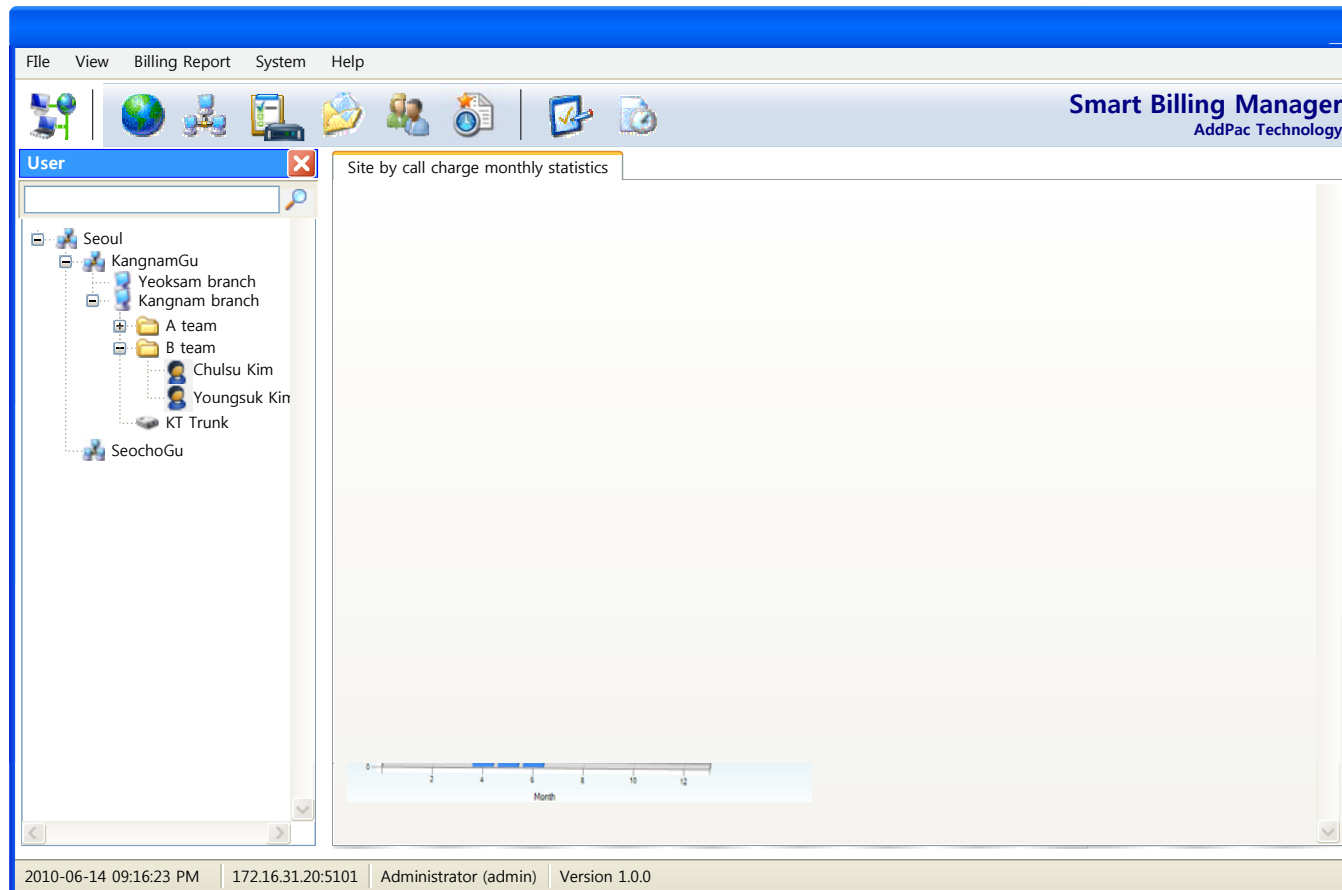




# Call Report Generation

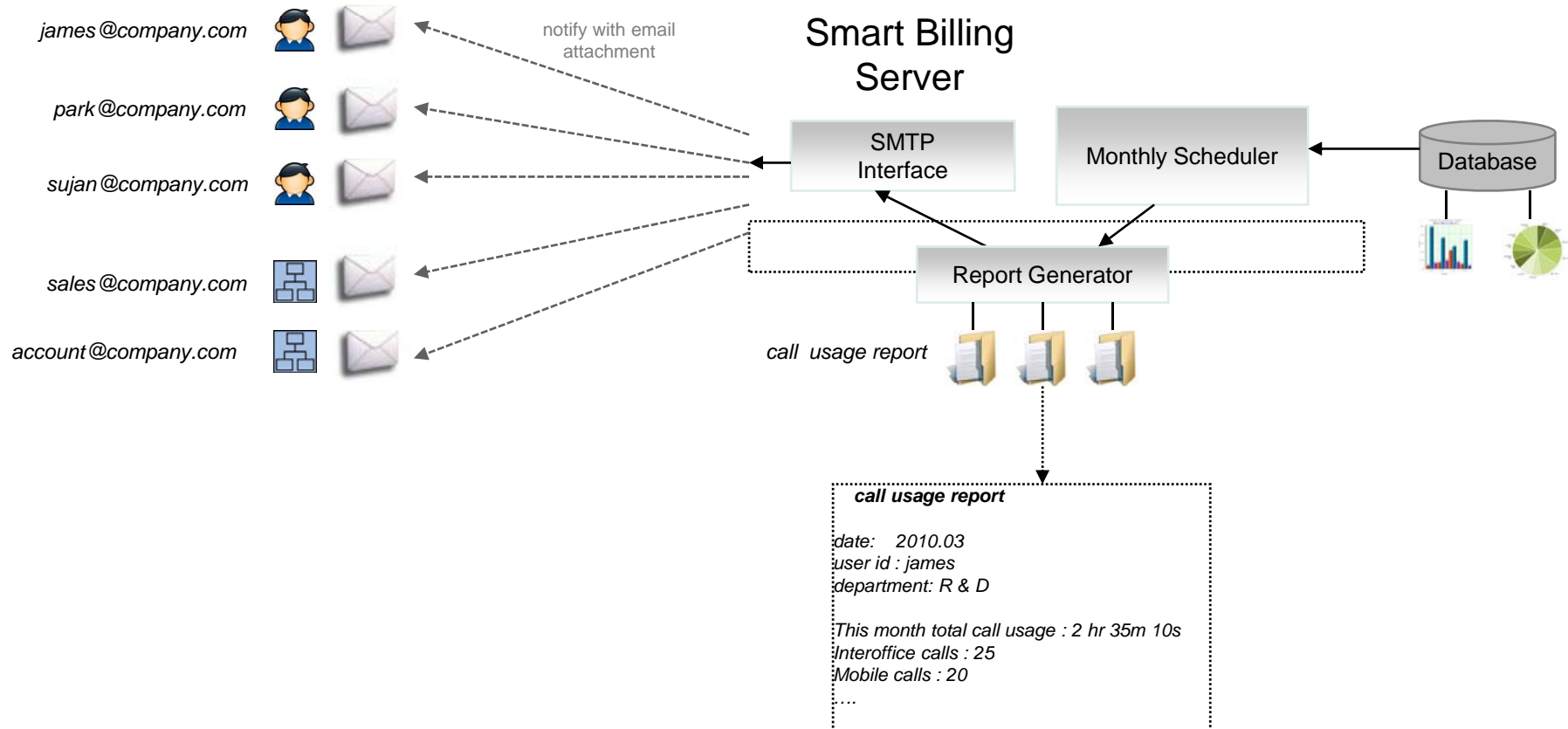


# Call Report Generation



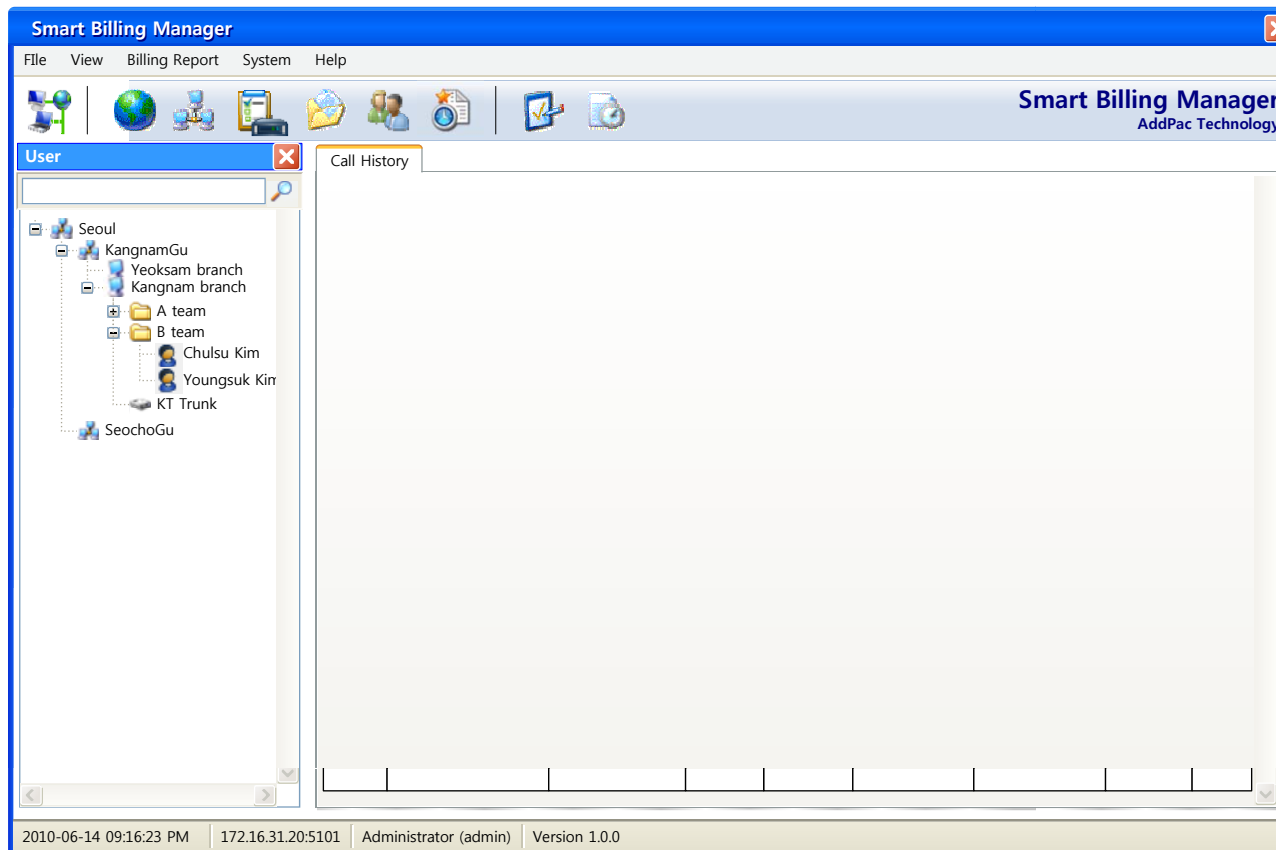
# Call Report Notification

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



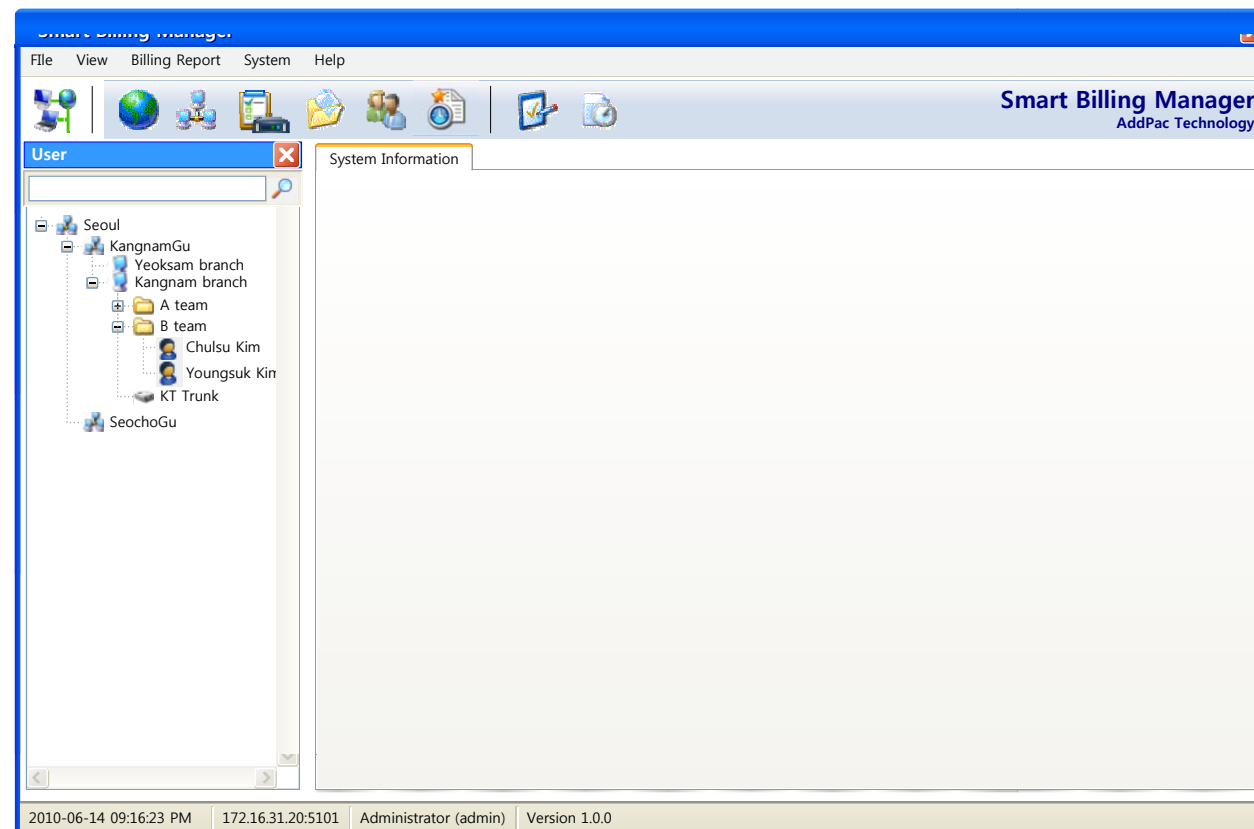
# Call History

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



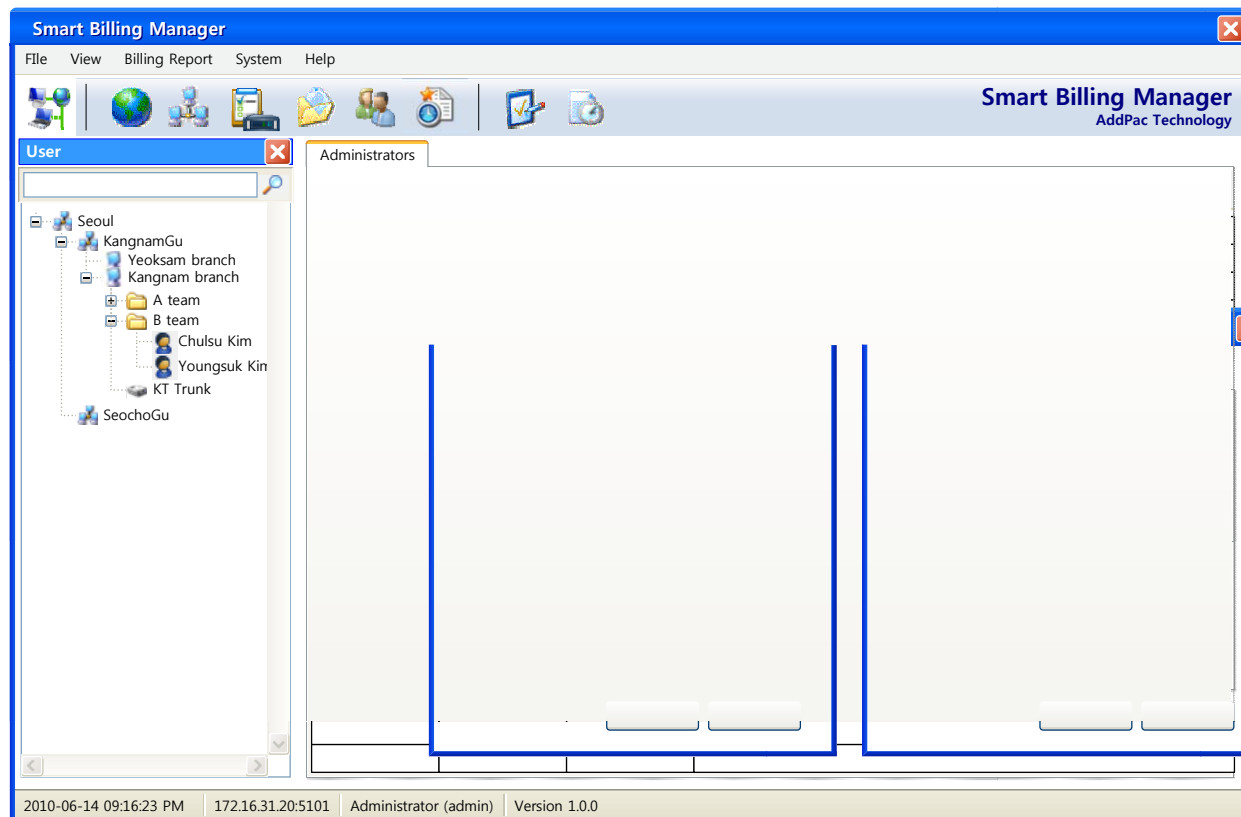
# System Performance Monitoring

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



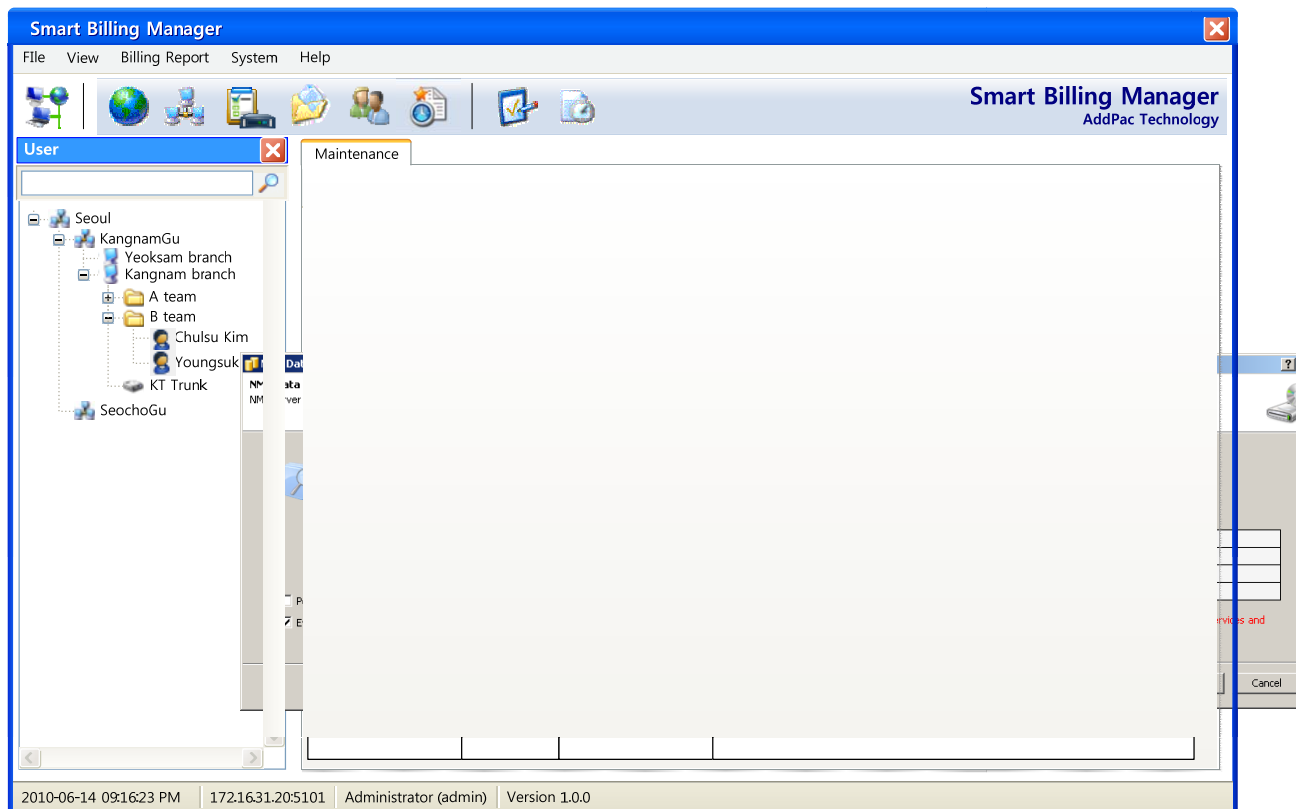
# Administrator Access Level Management

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



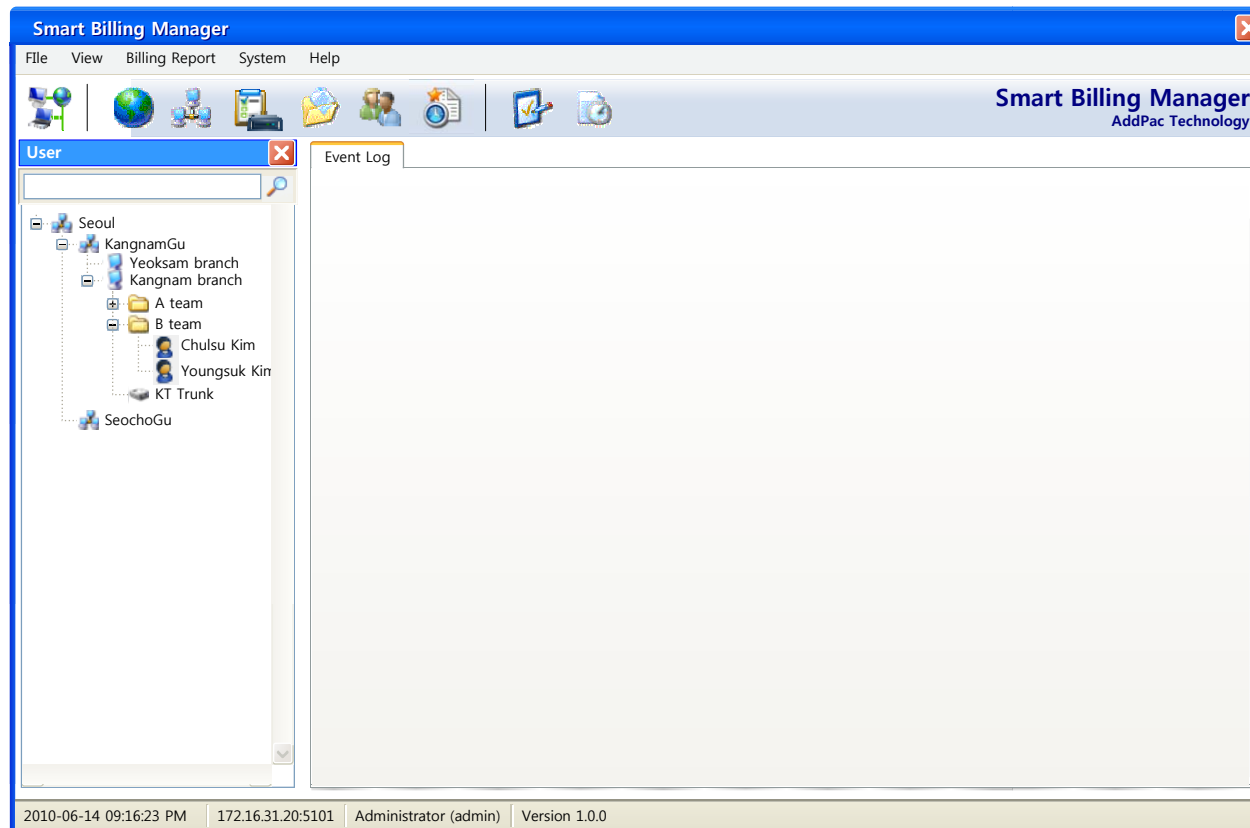
# Billing Operating Database Backup

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



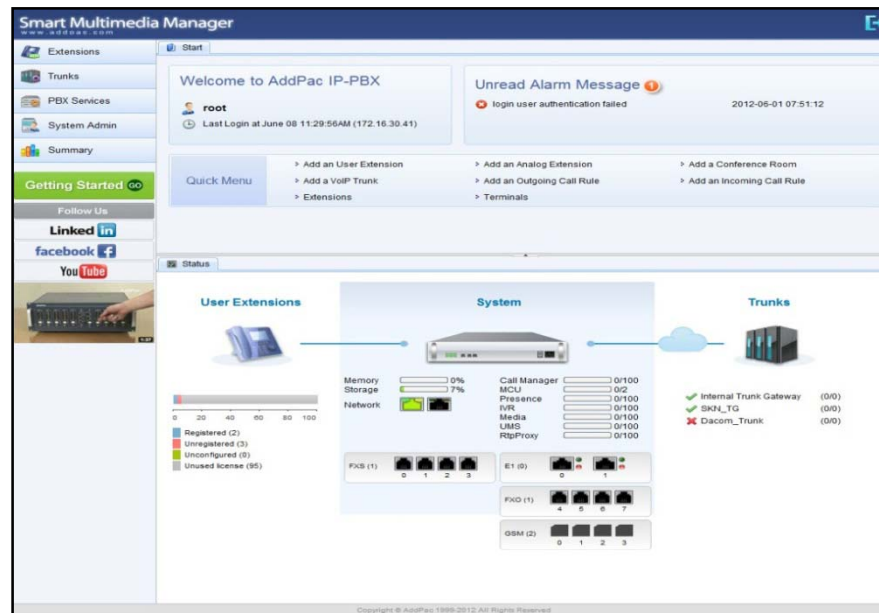
# View System Event Log

- 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 3<sup>rd</sup> party systems
- Integrated voice line management such as **FXS, FXO, E1, GSM, 3G**
- **User portal** to configure personal information, call forwarding
- **Diagnostic tool** to analyze SIP Call flow, current status and problems for terminal and trunk

# System Requirement

## **WSMM** (Web based Smart Multimedia Manager)

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

# Login

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

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

# Help

The screenshot shows the 'Smart Multimedia Manager' interface. The main content area is titled 'Add an User Extension' and contains a form with fields for Extension, First Name, Last Name, Voice Mail Password, User Password, Department, Title, Email, Home Phone, Mobile Phone, User ID, Photo, and Routing Access List. Below the form are 'Advanced Options' and 'General Settings' sections. A help window is open over the form, displaying the title 'Help :: User Extension' and the following content:

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

At the bottom of the screenshot, there is a yellow box with the following text:

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


# Related Links

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

**Modify the User Extension**

Apply Cancel Advanced Options

**User Extension**

|                       |  |                               |
|-----------------------|--|-------------------------------|
| Extension *           | 1009   | 3 - 8 digits                  |
| First Name            | ByoungGoo  |                               |
| Last Name *           | Choi   |                               |
| Voice Mail Password * | ****   | 4digits and user portal login |
| User Password *       | 1111   | For SIP registration          |
| Department            | root   | Search                        |
| Title                 |  | ex) manager                   |
| Email                 |  | ex) admin@addpac.com          |
| Home Phone            |  | ex) 123-456-7890              |
| Mobile Phone          |  | ex) 123-456-7890              |
| User ID               |  | SIP registration ID           |
| Photo                 |  (Maximum File Size: 100KB)<br>Select Photo |                               |

**Routing Access List**

Routing Access List: internal

**Advanced Options**

Terminal Profile: default

General Settings

Security Profile: default

Use RTP Proxy:

Back Tone at:

Representation: Default

**Description**

A user extension is an IP Phone (SIP / SSCP phone) or a soft phone for end user. It is composed of user profile, phone number and terminal belongs to the user.

**Related Links**

- WSMM User Portal
- Routing Access Lists
- Terminal Profiles
- Security Profiles
- Pickup Group

**Related Links**

- WSMM User Portal
- Routing Access Lists
- Terminal Profiles
- Security Profiles
- Pickup Group

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

# Diagnostic

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

**Terminal Diagnostic** 1009 (172.16.18.100)

You can check network connectivity from the PBX to the terminal by **Network Connectivity Test** and also you can check SIP awareness of the terminal by checking response message from the terminal by **SIP Aware Test**.

**Step 1.**

- 1. Network Connectivity Test** Successfully pinged 172.16.18.100 which is just provisioned to phone. Reply from 172.16.18.100: time=100ms loss=0% **Succeeded**
- 2. SIP Aware Test** This phone '172.16.18.100' is successfully responding SIP OPTIONS. **Succeeded**

At this step, you can make a test call on the diagnostic terminal to some destination number. If this terminal has problem on local call, the destination could be a local extension otherwise the destination could be mobile or PSTN number. The call trace shows information whether the call is properly handled or not. This test call can be traced only one administrator at same time and simultaneous test call will not be allowed.

1005 **Start Outbound Test**

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

2012-06-12 20:15:36 deviceId: 70 caller: 1009 callee: 1005 Call Test Start.  
----- From 1009 (172.16.18.100:5060) -----  
\*\*\*\*\*  
SIP/2.0 200 OK  
Via: SIP/2.0/UDP 172.16.17.30:5060;branch=z9hG4bKd84f0b0fa411  
From: <sip:dial-service@172.16.17.30>;tag=d84f0b0fa4  
To: <sip:1009@172.16.18.100>;tag=dc4fa2c5a4  
Call-ID: dca3d74f-519d-a2e8-80c5-0002a4038e2c@172.16.18.100  
CSeq: 11 INVITE  
Session-Expires: 1800;refresher=uac  
User-Agent: AddPac SIP Gateway  
Contact: sip:1009@172.16.18.100  
Require: timer  
Content-Type: application/sdp  
Content-Length: 179  
  
v=0  
o=1009 1339532254 1339532254 IN IP4 172.16.18.100  
s=AddPac Gateway SDP  
c=IN IP4 172.16.18.100  
t=1339532254 0  
  
/AVP 0  
0000/1  
(172.16.18.100:5060) -----  
  
172.16.17.30:5060;branch=z9hG4bKd84f0b0fa411  
dial-service@172.16.17.30>;tag=d84f0b0fa4  
172.16.18.100>;tag=dc4fa2c5a4  
f-519d-a2e8-80c5-0002a4038e2c@172.16.18.100  
  
1800;refresher=uac  
ac SIP Gateway  
@172.16.18.100  
  
lication/sdp  
179

**Step 2.**

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**Diagnostic**  
It provides to display terminal and trunk status inspection in IP-PBX

**Step 1.**

- Network Connection Test
- SIP Aware Test

**Step 2.**

- Outgoing Call Test



# Built-in IVR Scenario Editor

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

**IVR Scenarios**  
Apply Save Cancel

**IVR Scenario Properties**  
Name: addpac  
Description:

**IVR Scenario sequence**

- Start
  - Play
  - Menu (AddPac)
    - Multi
      - Check Extension
        - TRUE
          - Play
          - Transfer
        - FALSE
          - Play (Wrong Number)
          - Goto
      - 0
        - Play (Connect)
        - Transfer
      - 1
        - Play (Connect)
        - Transfer (Voice Mail)
      - 2
        - Play (Announcement)
        - Transfer
      - No Match
        - Play (Thank you )
        - Disconnect
      - No Input
        - Play (Please Press Number)
        - Goto

**Menu**

This action inputs a single digit or multiple digits from user phone and branches to an event handle by matching input digit.

Name \* AddPac  
File Path hello\_full .Open  
 Cancelable  
If this option is enabled, you can stop the sound by pressing any key.  
Initial Timeout 10  
Allowable Count 5

Single Digit  
Add Single Digit Event of 3

Multi Digit  
Add Multi Digit Event with Inter Digit Timeout 1 Sec  
and Max Digit Length 4

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

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**Built-in IVR Scenario Editor**  
WSMM is embedded with IVR Scenario Editor.  
An administrator may create/edit IVR scenario without using special tool  
**IVR Scenario Sequence**

- Start
- Menu / Play / Transfer / Check Extension / Goto / Disconnect

# Main

The screenshot shows the 'Smart Multimedia Manager' interface for an 'AddPac IP-PBX'. The interface is divided into several sections:

- Left Navigation Menu:** Contains 'Extensions', 'Trunks', 'PBX Services', 'System Admin', and 'Summary'. A red dashed box highlights this menu.
- Start Page:**
  - Welcome to AddPac IP-PBX:** Shows the user 'root' and their last login time: 'Last Login at June 08 11:29:56AM (172.16.30.41)'.
  - Unread Alarm Message:** Displays a message: 'login user authentication failed' with a timestamp of '2012-06-01 07:51:12'.
  - Quick Menu:** Lists actions such as 'Add an User Extension', 'Add an Analog Extension', 'Add a Conference Room', 'Add a VoIP Trunk', 'Add an Outgoing Call Rule', 'Add an Incoming Call Rule', and 'Extensions'.
- Status Page:**
  - User Extensions:** Shows a progress bar for 'Registered (2)', 'Unregistered (3)', 'Unconfigured (0)', and 'Unused license (95)'.
  - System:** Displays resource usage for 'Memory Storage' (0%), 'Network', and 'Call Manager' (0/100). It also lists other system components like MCU, Presence, IVR, Media, UMS, and RtpProxy.
  - Trunks:** Shows the status of various trunks: 'Internal Trunk Gateway (0/0)', 'SKN\_TG (0/0)', and 'Dacom\_Trunk (0/0)'.

Annotations on the left side of the screenshot:

- Main Menu:** Through left "Main Menu", setup IP-PBX policy.
- Alarm Message:** It displays IP-PBX system errors.
- Short Cut:** A short cut link.
- Status:** It displays current IP-PBX system major status.

# Main - Alarm History

The screenshot shows the Smart Multimedia Manager interface. The top section displays a welcome message for 'root' and an 'Unread Alarm Message' for 'login user authentication' on 2012-06-01 07:51:12. A red dashed box highlights this message, with a blue arrow pointing down to the 'Alarm History' page below. The 'Alarm History' page features a table with columns for Level, Messages, and DateTime, showing a list of system events. A yellow callout box provides instructions on how to access the alarm history from the main page.

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

| Level | Messages  | DateTime            |
|-------|---|---------------------|
| 1     | Minor NTP time sync service started !           | 2012-06-01 07:54:35 |
| 2     | Major ftp service disabled by operator          | 2012-06-01 07:54:02 |
| 3     | Minor network interface fastethernet 0/0 now up | 2012-06-01 07:53:35 |
| 4     | Major disk upper quota limit exceeded           | 2012-06-01 07:53:04 |
|       | Threshold exceeded !                            | 2012-06-01 07:51:55 |
|       | Authentication failed                           | 2012-06-01 07:51:12 |

# Main – Quick Menu

The screenshot displays the 'Smart Multimedia Manager' web interface. On the left is a navigation sidebar with options like 'Extensions', 'Trunks', 'PBX Services', 'System Admin', and 'Summary'. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root' and an 'Unread Alarm Message' regarding a failed login. Below this is a 'Quick Menu' section with several options: 'Add an User Extension', 'Add a VoIP Trunk', 'Extensions', 'Add an Analog Extension', 'Add an Outgoing Call Rule', 'Terminals', 'Add a Conference Room', and 'Add an Incoming Call Rule'. A red dashed box highlights the 'Quick Menu' and the 'Add an User Extension' form below it. A blue arrow points from the 'Add an User Extension' link in the Quick Menu to the form. The form includes fields for Extension, First Name, Last Name, Voice Mail Password, User Password, Department, Title, Email, and Home Phone. A 'Description' box explains that a user extension is an IP Phone or soft phone. A 'Related Links' section lists 'Routing Access Lists', 'Terminal Profiles', 'Security Profiles', and 'Pickup Group'. A yellow callout box at the bottom left provides a definition for the 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

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

**Getting Started** **GO**

**Follow Us**

- LinkedIn
- facebook
- YouTube

**Start**

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

Add a Conference Room

Add an Incoming Call Rule

**Trunks**

**Registered (2)**

**Unregistered (3)**

**Unconfigured (0)**

**Unused license (95)**

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



# Main – Status Monitoring

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

**Start**

Welcome to AddPac IP-PBX

**Unread Alarm Message**  
login user authentication failed 2012-06-01 07:51:12

**Quick Menu**

- Add an User Extension
- Add an Analog Extension
- Add a Conference Room
- Add a VoIP Trunk
- Add an Outgoing Call Rule
- Add an Incoming Call Rule
- Extensions
- Terminals

**Status**

**User Extensions**

**System**

Memory Storage 0%  
Network 7%

Call Manager  
MCU Manager  
Presence  
IVR  
Media  
UMS  
RtpProxy

**Trunks**

**Voice Lines**

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

| Slot / Port | Type | Status | Pattern | Connection PLAR | Input Gain | Output Gain | Check Source | Protocol Emul | Modify |
|-------------|------|--------|---------|-----------------|------------|-------------|--------------|---------------|--------|
| 1 0/0/0     | E1   | down   |         |                 | 0          | 0           | Master       | Network       |        |
| 2 0/1/0     | E1   | down   |         |                 | 0          | 0           | Master       | Network       |        |

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

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

# Extension - Extensions

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

Start

Welcome to AddPac IP-PBX

Unread Alarm Message  
No Unread Alarm Message

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

Quick Menu

Status

User Extension

Getting Started

Follow Us

Linked in

facebook

YouTube

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

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

Start Extensions

Extensions

All Extensions Input an Extension Search Add an Extension Refresh

|   | Modify | Delete | Extension Number | Type             | Name            | Date Created        |
|---|--------|--------|------------------|------------------|-----------------|---------------------|
| 1 |        |        | 1007             | User Extension   | Jinsuk Choi     | 2012-06-08 17:54:53 |
| 2 |        |        | 1009             | User Extension   | ByoungGoo Choi  | 2012-06-08 17:58:05 |
| 3 |        |        | 3000             | User Extension   | BongYoung Jeong | 2012-06-08 17:59:14 |
| 4 |        |        | 1008             | User Extension   | SeongHyun Lee   | 2012-06-08 18:59:48 |
| 5 |        |        | 1010             | Analog Extension | JongHwee Kwon   | 2012-06-08 18:36:34 |
| 6 |        |        |                  | Conference Room  | Ad-Hoc Defaults | 1999-11-30 08:00:00 |
| 7 |        |        | 0001             | Voice Mail       | vmal_rec        | 2012-06-08 17:49:53 |
| 8 |        |        | 0002             | Voice Mail       | vmal            | 2012-06-08 17:49:54 |
| 9 |        |        | 0003             | Voice Mail       | vmal_noauth     | 2012-06-08 17:49:54 |

Add an Extension

Cancel

**Analog Extension**  
An analog extension is a kind of user extension who has FXS (Foreign eXchange Station) analog voice line. Normal analog phone is connected at this extension.

**Hunt Group**  
A hunt group has members of user extensions. Within a hunt group, an available member (user extension) can receive a call to the hunt group extension. A hunt group has one of simultaneous, sequential or random call hunting mode.

**Pickup Group**  
A pickup group has members of user extensions who can pick up a ringing call within the group. The pickup group extension number is used for picking up a call by other group member.

**Conference Room**  
A conference room extension is used for making a conference room. In case of dial-out conference, when a privileged user calls to conference room extension, all conference participants receive call to join. In case of meet-me conference, conference participants call to conference extension to join.

**IVR Extension**  
An IVR (Interactive Voice Response) extension has a role of auto attendant for incoming calls from trunks. If incoming calls from trunk are routed to an IVR extension by incoming call rule, the interactive scenario will be proceed to transfer the call to a proper user extension.

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

# Extension - Directory

**Smart Multimedia Manager**

Welcome to AddPac IP-PBX

Unread Alarm Message  
No Unread Alarm Message

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

**Directory**

| Name             | Extension | Notes                       |
|------------------|-----------|-----------------------------|
| BongY Jeong      | 1101      |                             |
| SeongHyun Lee    | 1008      |                             |
| 연구소              |           |                             |
| Hardware         |           |                             |
| DongHee Jang     | 1020      |                             |
| Smart work       |           |                             |
| Smart Management |           |                             |
| BongYong Jeong   | 3000      | Hello ~ I am Jeong BongYong |
| Smart Framework  |           |                             |
| BY Jeong         | 1100      |                             |
| SangGyun Lee     | 1005      |                             |
| HyungSuk Oh      | 1006      | Have a nice day ~           |
| ByoungGoo Choi   | 1009      |                             |

**User Extension Status:**

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

**System Status:**

- Memory Storage: 1%
- MCU: 8%
- Network: [Icon]
- Call Manager: 0/100
- MCU: 0/2
- Presence: 0/100
- IVR: 0/100
- Media: 0/100
- UMS: 0/100
- RtpProxy: 0/100
- FXS (1): 0, 1, 2, 3
- E1 (0): 0, 1
- FXO (1): 4, 5, 6, 7
- Internal Trunk Gateway: (0/0)
- SKN\_TG: (0/0)
- Dacom\_Trunk: (0/0)

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



# Extension - Routing Access List

**Smart Multimedia Manager**

Start | Directory | Routing Access Lists

### Routing Access Lists

Add a Routing Access List Refresh

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

Status | Routing Access List

#### Add a Routing Access List

Add Cancel

**Routing Access List**

Name \*

Description

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

| Outgoing Call Rules |  |
|---------------------|--|
| Name                |  |

| Allowed Outgoing Call Rules |  |
|-----------------------------|--|
| Name                        |  |

Allowed Outgoing Call Rules

Description

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

Related Links

- Outgoing Call Rules

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

# Extension - Terminal Profile

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with the following items: Extensions, Directory, Routing Access Lists, Terminal Profiles, Terminals, Trunks, PBX Services, System Admin, Summary, Getting Started, Follow Us (LinkedIn, Facebook, YouTube), and a status bar showing Registered (2), Unregistered (3), Unconfigured (0), and Unused license (95). The main content area is divided into two sections. The top section, titled 'Terminal Profiles', includes a table with columns for Modify, Delete, Name, Description, and Date Created. The bottom section, titled 'Global Terminal Settings', contains various configuration options such as Calling Party Presentation, Language, Call Duration Limit, Off-net Transfer, Digit Timeout Settings, and Distinctive Ring Settings. A yellow callout box in the bottom left corner provides additional information about the Terminal Profile settings.

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

# Extension - Terminals

**Smart Multimedia Manager**

Welcome to AddPac IP-PBX

Unread Alarm Message  
No Unread Alarm Message

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

**Terminals**

|   | Modify | Delete | Diagnose | Extension | Name            | User Agent     | IP Address    | State        | MAC Address    | Create Time         |
|---|--------|--------|----------|-----------|-----------------|----------------|---------------|--------------|----------------|---------------------|
| 1 |        |        |          | 1007      | Jinsuk Choi     |                |               | Unregistered |                | 2012-06-08 17:54:53 |
| 2 |        |        |          | 1008      | SeongHyun Lee   |                |               | Unregistered |                | 2012-06-08 18:59:49 |
| 3 |        |        |          | 1010      | JongHwee Kwon   | AddPac SIP ... | 172.16.17.30  | Unregistered |                | 2012-06-08 18:36:35 |
| 4 |        |        |          | 1009      | ByoungGoo Choi  | AddPac AP-V... | 172.16.18.100 | Registered   | 0002.a403.8... | 2012-06-08 17:58:06 |
| 5 |        |        |          | 3000      | BongYoung Jeong | AddPac SIP ... | 172.16.18.101 | Registered   |                | 2012-06-08 17:59:15 |

**Modify the User Extension**

Apply Cancel Advanced Options

Extension \* 3000 3 ~ 8 digits

First Name BongYoung

Last Name \* Jeong

Voice Mail Password \* \*\*\*\* 4digits and user portal login

User Password \* 1111 For SIP registration

Department Search

Title ex) manager

Email ex) admin@addpac.com

Home Phone ex) 123-456-7890

Mobile Phone ex) 123-456-7890

User ID SIP registration ID

Photo (Maximum File Size: 100KB)

**Description**  
A user extension is an IP Phone (SIP / SSCP phone) or a soft phone for end user. It is composed of user profile, phone number and terminal belongs to the user.

**Related Links**

- WSMM User Portal
- Routing Access Lists
- Terminal Profiles
- Security Profiles
- Pickup Group

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

# Trunk - Trunks

The screenshot shows the Smart Multimedia Manager interface. The left sidebar contains navigation options: Extensions, Trunks, PBX Services, System Admin, Summary, Getting Started, and social media links. The main content area is divided into two sections. The top section, titled 'Welcome to AddPac IP-PBX', shows the user 'root' with a last login time of June 11 04:38:52AM (172.16.1.50) and an 'Unread Alarm Message' notification. The bottom section, titled 'Trunks', displays a table of existing trunks and an 'Add a Trunk' form.

| Modify | Delete | Diagnose | Name                   | Type             | IP Address    | State        | Description | Date Created      |
|--------|--------|----------|------------------------|------------------|---------------|--------------|-------------|-------------------|
| 1      |        |          | Internal Trunk Gateway | VoIP Trunk       | 127.0.0.1     | Registered   |             | 2012-06-08 17:... |
| 2      |        |          | Dacom_Trunk            | SIP Proxy Server | 172.16.19.201 | Unregistered |             | 2012-06-08 18:... |
| 3      |        |          | SKN_TG                 | VoIP Trunk       | 172.16.19.200 | Registered   |             | 2012-06-08 18:... |

**Add a Trunk**

- VoIP Trunk**  
This is a generic VoIP Trunk which can register to this PBX or communicate without registration. The VoIP Trunk could be VoIP gateway which has analog FXS, FXO, E&M line, digital E1, T1 line or mobile GSM line, or IP-PBX or other SIP / H.323 Trunk.
- SIP Proxy Server**  
This could be VoIP service provider who operates SIP Proxy Server and provides VoIP service to public telephone network or mobile network or other VoIP network. Also, this could be an IP-PBX who provides SIP server features. This PBX should register to the SIP Proxy Server for receiving incoming calls and sending outgoing calls.
- H.323 Gatekeeper**  
This could be VoIP service provider who operates H.323 Gatekeeper and provides VoIP service to public telephone network or mobile network or other VoIP network. Also, this could be an IP-PBX who provides H.323 Gatekeeper features. This PBX should register to the H.323 Gatekeeper for receiving incoming calls and sending outgoing calls.

**Description**  
Using the trunks, user extensions in this PBX can communicate with remote users in public telephone network or mobile network or other VoIP network including branches.

**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.

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

**Add an Outgoing Call Rule**

Name \*

Patterns \* {[0-9#\*]}|T

Trunks of Outgoing Call \*

Called Number Translation Add Rule

Number Translation

Calling Number Translation Add Rule

Description: An Outgoing Call Rule controls outgoing call routing to a specific trunk. An outgoing call from user extension can be routed to trunk by selecting an Outgoing Call Rule which has matched pattern with dialed digits of the call. Also, an incoming call from a trunk can be applied to Outgoing Call Rules by an Incoming Call Rule for routing to other trunk.

Related Links: • Trunks



# Trunk - Incoming Call Rules

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

Start

Welcome to AddPac IP-PBX

Unread Alarm Message  
No Unread Alarm Message

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

Quick Menu

Status

User Extension

Getting Started GO

Follow Us

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

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

Status Incoming Call Rules

Add an Incoming Call Rule

Add Cancel

Name \*

Internal Trunk Gateway  
SM\_SIP\_Provider  
ss  
jschoLgk

Incoming Call Rule

Trunks of Incoming Call \*

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

Route to an extension by called number + Add Rule

Transfer Rule Modify Delete

Single Extension Routing

Route to an extension by calling number + Add Rule

Transfer Rule Modify Delete

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

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

Route DID Rule Modify De

Multiple Extension Routing(DID)

The incoming call from a trunk can be routed to other trunks by applying Outgoing Call Rules.

Called Pattern to delete digits from the front and adding

Trunk Routing to outgoing call rules external rule

Description

The Incoming Call Rule controls incoming call routing from specific trunks by looking up calling party number and called party number of the call. Applying this rule, the incoming calls are routed to IVR extension, user extensions, or other trunks. Using malicious call filter, call might be dropped.

Related Links

- Trunks
- Outgoing Call Rules

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**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

The screenshot displays the Smart Multimedia Manager interface. On the left, a navigation menu highlights 'PBX Services' and 'Speed Button Profiles'. The main content area shows the 'Speed Button Profile' configuration page. A table lists existing profiles, and a form below allows for adding or editing a profile. A yellow starburst icon is placed over the 'Speed Button Profiles' menu item, with a blue arrow pointing to the 'Add a Speed Button' button in the configuration form.

| Modify | Delete | Name           | Description | Date Created        |
|--------|--------|----------------|-------------|---------------------|
|        |        | button profile |             | 2012-04-02 10:43:18 |

| Name | Extension | Type      | Type | Modify | Delete |
|------|-----------|-----------|------|--------|--------|
|      |           | Extension | N/A  |        |        |

**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

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

Start

Welcome to AddPac IP-PBX

root  
Last Login at June 11 01:51:12PM (127.0.0.1)

Quick Menu

Status

User Extension

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

**Announcement and Tones**

Language: Korean | Restore Default Ments | Global Media Setting | Refresh

| Modify | ID     | Announcement           | Description   | Custom File | Scheduled |
|--------|--------|------------------------|---------------|-------------|-----------|
|        | 400110 | Connect                | 연결 중 안내       |             |           |
|        | 400120 | Retry                  | 내선 번호 재 시도 안내 |             |           |
|        | 400130 | No Number              | 없는 내선 안내      |             |           |
|        | 400140 | Over Count             | 최수 초과 안내      |             |           |
|        | 410110 | Greeting               | 인사말           |             |           |
|        | 410120 | Connected to attendant | 안내원과 연결       |             |           |
|        | 410130 | Connect 2              | 연결 중 안내       |             |           |
|        | 410140 | No Number 2            | 번호 입력 오류      |             |           |
|        | 410150 | Over Time              | 입력 내용 오류      |             |           |
|        | 420110 | Busy                   | 통화 중 안내       |             |           |
|        | 420120 | No Answer              | 부재 중 안내       |             |           |
|        | 420130 | System Normal Fail     | 통화 실패 안내      |             |           |
|        | 420400 | Thank you              | 미용 감사 안내      |             |           |
|        | 430110 | Press Password         | 비밀번호 입력 안내    |             |           |
|        | 430120 | Over Count 2           | 최수 초과 안내      |             |           |
|        | 430130 | Connect 3              | 연결 중 안내       |             |           |
|        | 430140 | Over Time 2            | 번호 입력 오류      |             |           |

Status | Modify the Announcement

Apply | Cancel

**Announcement and Tones**

Announcement ID: 400110  
Description: 연결 중 안내  
Language: Korea

Upload

| File List                           | File name                | File type | Media type | Version | Upload |
|-------------------------------------|--------------------------|-----------|------------|---------|--------|
| <input checked="" type="checkbox"/> | 400110_kr.audio.ulaw.wav | package   | audio      | 8.50    | 2012   |

Create New Schedule

| No. | Name | Start date | End Date | Start Time | End T |
|-----|------|------------|----------|------------|-------|
|-----|------|------------|----------|------------|-------|

Description

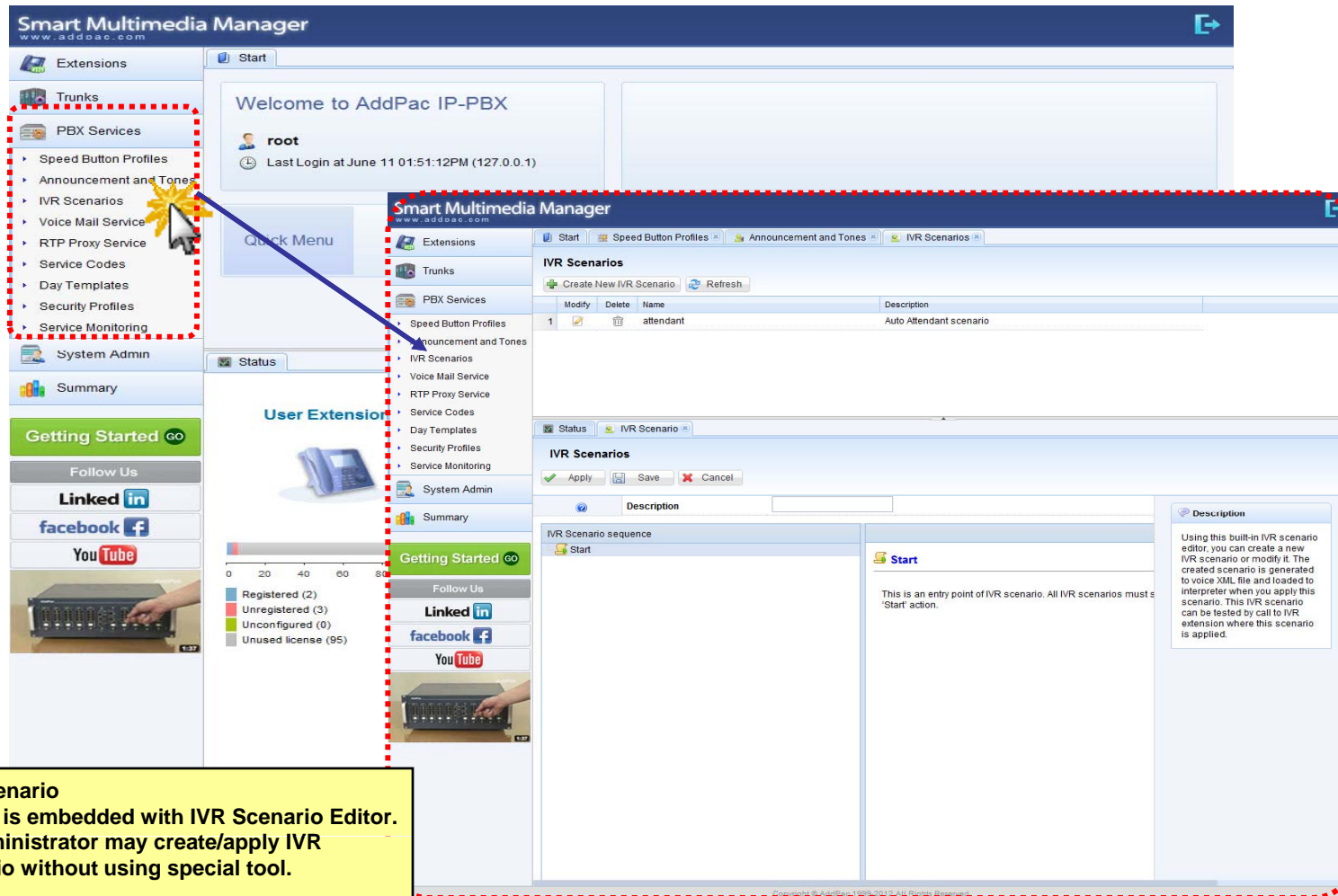
You can upload new announcements from your PC and each announcement can be assigned to time schedule.

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



# PBX Service - IVR Scenarios



The screenshot displays the Smart Multimedia Manager (WSMM) web interface. The left sidebar contains a navigation menu with 'PBX Services' expanded to show 'IVR Scenarios'. A red dashed box highlights this menu item, with a yellow starburst icon and a blue arrow pointing to the 'IVR Scenarios' section of the main content area. The main content area shows the 'IVR Scenarios' configuration page, which includes a table of existing scenarios, a 'Create New IVR Scenario' button, and a detailed view of an 'Auto Attendant scenario'.

| Modify | Delete | Name      | Description             |
|--------|--------|-----------|-------------------------|
|        |        | attendant | Auto Attendant scenario |

**IVR Scenarios**

Apply Save Cancel

Description

IVR Scenario sequence

Start

Start

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

Description

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

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

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

# PBX Service - Voice Mail Services

The screenshot displays the Smart Multimedia Manager interface. On the left, the 'PBX Services' menu is highlighted with a red dashed box, and a yellow starburst icon is placed over the 'Voice Mail Service' option. A blue arrow points from this menu item to the 'Voice Mail Service' configuration page on the right. The configuration page includes fields for 'Retrieving Extension by Other Phone', 'Retrieving Extension by Owner Phone', and 'Leave Extension'. It also features 'Advanced Options' for audio message length, per extension HDD quota, and notification settings. A 'Description' box on the right explains the purpose of the settings.

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

# PBX Service - RTP Proxy Service

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

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

**RTP Proxy Service**  
Apply Cancel

RTP Proxy Settings  
Idle Timeout: 600 (0-7200, default: 600sec)  
Packet Loss Event Count: 0 (0-65535, default: 0)

**\*IPv4**  
Add Network Domain

| Network Domain | Minimum | Maximum | DSCP | Modify | Delete |
|----------------|---------|---------|------|--------|--------|
|----------------|---------|---------|------|--------|--------|

**\*IPv6**  
Add Network Domain

| Network Domain | Minimum | Maximum | DSCP | Modify | Delete |
|----------------|---------|---------|------|--------|--------|
|----------------|---------|---------|------|--------|--------|

**Description**  
Manage RTP Proxy Service for NAT traversal. Normally, RTP proxying between private network and public network will be automatically handled by PBX. If you got problem to hear voice from remote side, enable option of RTP proxying in trunk setting or user extension setting.

**Getting Started** GO

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YouTube

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

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

# PBX Service - Service Codes

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

The screenshot shows the 'Smart Multimedia Manager' interface. On the left, the 'PBX Services' menu is highlighted with a red dashed box, and a mouse cursor points to 'Service Codes'. The main content area shows the 'Service Codes' configuration page with various settings for call forwarding, call reject, and call waiting. A 'Description' box on the right explains that a service code is a special digit (e.g., # or \*) used to activate services.

# PBX Service - Day Templates

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories like Extensions, Trunks, PBX Services, System Admin, and Summary. The 'PBX Services' category is expanded, and 'Day Templates' is highlighted. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root'. Below this, the 'Day Templates' section is active, displaying a table with one entry: 'holiday' created on 2012-03-30. A 'Day Template' dialog box is open, showing fields for 'Name' and 'Description', and a 'Description' tooltip that reads: 'Specify period or a special day(s) to apply in schedule policy.'

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



# PBX Service - Security Profiles

The screenshot shows the Smart Multimedia Manager web interface. The left navigation menu is highlighted with a red dashed box, and a blue arrow points from the 'Security Profiles' menu item to the main content area. The main content area displays the 'Security Profiles' configuration page, which includes a table of existing profiles and a 'Global Security Setting' section with a dropdown menu for 'TLS Cipher Suites'. The dropdown menu is open, showing a list of cipher suites.

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

**Global Security Setting**

Apply Cancel

**TLS Cipher Suites**

- N/A
- RC4\_40
- RC4\_128
- DES\_CBC
- 3DES\_CBC
- AES\_128\_CBC
- AES\_256\_CBC
- SEED\_CBC
- ARIA\_CBC

**Description**

In case of SIP, below cipher suites are used to negotiate with terminal for secure TLS. The cipher suites can have preferences as below three suites.

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**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

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

Extensions  
Trunks  
PBX Services

- Speed Button Profiles
- Announcement and Tones
- IVR Scenarios
- Voice Mail Service
- RTP Proxy Service
- Service Codes
- Day Templates
- Security Profiles
- Service Monitoring

System Admin  
Summary

Getting Started GO

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

Quick Menu

Service Monitoring  
Interval: 10 sec.

| ID | Established Time | Duration | Calling Number | Called Number | Audio Codec | Video Codec | Recording | Drop Call |
|----|------------------|----------|----------------|---------------|-------------|-------------|-----------|-----------|
|    |                  |          |                |               |             |             |           |           |

Active Calls | Conference

System Status

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

System

Memory Storage: 0%  
Network: 8%

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

Trunks

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

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

**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

The screenshot displays the 'Smart Multimedia Manager' web interface. The left sidebar contains a 'System Admin' menu with a red dashed box around it, and a yellow starburst icon pointing to the 'Network Interface' option. The main content area shows the 'Network Interfaces' configuration page, which is also highlighted with a red dashed box. The page includes a 'Welcome to AddPac IP-PBX' message, a 'User Extension' section, and a 'Getting Started' section. The 'Network Interfaces' configuration form is divided into two sections: 'WAN Interface' and 'LAN Interface'. The 'WAN Interface' section includes fields for 'Interface Mode' (DHCP or Static IP), 'IP Address', 'Subnet Mask', 'Default Gateway', 'Primary DNS Server', and 'Secondary DNS Server'. The 'LAN Interface' section includes fields for 'Interface Mode' (None, Bridge, IP Shared, NAT, or Static IP), 'IP Address', 'Subnet Mask', 'DHCP Server' (On or Off), 'DHCP Range', and 'IPv6 Address'. A 'Description' box on the right provides additional information about the interfaces.

**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

The screenshot shows the 'Smart Multimedia Manager' interface. The left sidebar contains a navigation menu with 'System Admin' highlighted. A red dashed box highlights the 'Network Services' option in the menu, with a yellow starburst icon and a blue arrow pointing to the configuration page. The main content area displays the 'Network Services' configuration page, which includes a 'Welcome to AddPac IP-PBX' message and a 'User Extension' status bar. The configuration page is divided into sections for NTP, TELNET, SNMP, HTTP, FTP, LDAP, and SYSLOG. Each section has a 'Service Enable' radio button and various configuration fields. A 'Description' box on the right states: 'You can change properties of system network services such as TELNET, SNMP, HTTP, FTP, LDAP, SYSLOG, and so on.'

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

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

# System Admin - Administrators

The screenshot shows the Smart Multimedia Manager interface. The left sidebar contains a 'System Admin' menu with sub-items: Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, and Show Command. The main content area displays the 'Administrators' page, which includes a table of existing administrators and a form for creating a new one.

| Modify | Delete | Name          | ID            | Level         | Description          |
|--------|--------|---------------|---------------|---------------|----------------------|
|        |        | root          | root          | Administrator | System Administrator |
|        |        | administrator | administrator | Administrator | Addpac Administrator |

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

# System Admin - Licenses

The screenshot shows the 'Smart Multimedia Manager' interface. The 'System Admin' menu is expanded, and the 'Licenses' option is selected. The 'Licenses' page displays a table of licenses and their values.

| Service | License      | Value                  |     |
|---------|--------------|------------------------|-----|
| 1       | Call Manager | Max Calls              | 100 |
| 2       | Call Manager | Max Devices            | 100 |
| 3       | Call Manager | Max Subscribers        | 100 |
| 4       | MCU          | Max Sessions           | 2   |
| 5       | MCU          | Max Party per Sessions | 4   |
| 6       | Presence     | Max Sessions           | 100 |
| 7       | IVR          | Max Sessions           | 100 |
| 8       | IVR          | Max Scenarios          | 100 |
| 9       | Media        | Max Sessions           | 100 |
| 10      | UMS          | Max Sessions           | 100 |
| 11      | UMS          | Max Mail-Boxes         | 100 |
| 12      | RtpProxy     | Max Sessions           | 100 |

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

# System Admin - Voice Lines

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

Start

Welcome to AddPac IP-PBX

root  
Last Login at June 11 01:51:12PM (127.0.0.1)

Click Menu

System Admin

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

Summary

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User Extension

Registered (2)

Unregistered (3)

Unconfigured (0)

Unused license (95)

Smart Multimedia Manager

Extensions Trunks PBX Services System Admin

Network Interfaces Network Services Administrators Licenses Voice Lines Alarm History Call History Show Command Summary

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Voice Lines

Apply Cancel

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

Analog & Mobile

| Slot / Port | Type | Status | Pattern | Connection PLAR | Input Gain | Output Gain | Clock Source | Protocol Emulat | Modify |
|-------------|------|--------|---------|-----------------|------------|-------------|--------------|-----------------|--------|
| 1 0/0/0     | E1   | down   |         |                 | 0          | 0           | Master       | Network         | ✓      |
| 2 0/1/0     | E1   | down   |         |                 | 0          | 0           | Master       | Network         | ✓      |

Digital

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

Extension Analog

Description

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

Related Links

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

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



# System Admin - Alarm History

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

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

**System Admin**

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

**Alarm History**

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

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

**System Status**

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

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

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

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

# System Admin - Call History

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

Start

Welcome to AddPac IP-PBX

root  
Last Login at June 11 01:51:12PM (127.0.0.1)

Click Menu

System Admin

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

Summary

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User Extension

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

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

Start Call History

Call History

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

Summary

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

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User Extensions

System

Trunks

Memory 0%  
Storage 8%  
Network

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

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

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

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**Call History**  
It displays device error which was occurred in IP-PBX System. User may check various information through Call Type (Unspecified / Inter-Site Call / PSTN Backup / Service Provider), Period, Number

# System Admin - Show Command

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a 'System Admin' menu with options like Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, Show Command, and Summary. A red dashed box highlights this menu and the 'Show Command' window. The 'Show Command' window shows a 'Request Command' field with the text 'show call-manager sscp'. Below this, the system output is displayed, including SSCP Timer Information, CM Servers SSCP Information, SSCP Policy Information, Client Auth Session Information, Client Session Information, and Servers Information. A yellow callout box at the bottom left contains the text: 'Show Command User may check the status of IP-PBX System through category and CLI (Command Line Interface)'. The interface also shows a 'User Extension' section with a bar chart and a 'Getting Started' button.

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



# Thank you!

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