Secure IP Telephony Solution
(TLS/SRTP Protocol)

AddPac

AddPac Technology

Sales and Marketing

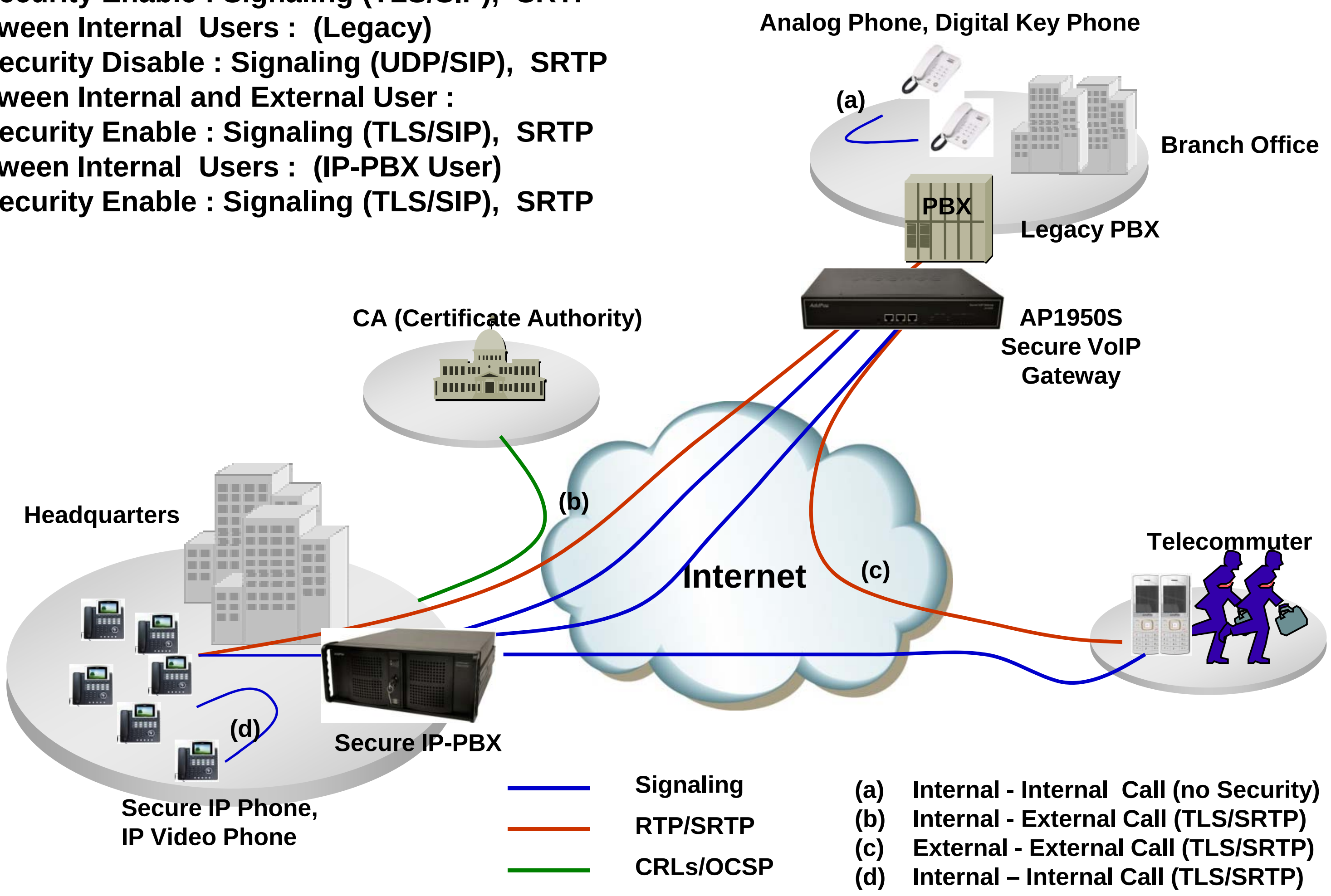
www.addpac.com

Contents

- Secure IP Telephony Service Diagram
- Secure VoIP Protocol & Algorithm (TLS & SRTP)
- AddPac Secure IP Telephony Solution Components
  - Secure IP-PBX Solution
  - Secure IP Phone Solution
  - Secure IP Video Phone Solution
  - Secure VoIP Gateway Solution
  - Secure Video MCU Solution
- NMS for AddPac Secure IP Telephony Solution

Secure IP Telephony Network Diagram

- **Between External Users :**
  Security Enable : Signaling (TLS/SIP), SRTP
- **Between Internal Users : (Legacy)**
  Security Disable : Signaling (UDP/SIP), SRTP
- **Between Internal and External User :**
  Security Enable : Signaling (TLS/SIP), SRTP
- **Between Internal Users : (IP-PBX User)**
  Security Enable : Signaling (TLS/SIP), SRTP

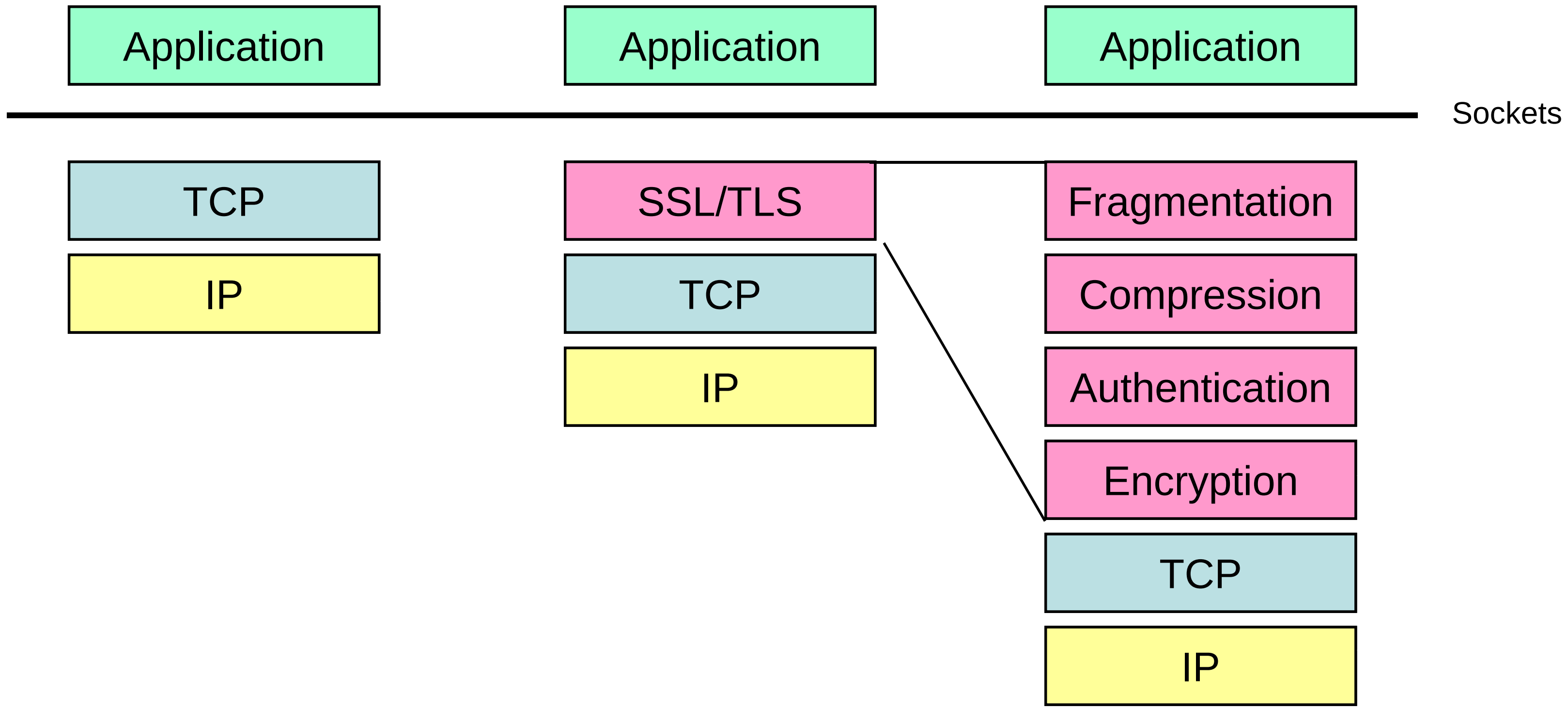


Secure IP Telephony Service Features

TLS Features for Secure VoIP Service

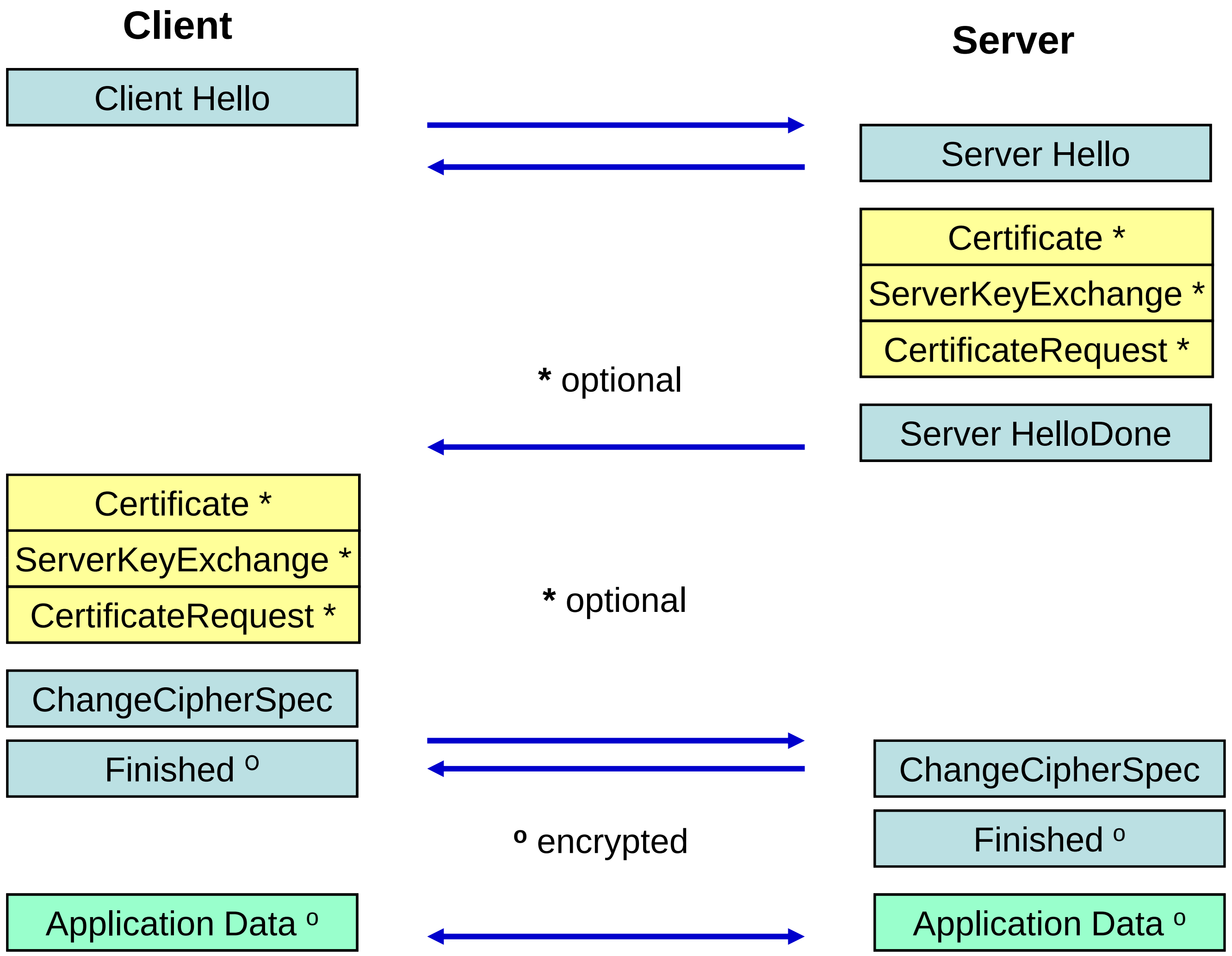
- Support for TLS 1.1, TLS 1.0 and SSL 3.0 protocols
- Since SSL 2.0 is insecure it is not supported.
- TLS 1.2 is supported but disabled by default.
- Support for TLS extensions: server name indication, max record size, opaque PRF input, etc.
- Support for authentication using the SRP protocol.
- Support for authentication using both **X.509 certificates** and OpenPGP keys.
- Support for TLS Pre-Shared-Keys (PSK) extension.
- Support for Inner Application (TLS/IA) extension.
- Support for X.509 and OpenPGP certificate handling.
- Support for X.509 Proxy Certificates (RFC 3820).
- Supports all the strong encryption algorithms (including SHA-256/384/512), including Camellia (RFC 4132).
- Supports compression (optional).
- CRLs
  - CRL (Certificate Revocation List)
  - OCSP (Online Certificate Status Protocol, RFC2560) (via HTTP)
- Hash Algorithm : SHA-1, MD5

SSL/TLS Protocol Layers



SSL/TLS Handshake

AP1950S Secure VoIP Gateway



TLS Comparison with OpenSSL

- **Protocol Support**

	SSLv2.0	SSLv3.0	TLSv1.0	TLSv1.1	TLSv1.2
AddPac	No	Yes	Yes	Yes	Yes
OpenSSL	Yes	Yes	Yes	No	No

- **Key Exchange Algorithms**

	Anon-RSA	RSA	RSA Export	DHE-RSA	DHE-DSS	SRP-DSS	SRP-RSA	SRP	PSK	ECC
AddPac	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No
OpenSSL	Yes	Yes	Yes	Yes	Yes	No	No	No	No	Yes

- **Encryption Algorithms**

(*1) **40-bit encryption is insecure**

	AES-256-CBC	AES-128-CBC	3DES CBC	DES CBC	RC4-128-CBC	RC4-40(*1)	RC2-40(*1)	Camellia	SEED	ARIA
AddPac	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
OpenSSL	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	No

SRTP (Secure Real-time Transport Protocol) Features

- RFC4568, Standards Track, Session Description Protocol (SDP) Security Descriptions for Media Streams
- RFC 3711, Proposed Standard, The Secure Real-time Transport Protocol (SRTP)
- RFC 3551, Standard 65, RTP Profile for Audio and Video Conferences with Minimal Control
- RFC 3550, Standard 64, RTP: A Transport Protocol for Real-Time Applications
- RFC 2104, Informational, HMAC: Keyed-Hashing for Message Authentication
- Cipher Algorithm : ARIA, SEED, AES, DES(*), 3DES(*)

* Support at AddPac Specific SRTP

Secure IP-PBX Solution (TLS/SRTP Protocol)

Contents

- IPNext IP-PBXs for Secure IP Telephony Service
  – IPNext10000 IP-PBX (Large Capacity)
  – IPNext2000 IP-PBX(Medium Capacity)
  – IPNext600 IP-PBX(Medium&Small Capacity)
  – IPNext190 Hybrid IP-PBX(Small Capacity)

IPNext10000 Large Capacity IP-PBX

Main Features

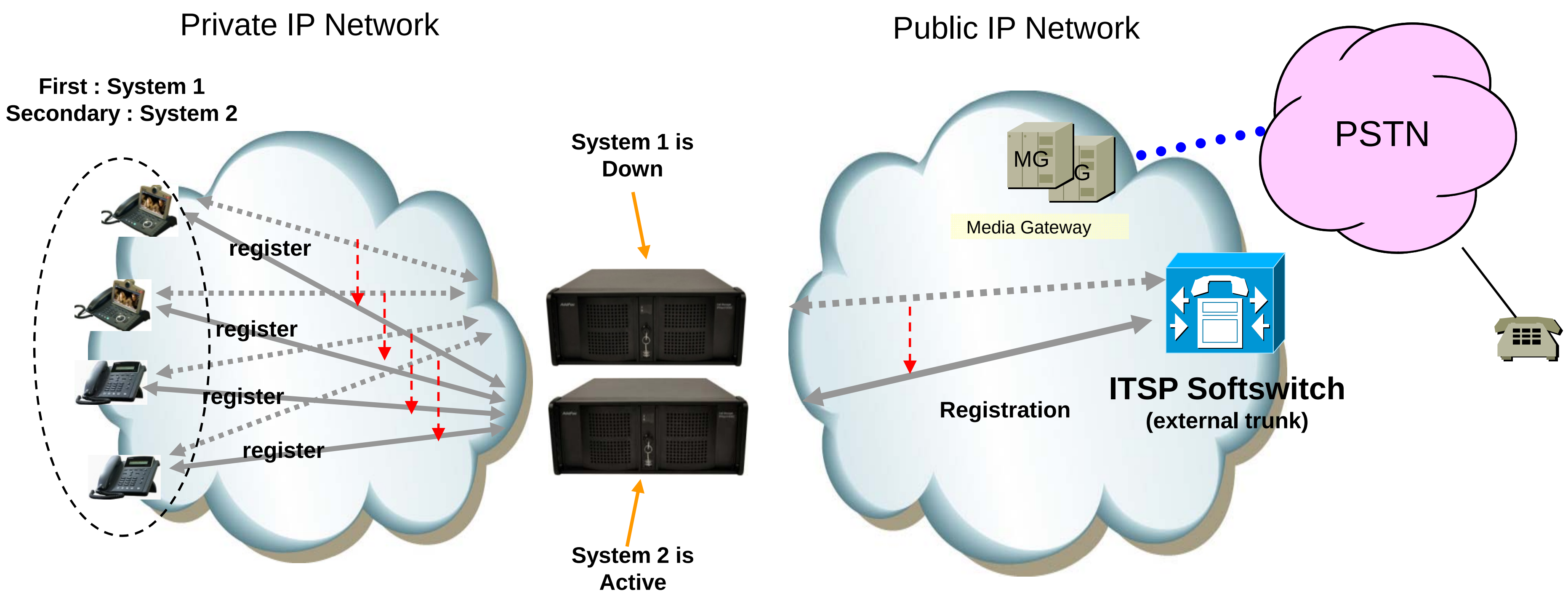
IPNext10000 Next Generation IP-PBX

- IP based Advanced PBX Solution
- IPv4/IPv6 based Dual Network Protocol Support
- External RTP Proxy Function Support
  - External RTP Proxy Server for Private Address : AP-RS3000
- External Application (IVR, RBT, UMS,etc) Function Support
  - External Application Server : AP-AS10000
- Powerful Management and User Friendly Features
- Fault Tolerant and Scalability Architecture
- High-performance Video, Audio, and Voice Service
- Firmware Upgradeable Architecture
- Linux Operation System

System Redundancy Features

IPNext10000 Next Generation IP-PBX

- Active– Active Duplication Scheme
- Active – Standby Duplication Scheme
- VRRP based Duplication Scheme

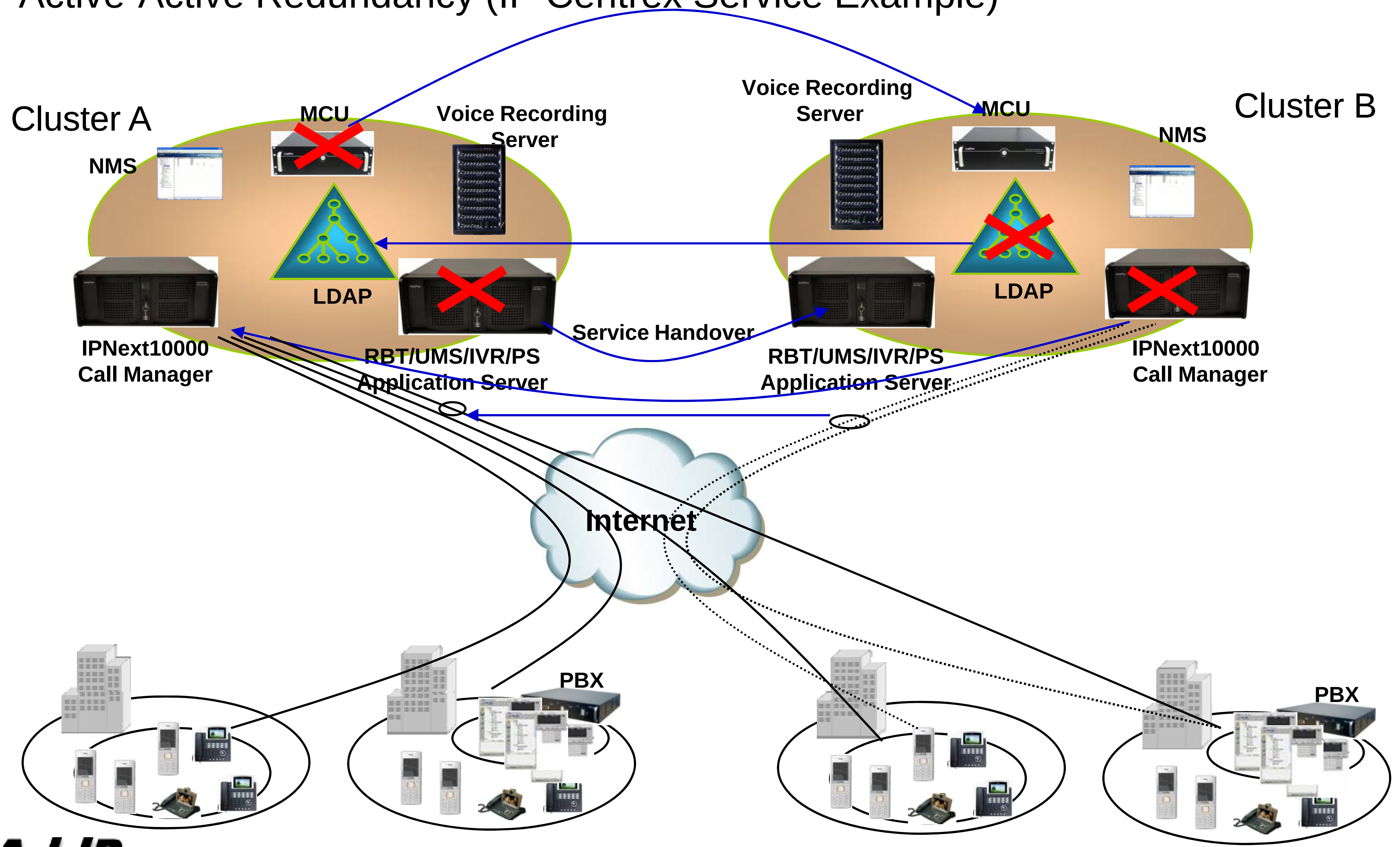


Active – Standby Duplication Scheme (example)

System Redundancy Features

IPNext10000 Next Generation IP-PBX

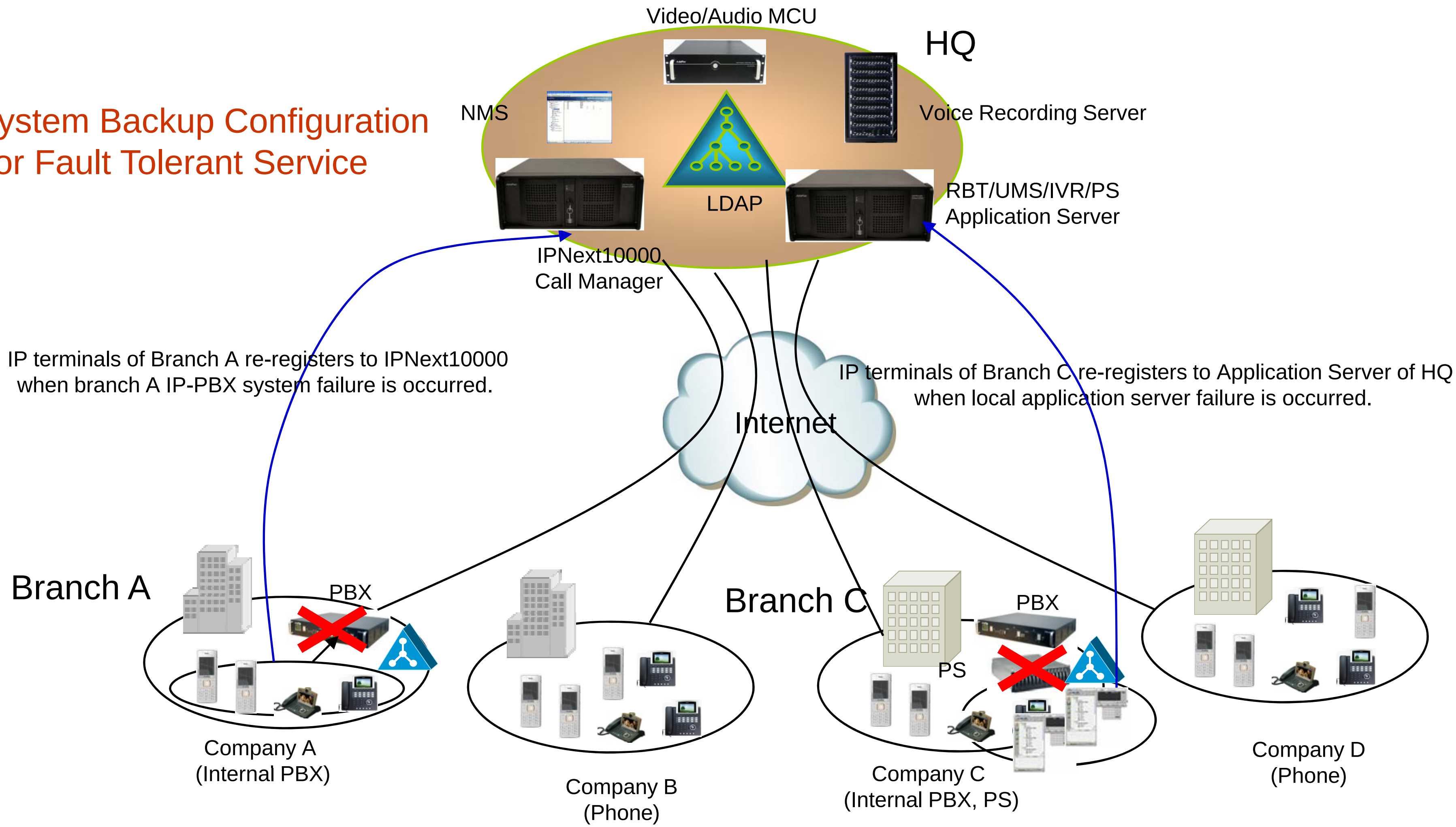
- Active-Active Redundancy (IP Centrex Service Example)



IP Centrex Service

IPNext10000 Next Generation IP-PBX

System Backup Configuration
for Fault Tolerant Service



IPNext2000 IP-PBX
(One System, Dual IP-PBX)

Main Features

IPNext2000 Next Generation IP-PBX

- IP based Advanced Mobile IP-PBX Solution
- IPv4/IPv6 Multimedia Telephony Solution for Large Office
- System Duplication Support (Dual System Board, Dual Power Supply)
- 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 Phone, Video 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 Multimedia Terminal Support

Hardware Specification

IPNext2000 Next Generation IP-PBX

RISC
CPU

- 64bit High-End Microprocessor Computing Power
- Main Chassis
  - Dual Redundancy CPU Boards for System Fault Tolerant
    - Two(2) 10/100/1000Mbps Gigabit Ethernet
    - One(1) RS-232C Console (RJ45)
    - Two(2) 3.5 Inch Hard Disk Interface Slot (RAID 1)
  - Dual Redundancy Power Supply Module
  - Hot-Swap Features

Hardware Specification

IPNext2000 Next Generation IP-PBX

RISC
CPU

IPNext2000 Front Side

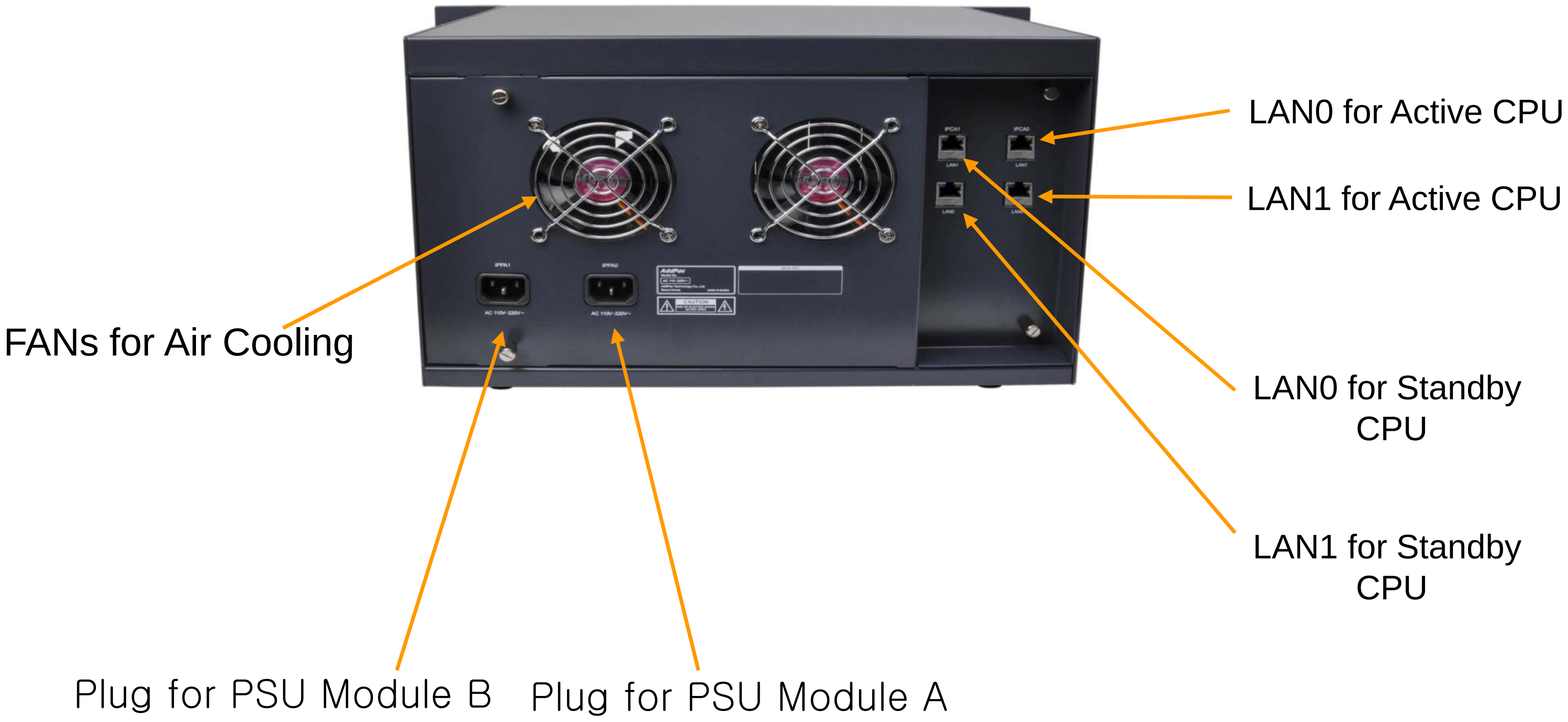
Power On/Off Switch
for System

CPU Board
(Hot-Swap)

Dual Power Supply
Modules
(Hot-Swap)

Dual 3.5 Inch
HDD Disks(RAID1)
: Hot-Swap

IPNext2000 Back Side



IPNext600 IP-PBX
(One System, Dual IP-PBX)

Main Features

IPNext600 Next Generation IP-PBX

- SIP Application Server, Proxy, Registrar and Location Server
- Multiple ITSP Trunk with SIP & H.323 Accounts Support
- Dual System Redundancy Architecture
  - Two(2) Fast Ethernet Interface / System
- High Performance RISC Architecture
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- IPv4/IPv6 Dual Stack
- RTP Proxy Function Embedded for Private IP and IPv6 Address Interworking
- User Presence Service Features for Smart Multimedia Messenger and Smart IP Phone
- IVR Scenario Editor, Voice Mail, Media Service (Coloring), Conference
- Firmware Upgradeable Architecture
- Smart Multimedia Manager for IP-PBX Management
- Smart Messenger Service (click to dial) for Unified Communication
- VPMS (VoIP Plug&Play Management System) & Smart NMS for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Dual Redundancy Power Module

Hardware Specification

IPNext600 Next Generation IP-PBX

RISC
CPU

- 64bit High-End Microprocessor Computing Power
- Main Chassis
  - Dual Redundancy CPU Boards for System Fault Tolerant
    - Two(2) 10/100Mbps Gigabit Ethernet
    - One(1) RS-232C Console (RJ45)
  - Dual Redundancy Power Supply Module
  - Hot-Swap Features

Hardware Specification

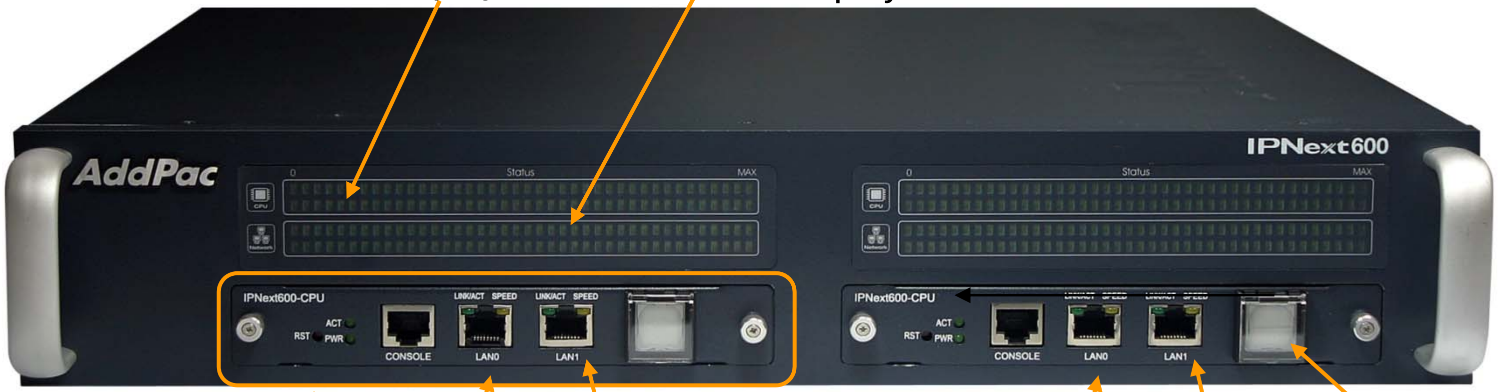
IPNext600 Next Generation IP-PBX

RISC
CPU

IPNext600 Front Side

CPU Utilization Display

Network Utilization Display



CPU Board (Hot-Swap)

LAN0 for Active CPU

LAN1 for Active CPU

LAN0 for Standby CPU

LAN1 for Standby CPU

Hot-Swap Button

IPNext600 Back Side

Dual Power Supply Modules (Hot-Swap)

PSU Module A

PSU Module B

Power On/Off Switch for System

IPNext190 Hybrid IP-PBX

Main Features

IPNext190 Hybrid IP-PBX

- SIP Application Server, Proxy, Registrar and Location Server
- Multiple ITSP Trunk with SIP & H.323 Accounts Support
- High Performance RISC & Programmable DSP Architecture
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- 4 Module Slots for VoIP Gateway Interface, Up to 32 Port
- FXS, FXO Interface (8 Port VoIP Module)
- VoIP Gateway : G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- IPv4/IPv6 Dual Stack
- RTP Proxy Function Embedded for Private IP and IPv6 Address Interworking
- User Presence Service for Smart Multimedia Messenger and Smart IP Phone
- IVR Scenario Editor, Three Party Conference (G.711)
- Unified Messaging Service (Voice Mail, etc)
- Firmware Upgradeable Architecture
- Smart Multimedia Manager for IP-PBX Management
- Smart Messenger Service for UC
- Advanced Voice QoS Mechanism
- Various Call Scenario (Call Pickup, Call Park, Call Transfer, etc)
- Various IP Terminal Support (Video Phone, IP Phone, WiFi Phone, Soft Phone)

Hardware Specification

IPNext190 Hybrid IP-PBX

RISC CPU

High-end DSP

- RISC Microprocessor Computing Power
- Main Chassis
  - Network Interface
    - Two(2) 10/100Mbps Fast Ethernet
    - One(1) RS-232C Console (RJ45)
  - Four(4) VoIP Module Slots for FXS, FXO etc

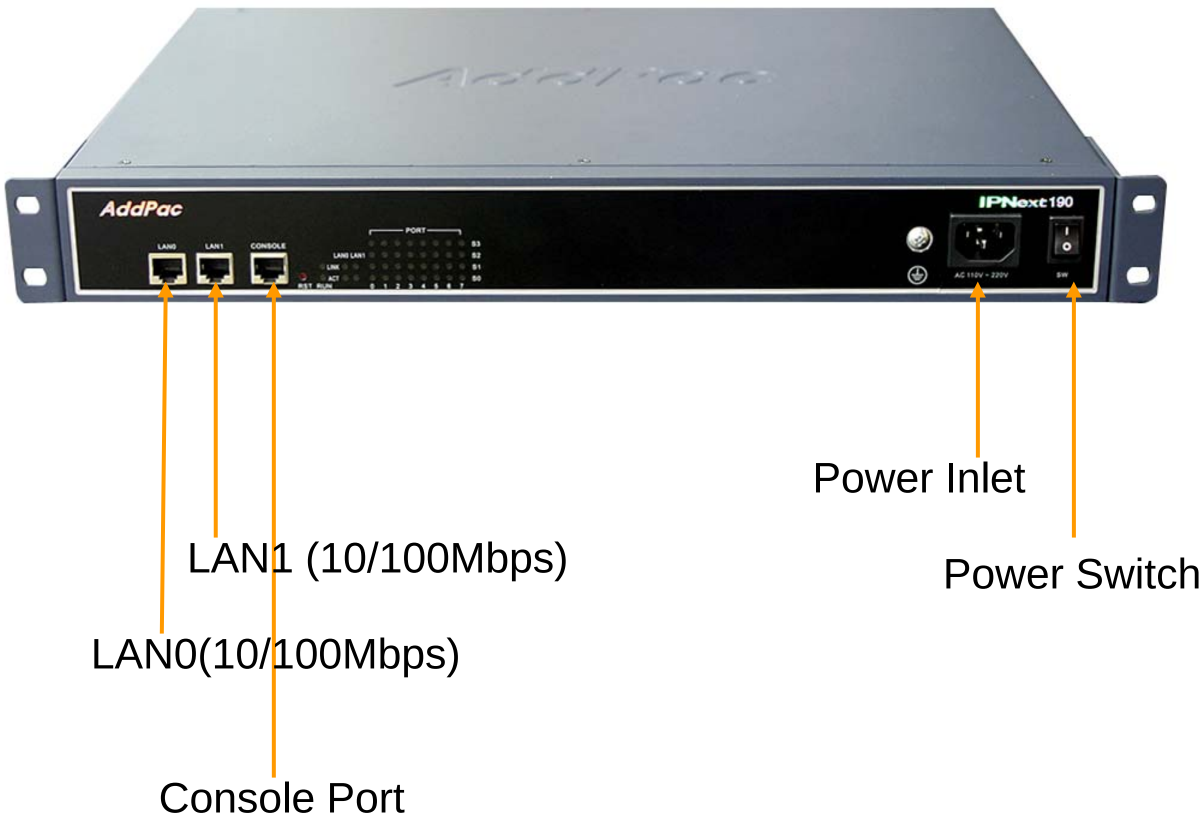
Hardware Specification

IPNext190 Hybrid IP-PBX

RISC CPU

High-end DSP

IPNext190 Front Side



Hardware Specification

IPNext190 Hybrid IP-PBX

RISC
CPU

High-end
DSP

IPNext190 Back Side

PSTN Interface Module

Hot-Swap Switch & LAMP Indication

Secure IP Phone Solution (TLS/SRTP Protocol)

Secure IP Phone Comparison Table

		AP-IP250 	AP-IP230 	AP-IP160 	AP-IP120 	AP-IP90 
LCD Size	4.3 Inch Color LCD	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	Support	N/A	N/A	N/A
Speed-Dial Keys	25 Key with Presence LED	Touch Screen based 25 Keys	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	G.711/G.726/ G.729/G.723
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
3-Party Conversation	Support	Support	Support	Support	Support	Support
Security Algorithm	TLS, SRTP	TLS, SRTP	TLS, SRTP	TLS, SRTP	TLS, SRTP	TLS, SRTP
LAN Port	2	2	2	2	2	2
PoE(Option)	Support	Support	Support	Support	Support	Support
FXO(Option)	Support	Support	Support	Support	Support	Support

Secure IP Video Phone Solution (TLS/SRTP Protocol)

Secure IP Video Phone Comparison Table

	AP-VP500	AP-VP300N	AP-VP280	AP-VP250	AP-VP230	AP-VP150	AP-VP120
							
LCD Size	12.1 Inch Touch Screen	7Inch Touch Screen	7Inch Touch Screen	4.3Inch Touch Screen	5Inch Touch Screen	4.3Inch Touch Screen	4.3Inch
Camera	CCD	CCD	CMOS	CMOS	CMOS	CCD	CMOS
Video Codec	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264
Signaling	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP	H.323/SIP
Security Protocol	TLS/SRTP	TLS/SRTP	TLS/SRTP	TLS/SRTP	TLS/SRTP	TLS/SRTP	TLS/SRTP
Voice MCU	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party	3-Party
LAN Port	2	2	2	2	2	2	2
PoE	N/A	Support	N/A	Support	Support	Support	Support

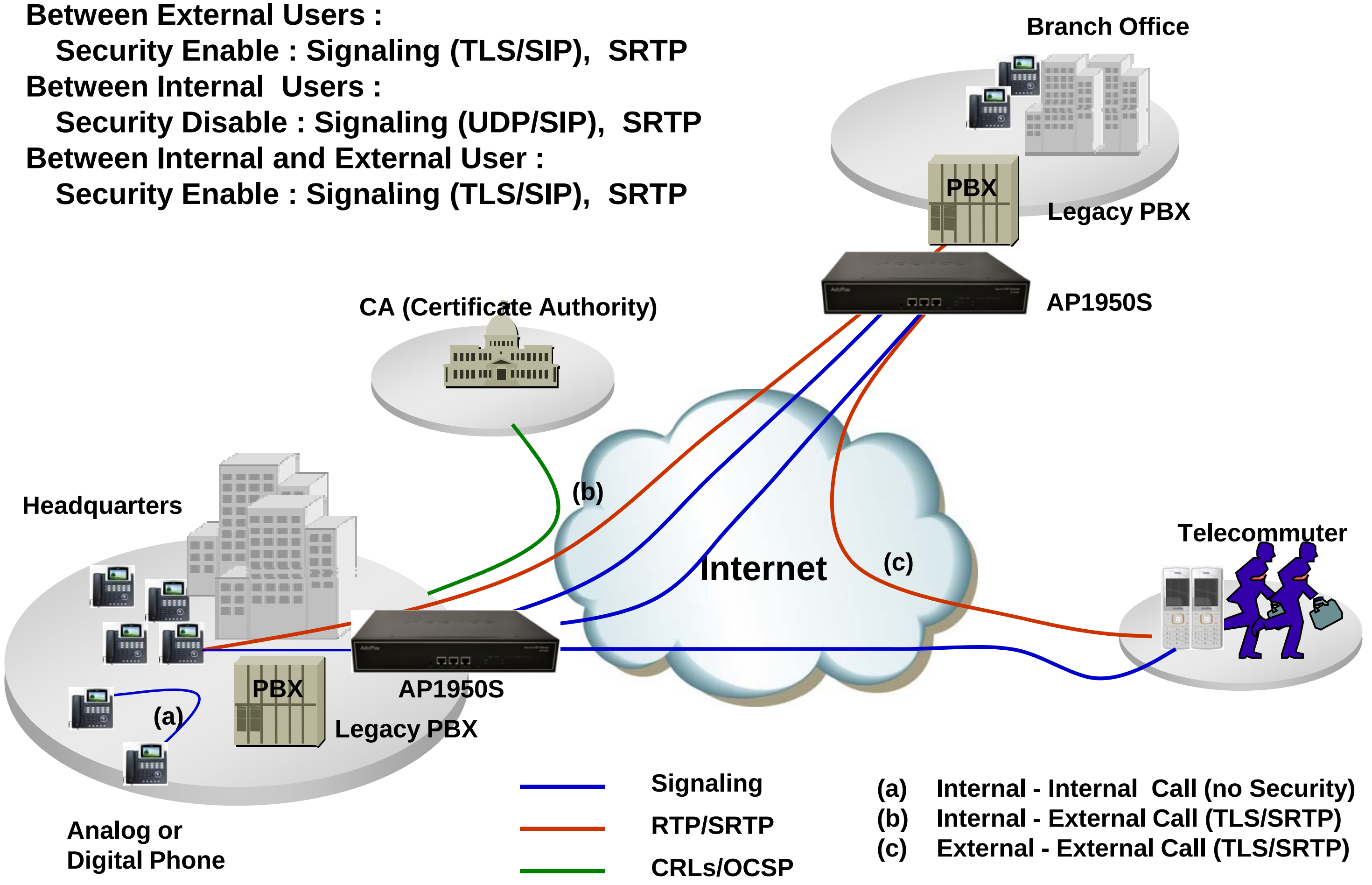
Secure VoIP Gateway Solution (TLS/SRTP Protocol)

Contents

- Secure VoIP Gateway Service Diagram
- Secure VoIP Gateway Comparison Table
    - Secure Analog VoIP Gateways
    - Secure Digital VoIP Gateways
- VoIP Modules for Rack Mountable Equipment
- VoIP Gateway Service Features

SRTP/TLS Network Diagram

- **Between External Users :**
  Security Enable : Signaling (TLS/SIP), SRTP
- **Between Internal Users :**
  Security Disable : Signaling (UDP/SIP), SRTP
- **Between Internal and External User :**
  Security Enable : Signaling (TLS/SIP), SRTP



Secure Analog VoIP Gateways (~32 Port)

Product	AP2330S 	AP2340S 
Available Modules	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4
Analog Ports	Up to 24	Up to 32
Signaling	SIP, H.323	SIP, H.323
TLS/SRTP Support	Yes	Yes
Module Slots	3	4
Module Slot	Three(3)	Four(4)
LAN Port	2	2
Console	1	1
Power	Single PSU	Single PSU

Secure Digital VoIP Gateways (1~2 E1/T1)

Product	AP1900S 	AP1950S 
Available Modules	AP-N1-E1 AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4	AP-N1-E1 AP-N1-2E1 AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4
VoIP Signaling	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 1E1	Up to 2E1
Digital Signaling	ISDN PRI, R2	ISDN PRI, R2
TLS/SRTP Support	Yes	Yes
Module Slot	Two(2)	Two(2)
LAN Port	2	2
Console	1	1
Power	Single PSU	Single PSU

VoIP Modules

Target :
AP1900S, AP1950S,
AP2330S, AP2340S

VoIP Modules

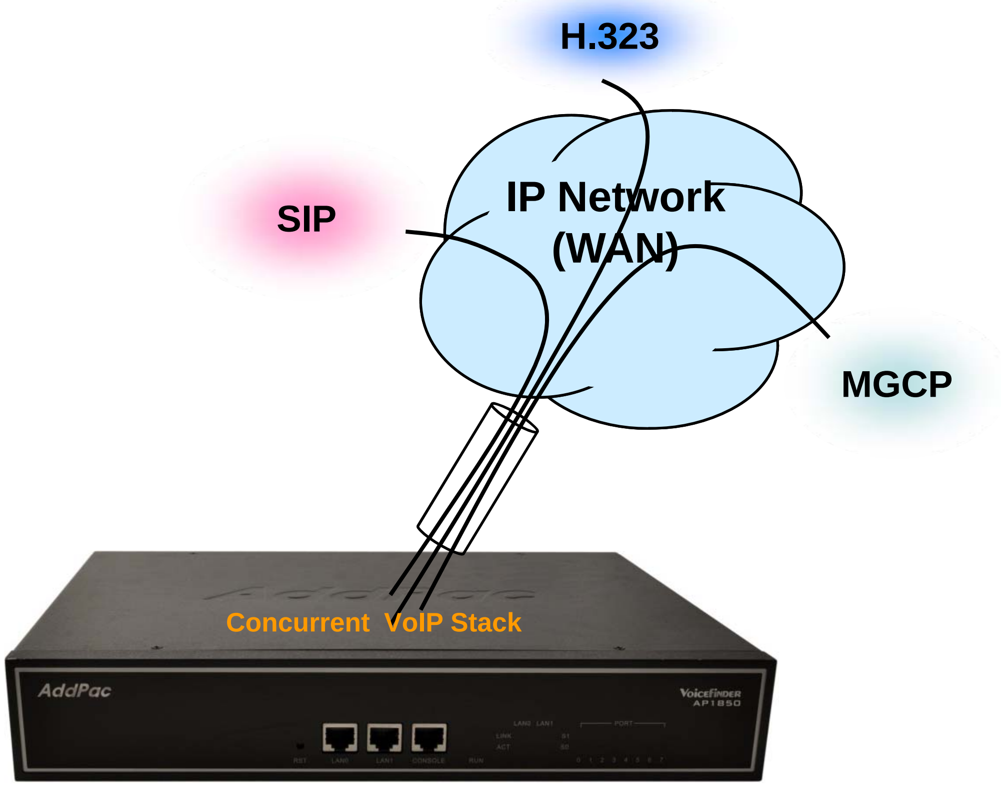
DSP

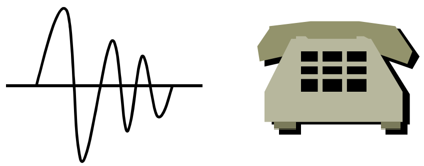
Target	VoIP Modules	Module Features	Module Picture
AP19x0S AP2330S AP2340S	AP-N1-FXS8	8-Port FXS Module	
AP19x0S AP2330S AP2340S	AP-N1-FXO8	8-Port FXO Module	
AP1800 AP2330S AP2340S	AP-N1-FXS4O4	4-Port FXS&4-Port FXO Module	
AP1900S AP1950S	AP-N1-E1	1-Port Digital E1/T1 Module	
AP1950S	AP-N1-2E1	2-Port Digital E1/T1 Module	

VoIP Gateway Service Features

VoIP (Voice over IP) Service

- **H.323, SIP Concurrent VoIP 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





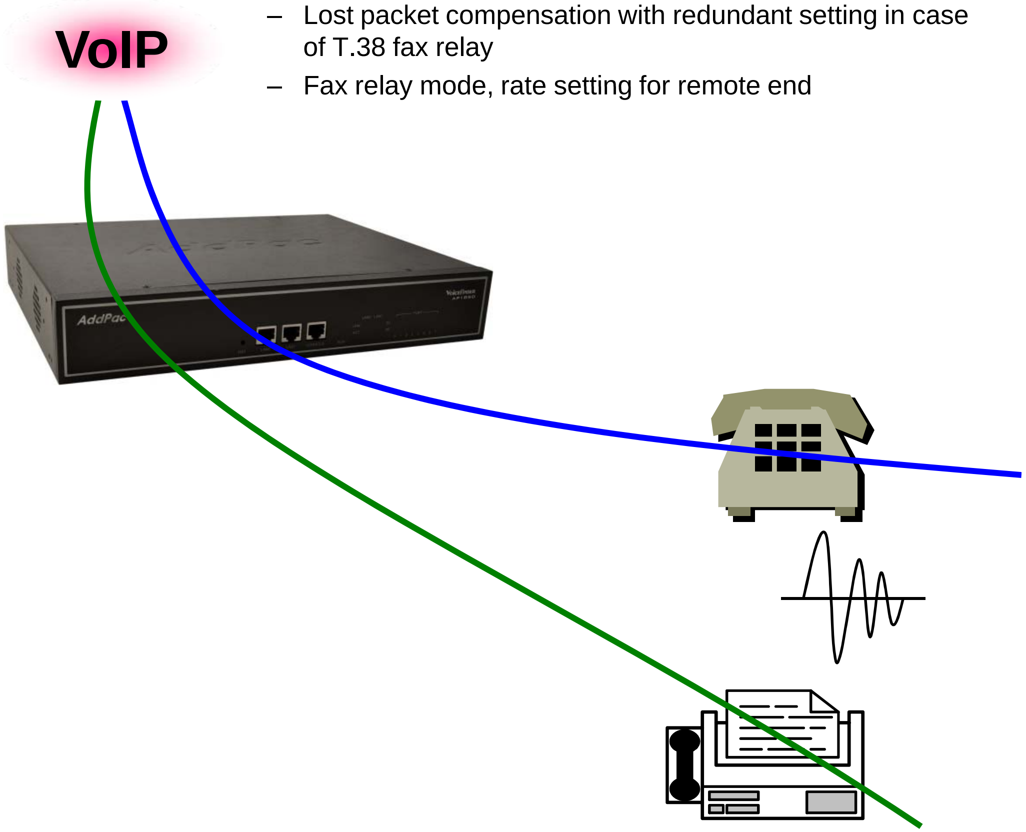
VoIP (Voice over IP) Service

- **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

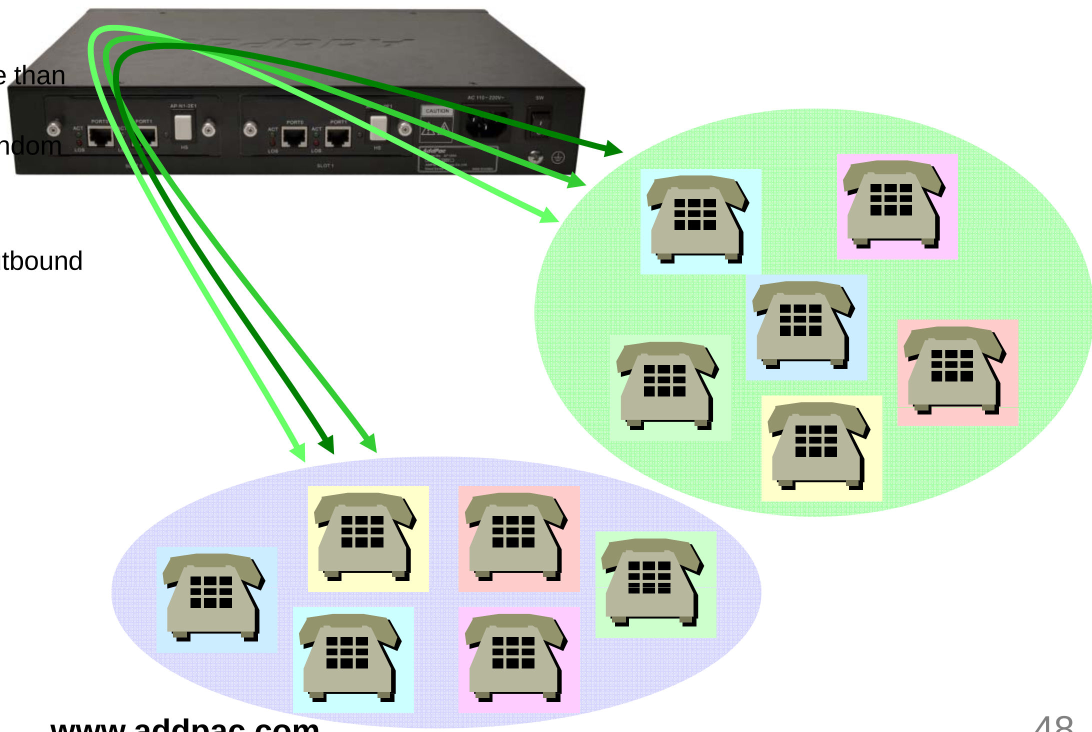
- **Voice Codec**
  - G.711 A-Law, G.711 U-Law
  - G.726 r16, G.726 r32
  - G.729A
  - G.723.1 r63, G.723.1 r53
  - VAD (Voice Activity Detection) function support
  - DTMF relay support (H.323, SIP, MGCP common) based on RFC2833
- **RTP**
  - Redundant RTP packet transmission in case of severe packet 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

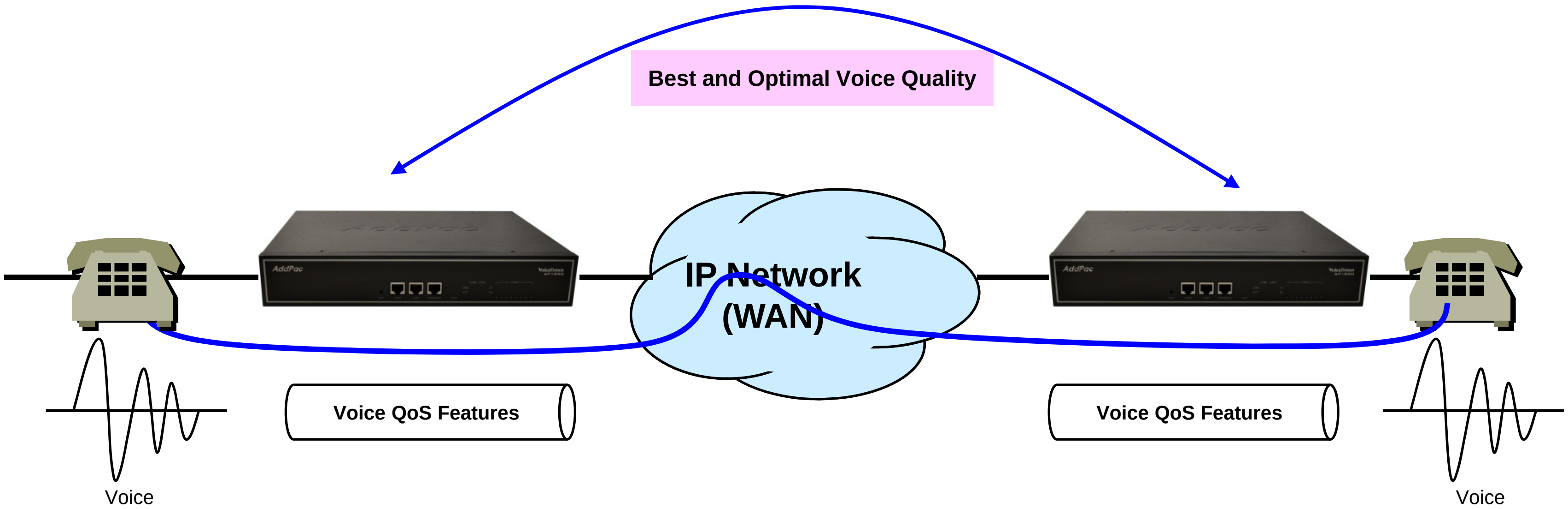
- **VoIP Call Controls**
  - Hot line connection function with PLAR (Private Line Auto Ring Down)
  - Leased line emulation function
  - Connection monitoring function
  - Fault tolerant with Redundancy and Call Distribution among Gateways for load balancing
  - Call attempt with IP address
  - H.323, 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

- **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

- **Basic Network Protocols**

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

- **Routing Protocol**

- IPv4 : Static

- **Service Protocol**

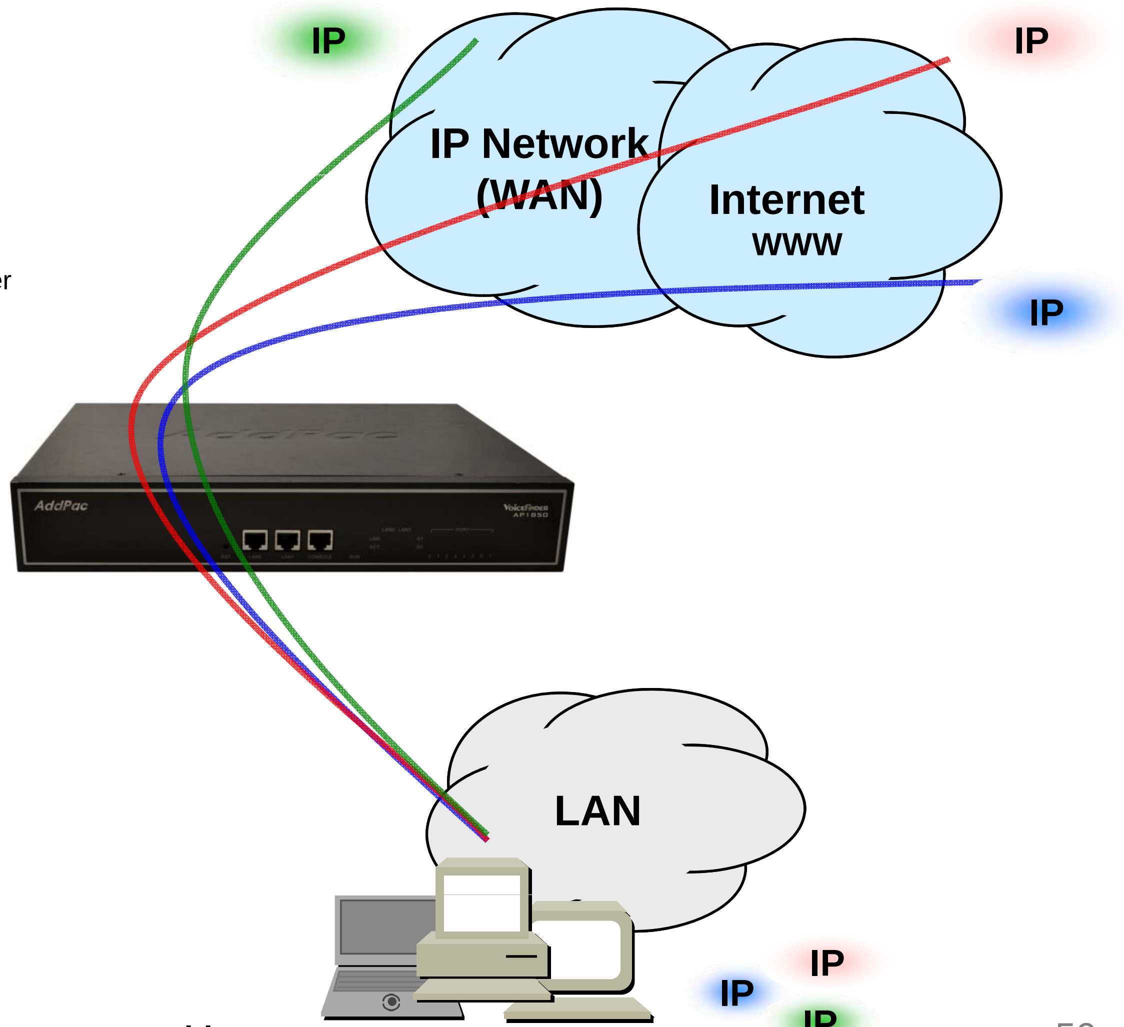
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

- **IPv4 Address Configuration**

- Fixed (Static)
- DHCP
- PPPoE

- **Miscellaneous**

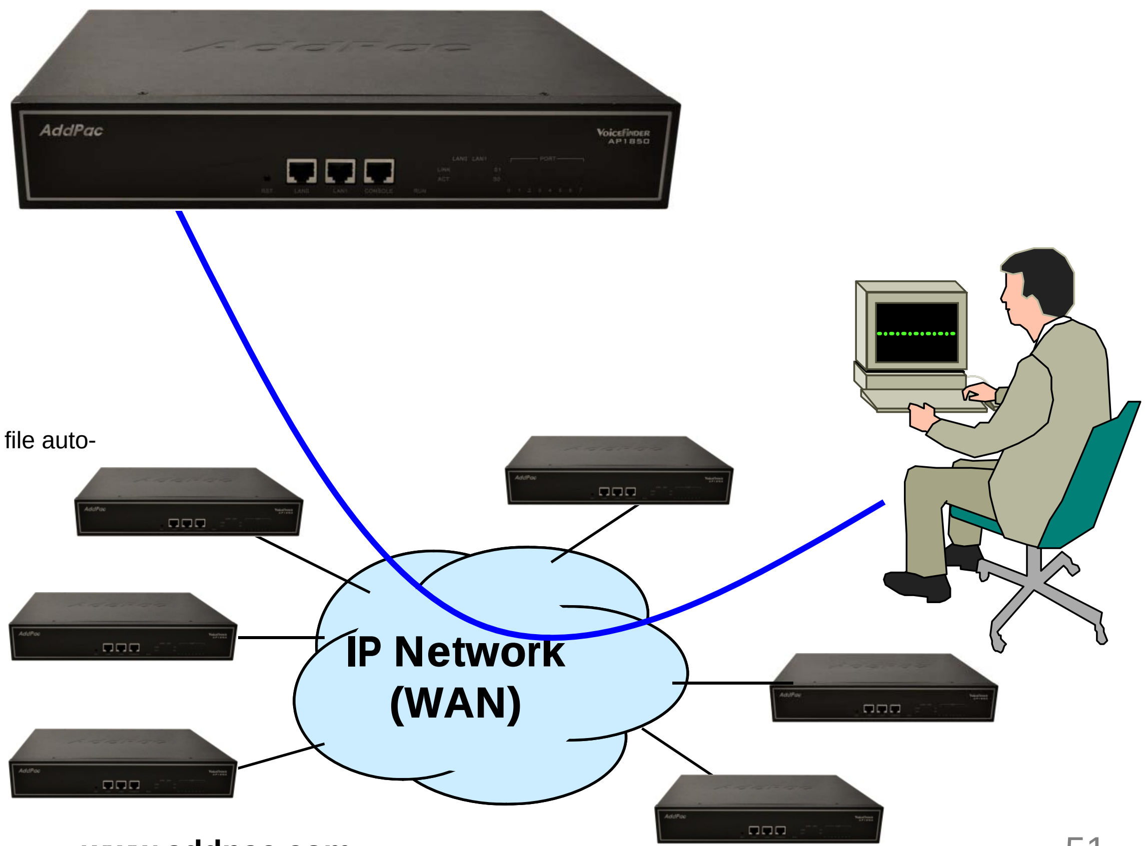
-Cisco Style CLI
- Standard & Extended IPv4 Access List
- Multi-level User Account Management
- IP accounting
- STUN Client



Network Management

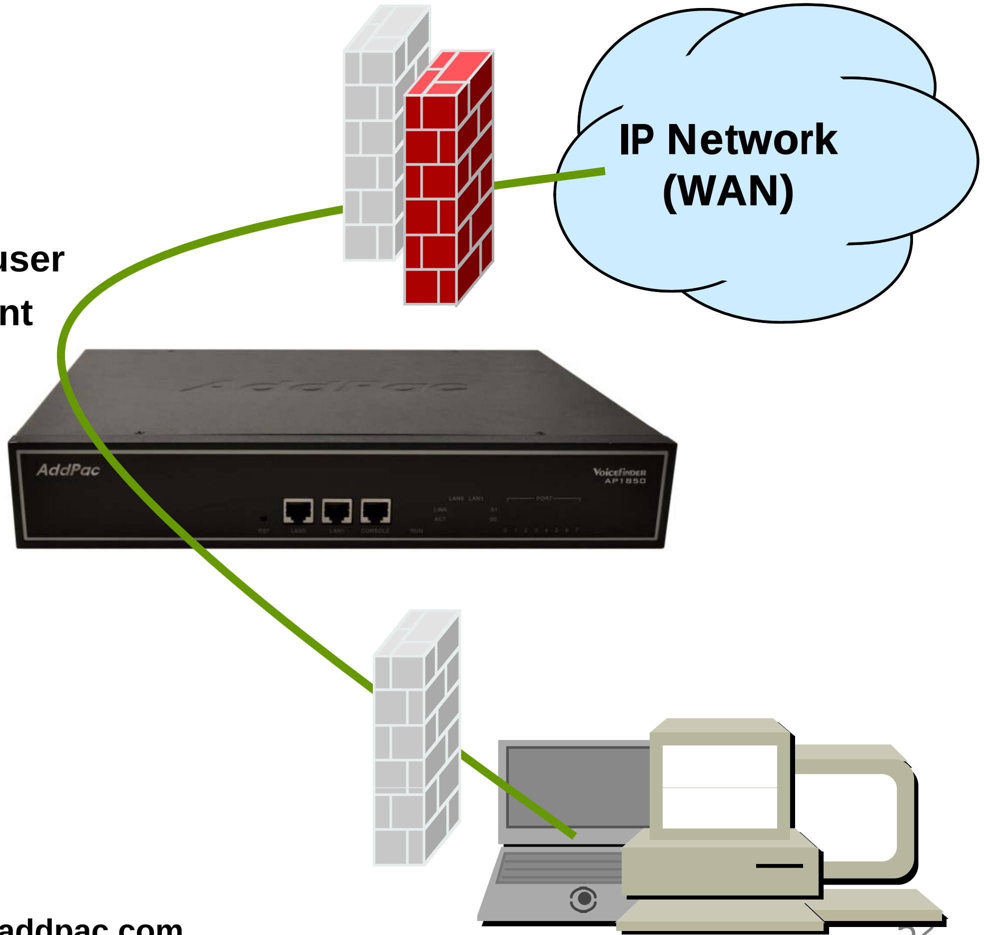
- SNMP
  - Standard Simple Network Management Protocol(SNMP) Agent support
  - MIB v1 and v2 Support
- Web-based Management
  - Smart Easy Setup
  - Standard Voice Interface
  - Standard PSTN Back-up Interface
- Watch-dog Function
  - Hardware, Software watch-dog services
- Remote Management
  - Telnet
  - 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

- **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**



Secure IP Video MCU (TLS/SRTP Protocol)

Secure IP Video MCU Comparison Table

	AP-MC3000	AP-MC2000	AP-MC1800
			
Video Codec	H.263 MPEG4 H.264	H.263 MPEG4 H.264	H.263 MPEG4 H.264
Signaling	H.323/SIP	H.323/SIP	H.323/SIP
Voice Codec	G.722/G.711/G.726/G.729/G.723, etc	G.722/G.711/G.726/G.729/G.723, etc	G.722/G.711/G.726/G.729/G.723, etc
Party Number	32	16	4/8
Multi-Session	Support	Support	Support
Conference Mode	Add-Hoc, Dial-Out, Meet Me	Add-Hoc, Dial-Out, Meet Me	Add-Hoc, Dial-Out, Meet Me
H.323 GateKeeper	Support	Support	Support
Security Protocol	TLS/SRTP	TLS/SRTP	TLS/SRTP
Conference Video Recording	Support	Support	Support
LAN Port	2	1	1
Power Duplication	N/A	N/A	N/A

Video MCU Service Features

Contents

Video Display Layout

- Various Layout
  - 31 types (Symmetric Layout, Asymmetric Layout)
    - Symmetric Layout : same participant picture size
    - Asymmetric Layout : asymmetric participant picture size
  - Auto, Manually : can choice a specific video layout when a conference is started
- Dynamic control
  - Dynamic layout change
  - Dynamic participant movement
- Floor
  - Can distinguish a participant by using the concept of the right of a speaking participant
  - Floor to full screen
- Name display
  - Display or hide the name of a participant dynamically
- Border
  - Three kind of a participant picture border : empty border, a participant boarder, a speaker boarder

Personal Feature*

- Individual (Per Connection) Rate Control for Down Stream
  - Codec, Picture Size, etc
- Personal Layout
  - Example, Zooming for Detailed View

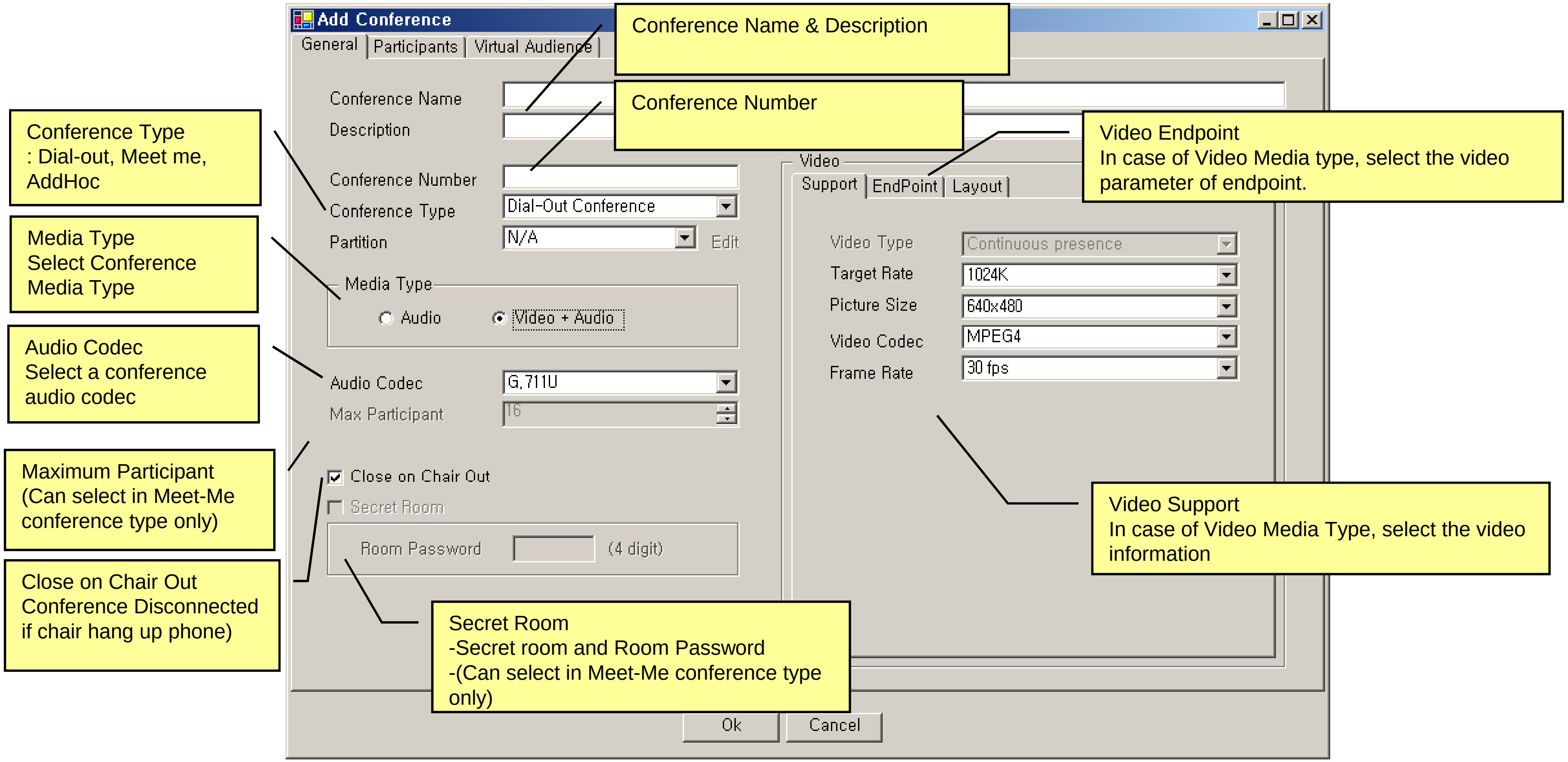
Video Conference Signaling

- Dial-in
  - Even in Dial-out started Video Conference, a participant can join the Video Conference if a participant knows the conference room number in outside.
- Mic off of invisible participant
  - Can turn off MIC of a invisible participant.
- Forced Mute (audio, video)
  - Can mute Audio/Video Capability of a participant via SMM or in Chair, Operator Terminal (Video Phone).
- Virtual Audience
  - Broadcasting solution
  - Can monitor the video conference via inter-working with AddPac Broadcasting Server.
- H323 Support *
  - Floor control(H.243)
  - Dual Video (H.239)

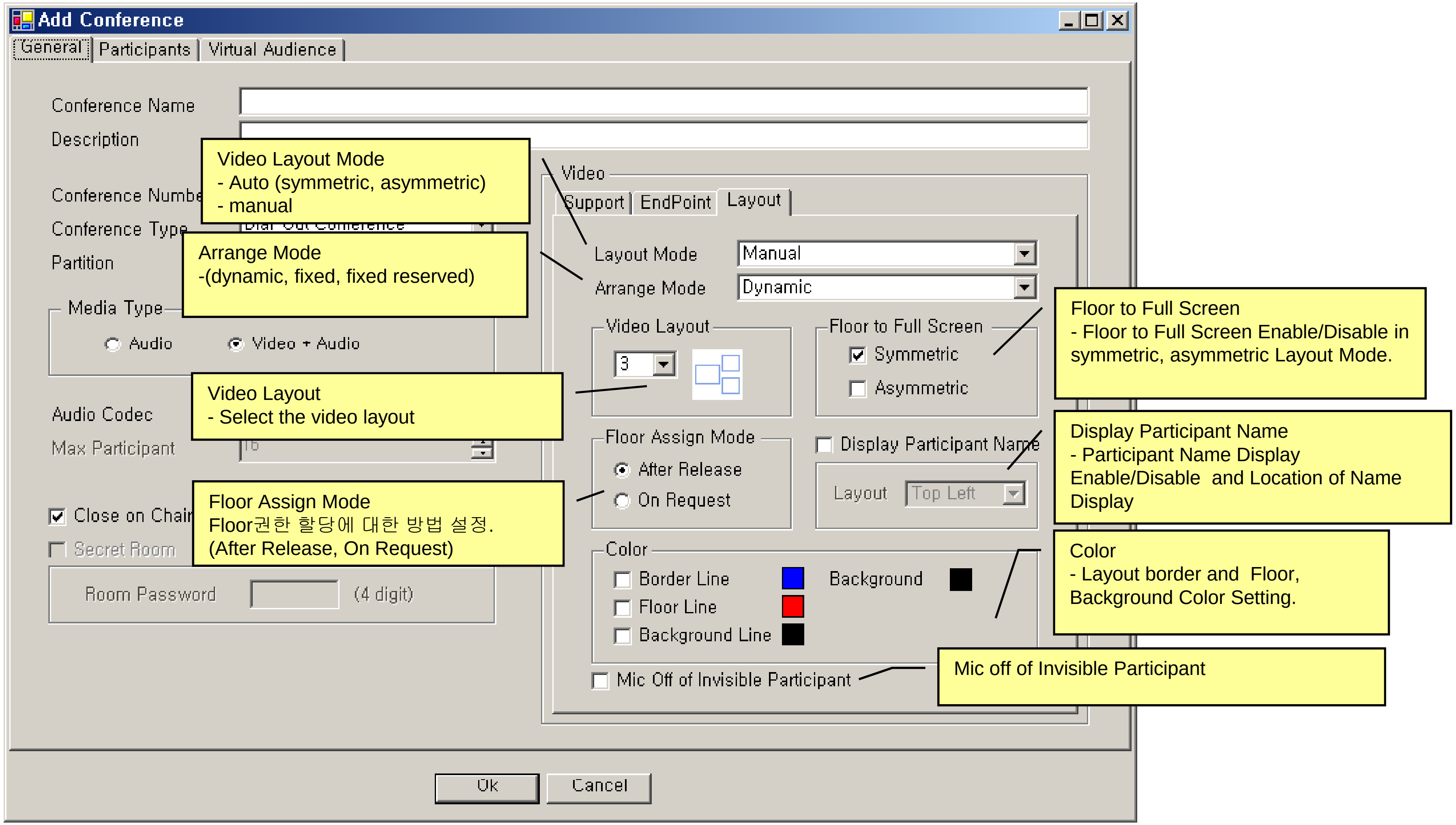
SMM (Smart Multimedia Management) for MCU

- Initial Setting
  - Video Layout Setting
  - User class
    - Chair, Operator, Participant, Audience
  - Initial position
- Active Monitoring
  - Monitoring
    - Snapshot
  - Control
    - Layout
    - Move party
    - Floor
    - Mute
- Video Conference Scheduling*
  - Scheduled Dial-Out Conference

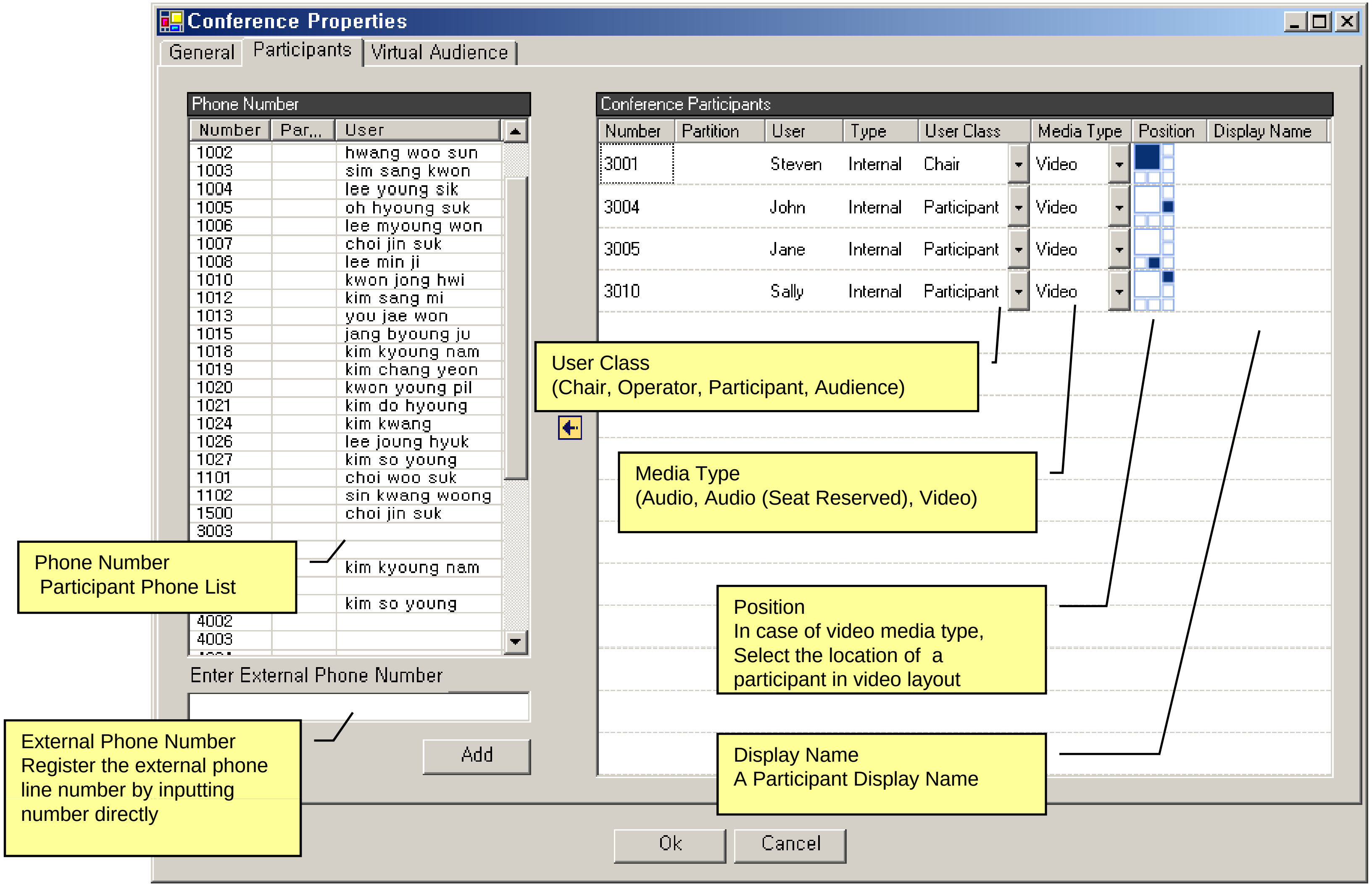
SMM : Conference Room



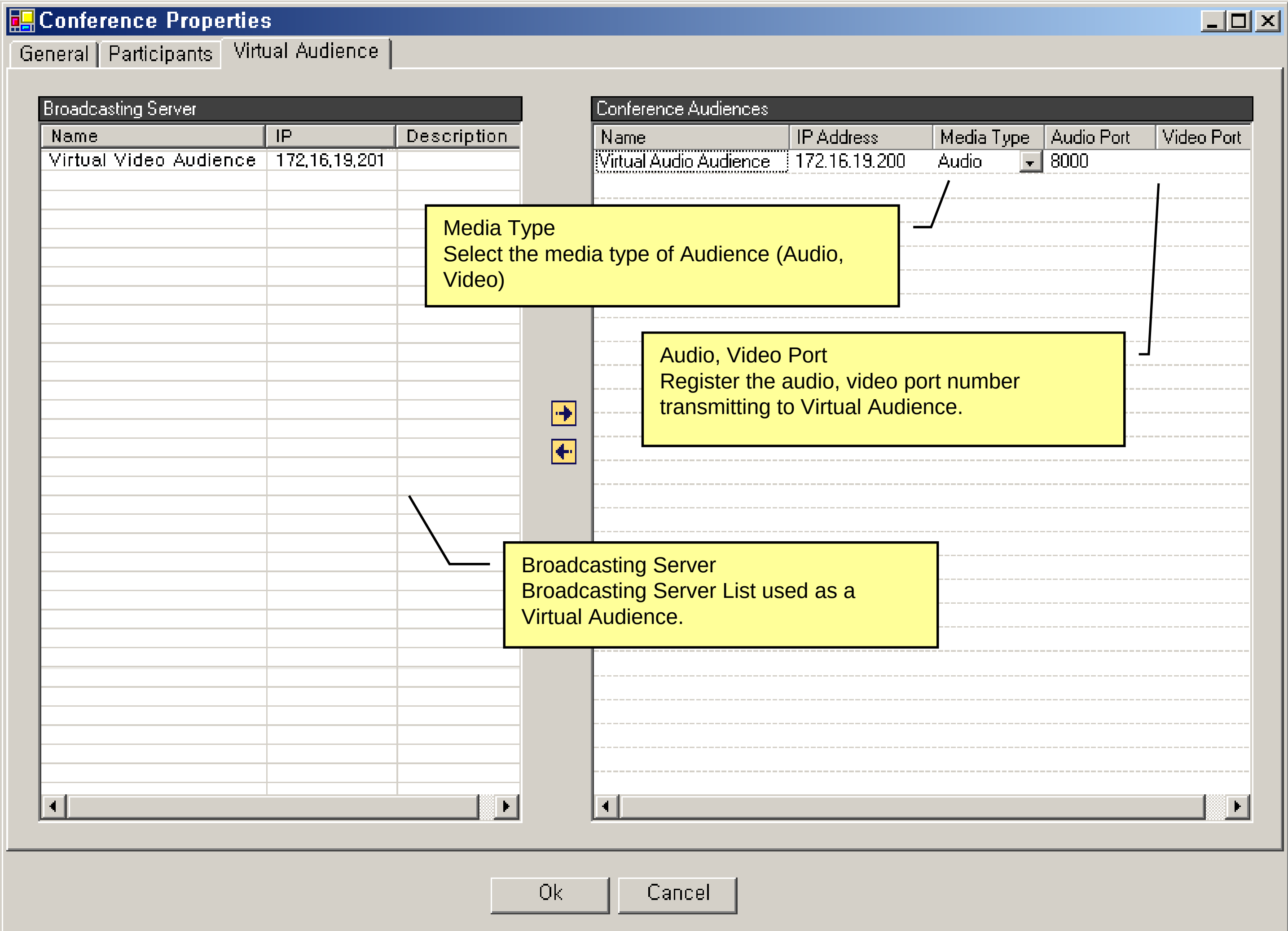
SMM : Conference - Layout



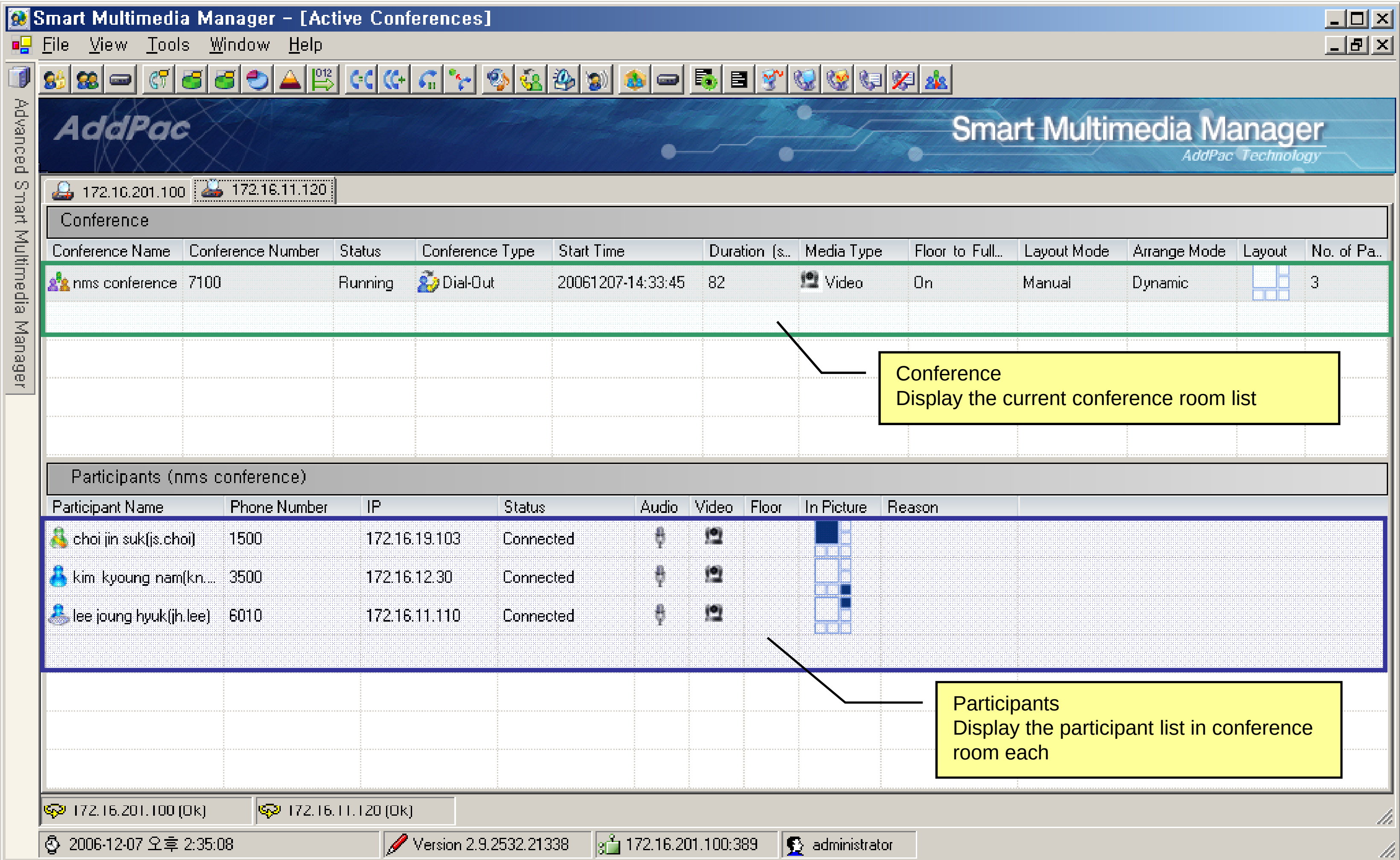
SMM : Conference - Participants



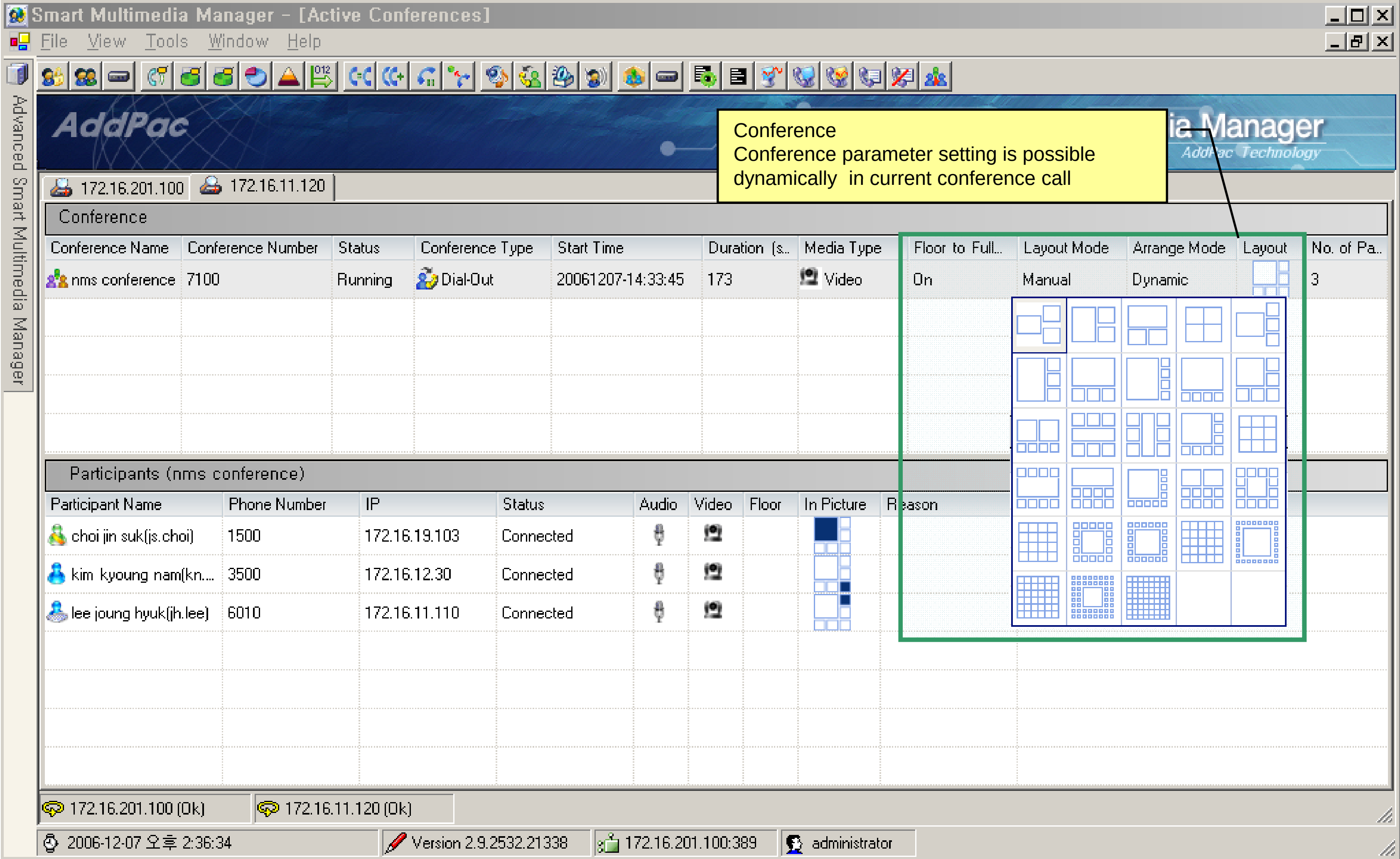
SMM : Conference – Virtual Audience



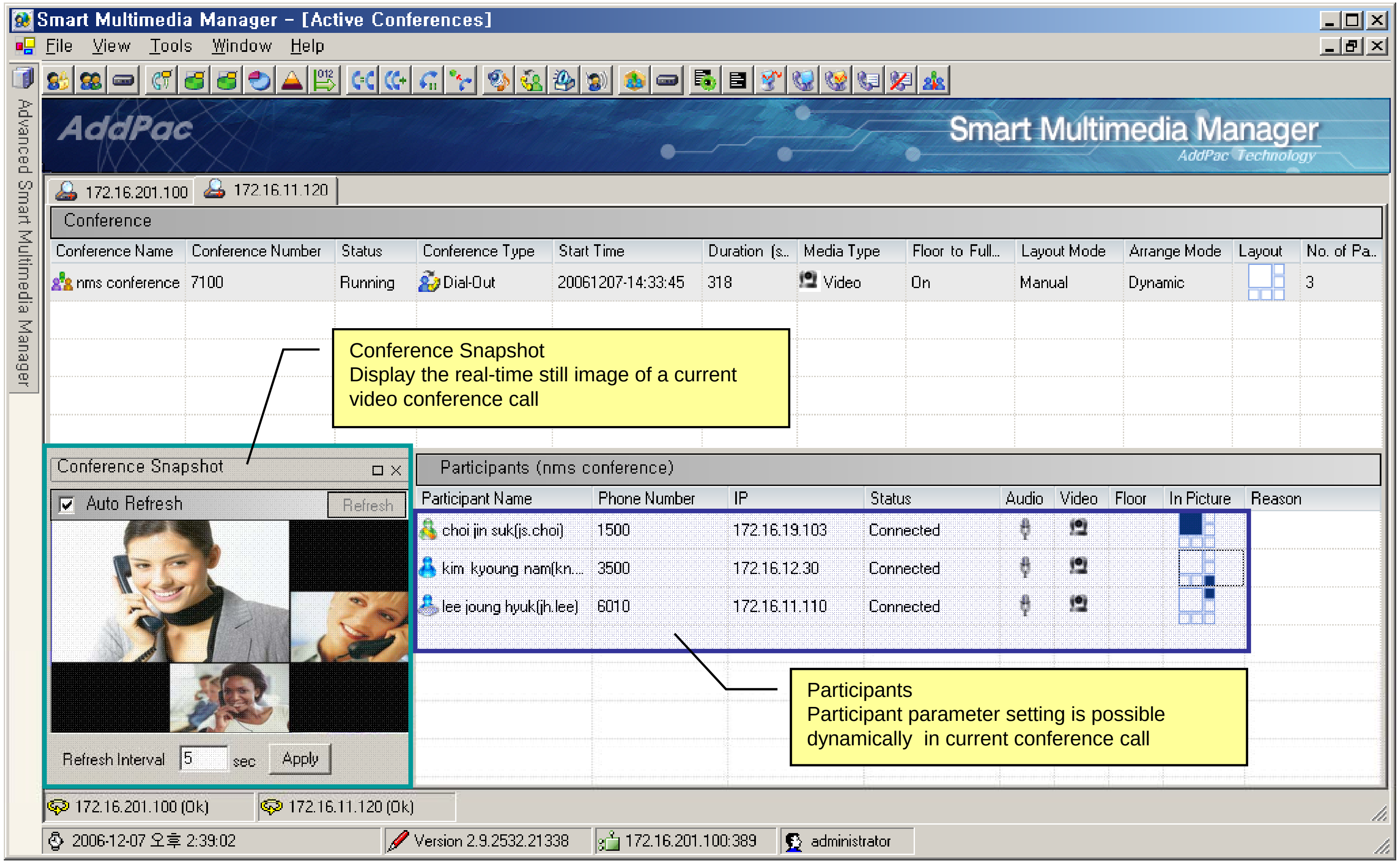
SMM : Active Conference (1)



SMM : Active Conference (2)



SMM : Active Conference (3)



OSD (Video Phone, Video Terminal, etc)

- Conference Management (Chair, Operator)
  - Layout
  - Move party
  - Floor
  - Mute
- Indicator
- Floor

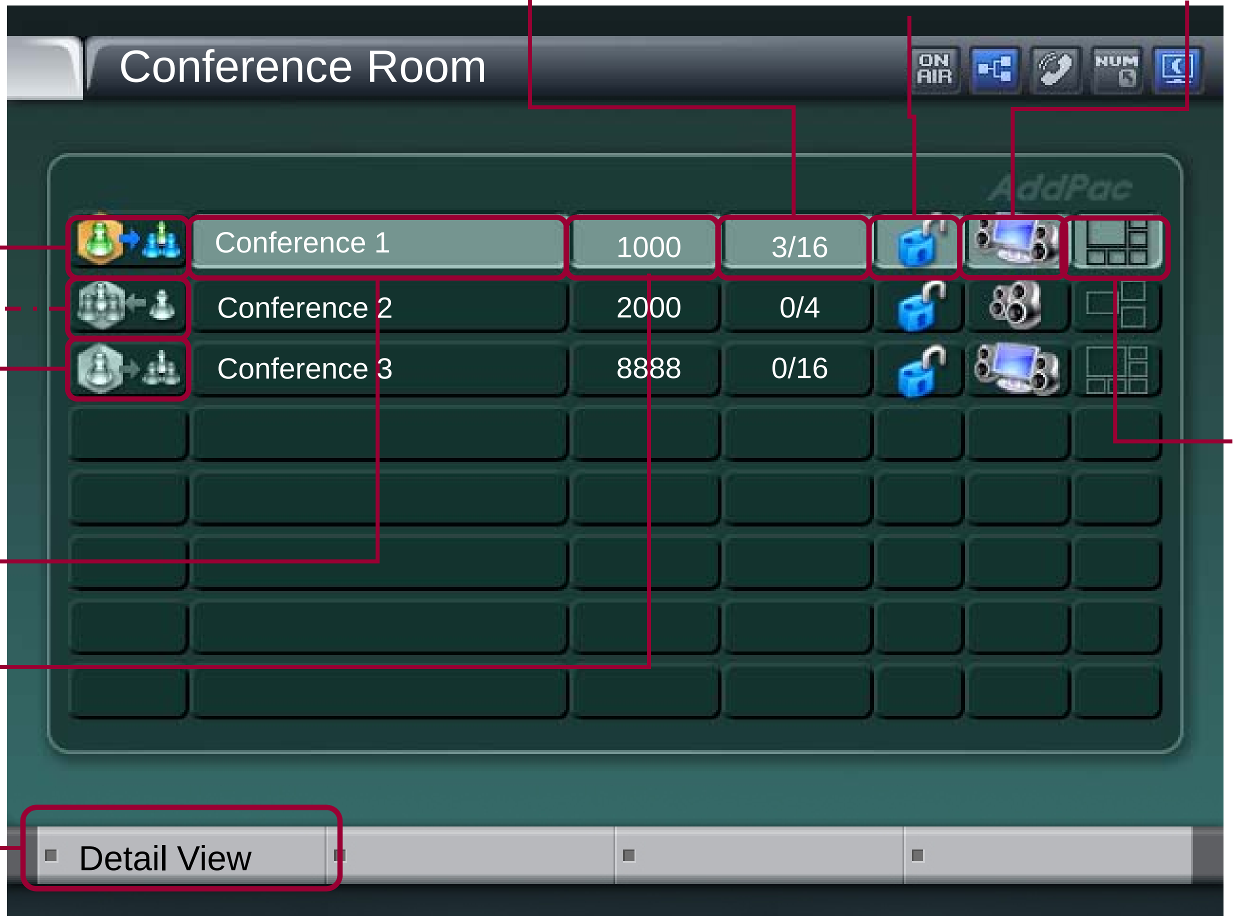
OSD : Conference Room

- Application >> Conference Room
  - conference list

User in conference / maximum

Conference room
-Lock
-unlock

Conference type
-video
-audio



Dial-in conference
- gray : not active
- Colored : active

Meet-me conference
- gray : not active
- Colored : active

Conference name

Conference number

Conference layout

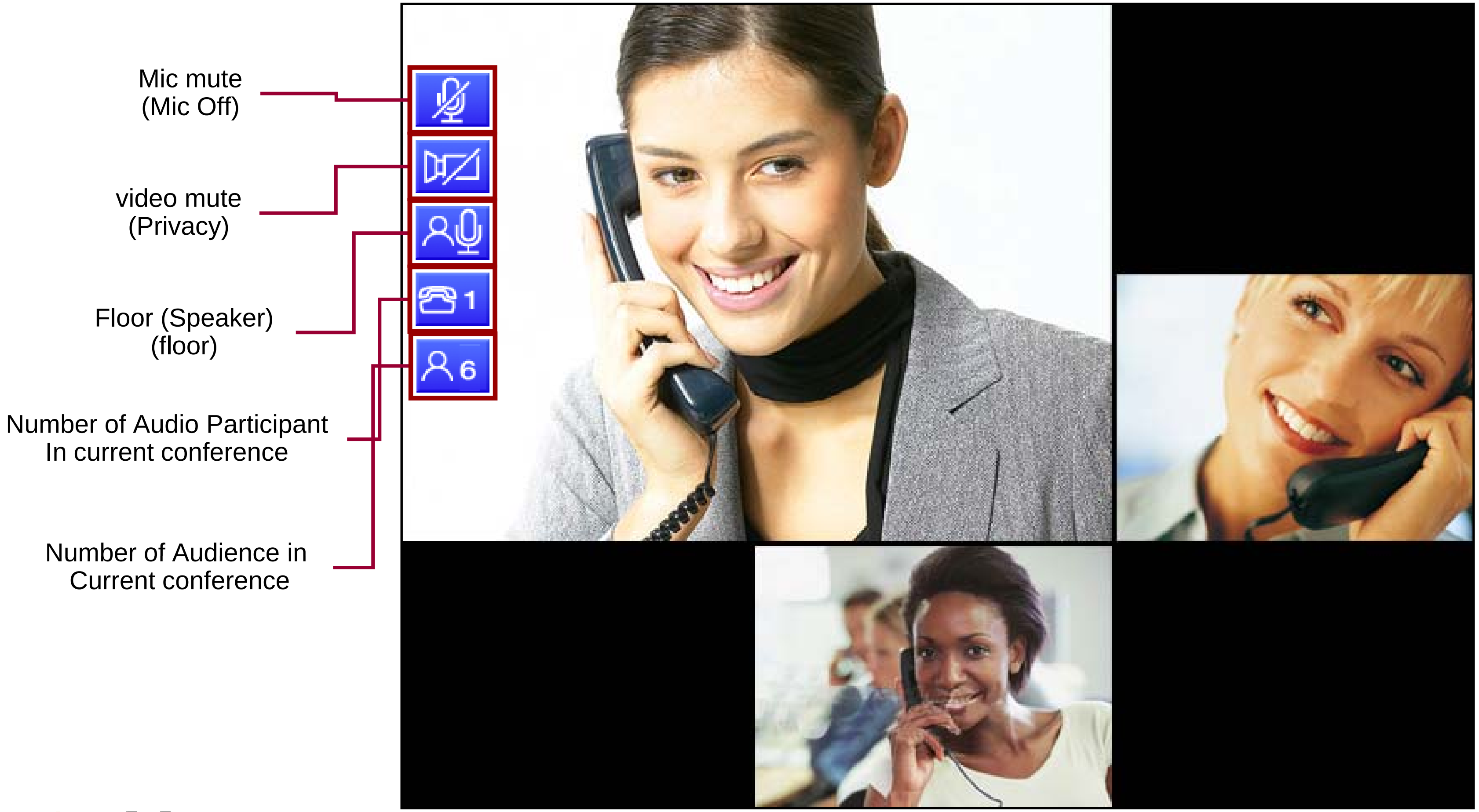
See Detail Participant list

OSD : Conference Open

- Conference open
  - OK or Send Key in Conference Room List
  - Dialing using Conference Room List

OSD : Indicators

- User conference status information
  (function key)



OSD : Add Party

- New User Invite in Current Conference

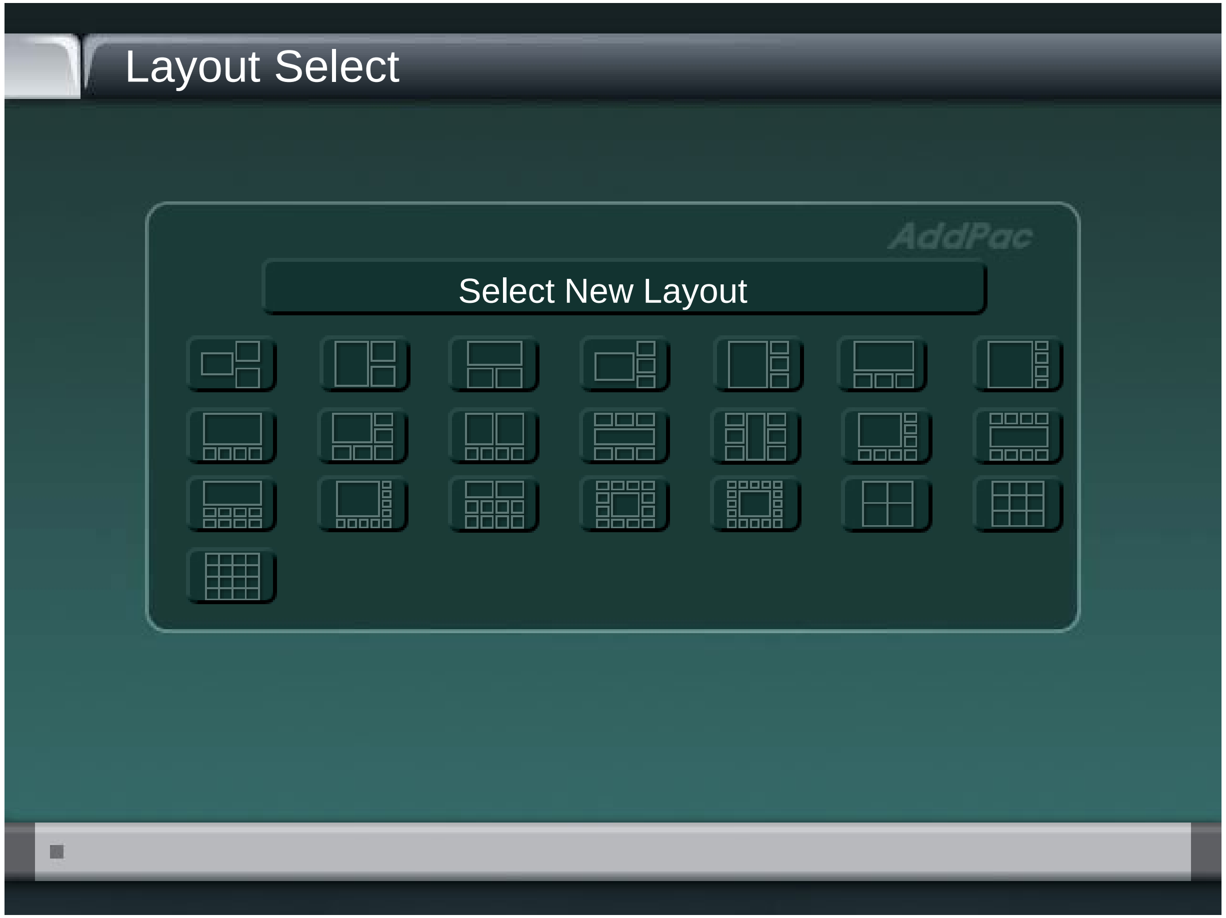
Invite new user

Select User class
-participant
-audience

Select Media type
-audio call
-video call

Layout

- Video Layout Change in current conference



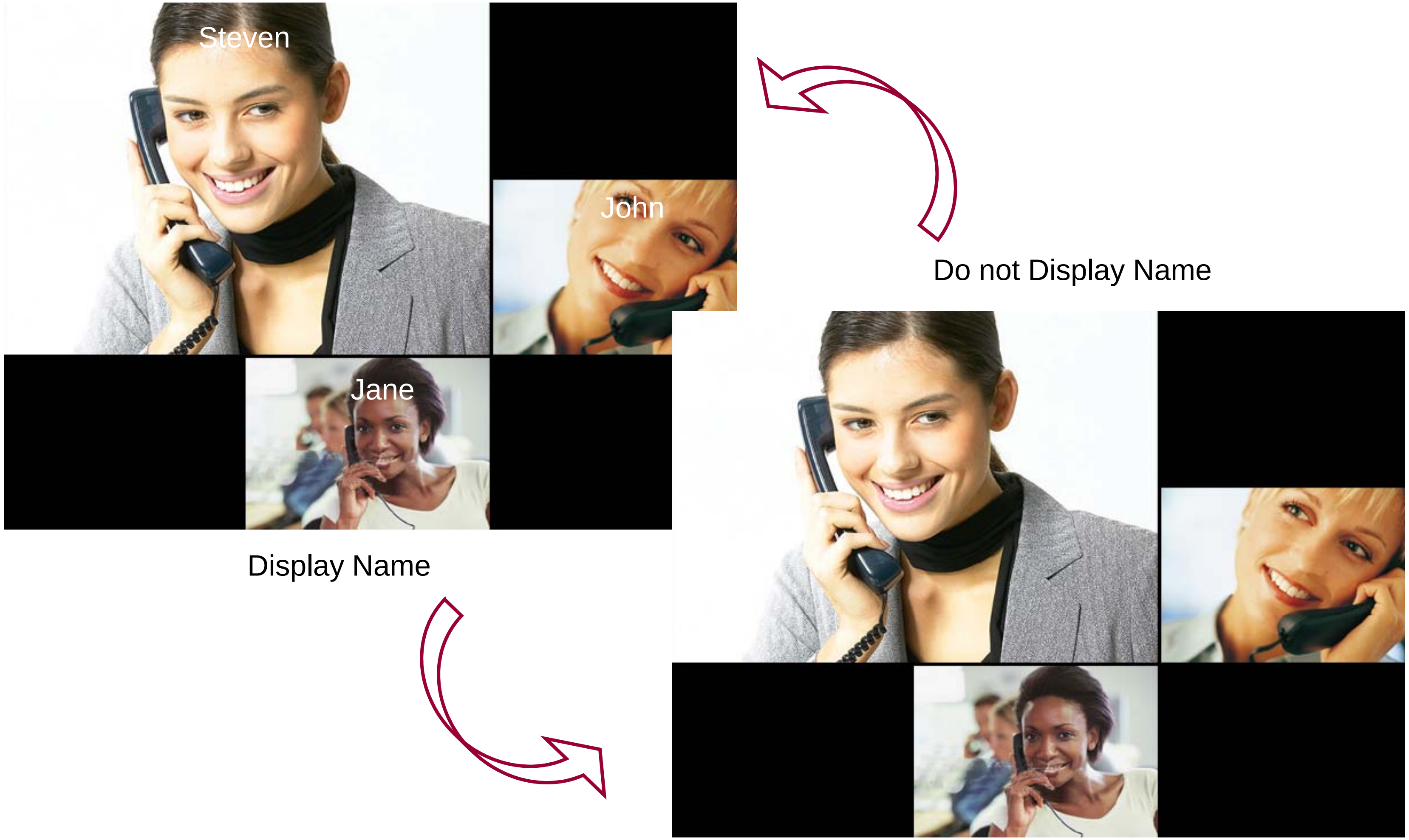
Floor To Full Screen

- Large view display mode : Floor Participant

Floor (speaker)
(floor)

Display Name

- Conference Participant Name Display Enable/Disable



Accept

- User unregistered want to join the conference
  - User can join the conference by Chair or Operator's Permission,

User class
-participant
-audience

Conference Join
Accept

Media type
-audio call
-video call

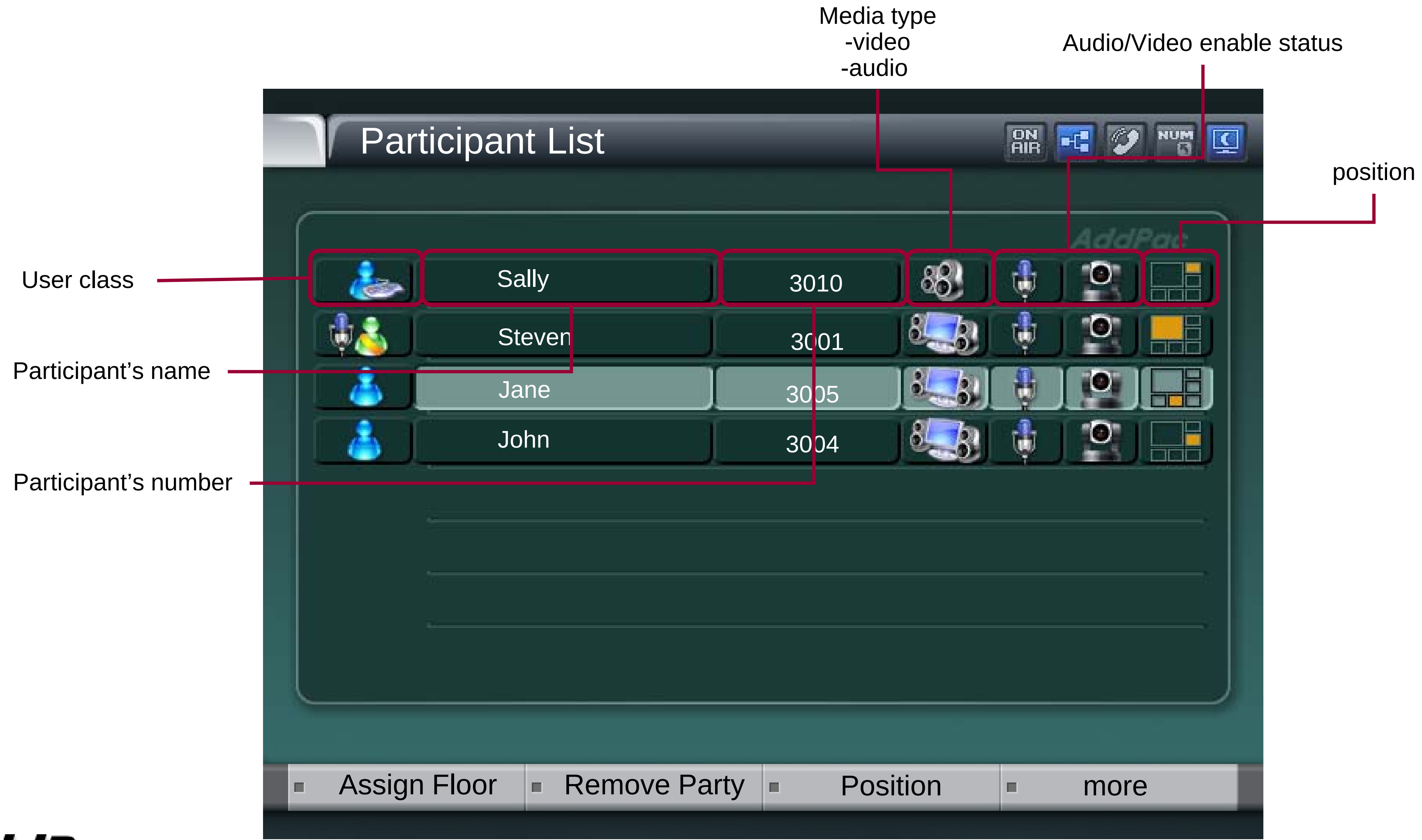
Conference Join
Reject

Participant info

- Party Info
  - Simple Participant info list View/Modification
  - Chair, Operator can modify the status of participant
  - Participant, Audience can read only the status of participant

Detail participant list

- Can view the status information of member In conference



Assign floor

- Give floor (Right of Speaking) to Participant selected in List
  - The ICON of Participant having Floor is changed as MIC ICON

Have a right of speaking

Don't have a right of speaking

Give Floor (Speaker) to Participant

Remove Party

- Conference Call Disconnect of a Participant selected in List

Participant Select

Conference call disconnect of a participant selected

Position

- Participant Location Change In Conference View Layout
- Display the Position List in current conference view layout

User Class

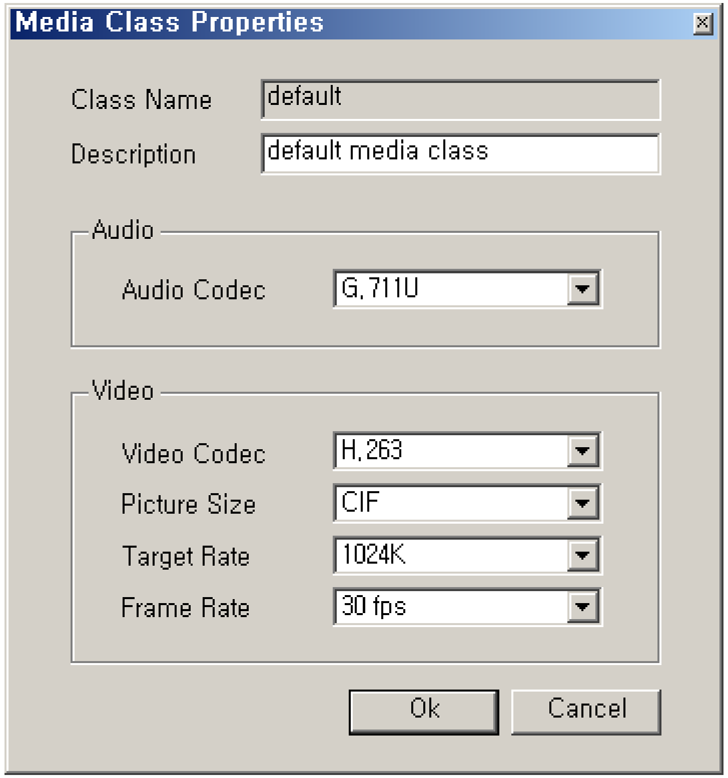
- Change the Right of Participant selected in List
- User class icon change

Audio/Video Mute

- audio, video mute enable/disable

New SMM Feature : Media Class

- Media Class represents the profile information about Audio, Video Codecs, and is used for configuration at conference setup



New SMM Feature : Conference Room

*Blue Box is newly implemented functions.

Conference Name & Description

Conference Number

Conference Type
: Dial-out, Meet me, AddHoc

Media Type
Select Conference Media Type

Media Class
Select a Media Class for a conference room. Second, third media class are reserved for first media class Mismatch.

Speaking Mode
Set up Who can speak out at conference

Encryption Mode
Encryption mode setup for secure conference

Enable Voice Switching
Provide the Voice Activity Switching Function

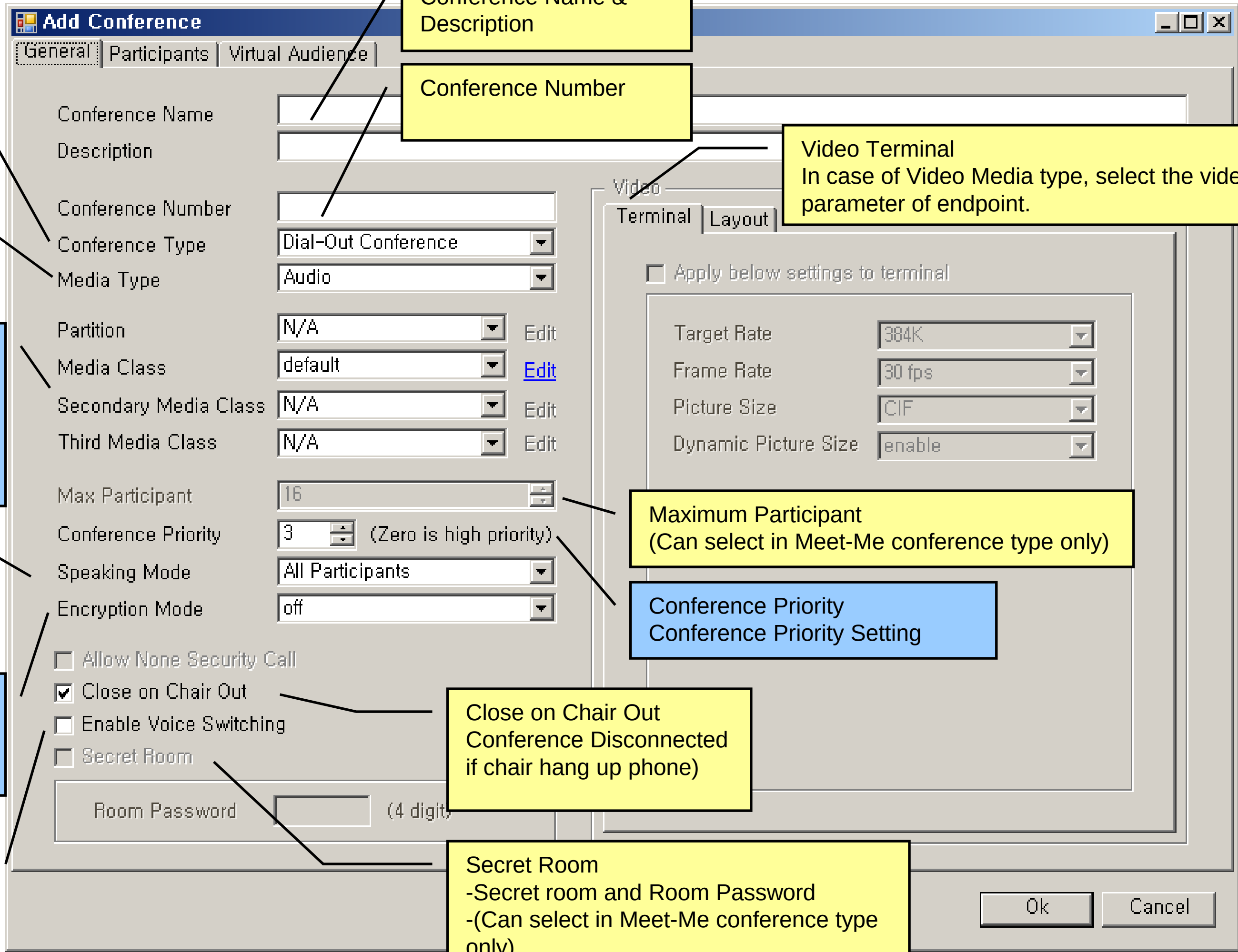
Video Terminal
In case of Video Media type, select the video parameter of endpoint.

Maximum Participant
(Can select in Meet-Me conference type only)

Conference Priority
Conference Priority Setting

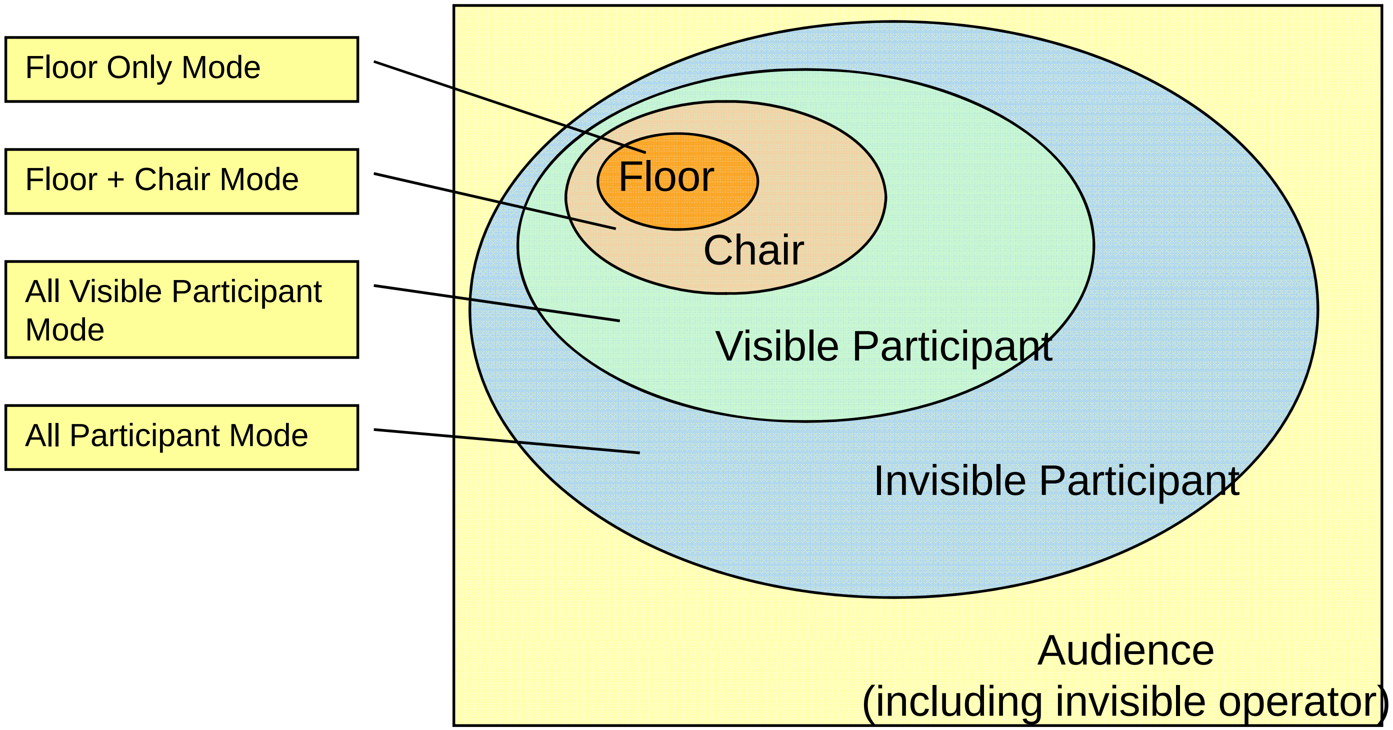
Close on Chair Out
Conference Disconnected if chair hang up phone)

Secret Room
-Secret room and Room Password
-(Can select in Meet-Me conference type only)



New SMM Feature : Speaking Mode

- Speaking Mode
  - Determine the scope of participant who can speak out at conference.

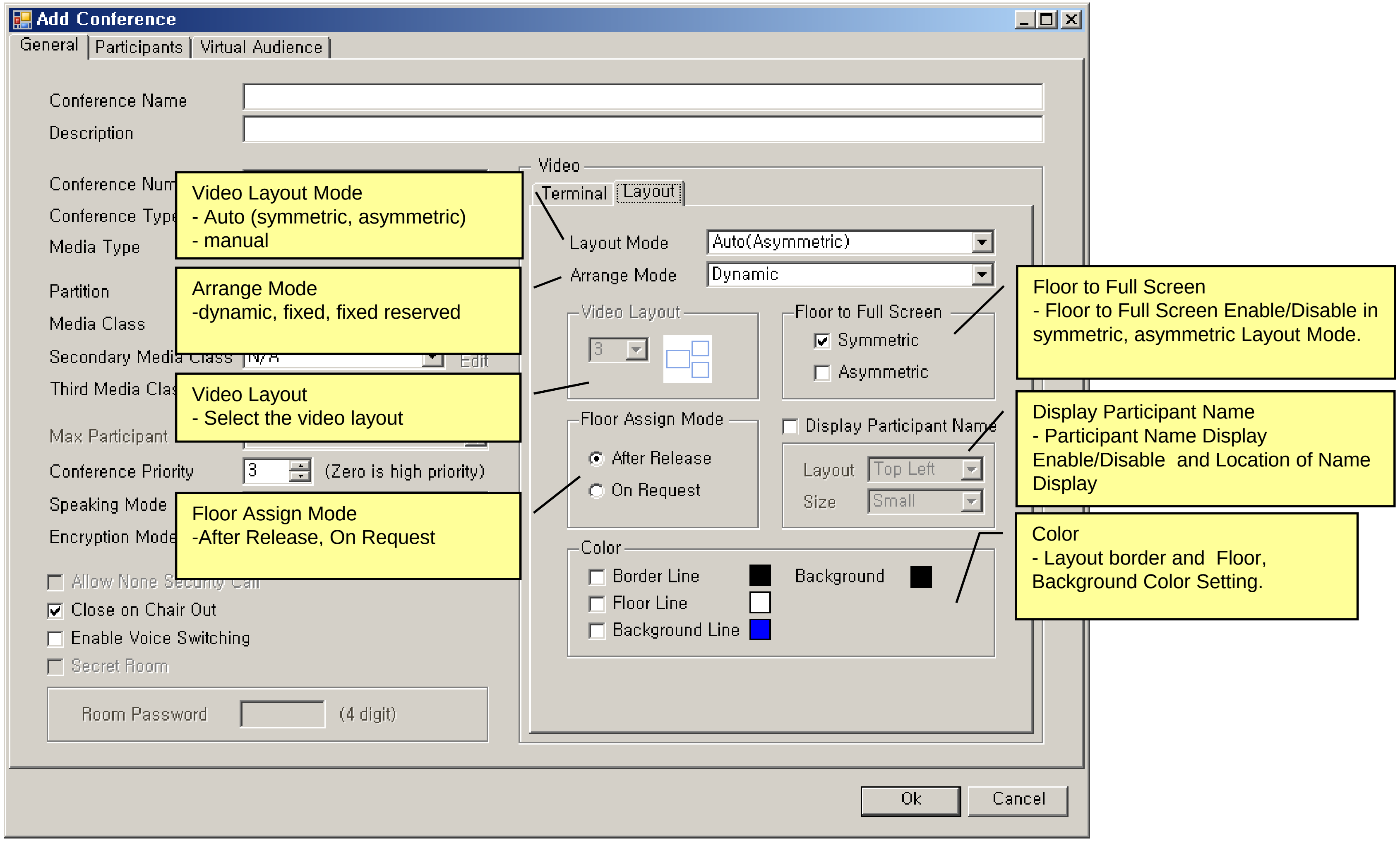


New SMM Feature : Voice Switching

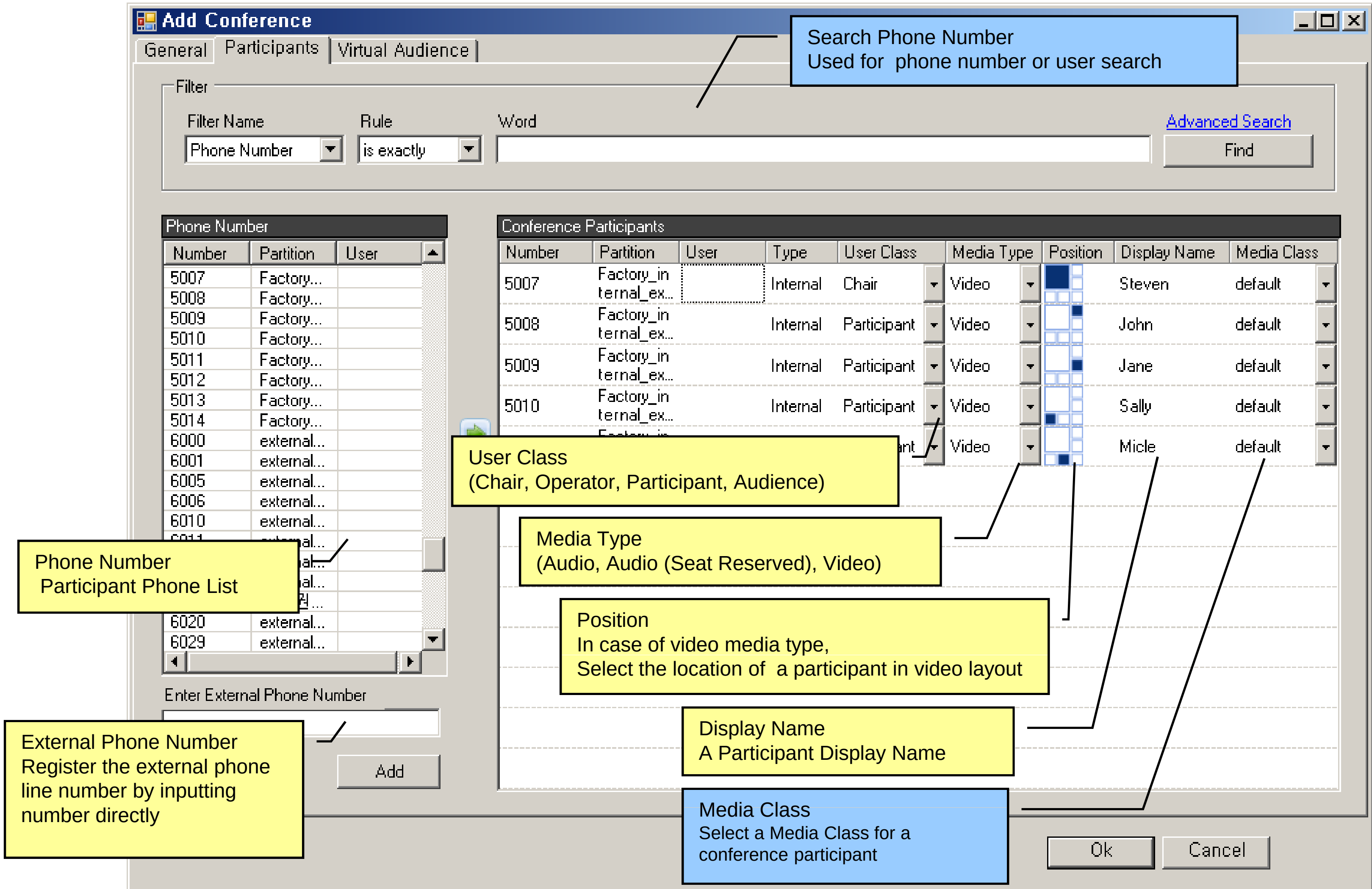
- Voice Switching
  - Voice Activity Switching or Voice Detect Switching
  - Detect the Voice Activity of participants during video conference, and dynamically change the MCU display layout mode to display a participant who is speaking out currently
  - If one participant obtains Floor, Voice Switching is inactive automatically till Floor is released.

- Display Priority Control
  - Floor > Voice Switching > Chair

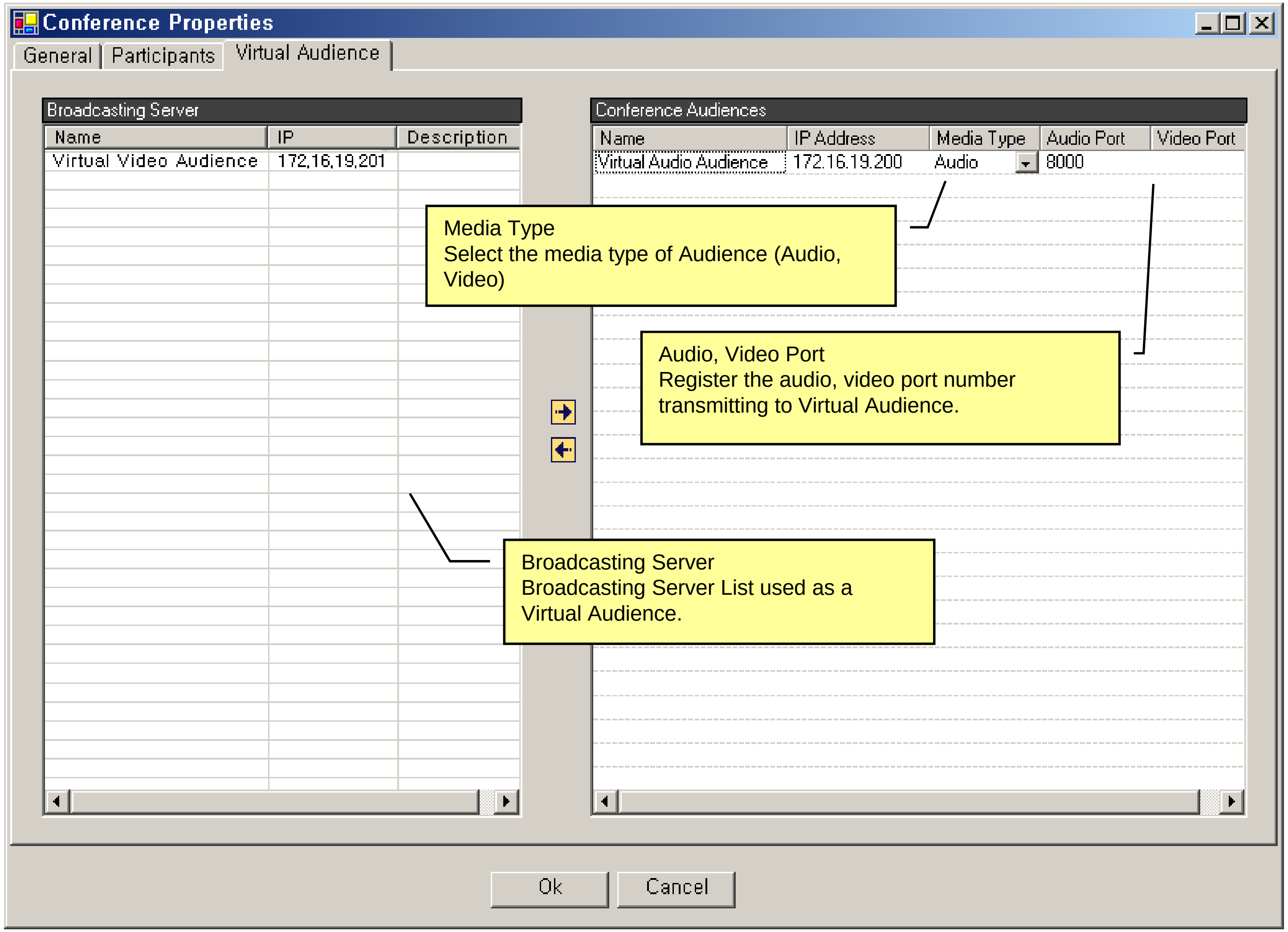
New SMM Feature : Conference Room - Layout



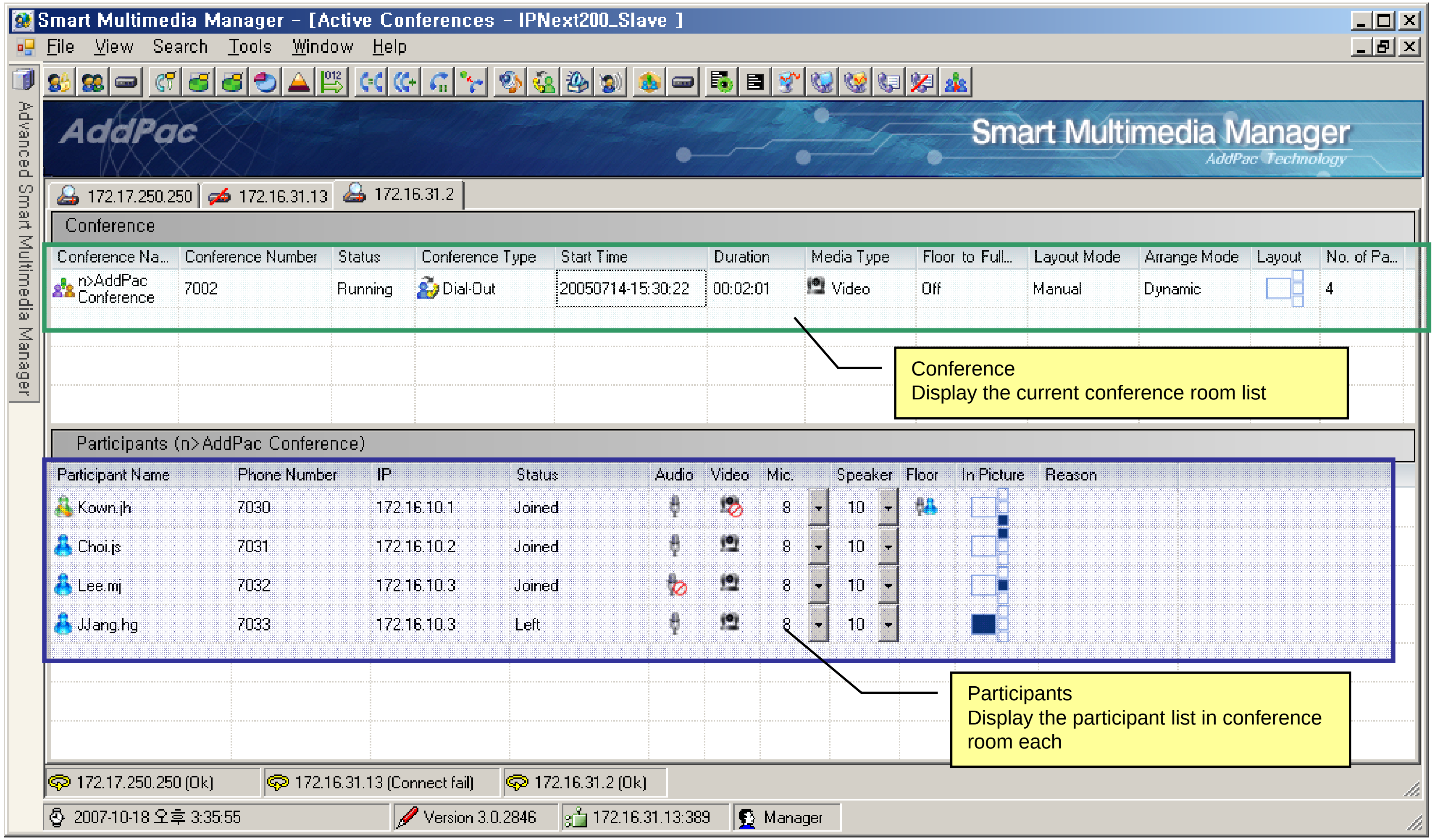
New SMM Feature : Conference Room - Participants



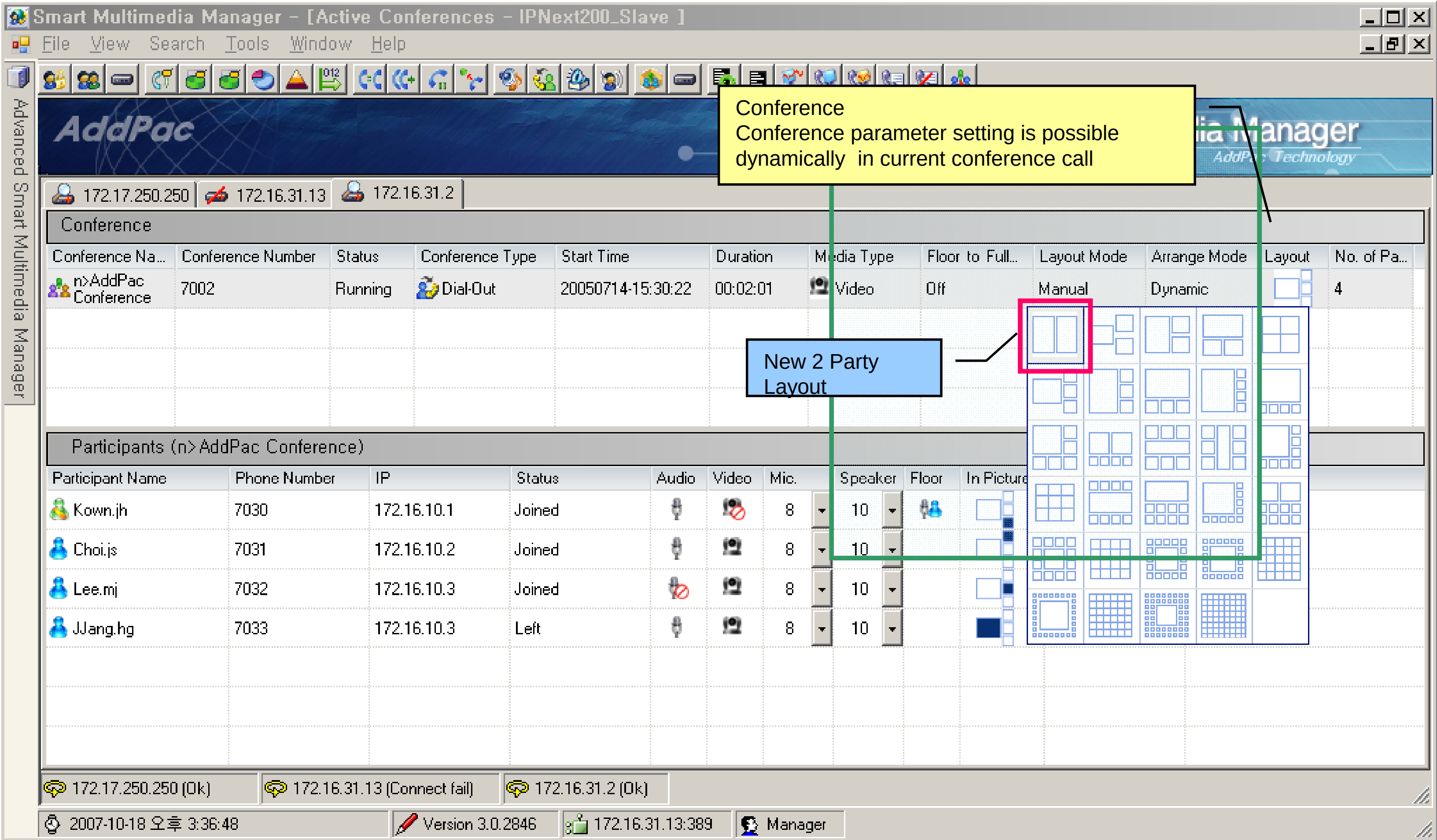
New SMM Feature : Conference Room – Virtual Audience



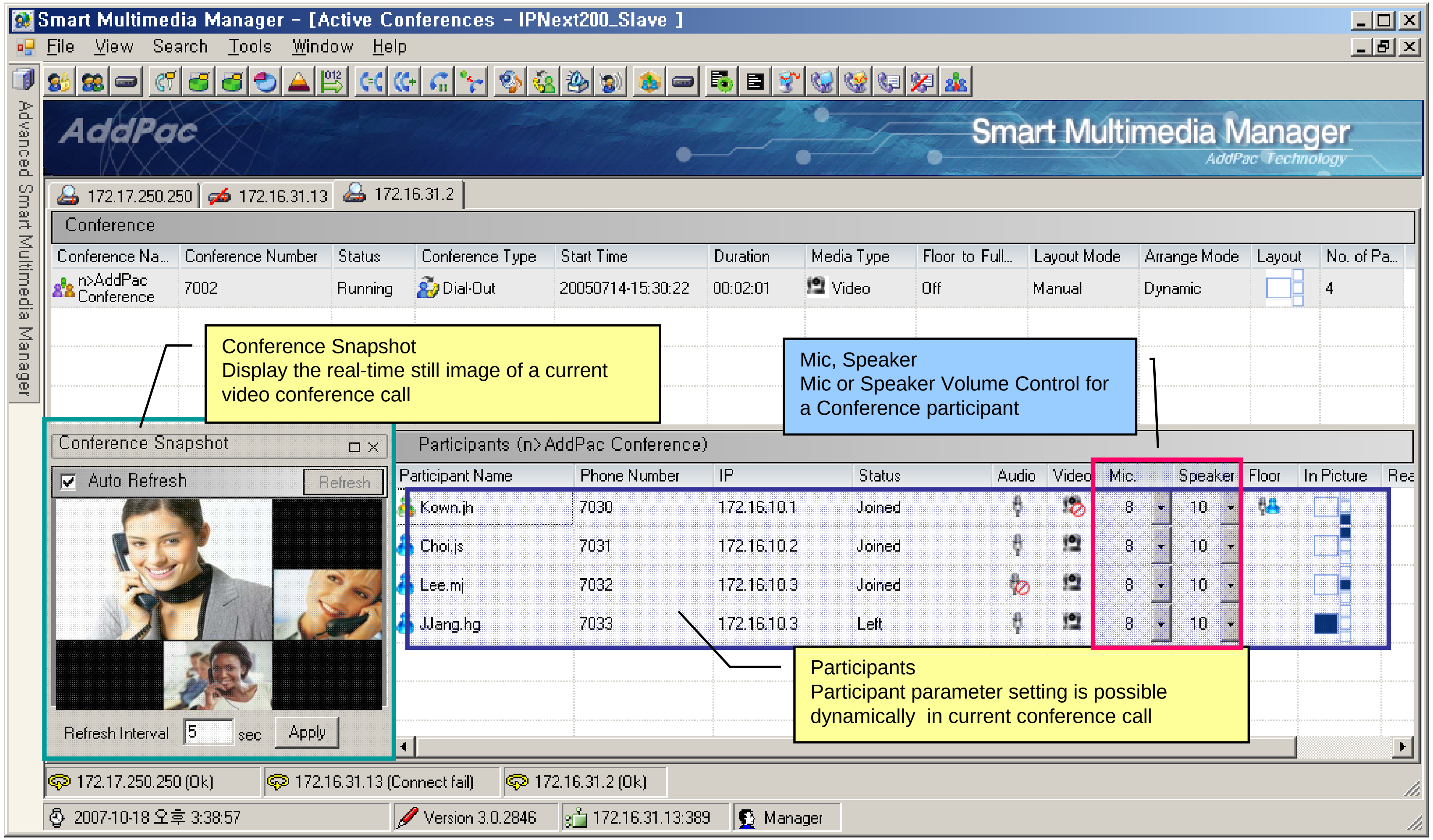
New SMM Feature : Active Conference (1)



New SMM Feature : Active Conference (2)

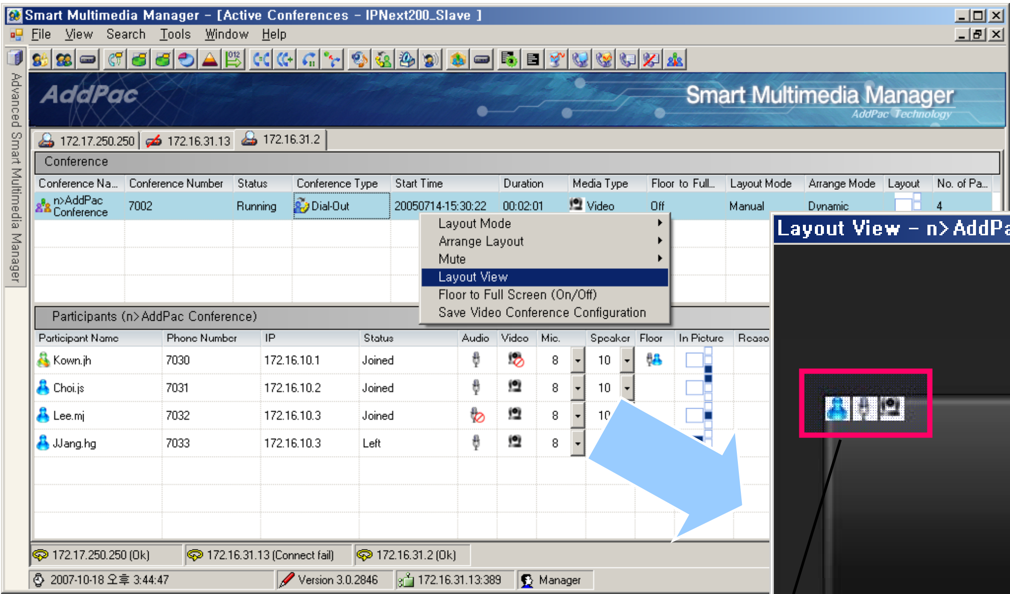


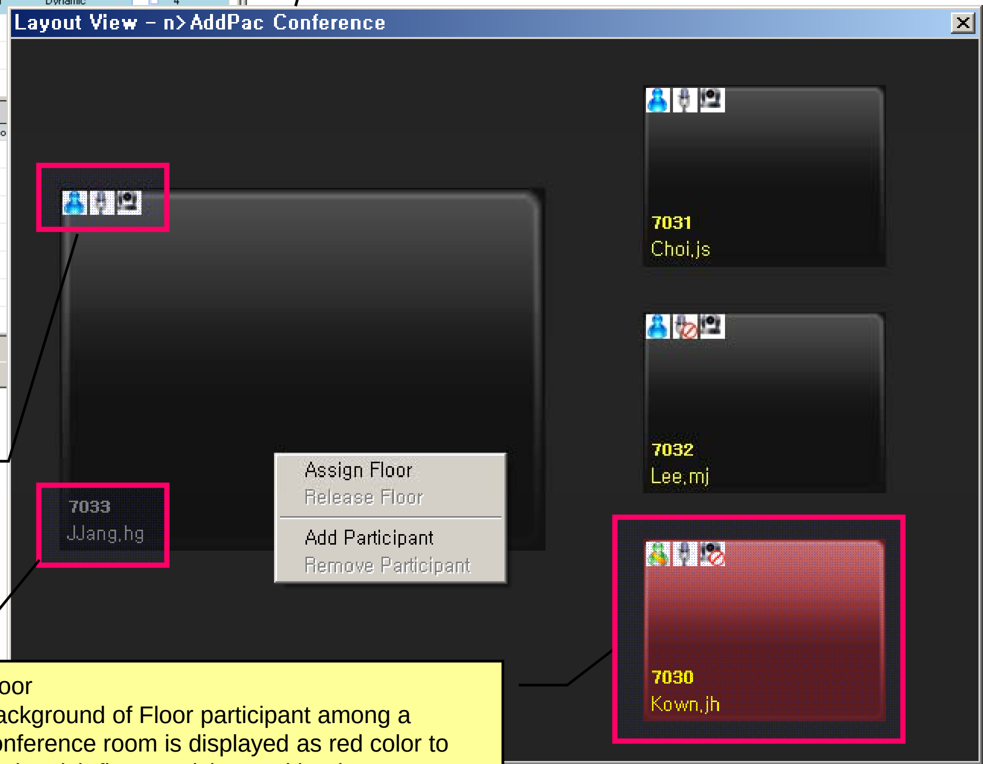
New SMM Feature : Active Conference (3)



New SMM Feature : Active Conference (4)

Layout view function provide the layout status of a conference participant





Layout View
User can change a participant layout easily via Mouse Drag & Drag at Layout view window. Floor configuration, participant add or delete function is also provided.

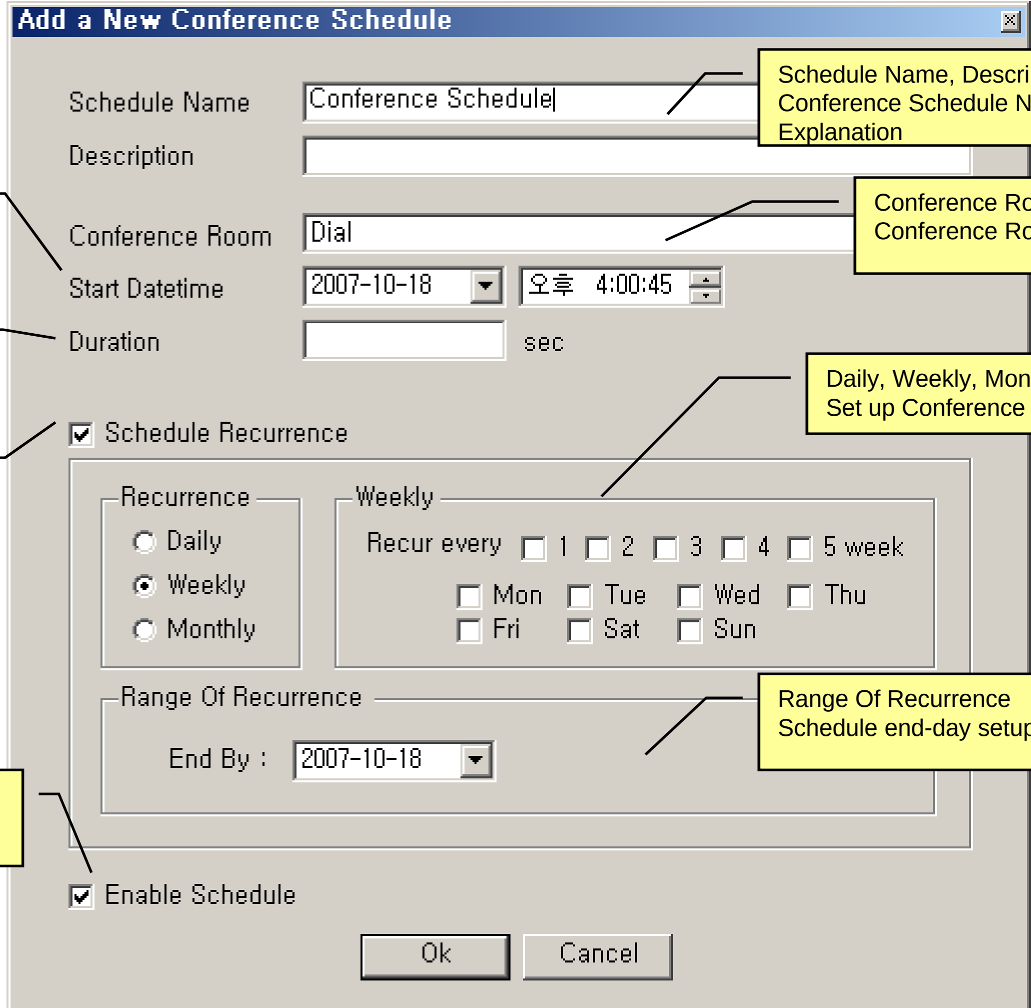
Indicator
User class configuration, audio/video on/off and configuration is possible

Status
Display the status of each participant such as phone number, nick name, alias. Using font color, show whether a user participate conference room or not. (yellow color : participating , grey color : not participating)

Floor
Background of Floor participant among a conference room is displayed as red color to distinguish floor participant with other participant.

New SMM Feature : Conference Schedule

Conference scheduling is used for configuration of repeated conference or a specific day conference



Schedule Name, Description
Conference Schedule Name and Explanation

Start Date time
Schedule start day and time setup

Conference Room
Conference Room Setup

Duration
Conference duration time (sec)

Daily, Weekly, Monthly
Set up Conference schedule

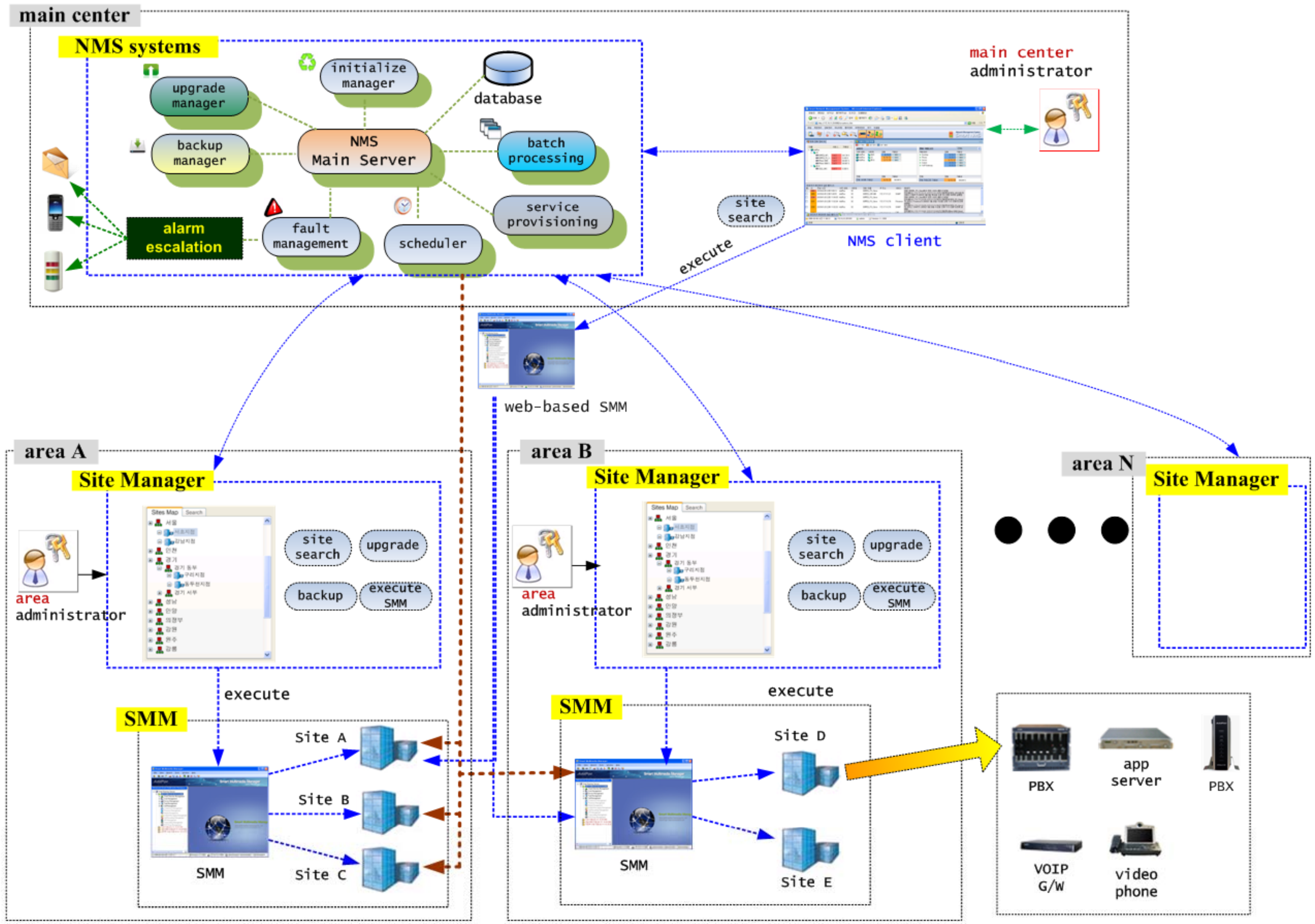
Schedule Recurrence
Repeated conference setup

Range Of Recurrence
Schedule end-day setup

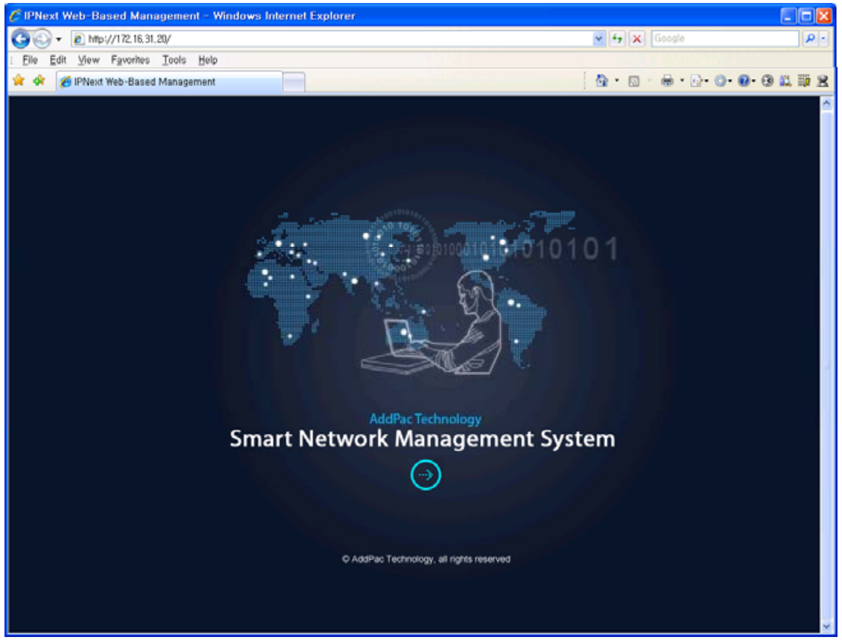
Enable Schedule
Schedule enable or disable Setup

NMS for AddPac Secure IP Telephony Solution

NMS System for Whole Site Management



Contents



IPNext Web-Based Management - Windows Internet Explorer
AddPac Technology
Smart Network Management System
© AddPac Technology, all rights reserved

System Requirement

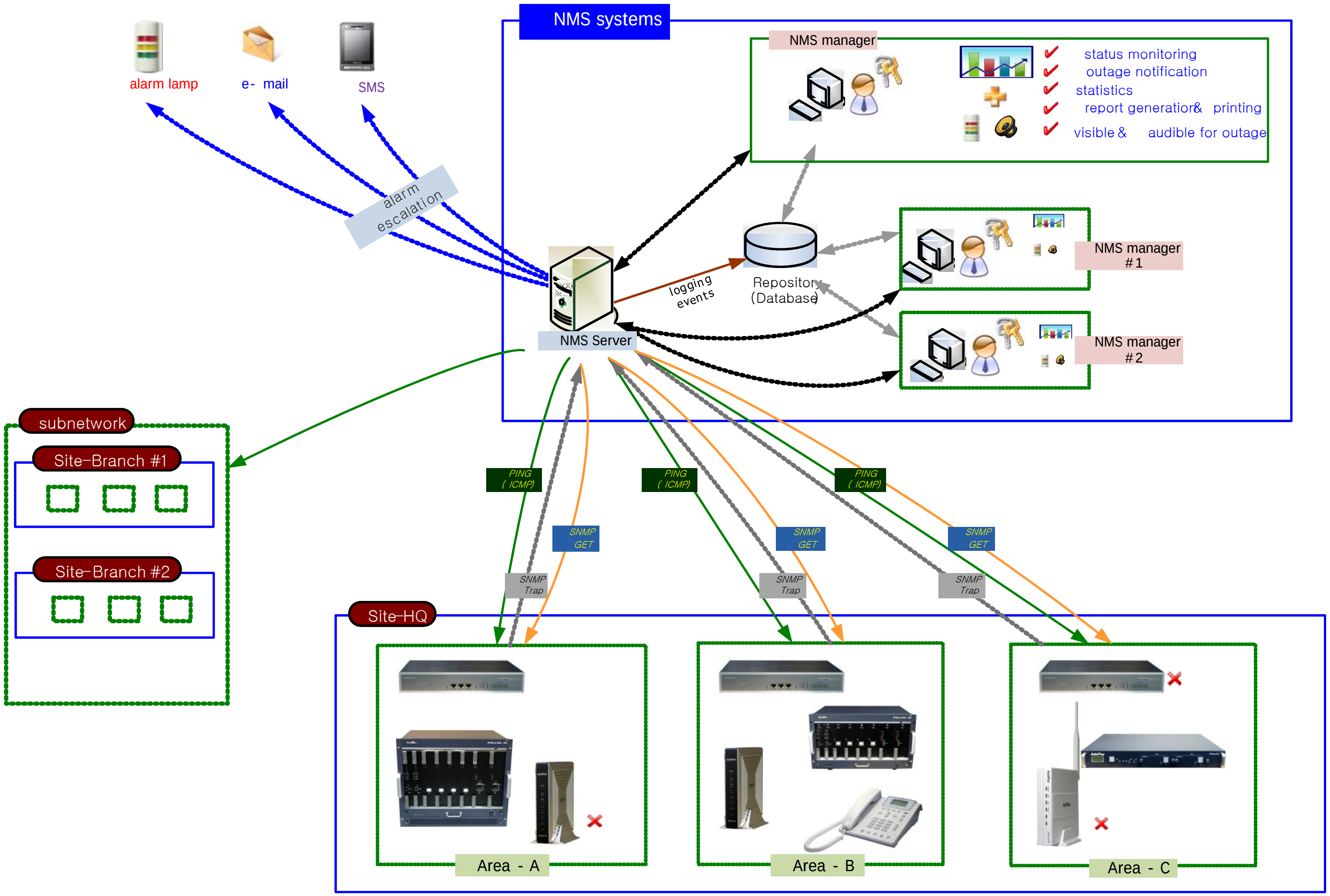
NMS Server

- OS : RHEL (Redhat Enterprise Linux) 5.0 or higher
- CPU : Quad-Core 2.0 GHz / 1333MHz FSB 2x4 MB cache
- Physical Memory : 4 GB
- HDD : 300 G
- JRE (Java Runtime Environment) 1.5.1 or Higher
- Database : PostgreSQL 8.1.11

NMS Client

- Windows XP, Vista, Windows Server 2000/2003
- Microsoft Internet Explorer 6.0 or higher

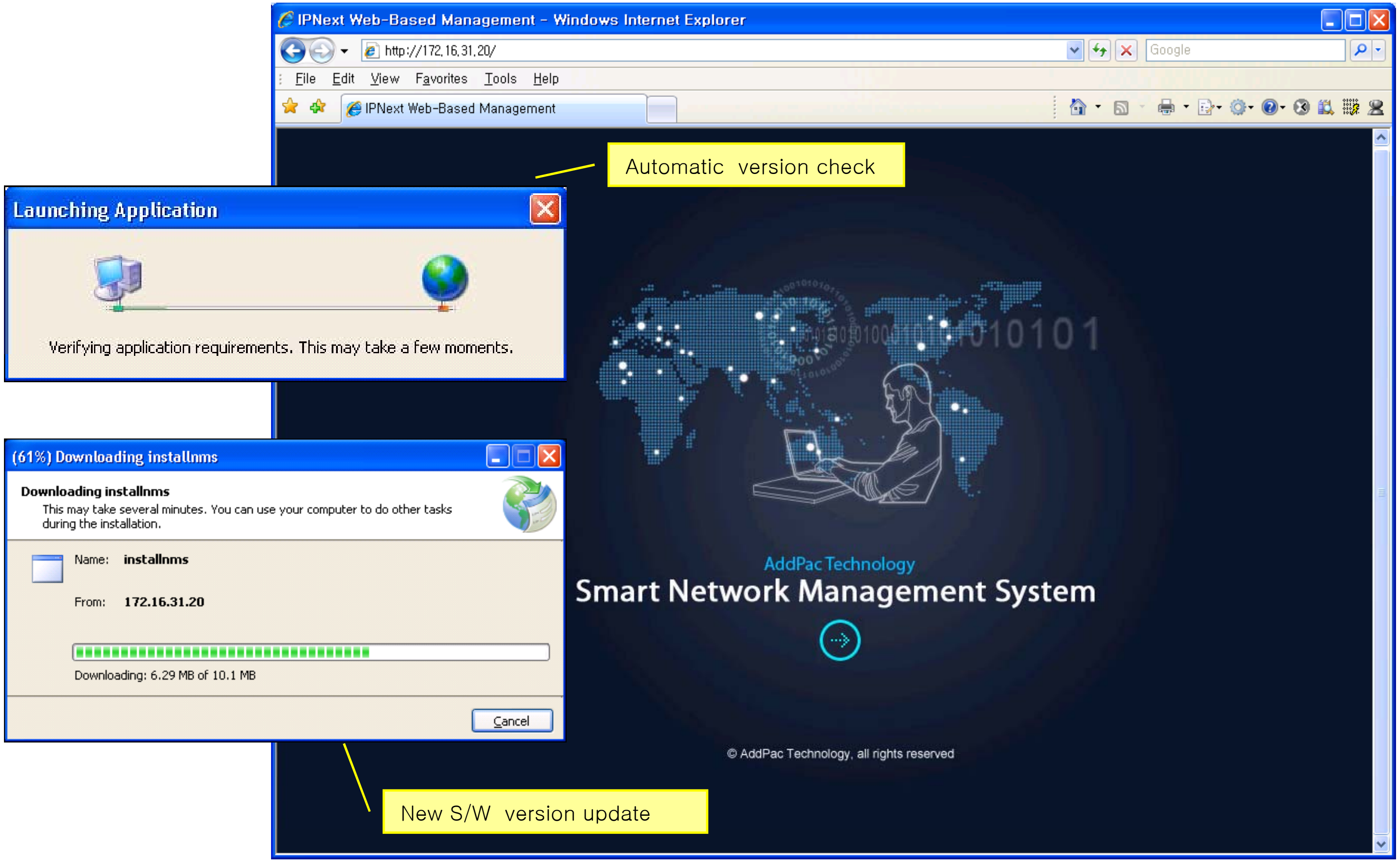
NMS Networking Diagram



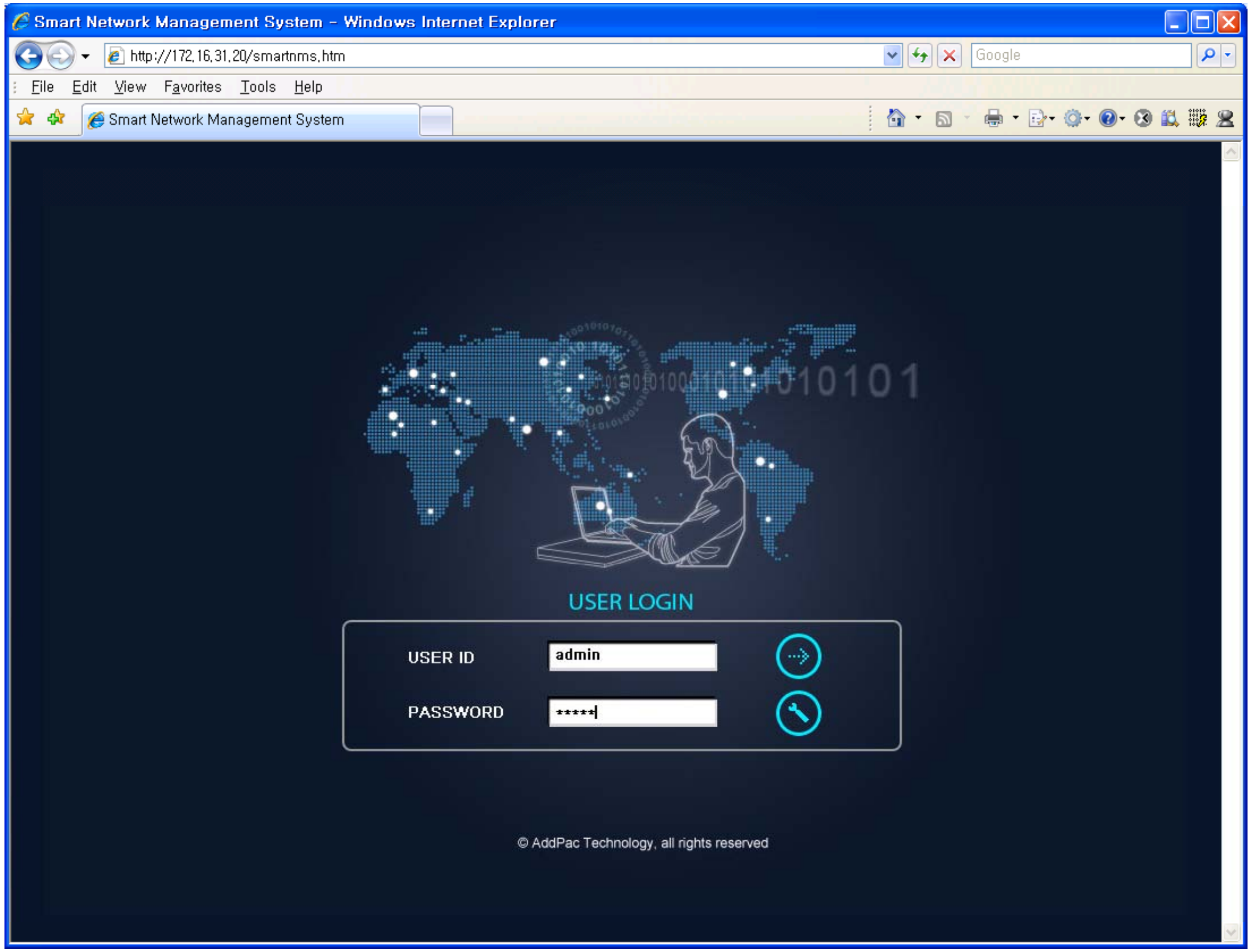
Web-based Management

- **Easy Access via Web browser**
  - Microsoft Internet Explorer 6.0 or higher compatible
- **Version Control**
  - Automatic version check
  - New version software download feature
- **UI control**
  - User friendly GUI management

Version Control

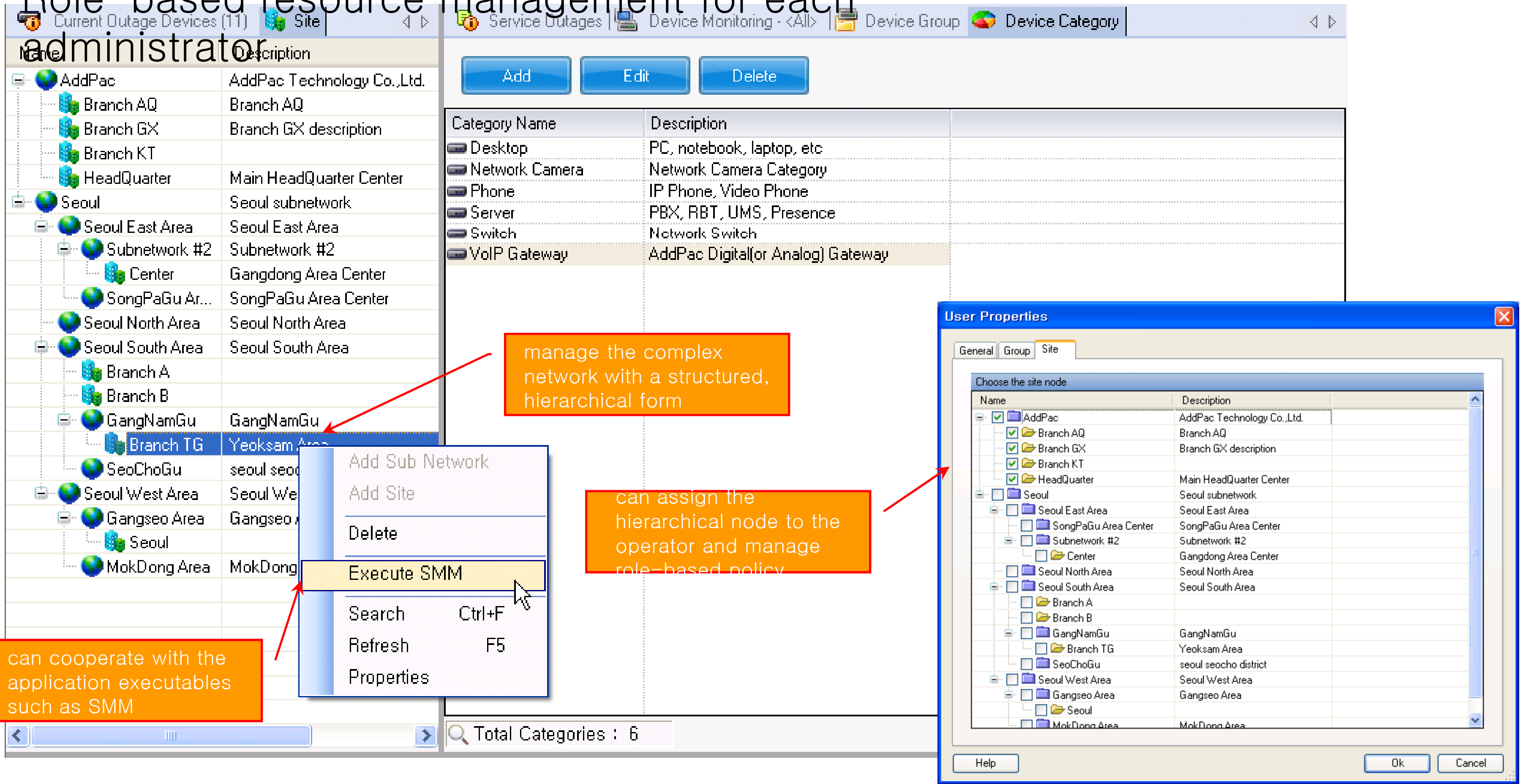


Web-based Login



Network Resource Management

- Network resource management with hierarchical structure
- Role-based resource management for each administrator



Device Fault Management

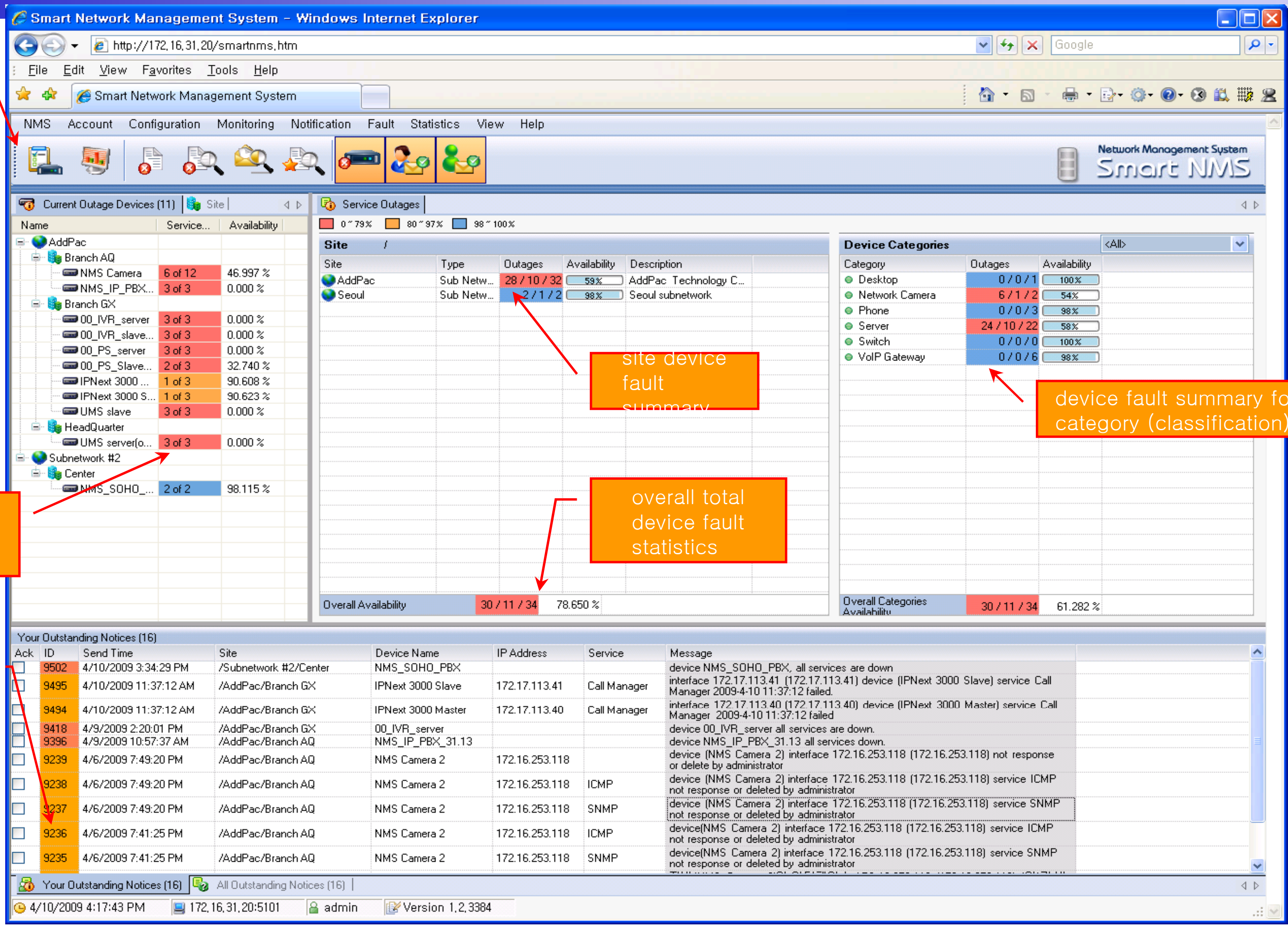
- Centralized fault summary information in main window
- Display current fault device through tree view
- Notify administrator with detailed fault information
- Provide device availability information for 24hrs

Device Fault Management

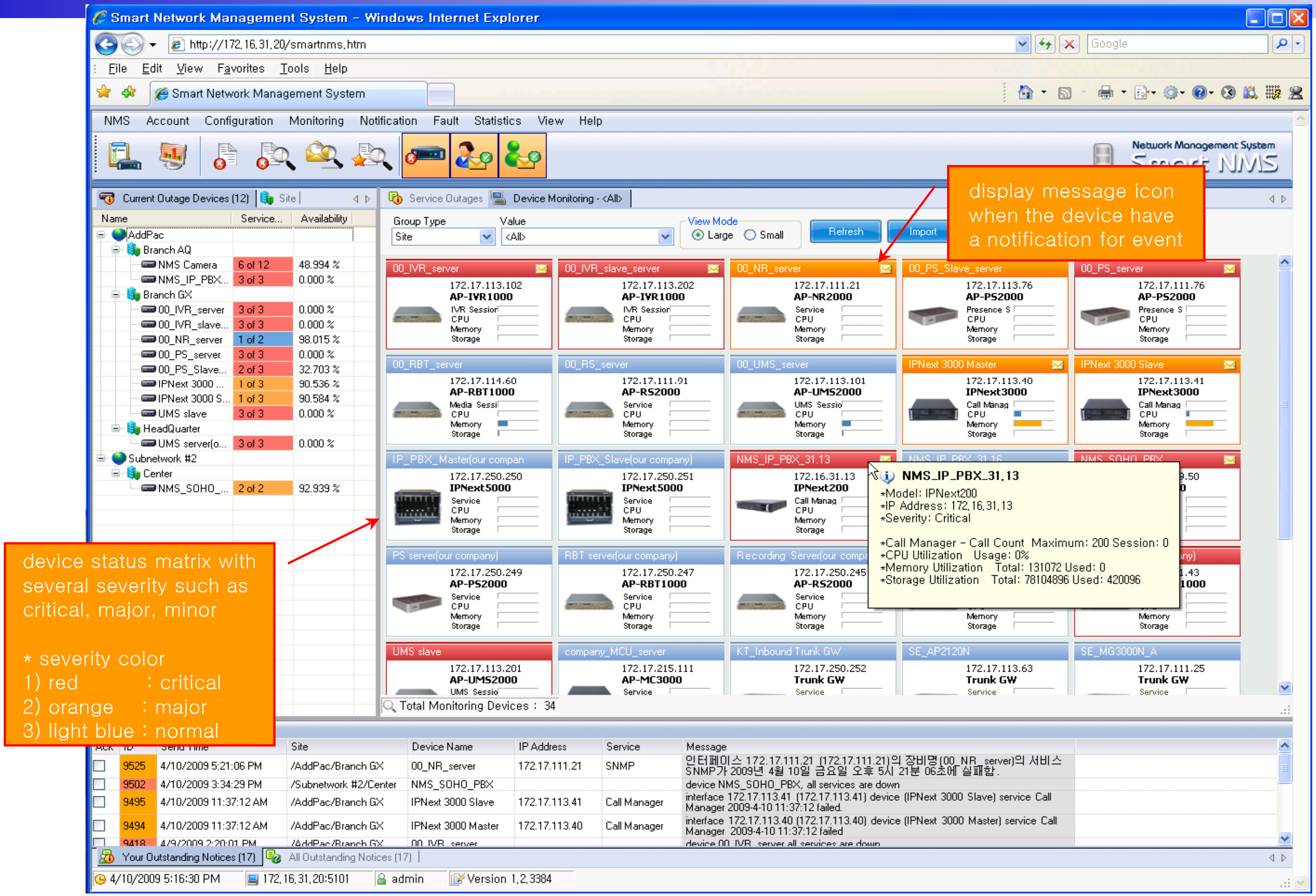
main window

current device fault list with hierarchy view

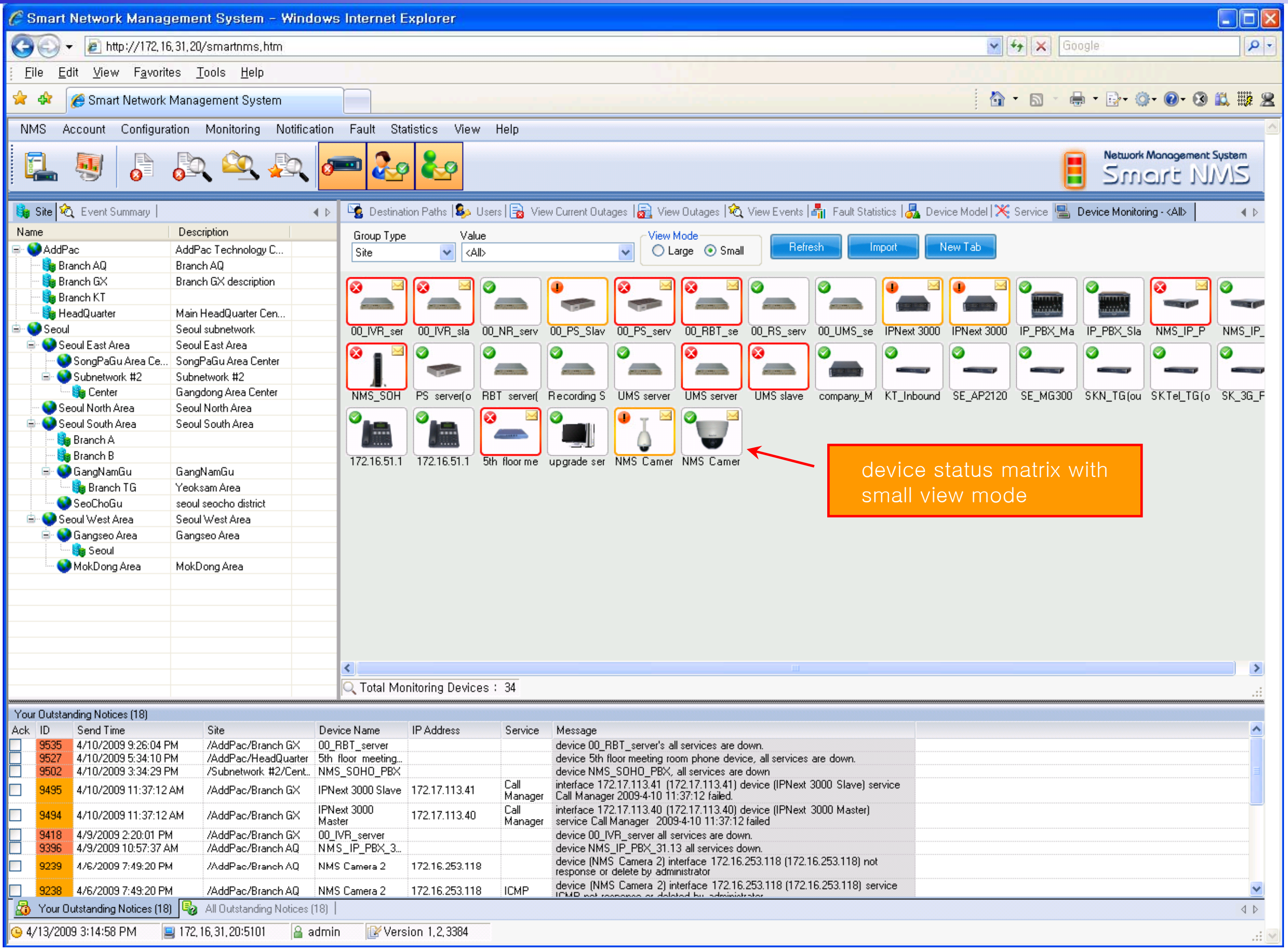
current device fault event message are shown as below



Device Fault Management



Device Fault Management



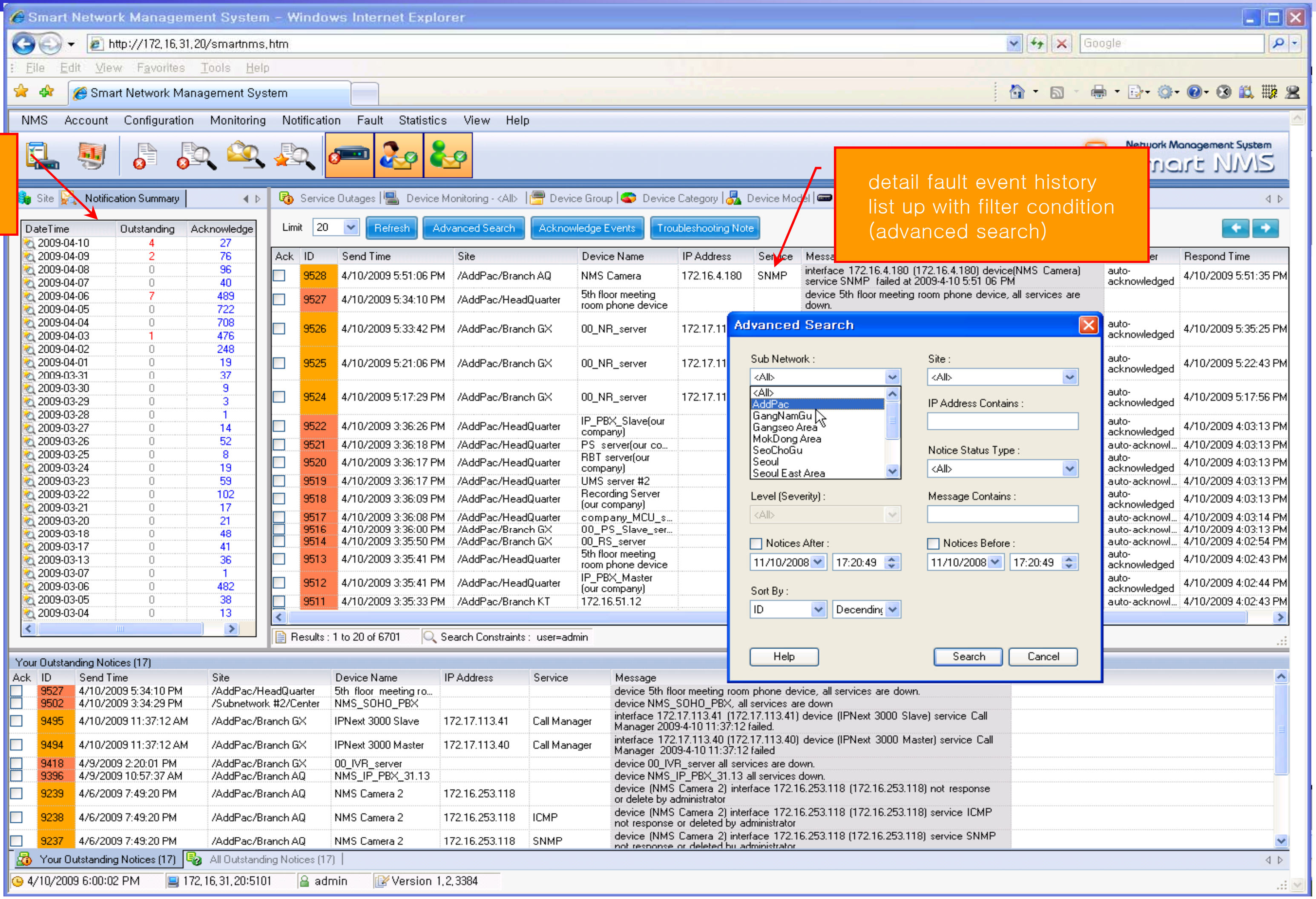
Device Fault History Management

- Provide both summary view and detailed event message
- Can Write troubleshooting job note for each event manually
- Administrator can query for a history fault with search condition
- Each fault is related to the several raw events

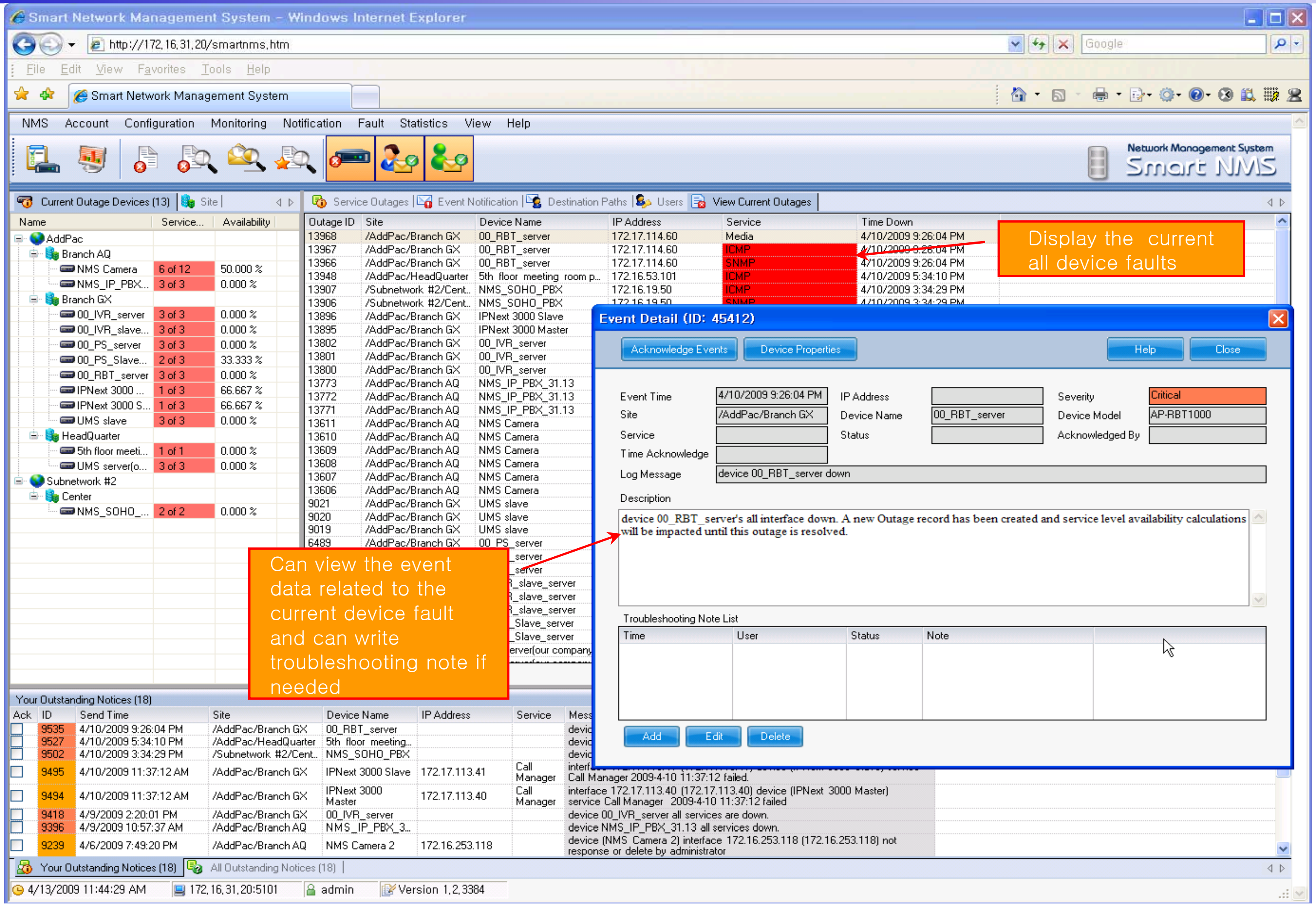
Device Fault History Management

daily fault event summary statistics information

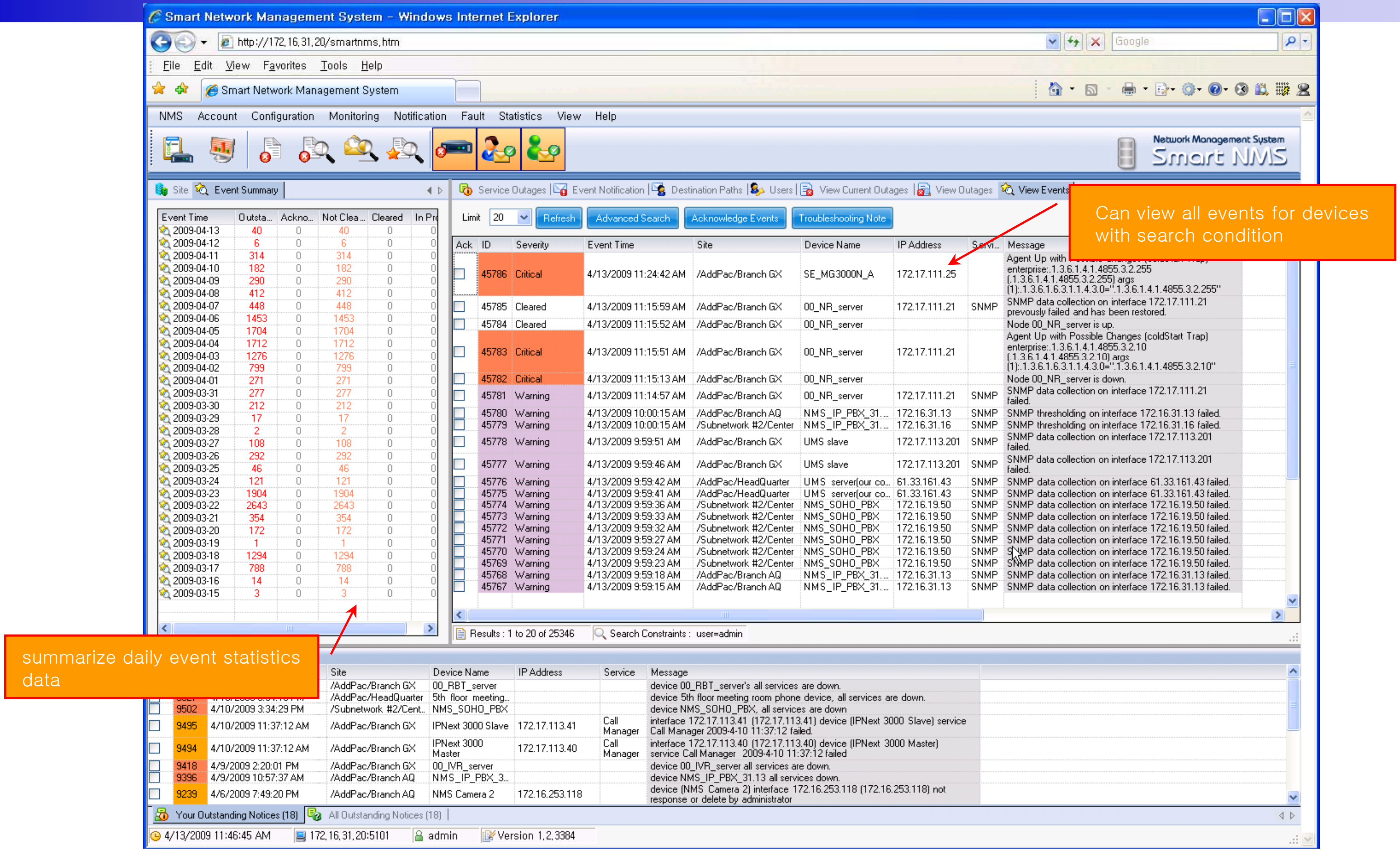
detail fault event history list up with filter condition (advanced search)



Current Device Fault (Outage)



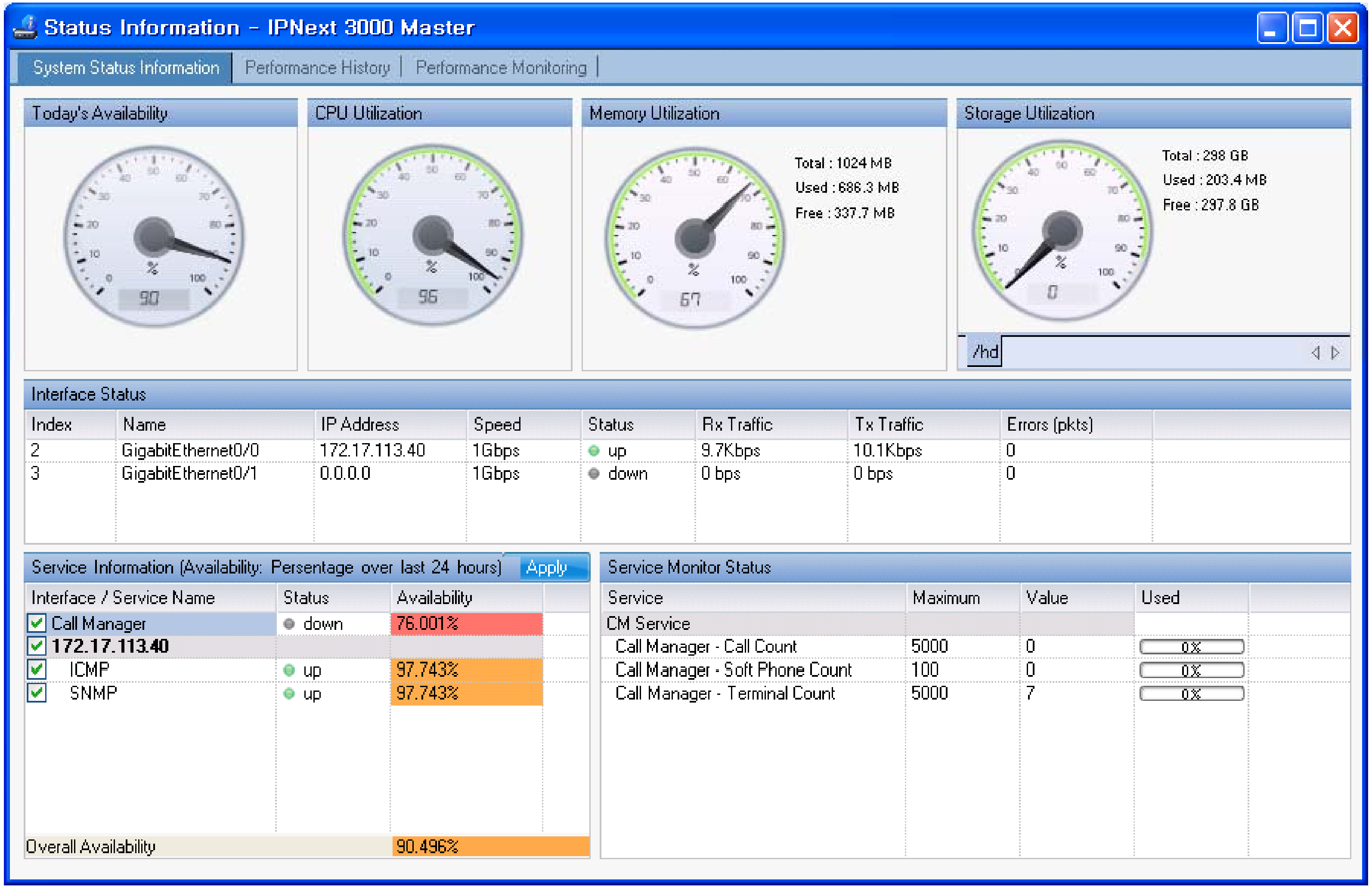
Device Event History



Device Status Information

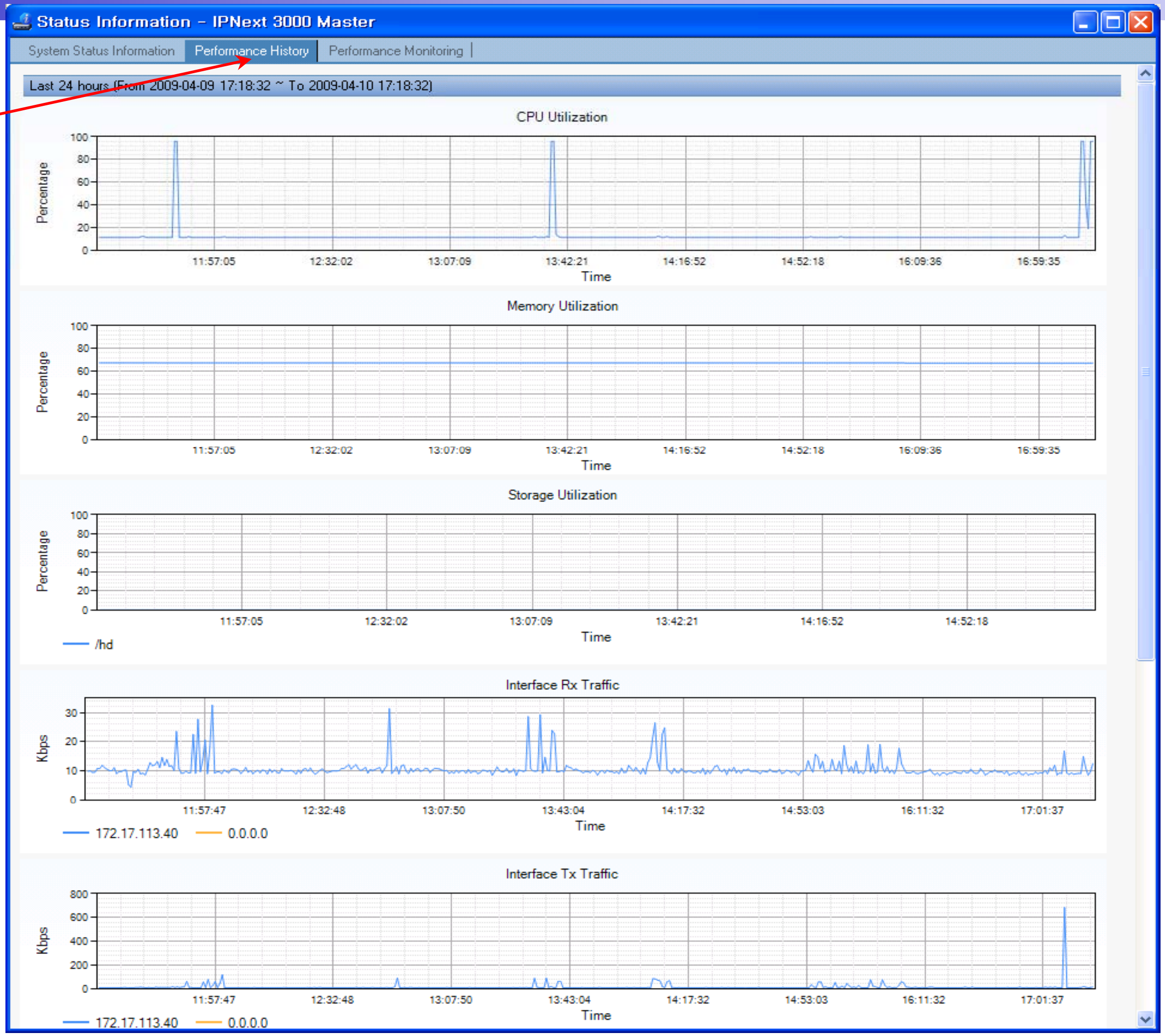
- System Performance Information (CPU, HDD, Memory,…)
- Provide device current service status (up/down)
- Provide device main status (max value vs current value)
- Display Graph Series with System Performance Information
- Monitor Main Status Flow with System Monitoring View

Device Status Information

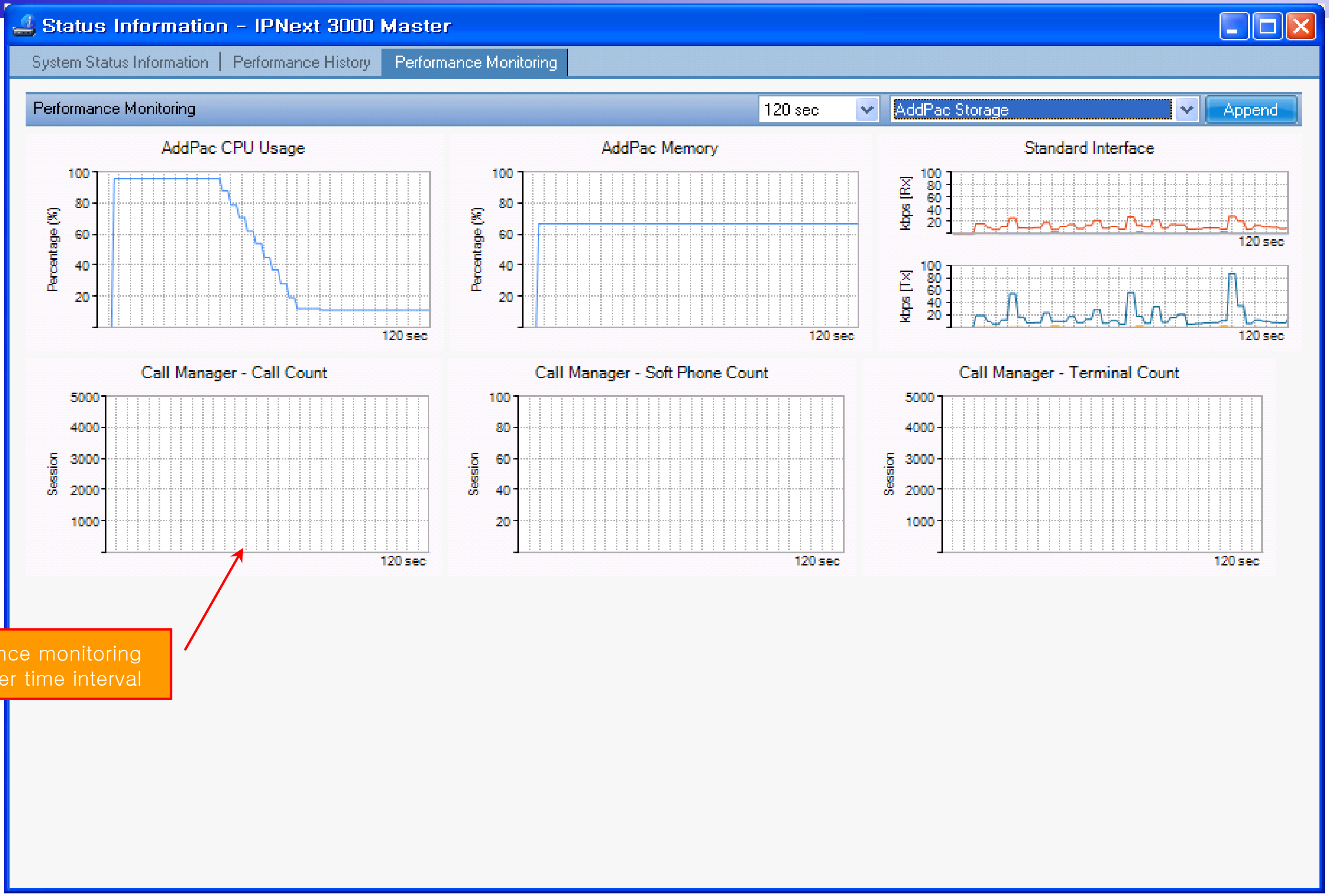


Device Status Information

performance analysis
graph for last 24 hours



Device Status Information

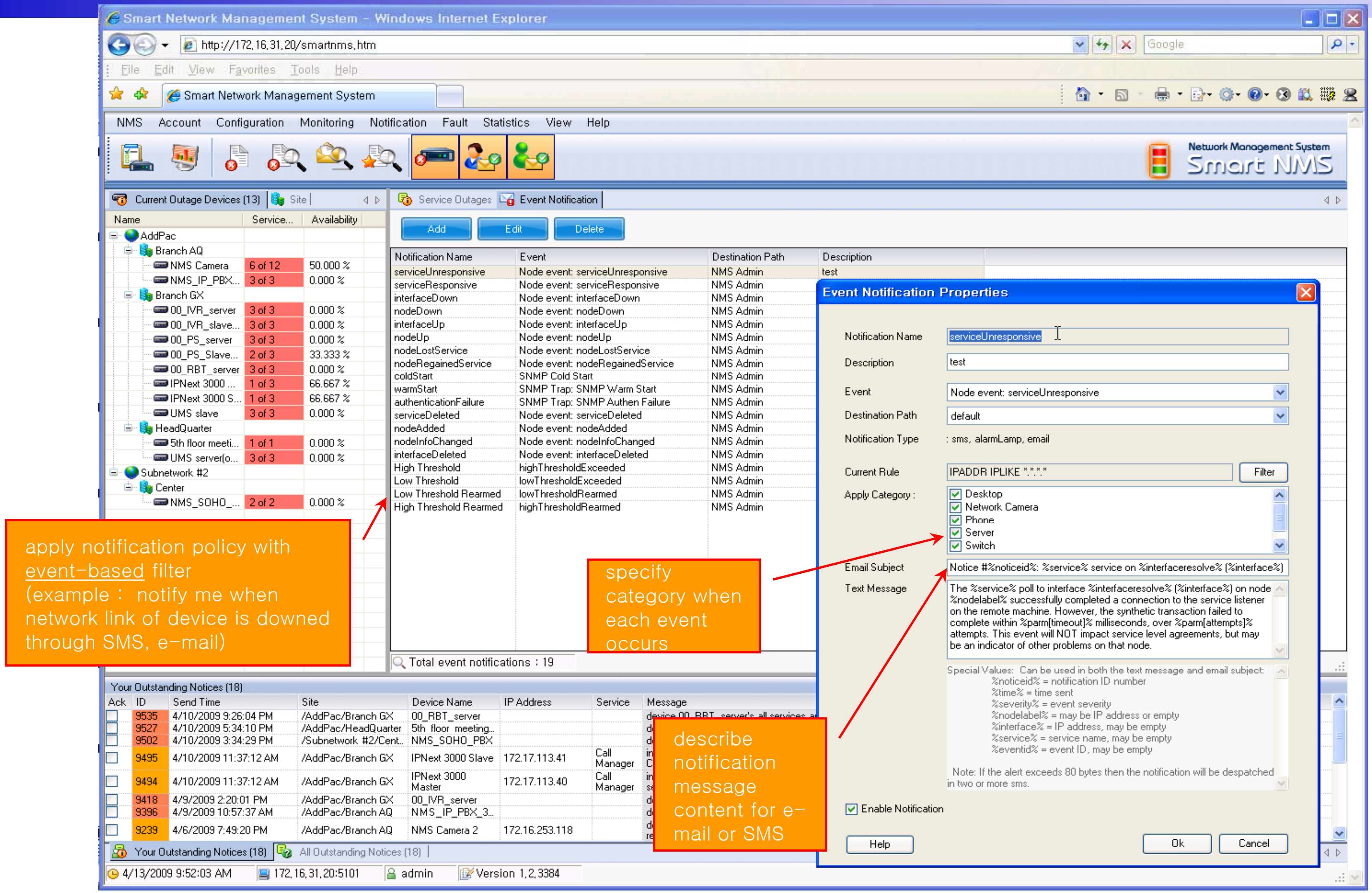


performance monitoring
with proper time interval

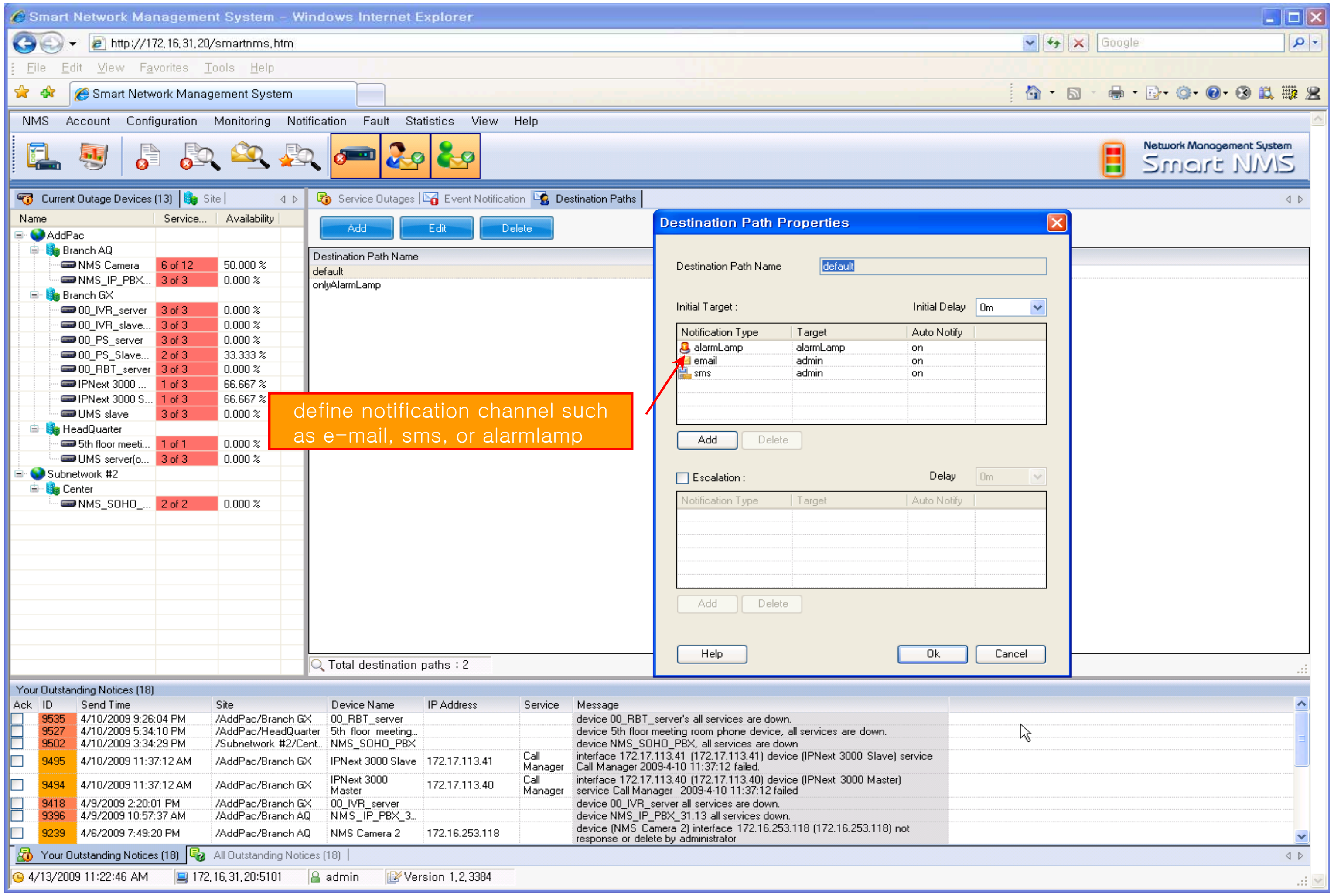
Notification Management

- Notify administrator for important event such as critical device fault when proper action needs
- Provide several notification channel such as SMS, e-mail, alarm lamp
- Notification channel configuration for each event
- Manage notification with device category such as Server, Terminal, PC, etc
- Provide Alarm with audible (play sound), visible (alarm lamp) form

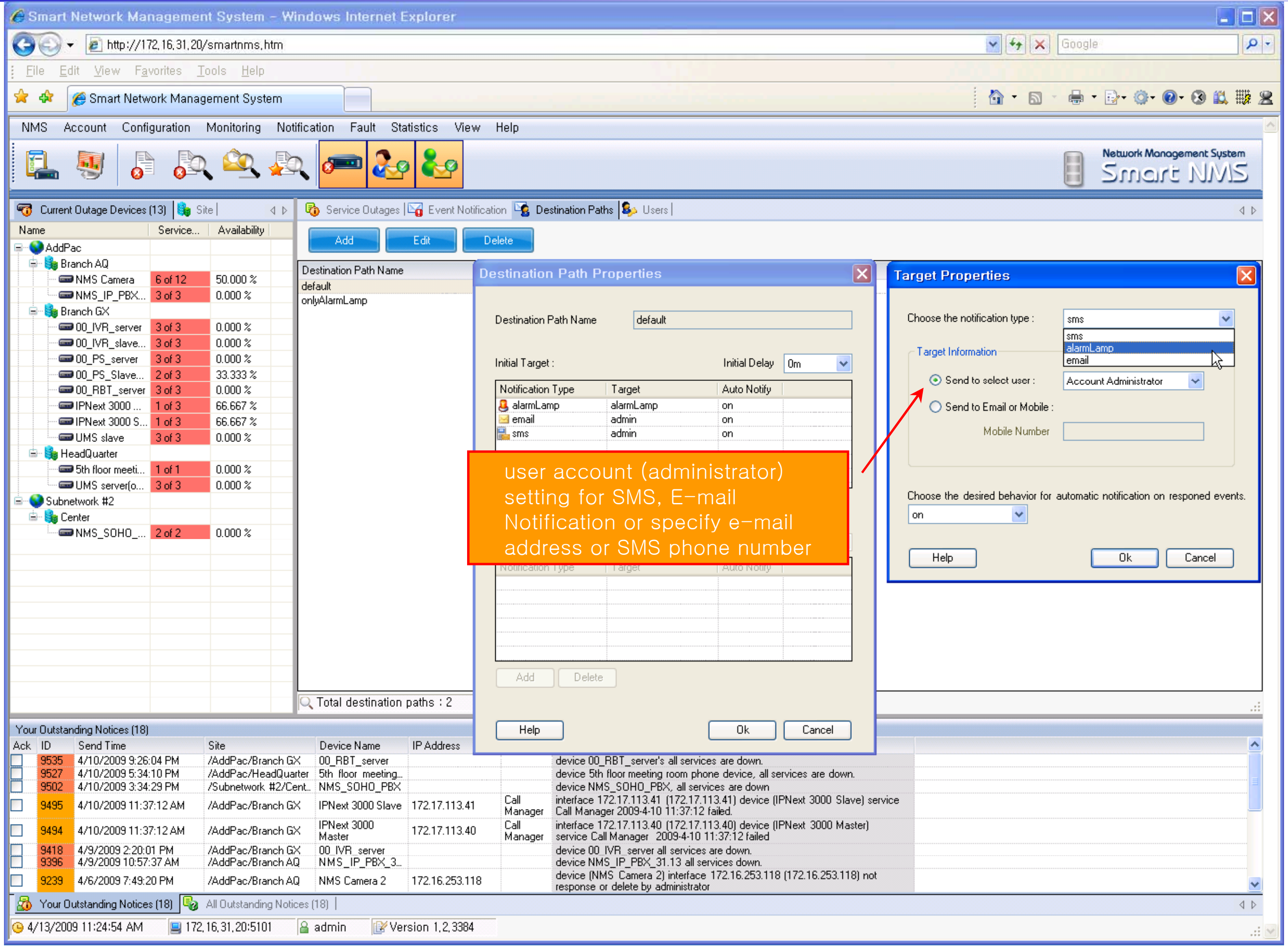
Event Notification Management



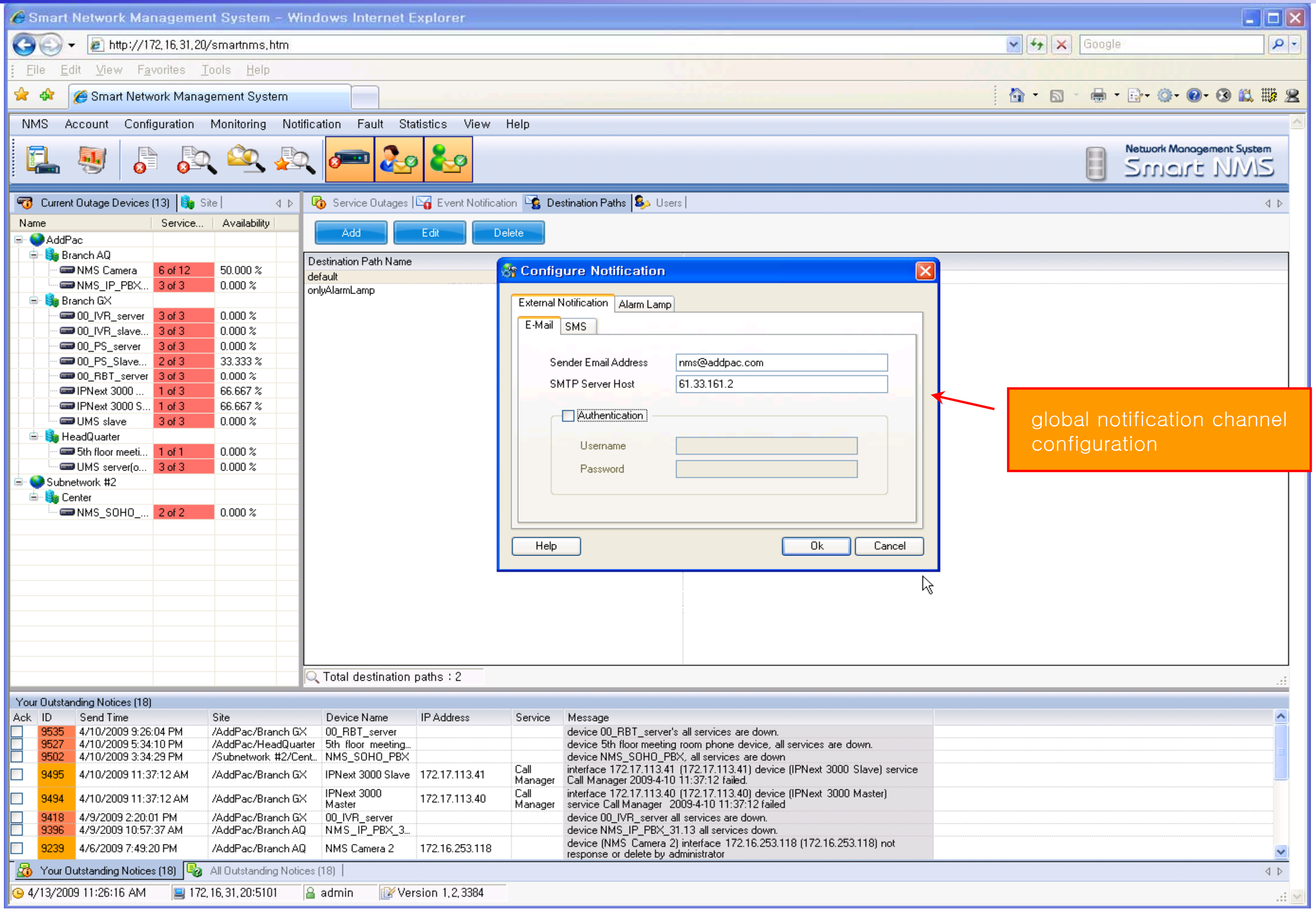
Event Notification Management



Event Notification Management

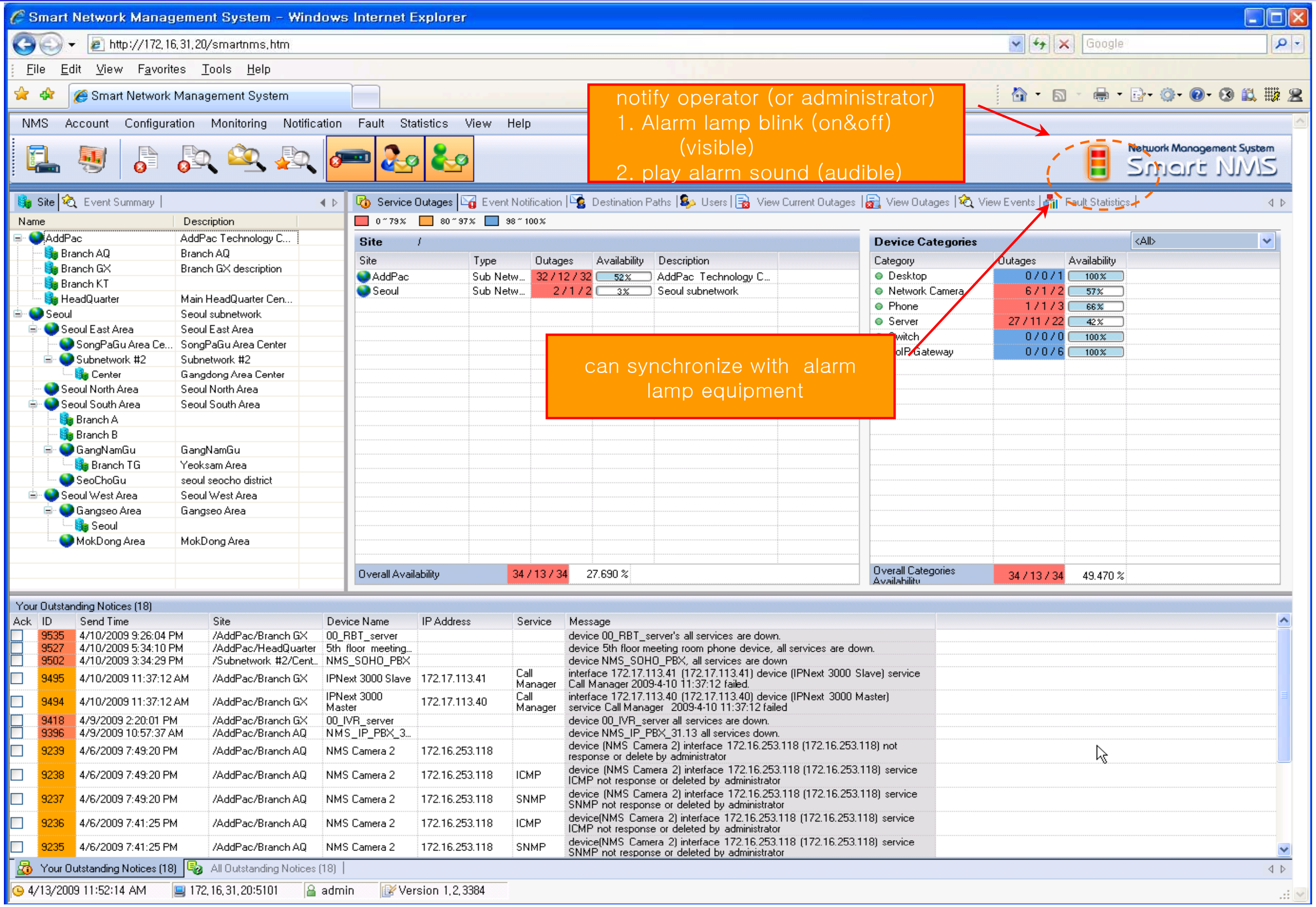


Configuration



global notification channel configuration

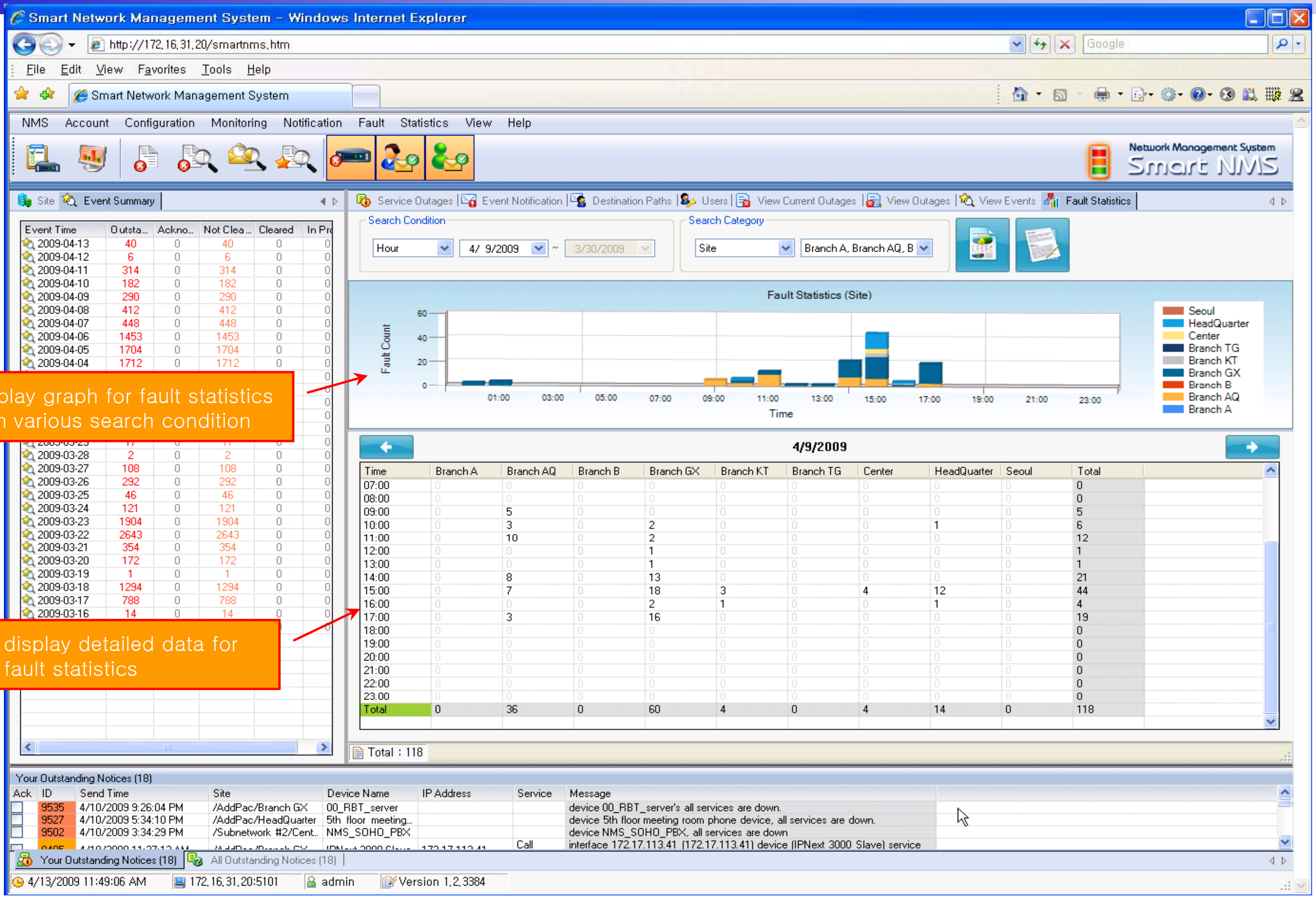
Audible & Visible Alarm



Fault Statistics

- analyze for a fault event with graph and detailed list data
- Report form generation and print out for statistics result

Fault Statistics



display graph for fault statistics with various search condition

display detailed data for fault statistics

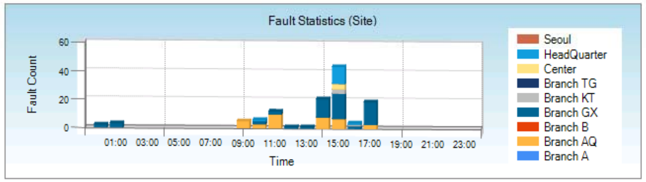
Fault Statistics – Report Generation

report generation for fault statistics and print out the result

Statistics Report

Fault Statistics AddPac

Search Condition 4/9/2009
Search Category Site



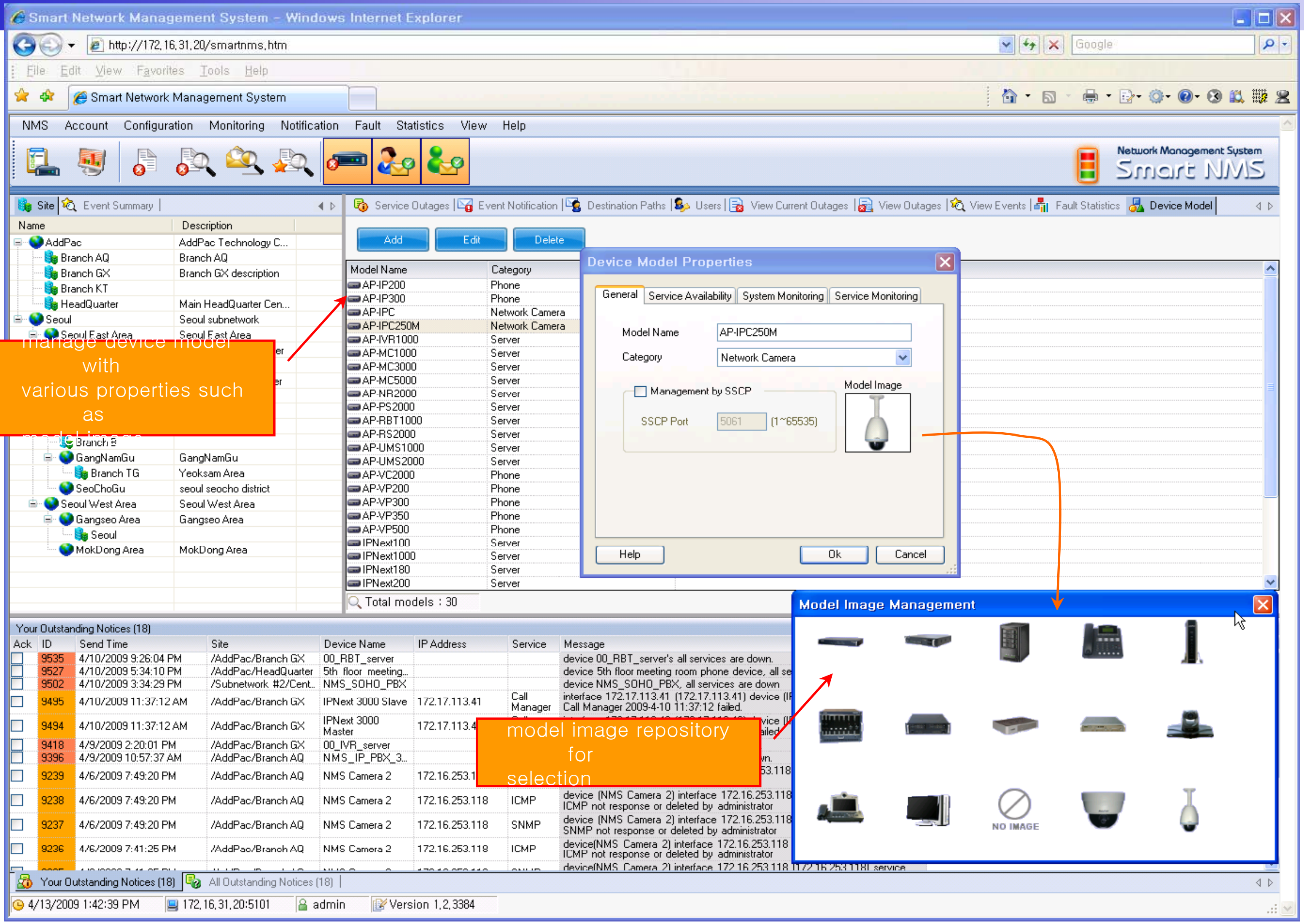
Time	Branch A	Branch AQ	Branch B	Branch GX	Branch KT	Branch TG	Center	HeadQuarter	Seoul	Total
00:00	0	0	0	2	0	0	0	0	0	2
01:00	0	0	0	3	0	0	0	0	0	3
02:00	0	0	0	0	0	0	0	0	0	0
03:00	0	0	0	0	0	0	0	0	0	0
04:00	0	0	0	0	0	0	0	0	0	0
05:00	0	0	0	0	0	0	0	0	0	0
06:00	0	0	0	0	0	0	0	0	0	0
07:00	0	0	0	0	0	0	0	0	0	0
08:00	0	0	0	0	0	0	0	0	0	0
09:00	0	5	0	0	0	0	0	0	0	5
10:00	0	3	0	2	0	0	0	1	0	6
11:00	0	10	0	2	0	0	0	0	0	12
12:00	0	0	0	1	0	0	0	0	0	1
13:00	0	0	0	1	0	0	0	0	0	1
14:00	0	8	0	13	0	0	0	0	0	21
15:00	0	7	0	18	3	0	4	12	0	44
16:00	0	0	0	2	1	0	0	1	0	4
17:00	0	3	0	16	0	0	0	0	0	19
18:00	0	0	0	0	0	0	0	0	0	0
19:00	0	0	0	0	0	0	0	0	0	0
20:00	0	0	0	0	0	0	0	0	0	0
21:00	0	0	0	0	0	0	0	0	0	0
22:00	0	0	0	0	0	0	0	0	0	0
23:00	0	0	0	0	0	0	0	0	0	0
Total	0	36	0	60	4	0	4	14	0	118

Print Time : 4/13/2009 11:50:15 AM

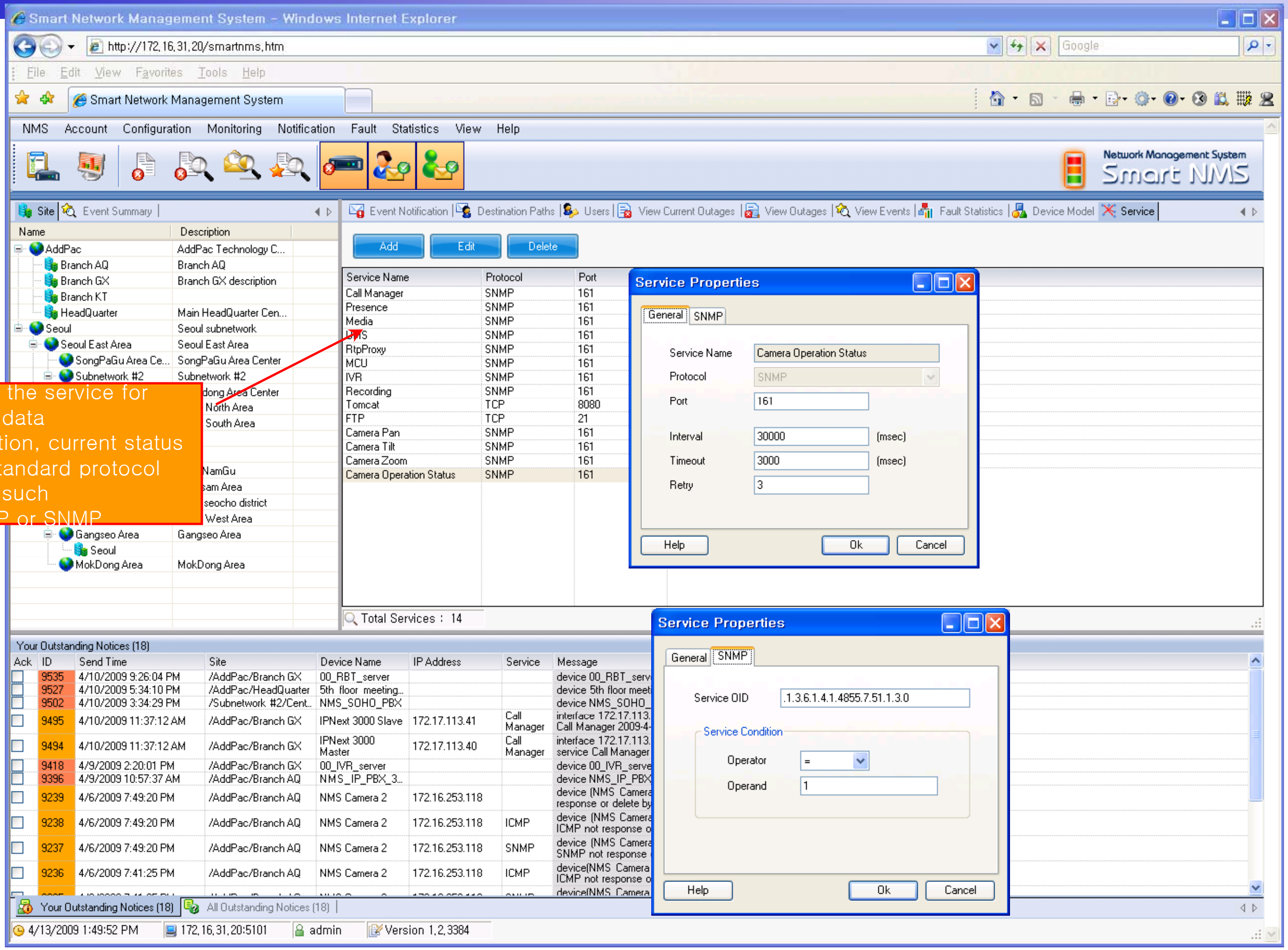
Model & Service Management

- Define new model with provided template image & properties
- Customize data collection with standard protocol such as TCP, SNMP

Device Model Management



Service Definition



define the service for data collection, current status with standard protocol such as TCP or SNMP

Thank you!

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