IPNext PBX Series

[Smart Multimedia Manager Operation Guide]

Edition 2.30
(For IPNext PBX v8.43)
(For SMM v3.0)
Dec. 2006

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IPNext 200 IP-PBX

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The revision history of the IPNext PBX Installation and Operation Manual is listed in the followings:

Revision	Date	Change Summary	Prepared by
Version 1.00	November 28,2005	Initial Released	R&D of AddPac Technology
Version 1.10	February 2, 2006		R&D of AddPac Technology
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Version 2.20	December 28, 2006	SMM v2.9	R&D of AddPac Technology

Chapter 1. Installing SMM

Installing and Operating SMM

The AddPac SMM (Smart Multimedia Manager) is GUI-based, easy-to-use program to operate and manage IPNext PBX Series.
The AddPac SMM features allow Smart Directory Server of IPNext Series PBX to set the user, equipment and telephone numbers and other extra features. The Smart Directory Server also can verify the call-on-line and all kinds of Event and Log in real time basis.

SMM Installation

- Perform smm_setup.exe which is installed in PC
- Start Step 1 of SMM Install in the following Figure 1-1

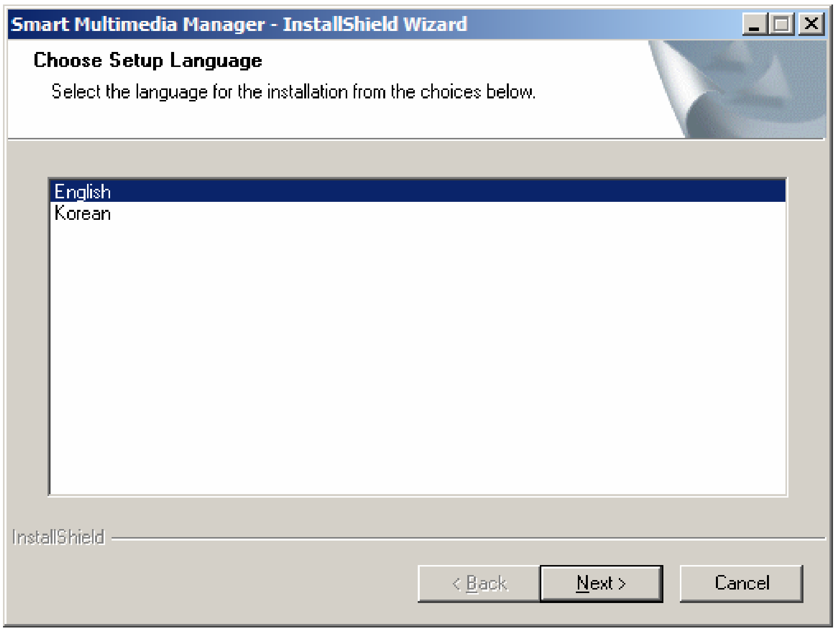

Figure 1-1 SMM Installation - Step 1

Select the choice of the given language either English or Korean, then click (N).
To stop installation, click ‘Cancel’.

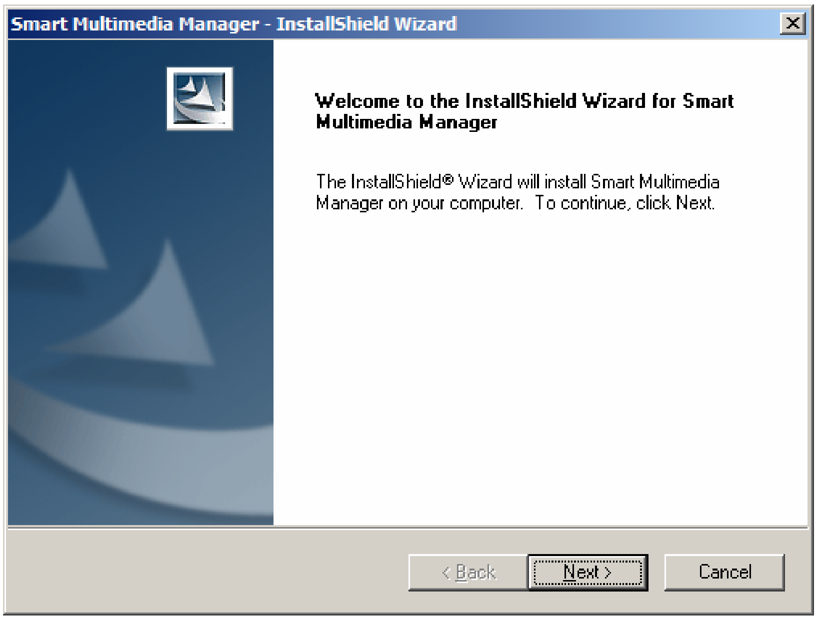

Figure 1-2 SMM Installation - Step 2

Click 'Next' (N) to continue installation. Click 'Cancel' to stop installation

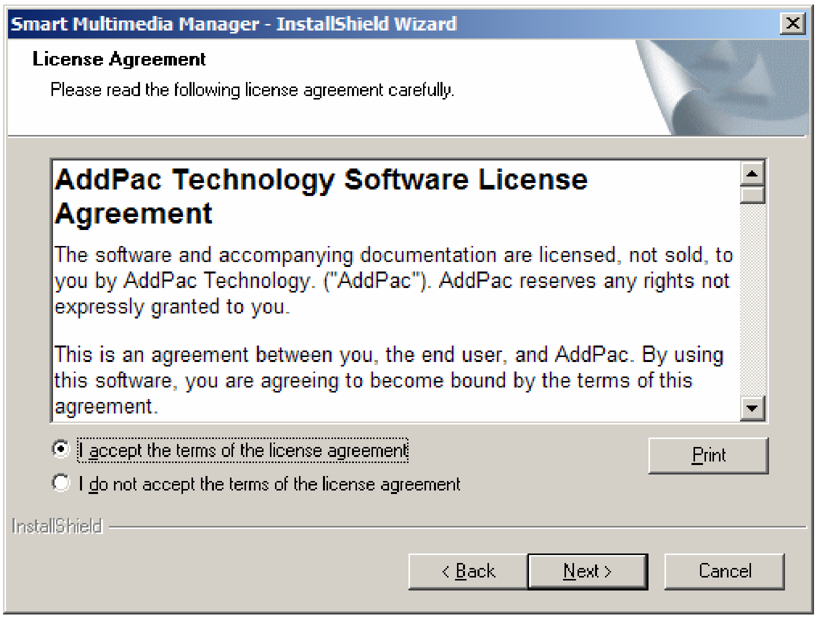

Figure 1-3 SMM Installation - Step 3

After reading the license agreement, click 'I accept the terms of this agreement' (A) then click (N) to continue installation. To stop installation, click 'Cancel'.

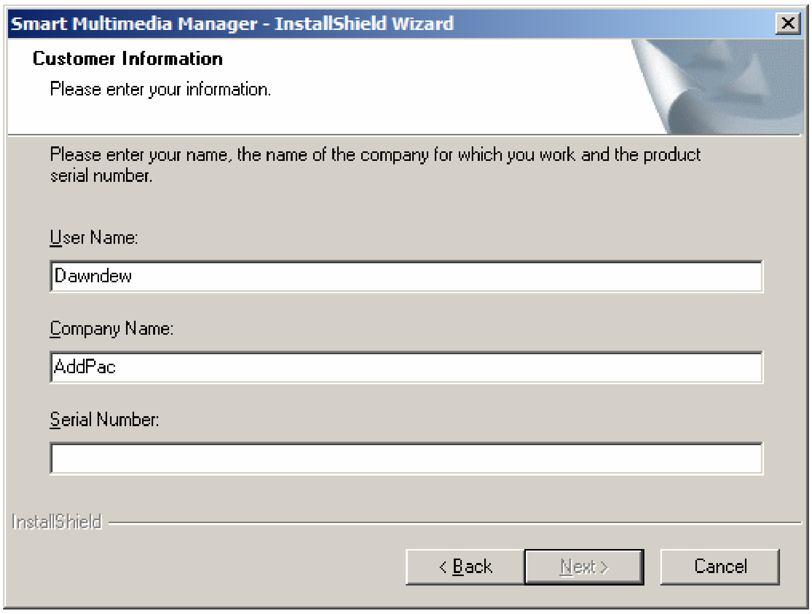

Figure 1-4 SMM Installation - Stage 4

After entering a user name, company name and serial number, click (N). To stop installation, click ‘Cancel’.

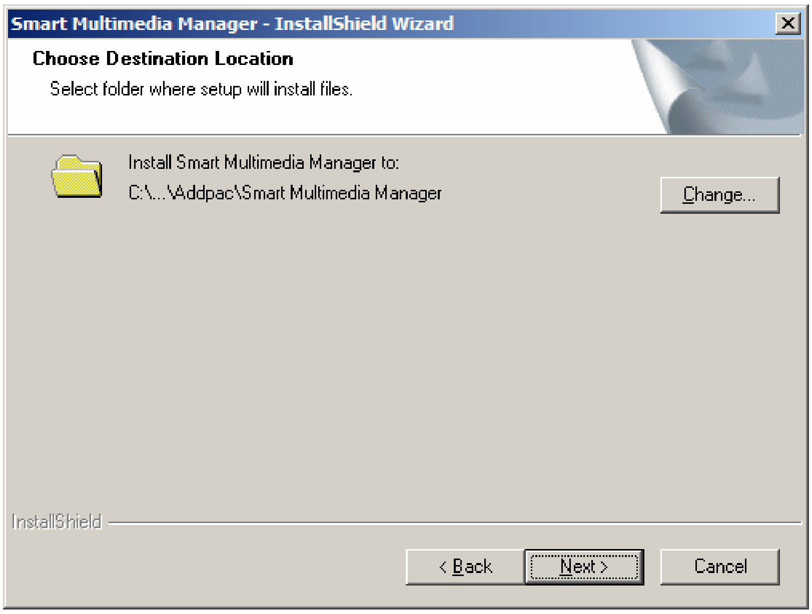

Figure 1-5 SMM Installation – Stage 5

To change the Directory of the location to where Smart Multimedia Manager to be installed, click ‘Change’, then enter the new Directory to be installed.

To install the Default Directory provided by SMM Install Program, click (N). Installation will processed to the folder of AddPac\Smart Multimedia Manager in the Directory of Program Files of the Drive installed with the basic OS.

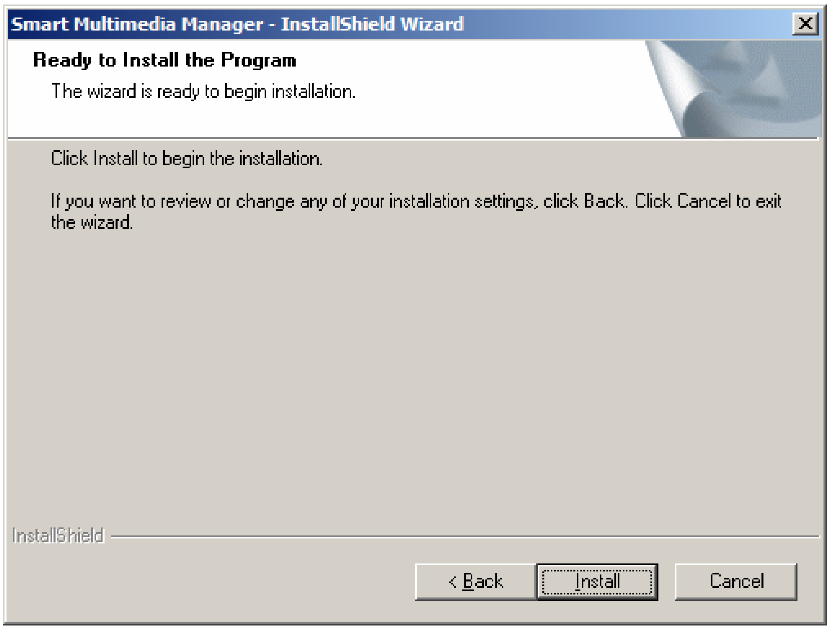

Figure 1-6 SMM Installation – Step 6

This is the screen with an option whether to start or stop the installation.
Click ‘Install’ (i) to process installation automatically or click ‘Cancel’ to stop the installation.

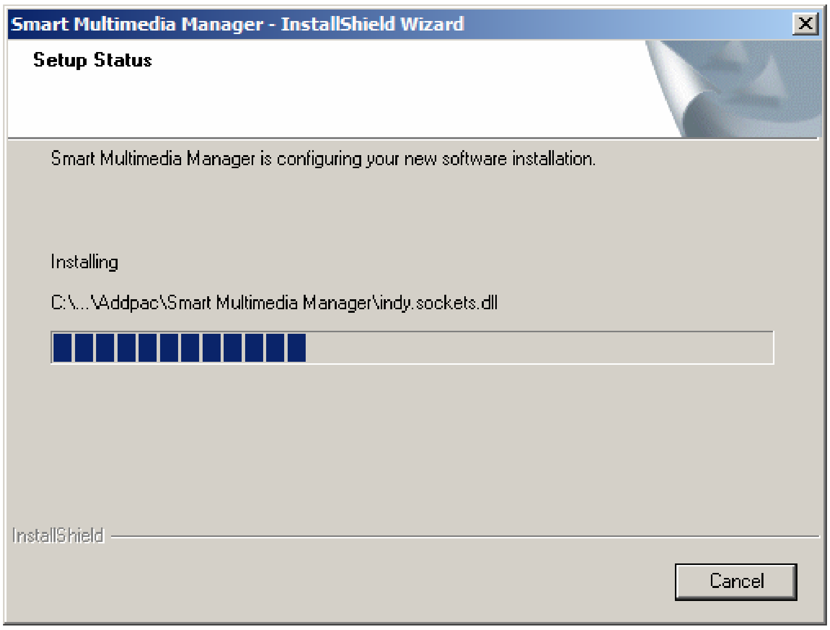

Figure 1-7 SMM Installation - Step 7

This is the screen to inform the installation process to the user. To stop the installation, click ‘Cancel’.

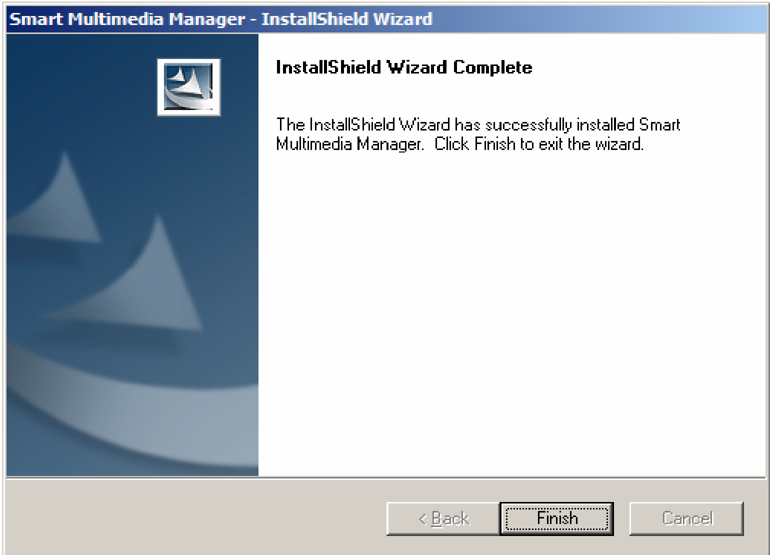

Figure 1-8 SMM Installation – Step 8

After completing the installation, the screen is shown as Figure 1-8. Click 'Finish' to finish the installation.

Running SMM

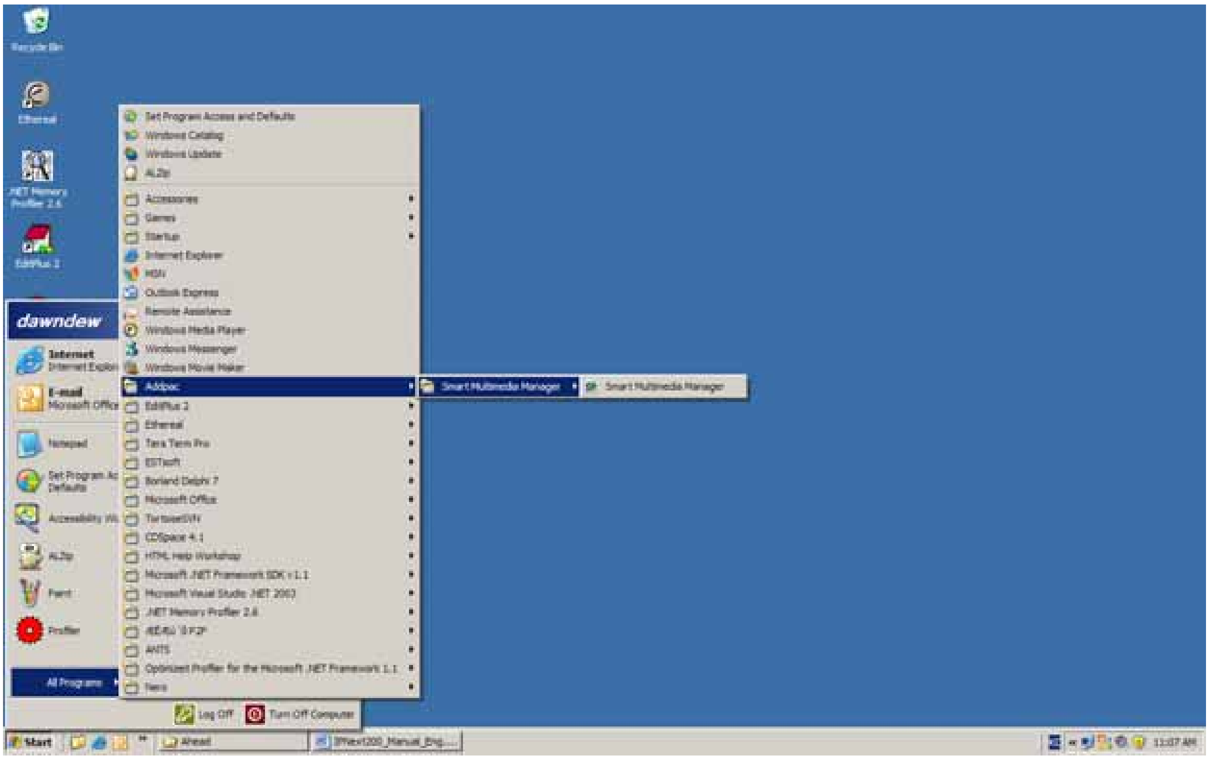

Figure 1-9 Process of Performing SMM Program

After the installation of Smart Multimedia Manager Program is completed, the user may find the installed program in 'Start' Screen of Windows. Then click the icon of Smart Multimedia Manager to start the program.

Figure 1-10 SMM Login Screen

After running the Smart Multimedia Management Program, the user may find the prompt screen to log-in in Figure 1-10. This is the process of user password authentication to access the program. You can run the program by entering the same password entered and saved in the beginning. After running the program, you can change the password by 'Tools > Change Password', from the main menu and you can also select an option to cancel authentication by 'Tools > Preferences > SMM Login > Authentication'. More details will be explained later this manual.

SMM v3.0 or later version allows the program to run by local authentication without accessing to Smart Directory Server.

Figure 1-11 shows registering the first password and completing authentication by reentering the password. The SMM program starts running.

Figure 1-11 Confirming SMM Login Password

Chapter 2. Operating SMM

Overview

As Smart Multimedia Manager (SMM) is the consolidated management tool for AddPac IP Multimedia Telephony Solution, SMM provides the features of IPNext PBX, Smart Directory Server, Multipoint Conference Unit (MCU), Presence Server, Auto Upgrade Server, IP Phone, Video Phone and others.

Layout Display of Window Screen

All the setups of Menu for SMM Program are categorized in the left side screen. When one of the categories of the Menu is selected, Window List lists the functions to be configured on the right side of the screen. You can perform registering, editing, deleting, renewingby pressing the right button of the mouse.

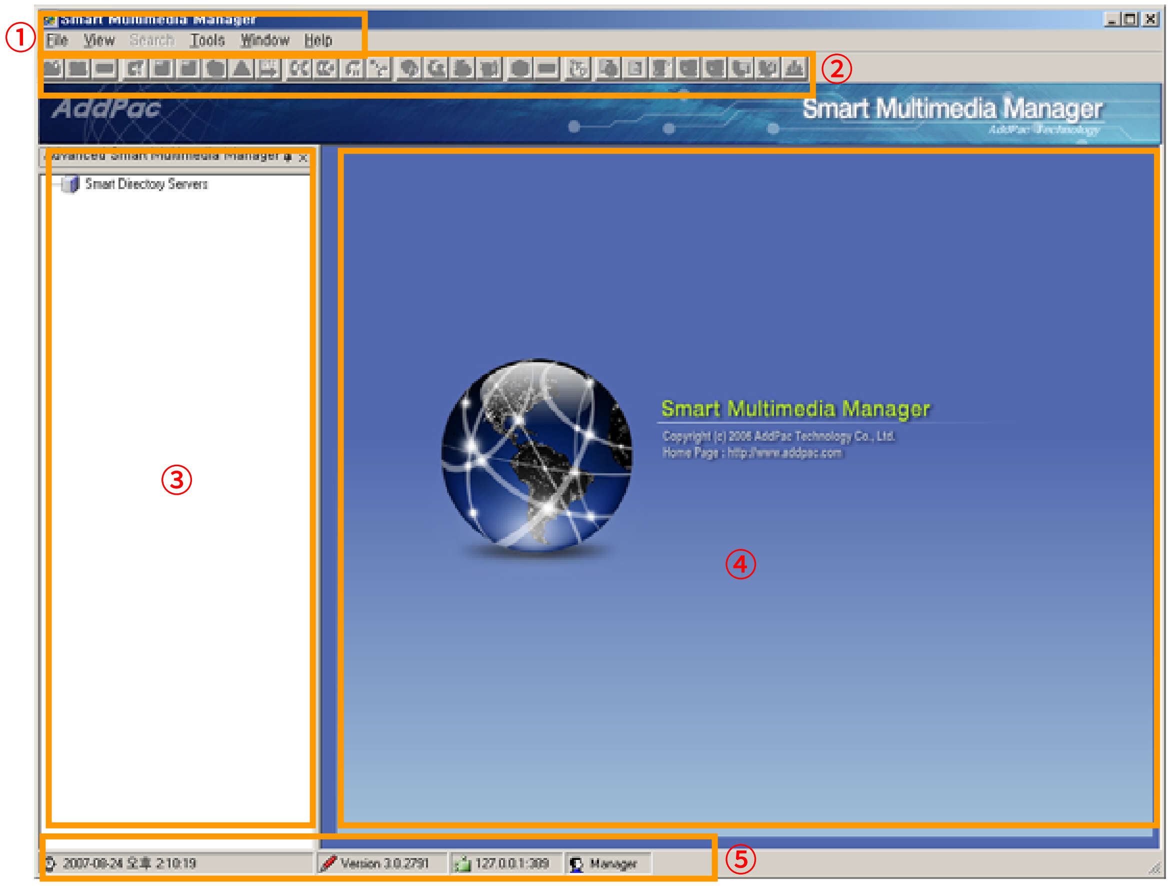

Figure 2-1 Layout Display of SMM

Table 2-1 Description for Layout Display of SMM

Components	Description
1	Management and Monitoring, Data Backup/Restore functions can be set
2	This is the short cut icon for the frequently used functions
3	This is the tree structure to set the related information of IP-PBX
4	The selected category of the tree Menu is to be listed and it can be added, deleted or monitored
5	The present time, IP address and connection status, Version, Login User of Smart Directory Server can be identified.

SMM Main Menu

The followings describe the features of SMM main menu :

Figure 2-2 SMM Main Menu

Table 2-2 SMM Main Menu Details

File	View	Search	Tools	Window	Help
Exit	SMM Mode	User	Backup/Restore (FTP)	Cascade	About
	Show SMM Menu	Terminal	Export/Import (ldif)	Tile Horizontally	
		Phone Number	Export/Import (xls)	Tile Vertically	
			Change Password		
			Preferences		

Table 2-3 Description of SMM Menu

Main Menue	Menu	Sub-Menu	Description
File	Exit		Ends SMM program
View	SMM Mode	Easy Mode	Changes the program menu mode to Easy Mode. In the easy mode, you can configure only the simple setting for a small number of IP-PBX.
		Advanced Mode	Change the program menu mode to Advanced Mode. In this advance mode, all the features of IP-PBX can be configured and administered by an administrator who has advanced level of experience.
	Show SMM Menu		Displays the left tree menu again, when the menu is closed
Search	User		Searches all the users who are registered to the selected Smart Directory Server
	Terminal		Searches all terminals who are registered to the selected Smart Directory Server

	Phone Number		Searches all the phone numbers which are registered to the selected Smart Directory Server.
Tools	Backup/Restore (FTP)	Backup	Downloads IP-PBX basic configuration file or database through FTP to PC
		Restore	Restores by uploading the data, which has been downloaded through FTP, to IP-PBX. Initializes (the fundamental files and data initialization) for the first installation
	Export/Import(ldif)	Export	Exports all the data registered to SMM in LDAP Data Interchange Format (LDIF)
		Import	Imports all the data saved in LDIF to Smart Directory Server (IP-PBX)
	Export/Import(xls)	Export	Exports all the data registered to Smart Directory Server (IP-PBX) in Microsoft Excel file (xls) format
		Import	Imports all the saved file, in Microsoft Excel (xls) file format, to Smart Directory Server (IP-PBX)
	Change Password		Changes SMM authentication password
	Preferences		Performs the environmental settings of SMM program
Window	Cascade		Tiles out the serve windows in cascaded style
	Tile Horizontally		Tiles out the serve windows in horizontal style
	Tile Vertically		Tiles out the serve windows in vertical style
Help	About		Displays the program information

SMM Administration Mode

Smart Multimedia Manager(SMM) provides 2 kinds of mode (Easy, Advanced). Easy Mode displays only the necessary configuration details, so the user can approach easily. Advanced Mode administers all the services supported by IPNext PBX. The following figure shows how to enter the SMM Mode (View > SMM Mode).

Figure 2-3 SMM – Easy Mode

Figure 2-4 SMM – Advanced Mode

Search

Search menu looks up all the registered users, terminals and telephone numbers.

User Search

This feature searches all the users for the selected Smart Directory Server. You can enter the conditions such as user name, ID, telephone number and others to look up the subscriber whom you want to find. If no search word is entered, it searches all the users.

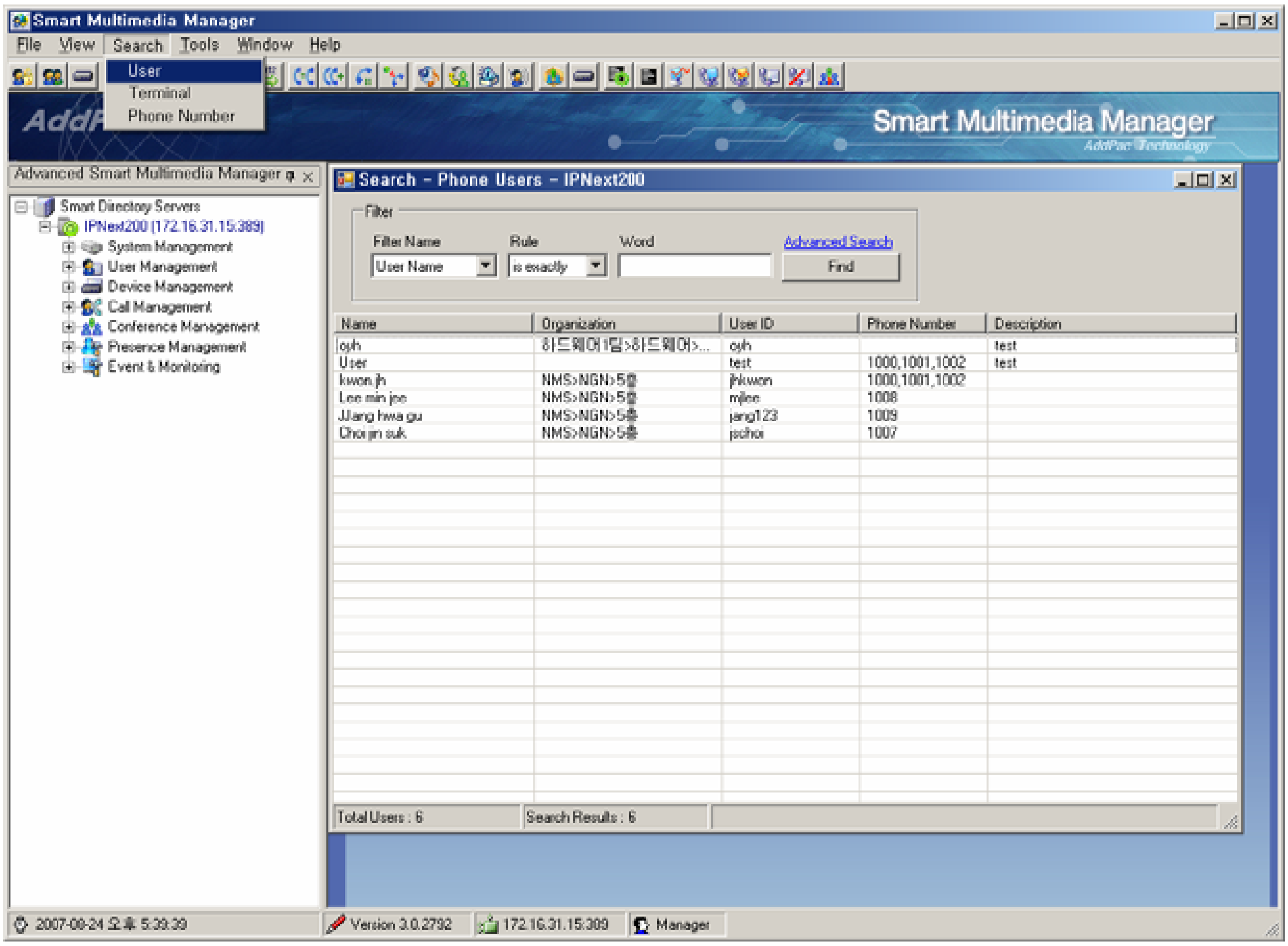

Figure 2-5 Search – User Search

Terminal Search

This feature searches all the terminals for the selected Smart Directory Server. Device name, model. IP address, version are the search conditions, so you can find the terminal that you want easily. If no search word is entered, it searches all the terminals.

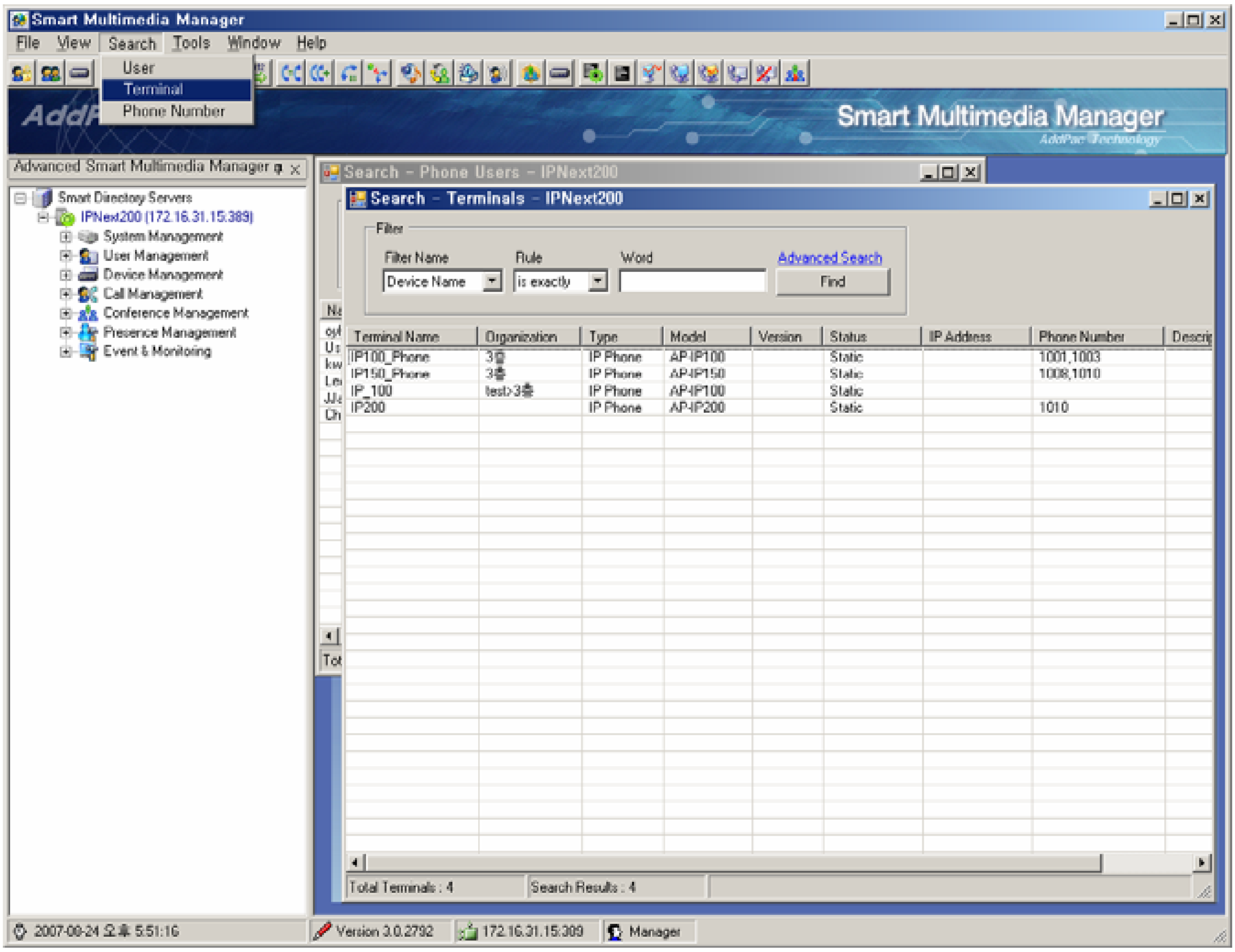

Figure 2-6 Search – Terminal Search

Phone Number Search

This feature searches all the phone numbers for the selected Smart Directory Server. You can find the terminal that you want easily with the search conditions such as Phone Number, Partition, Park Group, Pickup Group. If no search word is entered, it searches all the phone numbers.

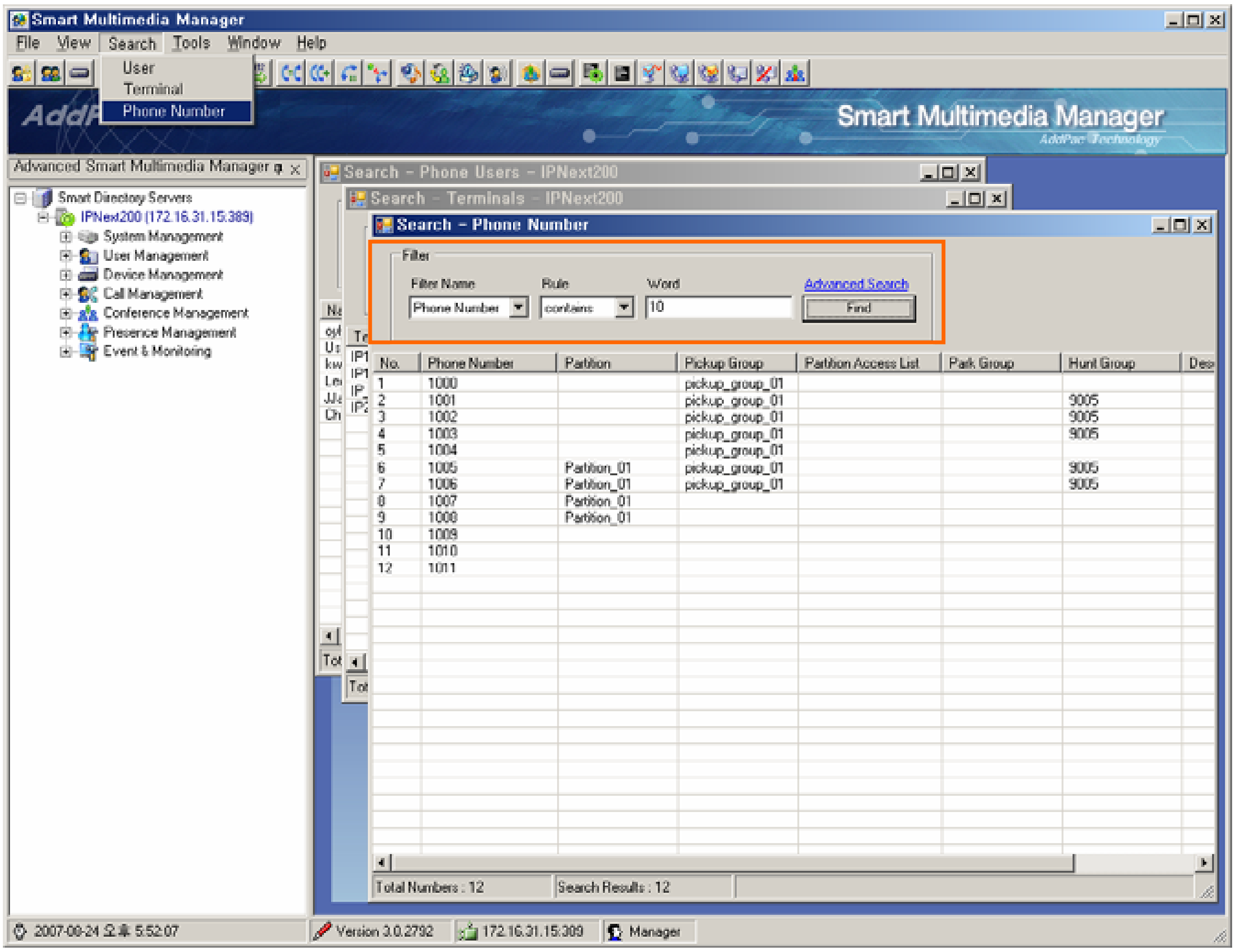

Figure 2-7 Search – Phone Number Search

SMM Preferences

SMM(Smart Multimedia Manager) allows the user to set Event, Call, Conference Monitoring and other additional setup options. Selecting the option is shown as Tools > Preferences Menu in the Figure 2-8.

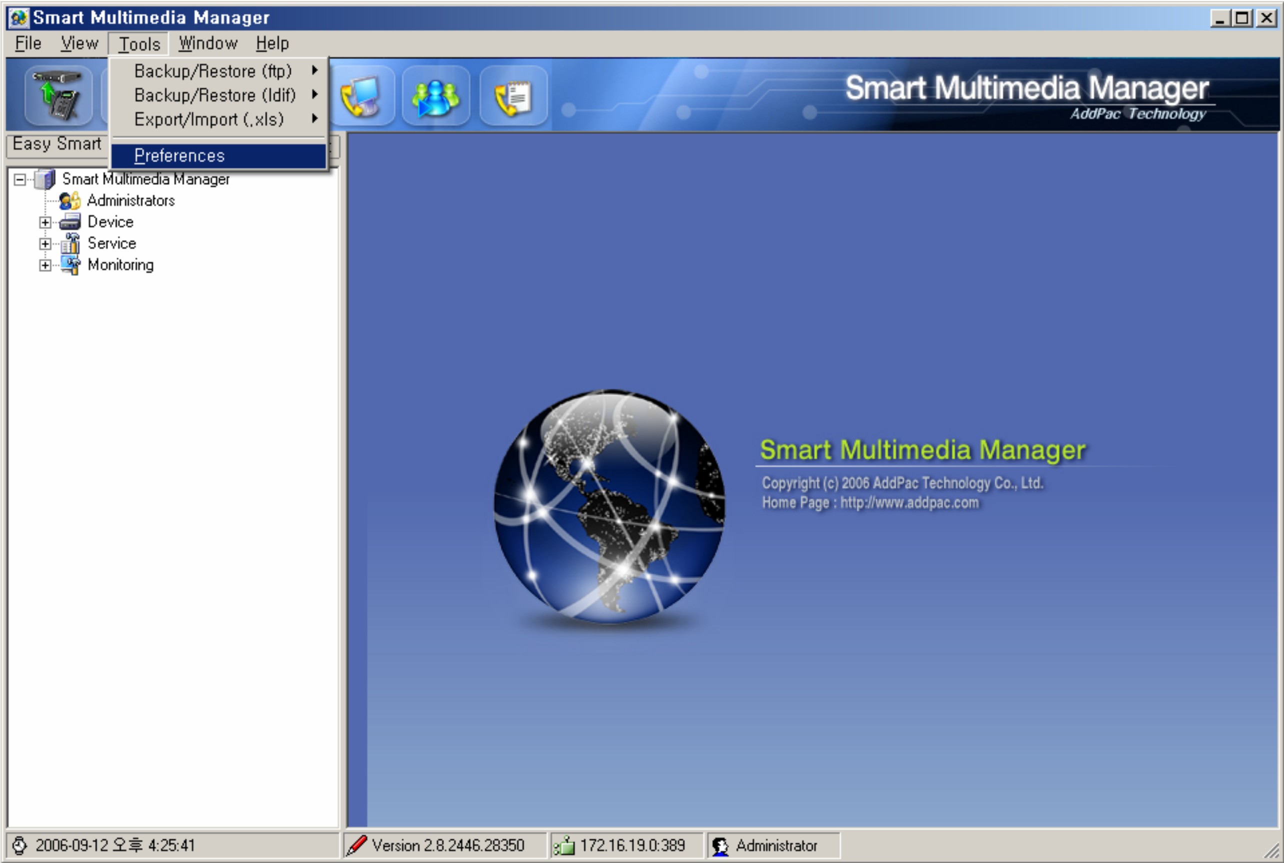

Figure 2-8 SMM Configuration Settings

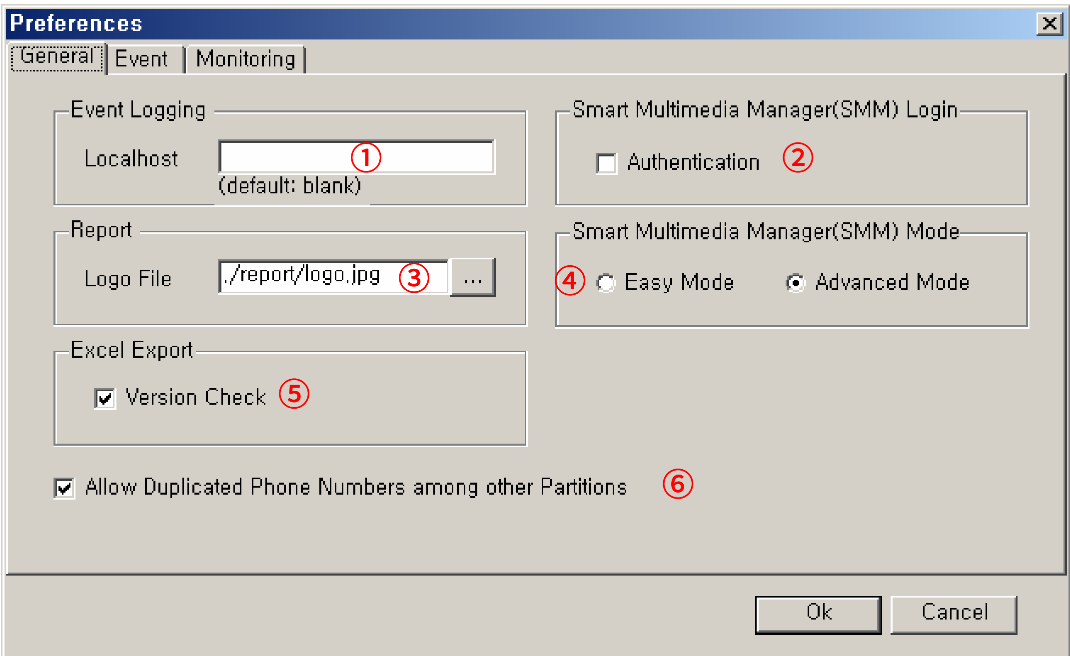

Figure 2-9 Preferences – General Registration

Table 2-4 Preferences– Description of General Registration

No.	Description

1	To set Event Logging of SMM to Call Manager, enter an IP address of the PC. When the configuration setting is in IPv6, register the IPv6 address to only Localhost of the PC (blank at default).
2	Choose the option for authentication of SMM login
3	Register an image file for company logo to be printed when to generate statistic data report
4	Set up the initial access mode of SMM. The basic setup is Advanced Mode.
5	This is an option to execute export when the current SMM data version matches with the data version of the equipment for Excel Export.
6	Select this option to register a phone number when the same number exists in other partitions

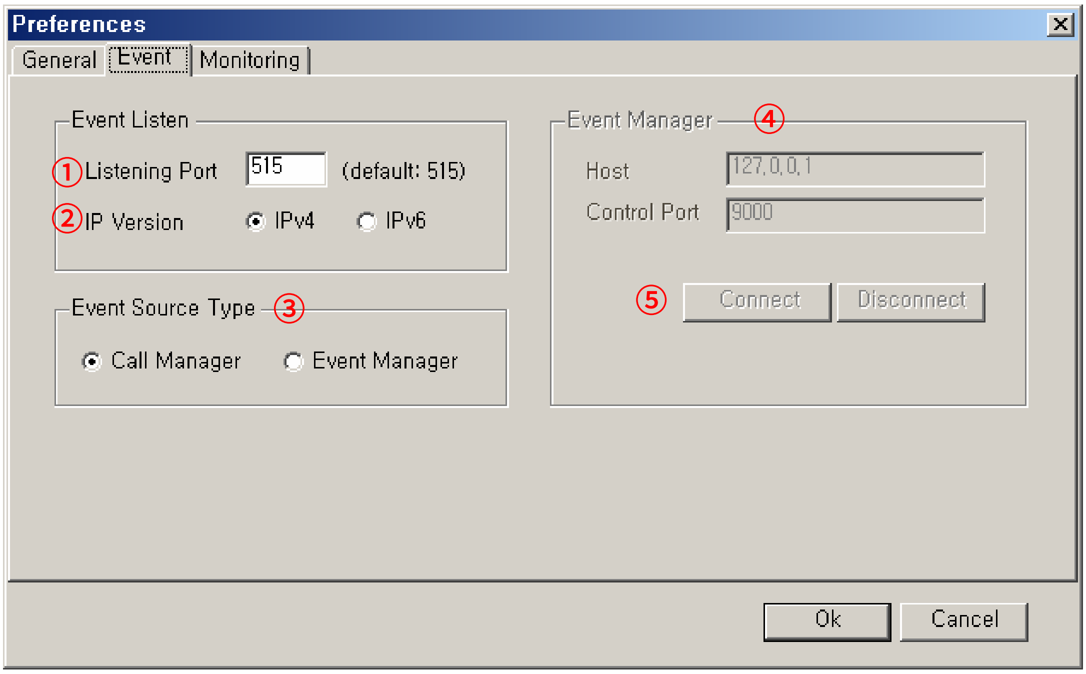

Figure 2-10 Preferences – Event Properties

Table 2-5 Description of Preferences – Event Properties

Section	Description
1	Setting the interval of the maximum response time when the information is requested from Smart Directory Server
2	Select this option to login automatically to Smart Directory Server when to operate SMM.
3	This is the space for the Localhost to be filled with and to register the IPv6 address of the PC only when the setting of IP address of the PC is IPv6 to set the Event Logging of SMM to the Call Manager (this space is left blank as Default)
4	To set the initial access mode of SMM. The basic setting is Advanced Mode.
5	Select this option to perform Export only when the data version of the equipment is same as the data version of SMM (when this is not selected, Export is to ignore the data version.

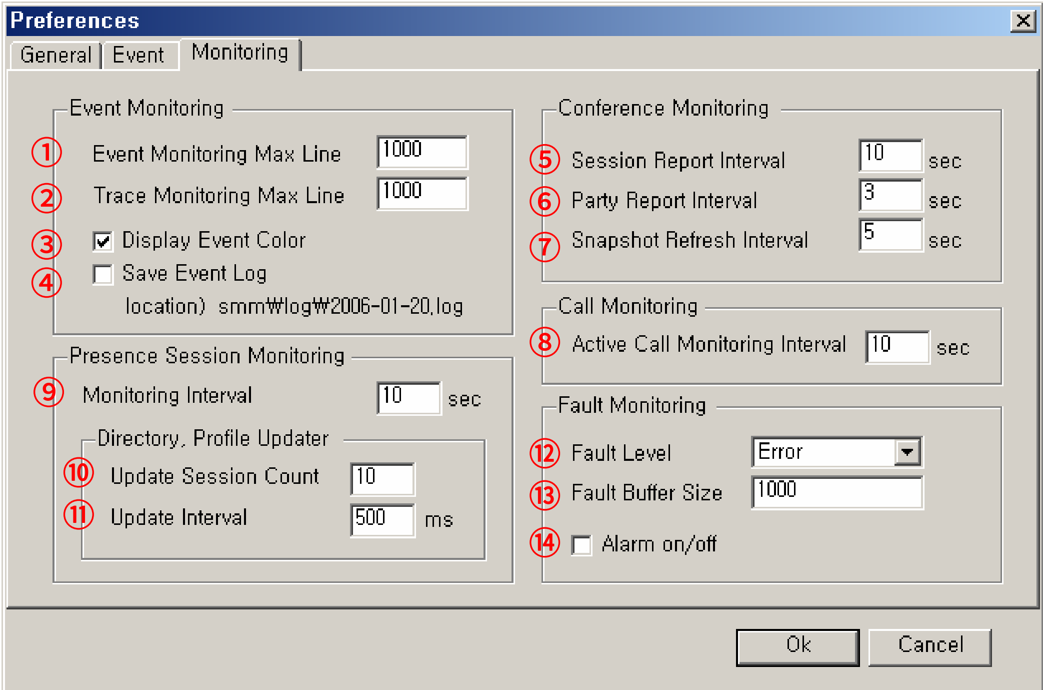

Figure 2-11 Preferences – Monitoring Properties

Table 2-6 Preferences – Monitoring Properties

No.	Description
1	Setting the maximum number of the lines for displaying Event information, in Event Monitoring, which is transmitted from SEM to Call Manager
2	Setting the maximum number of lines to be displayed for the information of Trace Monitoring which is transmitted from SEM or Call Manager
3	Select this option to organize the Event information to be displayed in Event Monitoring for organizing the information and to see the information of Event Monitoring in color
4	Select this option to organize the Even information to be displqayed Event Monitoring and to save the information of the table to the Directory in where SMM is installed
5	Setting the frequency interval to create the table for the Conference List to be processed
6	Setting the frequency interval to display the participant's information of the Conference List
7	Setting the frequency interval of refreshing the snapshot of the Conference
8	Setting the frequency interval to create the table for monitoring the calls being processed by IPNext PBX
9	Set a time interval to display the session registered to Presence Server for monitoring
10	Set the maximum number of session count to be updated with profile or directory
11	Set a time interval to update the session of the profile or directory
12	Set a level for Fault Event for Fault Monitoring
13	Set the maximum number of lines for Fault Monitoring

14	Set alarm when Fault Event is received

Access Management of Smart Directory Server

Smart Multimedia Manager (SMM) is capable of registering and managing many Smart Directory Servers, which are the basic access unit, for managing many IP-PBX's. SMM can set each Smart Directory Server to each IP-PBX and determines a status of Smart Directory Server.

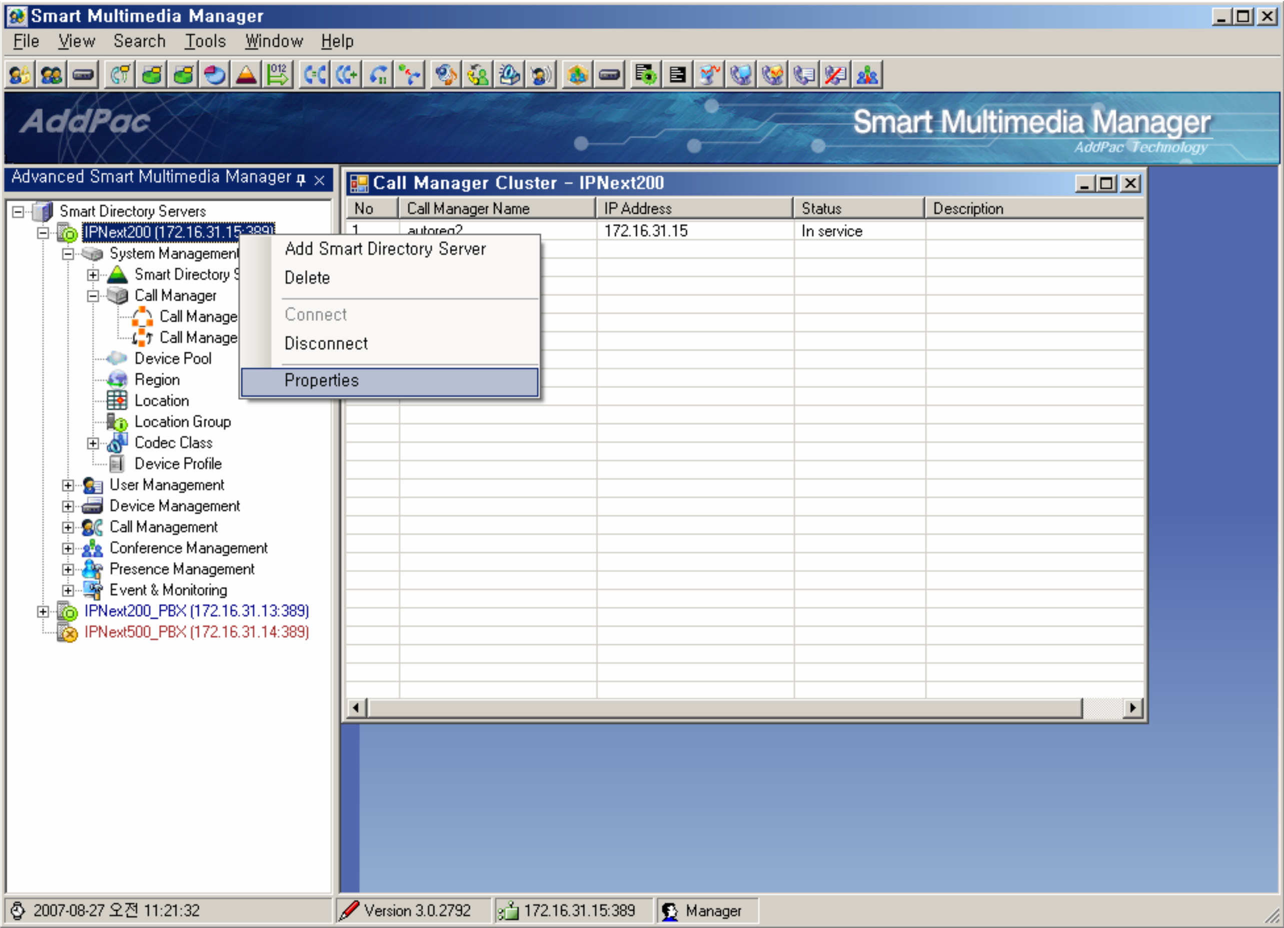

Figure 2-12 Smart Directory Server Menu

Smart Directory Server can register, delete and edit through pop-up menu. The server also can choose connect and disconnect of the menu to change the connection status of the server. The server can be accessed, in a same way, through automatic connection. When the server is connected, the icon indicates the status and automatically added to the setting menu. The above figure shows the registration status and menu of Smart Directory Server.

Smart Directory Server Properties

The below figure describes Smart Directory Server properties.

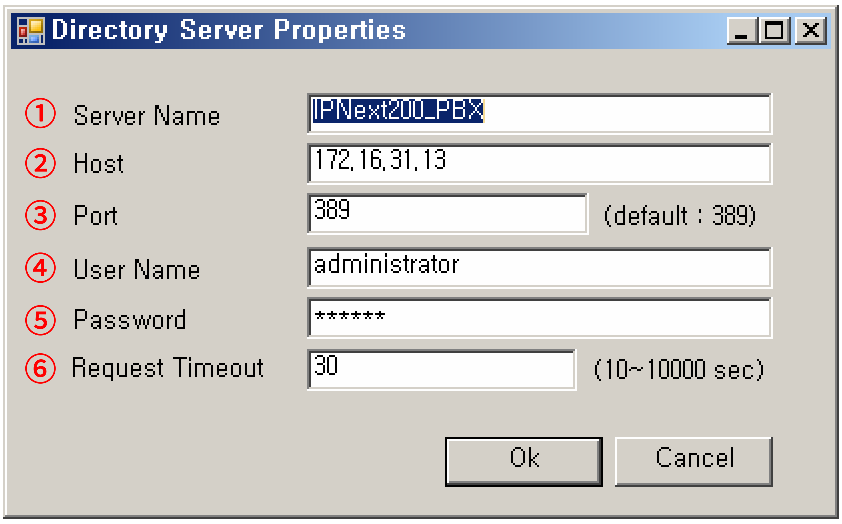

Figure 2-13 Smart Directory Server Properties

Table 2-7 Description of Smart Directory Server Properties

No.	Description
1	Enter a name of the server
2	Enter an IP address of Smart Directory Server (IPv4, IPv6 can be entered)
3	Enter a port number of Smart Directory Server (default: 389)
4	Enter a user ID (default: administrator)
5	Enter a password (default: router)
6	Set the timeout value in response to an access request to Smart Directory Server

Easy Configuration (Easy Mode)

The Easy Mode allows the configuration to be easy and simple for the IPNext PBX operation.
Therefore this section is limited to describe only the Trunk management function and Enduser Terminal which can be seen in the Easy Configuration. Of course, it is possible to register the Terminal and Trunk at the Advanced Mode, but the registration process is too complicated. The Easy Mode provides the general settings of Terminal, User, Phone information and others for registration. The rest of the functions can be described in Advanced Mode.

Configuring Enduser Terminal

The Enduser Terminal Menu is the function to configure the subscribers of the IPNext PBX allows to register only VoIP Gateway (1-port) and IP-Phone and just the minimum information (Terminal, User, Phone) is required to

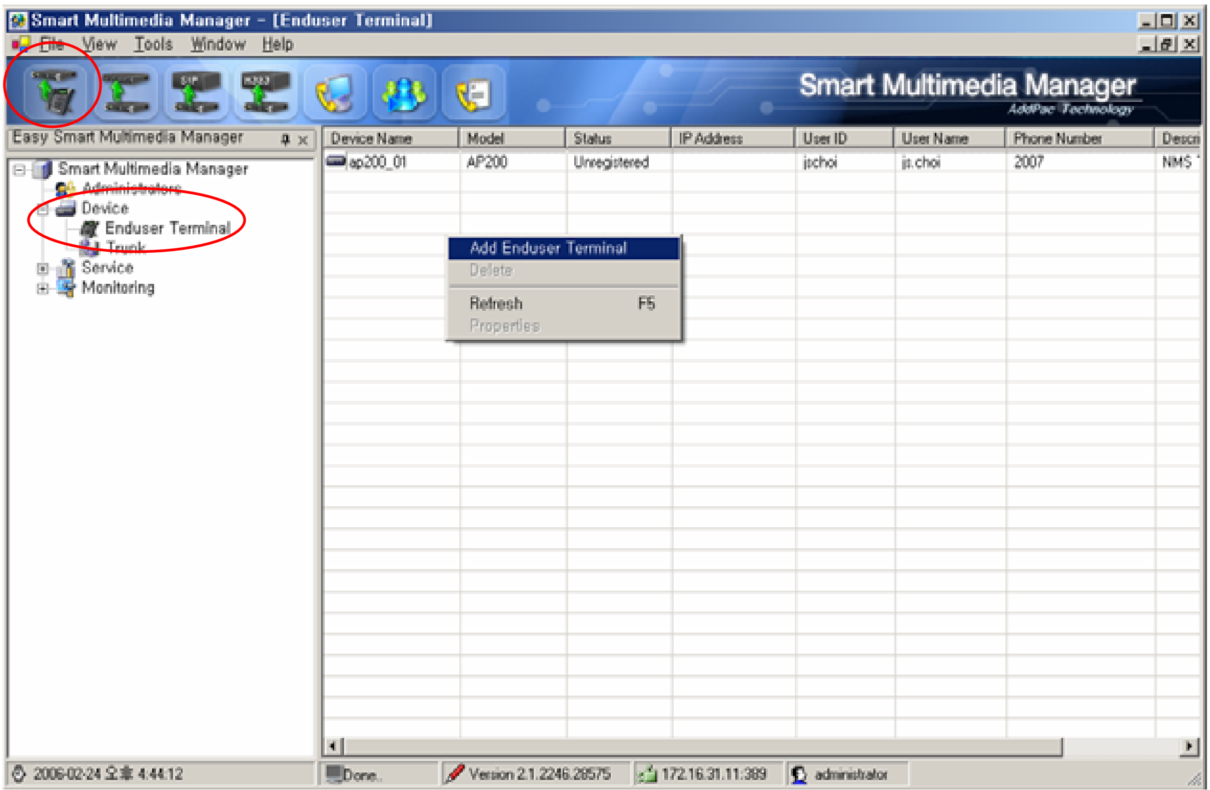

Figure 2-14 SMM Easy Mode – Enduser Terminal

The above Figure 2-7 is the Screen that performed ‘Device > Enduser Terminal Menu’ of Easy Mode. For the new Terminal registration, click one of the icons indicated on the top or right button of the mouse to perform the registration screen of Enduser Terminal.

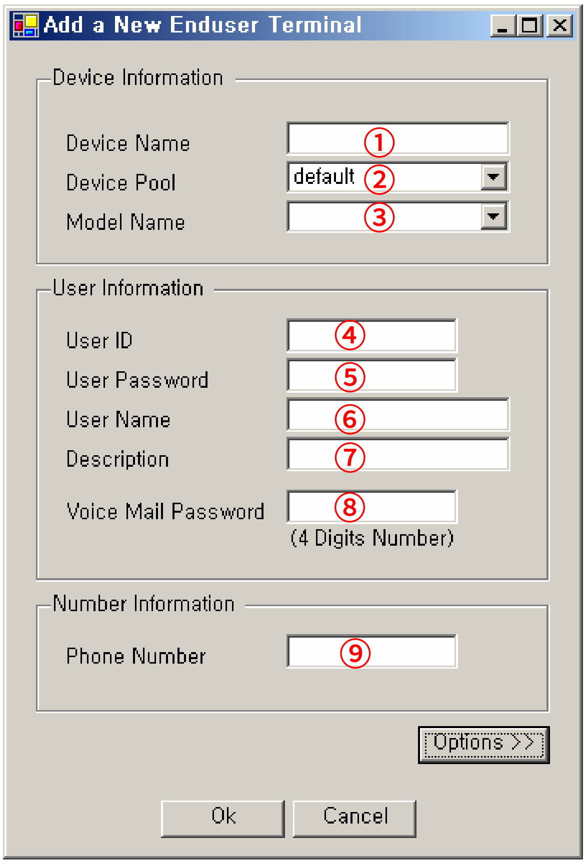

Figure 2-15 Enduser Terminal Properties

Table 2-8 Enduser Terminal Properties

Field	Description
1	Set the Device Name
2	Select one of the options for the Device Pool
3	Select the Model of the Device
4	Set the user ID (when the existing user is entered, the table for the user information is created automatically)
5	Set the use password
6	Set the use name
7	Set description of the use
8	Set the password for Voice Mail (4 numeric digits)
9	Set the telephone number

Click the Option button of Enduser Properties Screen to select the option for the terminal.

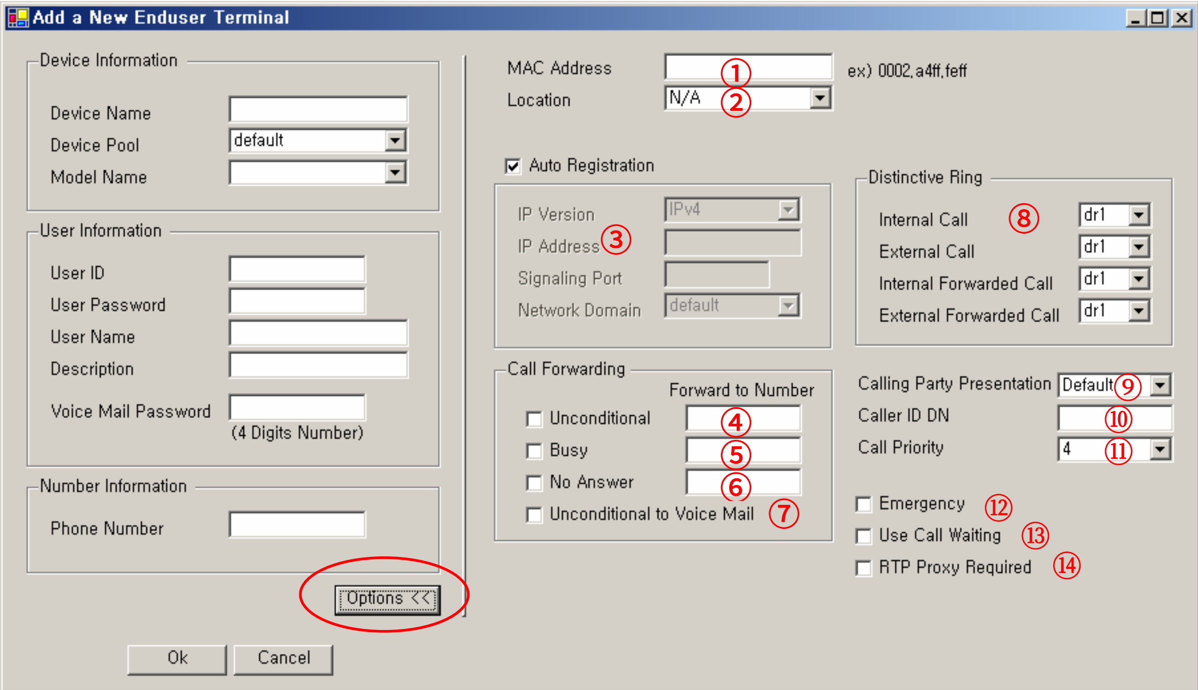

Figure 2-16 Enduser Terminal Option Properties

Table 2-9 Description of Enduser Terminal Option Properties

Section	Description
1	This option is to set the MAC address. When Auto Plug-in of IP Phone is performed, the MAC address is registered automatically. However, when an operator tries to register the information of the IP Phone first, before the terminal performs Auto plug-in, the MAC address must be entered in order for IP Phone to update the terminal information automatically.
2	This option is to set the Location for the Terminal.
3	This option is to set the automatic registration for IP address, port, domain information of the terminal. To register manually, set IP address, port and domain information directly. When Auto Registration is set and Auto Plug-in is performed, IP address, port, domain information is registered automatically
4	When the incoming call is received, the call is forwarded to the fixed number.
5	When the incoming call is received, the call is forwarded to the fixed number only when the line is busy.
6	When the incoming call is received, the call is forwarded to the fixed number only when the attendant is vacant.
7	This is the option to set the function of Voice Mail.
8	This is the option to set the different ring tone for each different kind of call. - Internal Call: To set a ring tone for the incoming from the inside. - External Call : To set a ring tone for the incoming call from the outside - Internal Forwarded Call : To set a ring tone for the incoming call from inside which is to be forwarded - External Forwarded Call : To set a ring tone for the incoming call received from the outside which is

	to be forwarded
9	Select the option for displaying the caller's phone number for the inbound call - Default: This is the basic setting which is set to Call Manager - Allowed: Displaying my phone number to the other party - Restricted: Not displaying my phone number to the other party
10	Displaying the caller's number all the time
11	Placing the priority level to the call as to apply different limitations on the functions
12	To set the highest priority on the call which has been set to Emergency
13	Call Waiting is a function to send a receiver a signal when another call arrives while the receiver is already on the phone. The receiver can place Hold on the call which is already on the line and gets through the new call by pressing Hook Flash
14	This is the option to get the call signal though RTP Proxy Server when the terminal in the inside of the private network is used to make a call to the outside of the network.

Configuring Trunk

TrunkMenu is to manage registration of Trunk Gateway, H.323 Gatekeeper, SIP Proxy Server as a function to configure Trunk of IPNext PBX.

Trunk Gateway is the equipment that takes the call when the inside caller sends the signal to the outside.

IPNext PBX is used as internal Call Manager. The outgoing signals (VoIP, PSTN) from other equipment are to be registered to Trunk Gateway first, then they can be sent to the outside. IPNext PBX provides 2 voice module slots, which allows the direct connection to VoIP or PSTN is possible externally without using other external terminal devices. At this time, it is absolutely necessary to configure the Trunk Gateway. The configuration at the Advanced mode is described in details on p77.

H.323 Gatekeeper is to register the Gatekeeper to interoperate the Gatekeeper with IPNext PBX. For instance, if a caller wants to make a call from the inside of the company to the outside, the company phone numbers can be registered to the outside Gatekeeper.
H.323 Gatekeeper is to configure the settings, so the call can get though the registered Gatekeeper. The Configuration for Gatekeeper Registration at the Advanced mode is described in details on p81.

SIP Proxy Server is a function to register SIP Proxy Server to IPNext PBX to interoperate with the outside SIP Proxy Server, in the same way as H.323 Gatekeeper. All the details of registering SIP Proxy Server is described in the section of SIP Proxy Server configuration

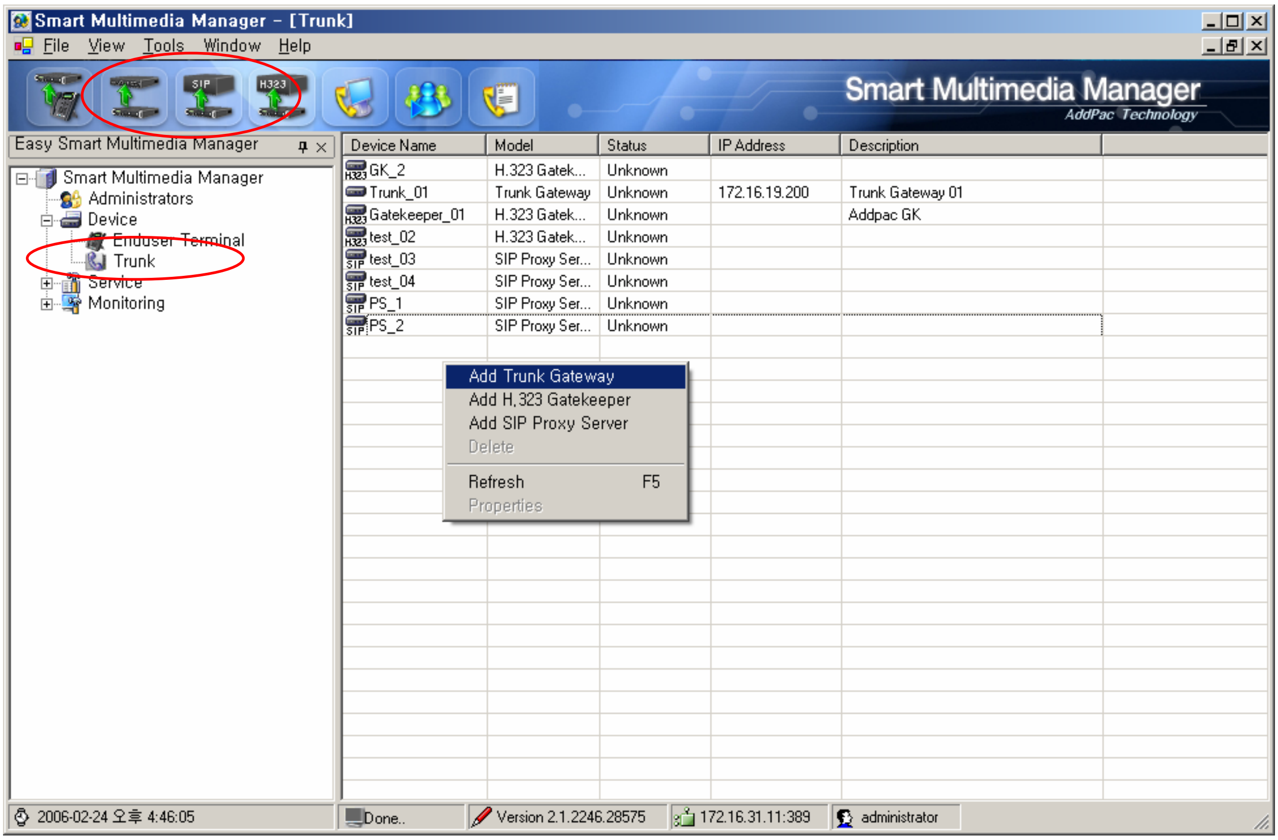

Figure 2-17 Easy Mode - Trunk

The above Figure 2-10 is the Screen that performed Device>TrunkMenu of Easy Mode. This Screen allows the user to register and manage the Trunk Gateway, H.323 Gatekeeper, SIP Proxy Server.

System Management

System Management includes the overall settings and management functions of Smart Directory Server, Call Manager, Device pool which are related to IPNext PBX.

Smart Directory Server – Configuring Smart Directory Cluster

Smart Directory Cluster is the listings of Smart Directory Server configured in redundancy, which is used from the Call Manager. Smart Directory Server is imbedded in IPNext PBX. So if the user wants to configure IPNext PBX in redundancy, Smart Directory Server registers Smart Directory Cluster and shares the database of IPNext PBX. In this way, IPNext PBX can be operated in redundancy.

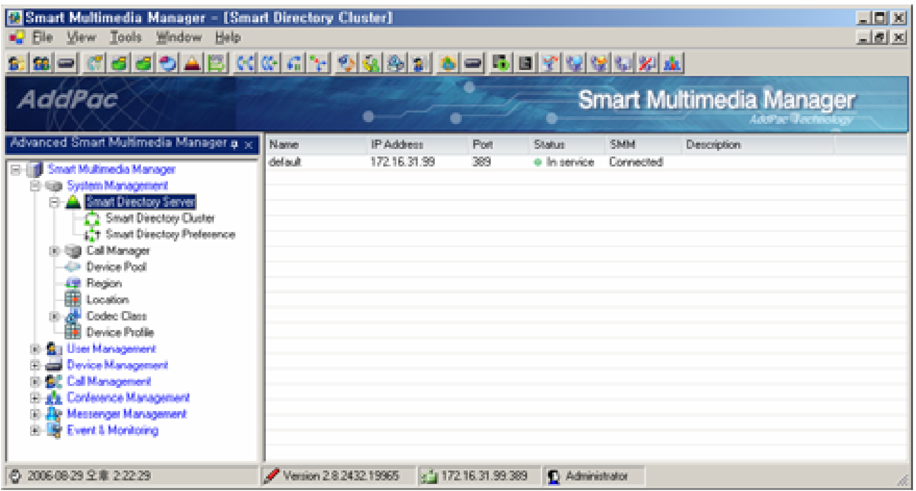

Figure 2-18 Smart Directory Cluster Screen 1

The below Figure 2-12 exhibits the screen of Smart Directory Cluster. In this Cluster list, the name, IP address, port, status, connectivity to SMM are represented.

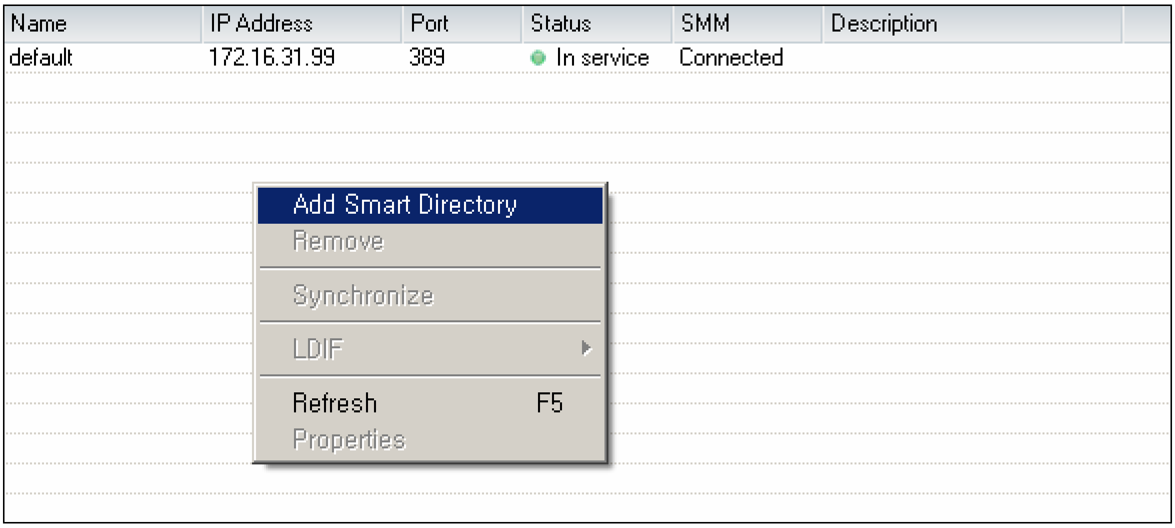

Figure 2-19 Smart Directory Cluster Screen 2

The following are the description of Smart Directory Cluster Menu:

Table 2-10 Smart Directory Cluster Menu Description

Menu 이름	Description
Add Smart Directory	Register a new Smart Directory Server (configured in redundancy)
Remove	Delete the existing Smart Directory Server
Synchronize	Synchronize the selected data of Smart Directory Server with the other Directory Server
LDIF Import	Import the saved ldif file to Smart Directory Server
LDIF Export	Export the data of Smart Directory Server as ldif file.
Refresh	Renew the information of Smart Directory Server.
Properties	Perform the selected Properties of Smart Directory Server.

The below Figure 2-13 exhibits the screen for registration of a new Smart Directory Server.

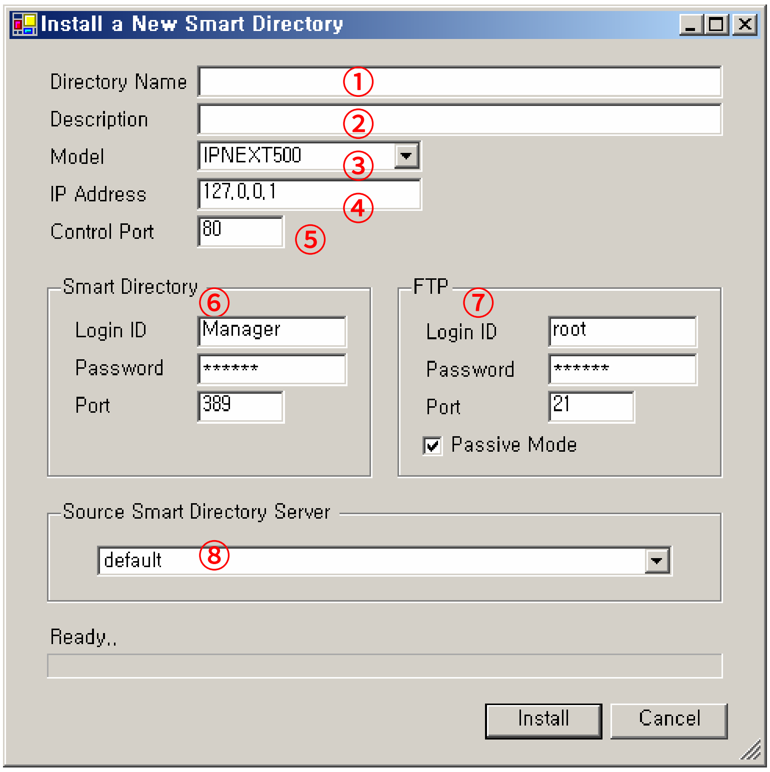

Figure 2-20 Registration Screen for Smart Directory Server

The followings are the description of Smart Directory Cluster Menu.

Table 2-11 Description for Properties of Smart Directory Cluster

Field	Description
1	Enter a name of the new Smart Directory Server
2	Enter a Description
3	Select a product model of the new Smart Directory Server (IPNext PBX)
4	Enter an IP address of the new Smart Directory Server
5	Enter Control Port (HTTP Port) number of the new Smart Directory Server
6	Enter an access information (ID, password, port number) of the new Smart Directory Server
7	Enter FTP access information (ID, password, port number, passive mode) of Smart Directory Server.
8	Select Smart Directory Servers which can be a source for copying the data

Smart Directory Server – Configuring Smart Directory Preference

Smart Directory Preference is a function to register and mange Smart Directory Server which is included in Directory Cluster basing on the order of priority. After registering Smart Directory Preference, the user can apply the configuration of the Preference to a Device Pool in a Device Configuration which will be described in the next section.

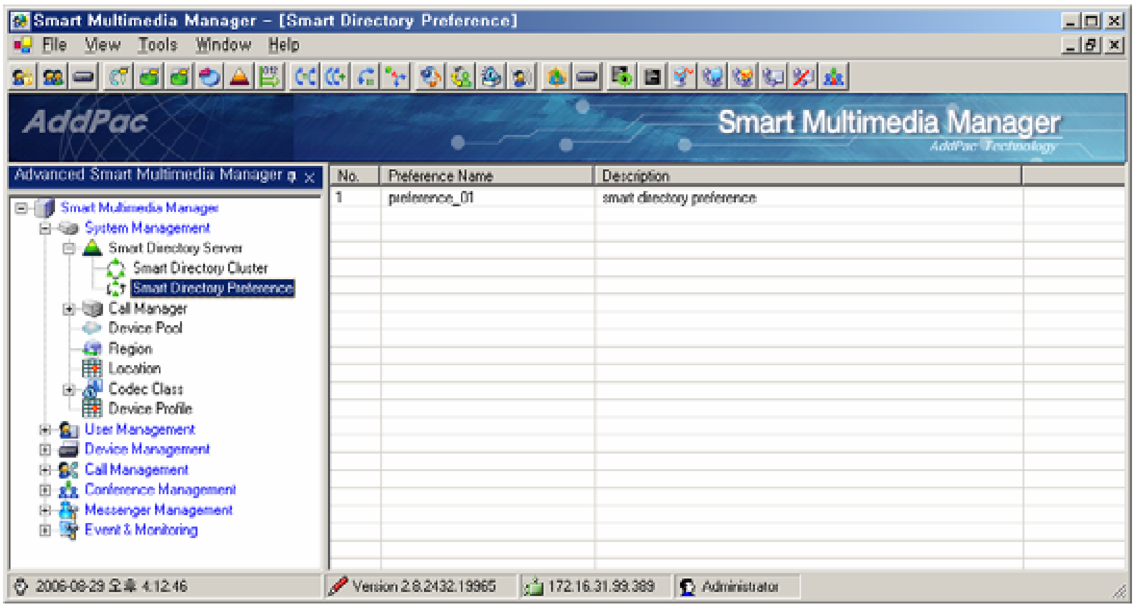

Figure 2-21 Smart Directory Preference Screen

The following Figure exhibits the Registration Screen for Smart Directory Preference Data:

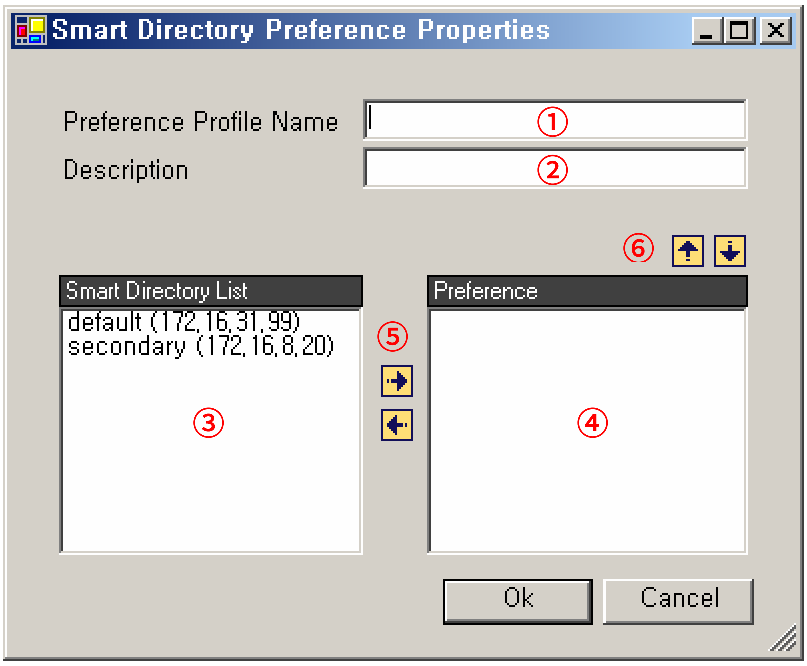

Figure 2-22 Smart Directory Preference Properties Screen

Table 2-12 Smart Directory Preference Properties Description

Field	Description
1	Enter a name of Smart Directory Preference
2	Enter a description.
3	This is the list of Smart Directory Servers which are not registered to Preference.
4	This is the list of Smart Directory Servers which are registered to Preference.
5	Delete registration from the list of Preference
6	Change the priority of Smart Directory Server from the list of Preference.

Call Manager – Configuring Call Manager Cluster

Call Manager Cluster produces the redundant list of Call Manager. When Smart Directory Server is registered, the new Call Manager is registered automatically to Call Manager Cluster through the procedure. The user can delete or edit the information of Call Manager Cluster from SMM. The Figure below represents the name, IP address and status of Call Manager.

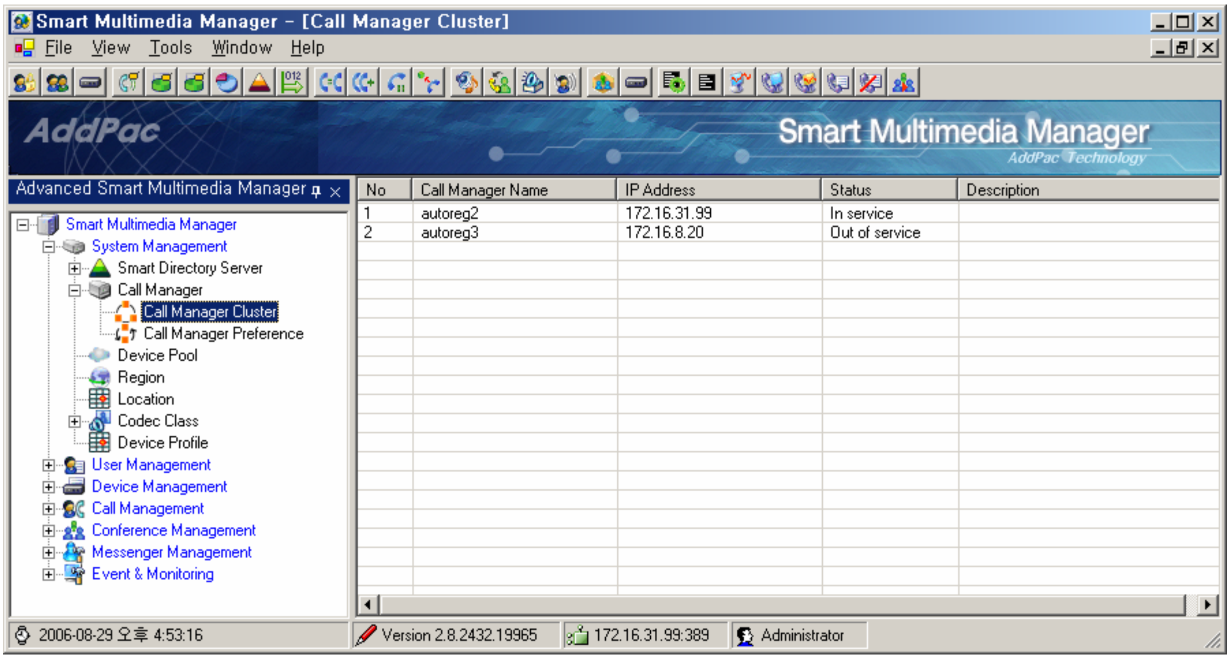

Figure 2-23 Call Manager Cluster Screen

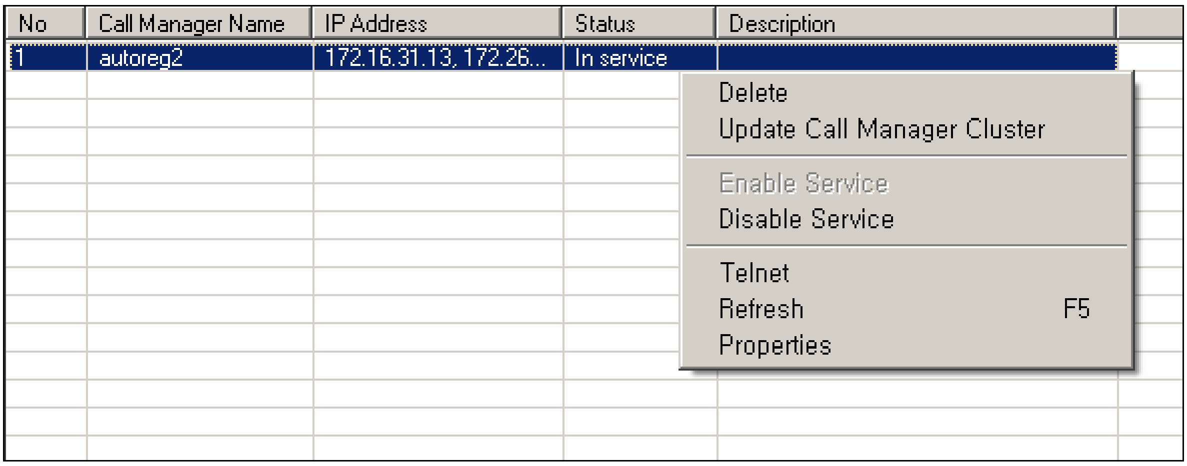

Figure 2-24 Call Manager Cluster Menu Screen

The followings are the description for the menu options of Call Manager Cluster.

Table 2-13 Description for the Menu Options of Call Manager Cluster

Menu Options	Description
Delete	Delete Call Manager from the Cluster
Update Call Manager Cluster	Update the information of Call Manager Cluster
Enable Service	Enable the service of the Call Manager
Disable Service	Disable the service of the Call Manager
Telnet	Access remotely to the selected Call Manager
Refresh	Renew the information of Call Manager Cluster
Properties	Perform Properties of the selected Call Manager

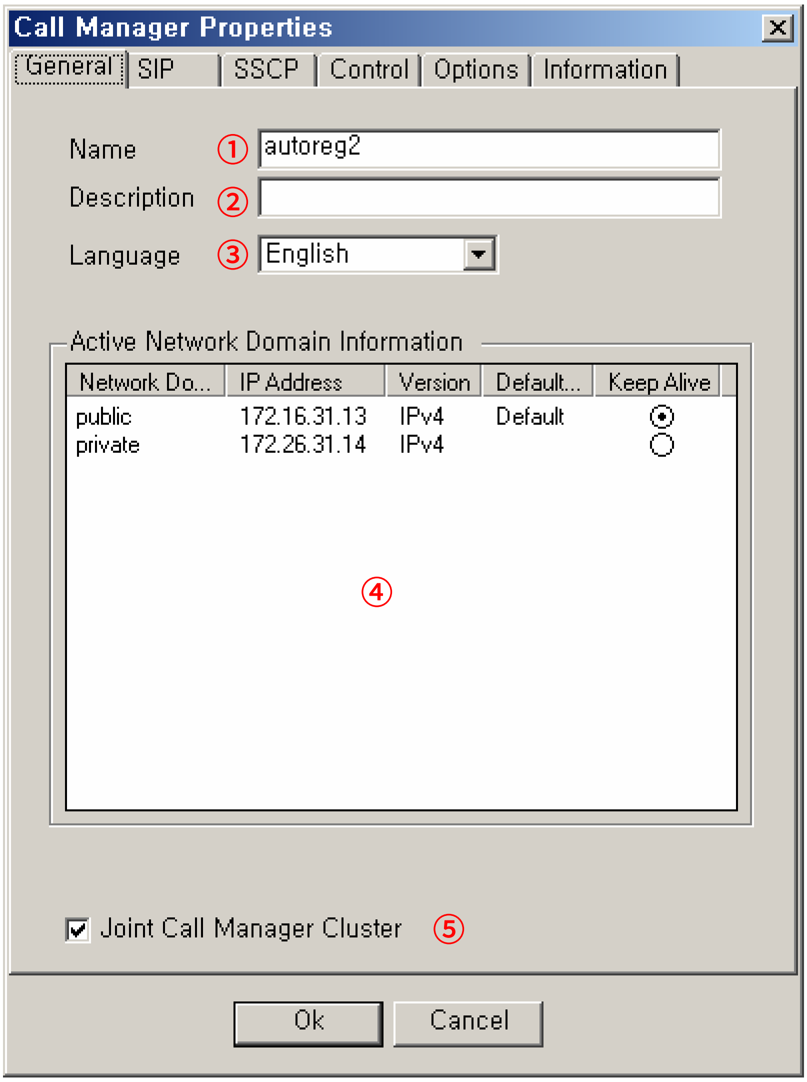

Figure 2-25Call Manager – General Properties

Table 2-14 Call Manager – Description for General Properties

Field	Description
1	Register a name of Call Manager
2	Enter a description of Call Manager
3	Select the language of Call Manager (if any change has been made, the system must be restarted)
4	Show the domain information of the present IPNext PBX which has been cofigured - Network domain: domain name - IP Address : IP address for each domain - Version: IP version - Select the Default domain for SMM to control - Select the Default domain to checking connections continually between the Call Managers
5	Select an option whether to include in Call Manager Cluster

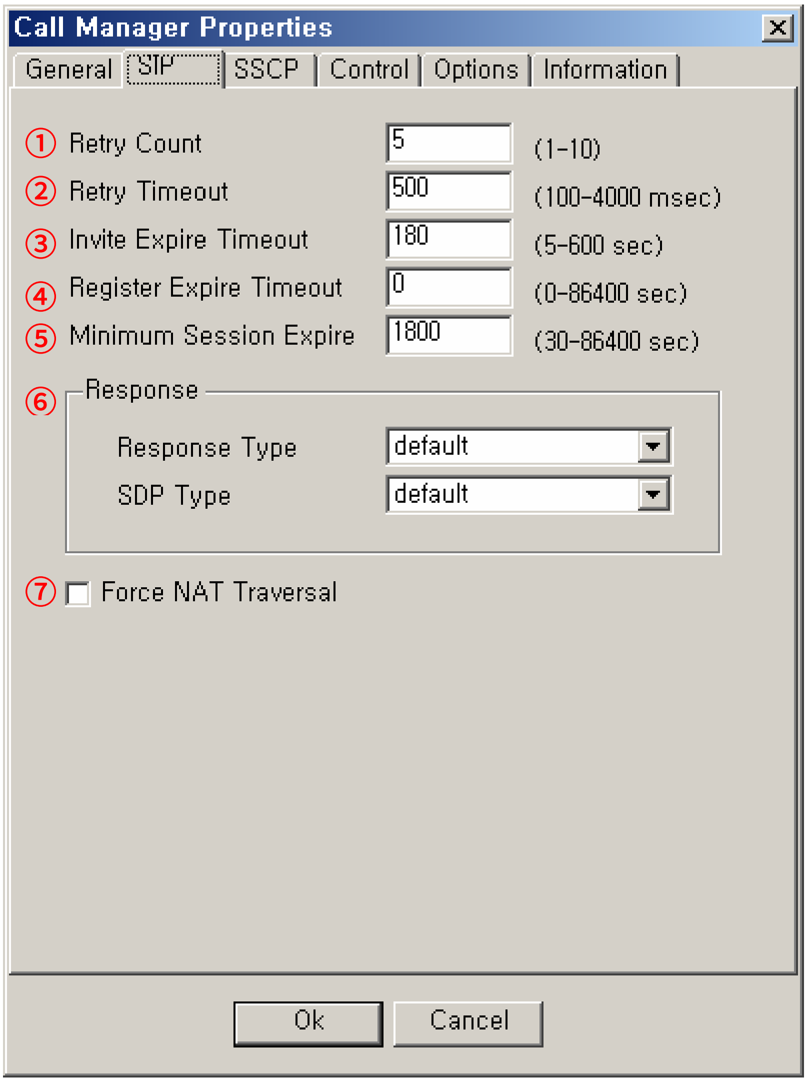

Figure 2-26 Call Manager – SIP Properties

Table 2-15 Call Manager – SIP Properties Description

Field	Description
1	Set the number of frequency for retransmitting SIP messages
2	Set the expiration time for retransmitting SIP messages
3	Set the expiration time for SIP Invite
4	Set the expiration time for registering SIP terminals
5	Set the minimum expiration time for a session
6	Response Type : Determine the type of response to Invite (180 or 183) Determine whether to include SDP
7	Determine whether to support the terminals under NAT settings [1]

[1] RFC 3581 Supporting NAT Transversal basing on “An Extension to the SIP for Symmetric Response Routing”

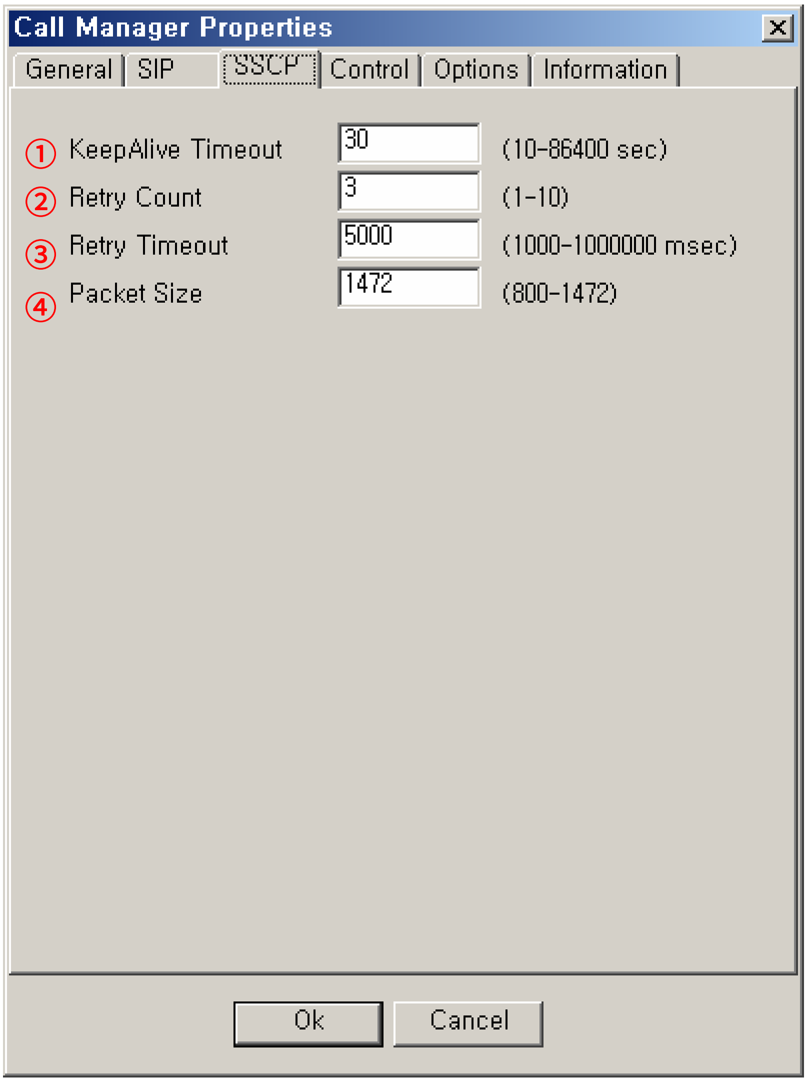

Figure 2-27 Call Manager – SSCP Properties

Table 2-16 Call Manager – Description of SSCP Properties

Field	Description
1	Set the number of frequency for exchanging KeepAlive messages between SSCP terminals
2	Set the number of frequency for retransmitting SSCP message
3	Set the number of frequency for retransmitting SSCP message
4	Determine the size of SSCP packet

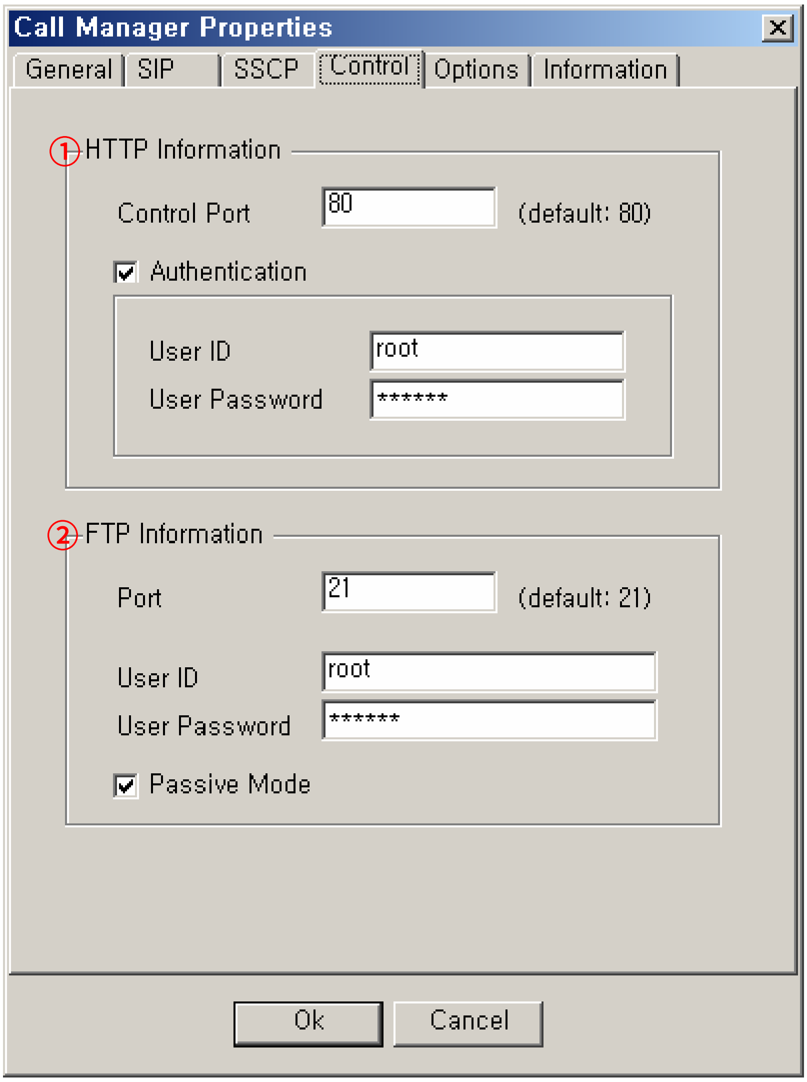

Figure 2-28 Call Manager – Control Properties

Table 2-17 Call Manager – Description of Control Properties

Field	Description
1	Configure the access information of HTTP (port, IP, password) to communicate with Call Manager, from SMM
2	Configure the access information of FTP to communicate with Call Manager, from SMM

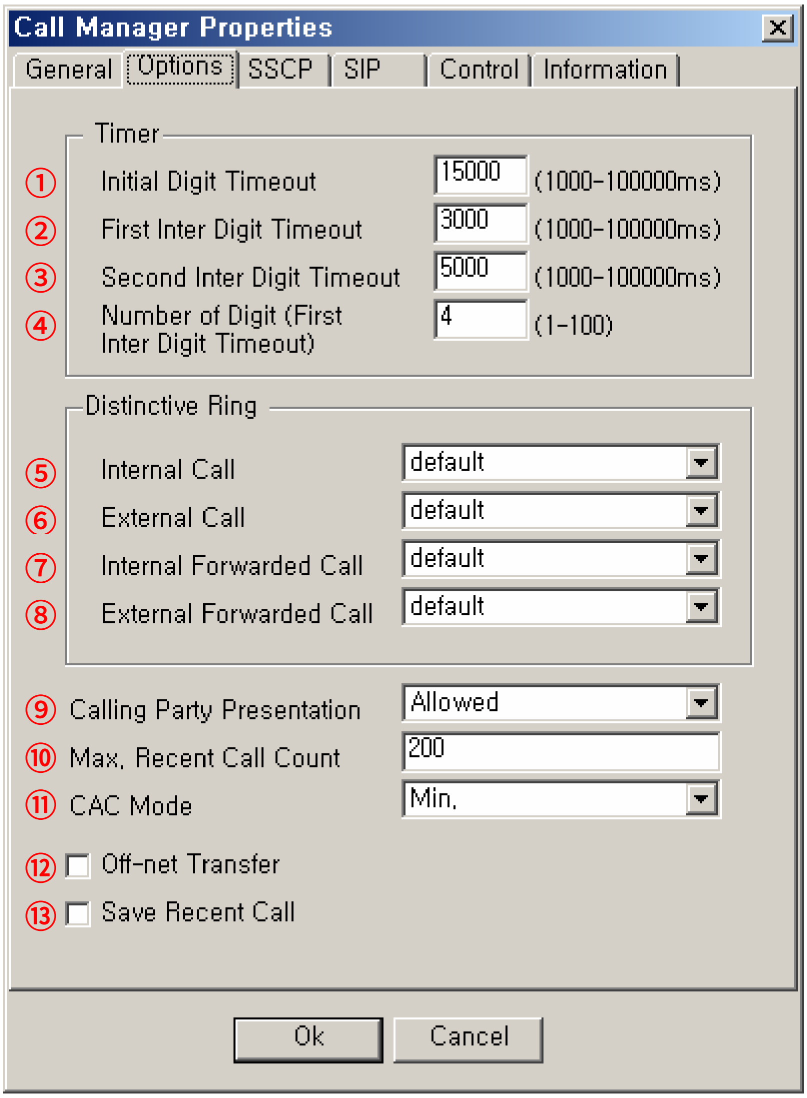

Figure 2-29 Call Manager – Options Properties

Table 2-18 Call Manager –Description of Options Properties

Field	Description
1	Set a timeout to determine DTMF entry
2	Set a timeout for entry of the initial number
3	Set a timeout to determine no entry for the second DTMF tone entry
4	Set a number of the first digits
5	Set the ringer sound for the incoming internal call
6	Set the ringer sound for the incoming external call
7	Set the ringer sound for the forwarded internal call
8	Set the ringer sound for the forwarded external call
9	This is the default setting for displaying a caller id. Allowed: Allow my number to be displayed

	Restricted : Restrict my number to be displayed
10	Set the maximum number of saving the recent calls per user
11	Set the minimum or maximum level for reserving a bandwidth for a call
12	Allow transferring a call to the outside while the call is con line with the outside
13	Select an option to save a list of telephone numbers for incoming and outgoing calls

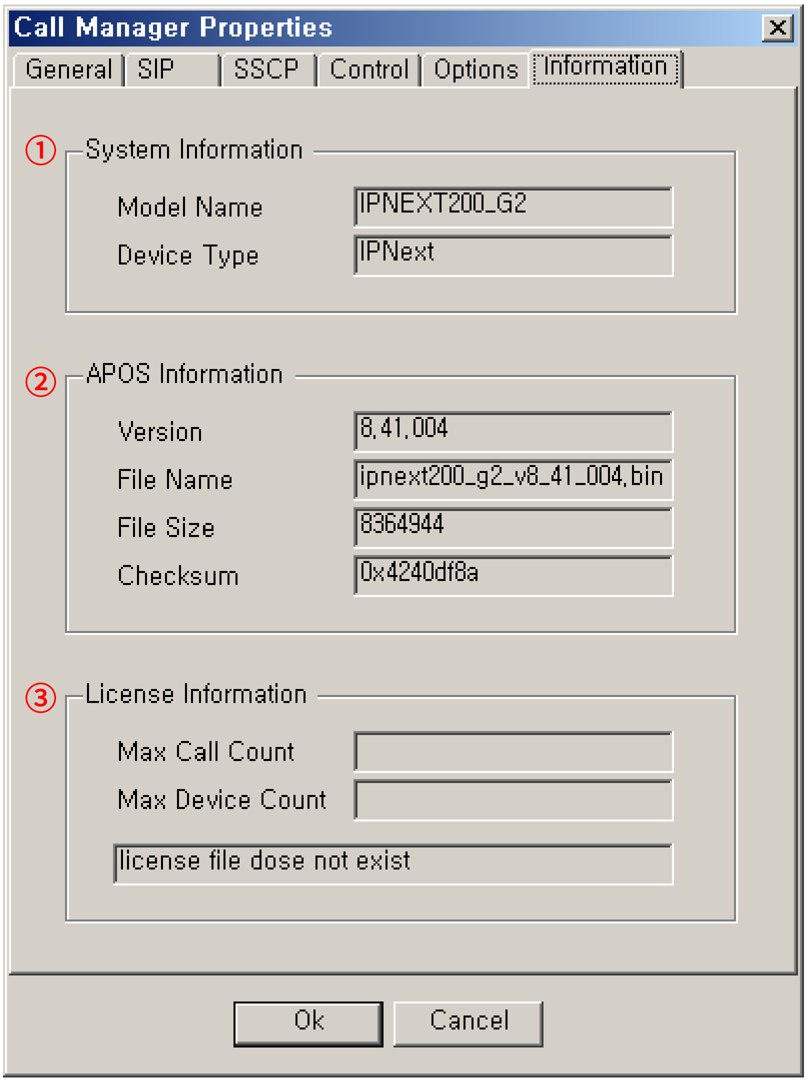

Figure 2-30 Call Manager – Information Properties

Table 2-19 Call Manager – Information Properties Description

Field	Description
1	Displays the System Information of Call Manager (Model, Device, Type)
2	Displays the information of APOS (Version, File Name, File Size, Checksum)

3	Displays the information of license (Max Call Count, Max Device Count)

Call Manager – Configuring Call Manager Preference

Call Manager Preference is a function to assign the priority level for each Call Manager to access in the following way:

- Assigns the priority level to each Call Manger which is configured by Call Manger Cluster
- Applies the policy of Preference to those terminals registered to Call Manager

This following Figure shows the screen for managing and registering Call Manager Preference:

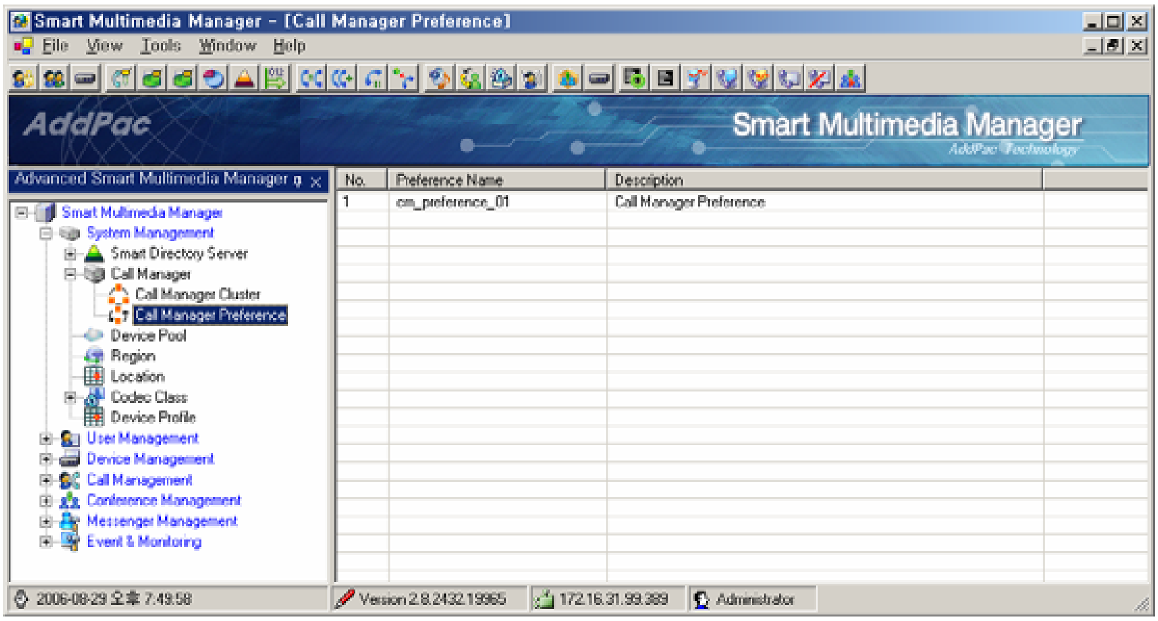

Figure 2-31 Call Manager Preference Screen

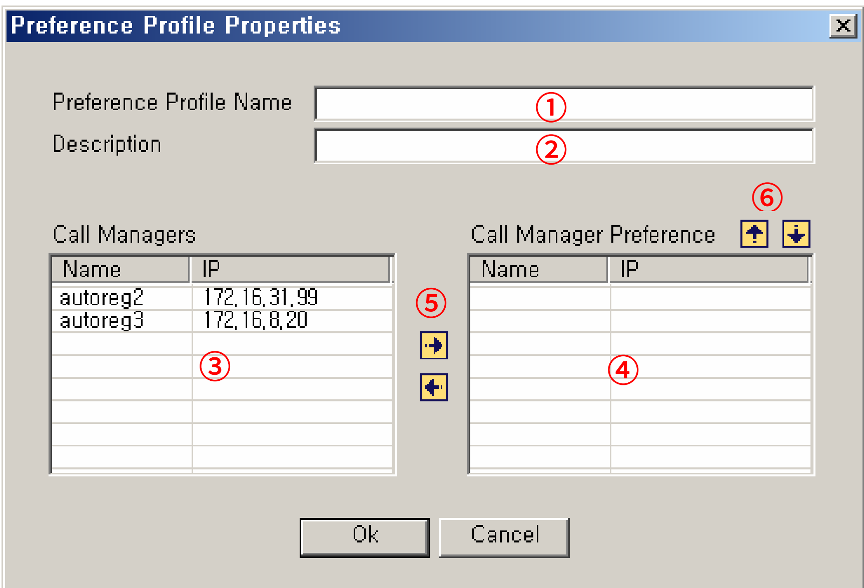

Figure 2-32 Call Manager Preference Properties

Table 2-20 Call Manager Preference Properties Description

Field	Description
1	Enter a name of Call Manager Preference
2	Enter a description of Call Manager Preference
3	This is the list of Call Manager Clusters
4	Enter a list of Call Managers included in the Preference.
5	Register or delete a Call Manager in the Preference
6	Change the priority level of Call Manager Preference

Device Pool Configuration

Device Pool is a function to configure the details of a device (terminal, trunk, servers and others) to be applied in common. All the devices must be included in Device Pool. The basic details of Device Pool are registered at the initial installation. In the default settings of Device Pool, the user can not change a name or delete. Only editing is allowed.

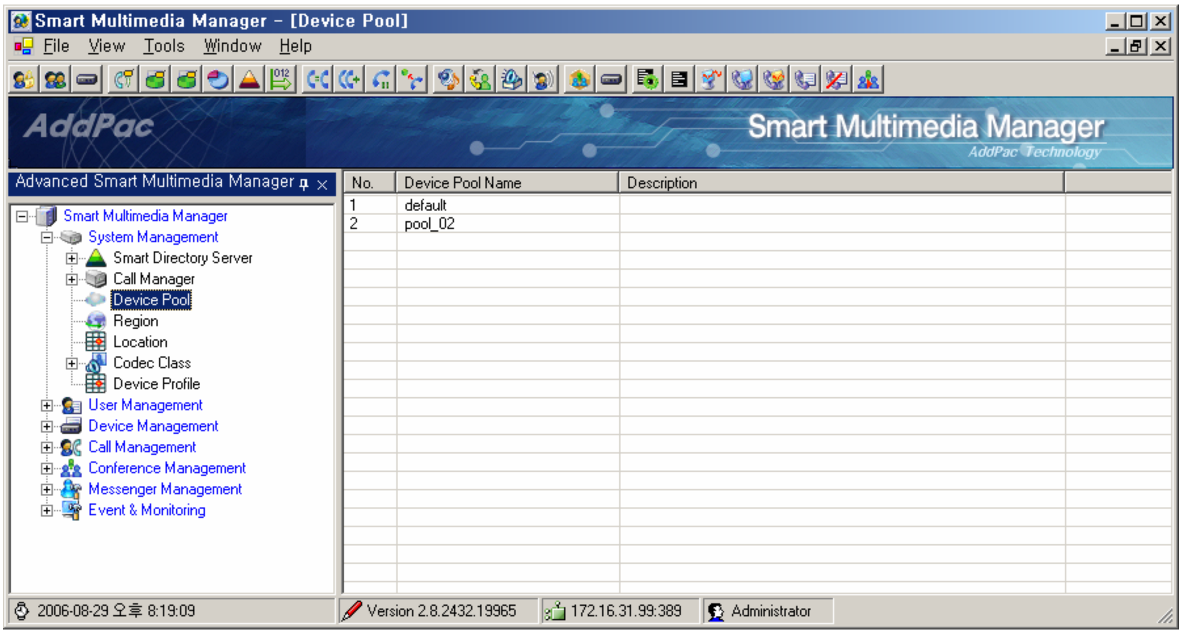

Figure 2-33 Device Pool Screen

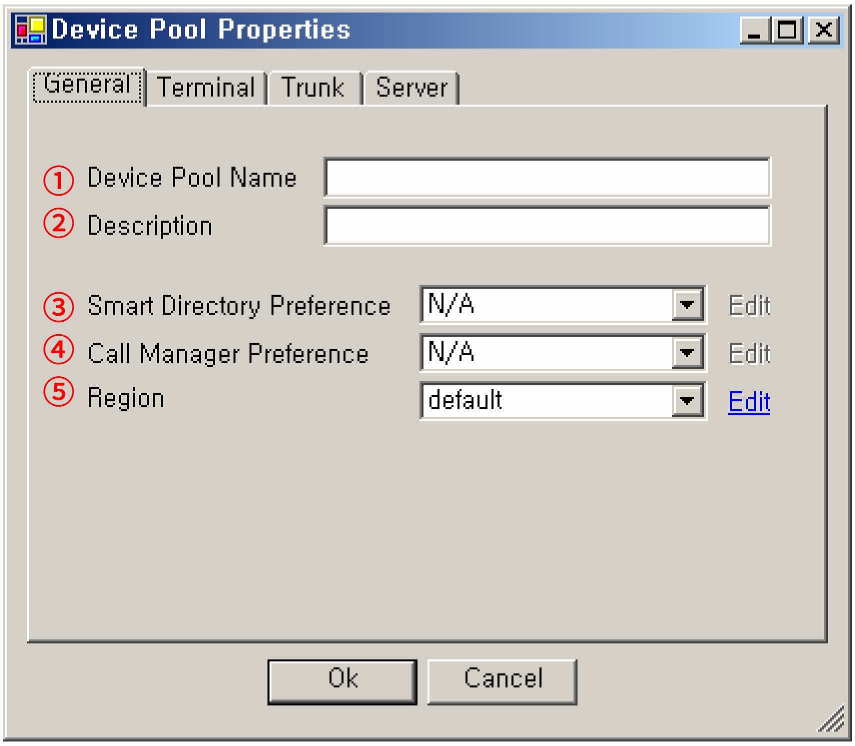

Figure 2-34 Device Pool - General Properties

Table 2-21 Device Pool – Description of General Properties

Field	Description
1	Enter a name of the new Device Pool that you are creating
2	Register a description of the Device Pool
3	Choose a Smart Directory Preference to be applied to the device (optional)
4	Choose a Call Manager Preference to be applied to the device (optional)
5	Choose a Region to be applied to the device (it is set to ‘default’ when no selection is provided)

The figure below shows each template of “Terminal’, ‘Trunk’ and ‘Server’ for the configuration of Device Pool to be applied to each device. These devices are applied to the present configuration of Device Pool.

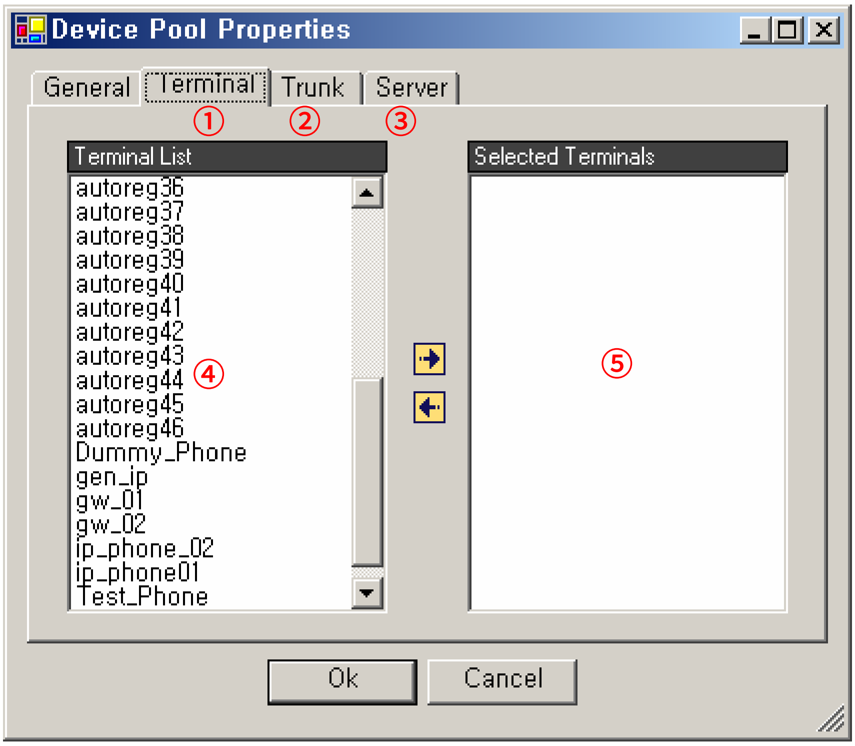

Figure 2-35 Device Pool Properties

Table 2-22 Device Pool Properties

Field	Description
1	This is the configuration template for the list of the Terminal to be applied to Device Pool.
2	This is the configuration template for the list of the Trunk to be applied to Device Pool.
3	This is the configuration template for the list of the Server to be applied to Device Pool
4	This is the list of the devices (Terminal, Trunk and Server) at the default setting of Device Pool.
5	This is the list of the devices (Terminal, Trunk, Server) to be applied to the present Device Pool

Region Configuration

Region is a function to allow the user to select a codec to be used for the calls between the different regions and to configure the related details in Device.

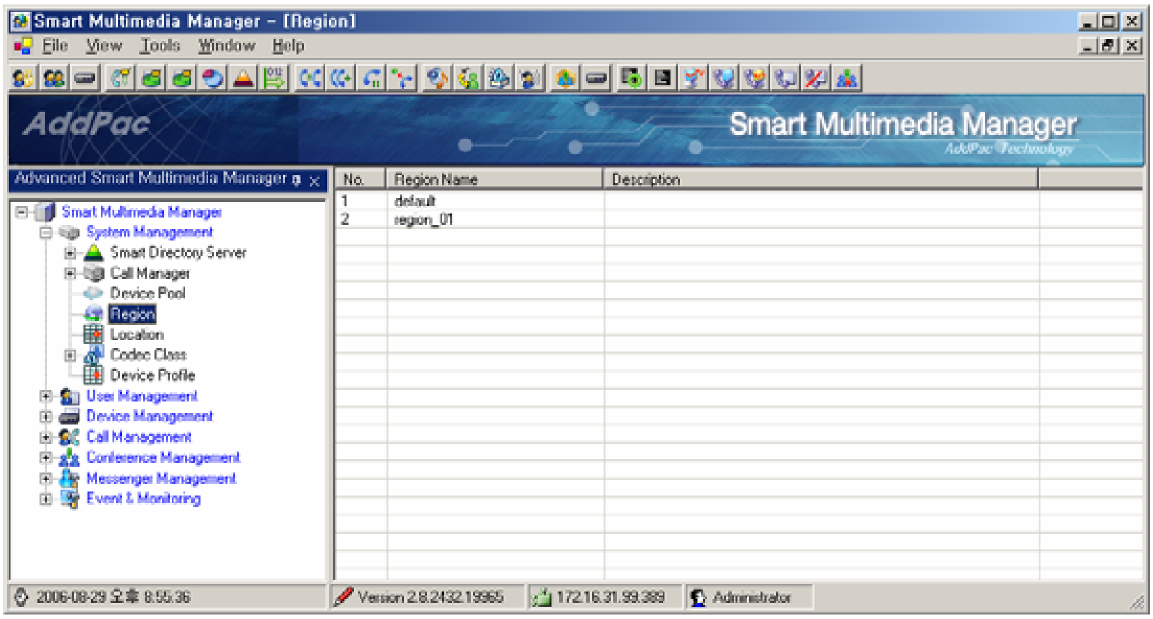

Figure 2-36 Region Screen

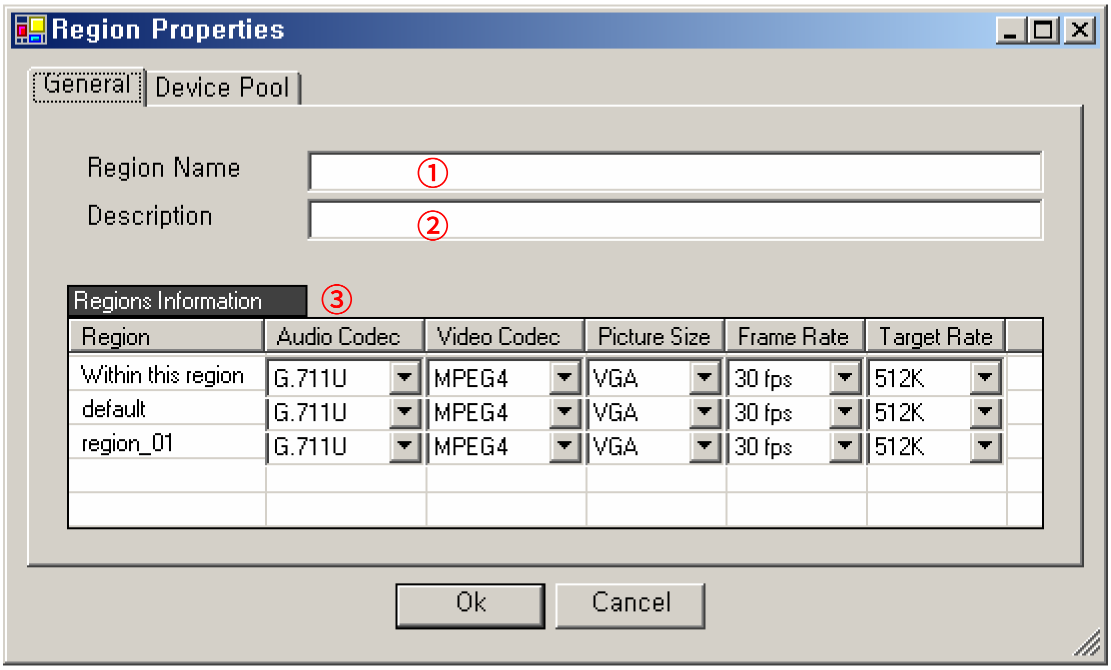

Figure 2-37 Region – General Properties

Table 2-23 Region - General Properties Description

Field	Description
1	Enter a name of the region
2	Enter a description of the region
3	'Within this region' is the information to configure the codec to be applied to the present region and the rest of the information is to be applied together with the configuration information of the other regions. Audio Codec : g.711u, g.711a, g.723, g.726, g.729 or audio codec class to be selected Video Codec : h.263, h.264, mpeg4 or video codec class to be selected Picture Size : this can be changed depending on a type of video codec (QCIF, CIF, 4CIF, QVGA, HVGA, VGA) Frame Rate : 9 ~ 30 fps Target Rate : 64K ~ 2048K

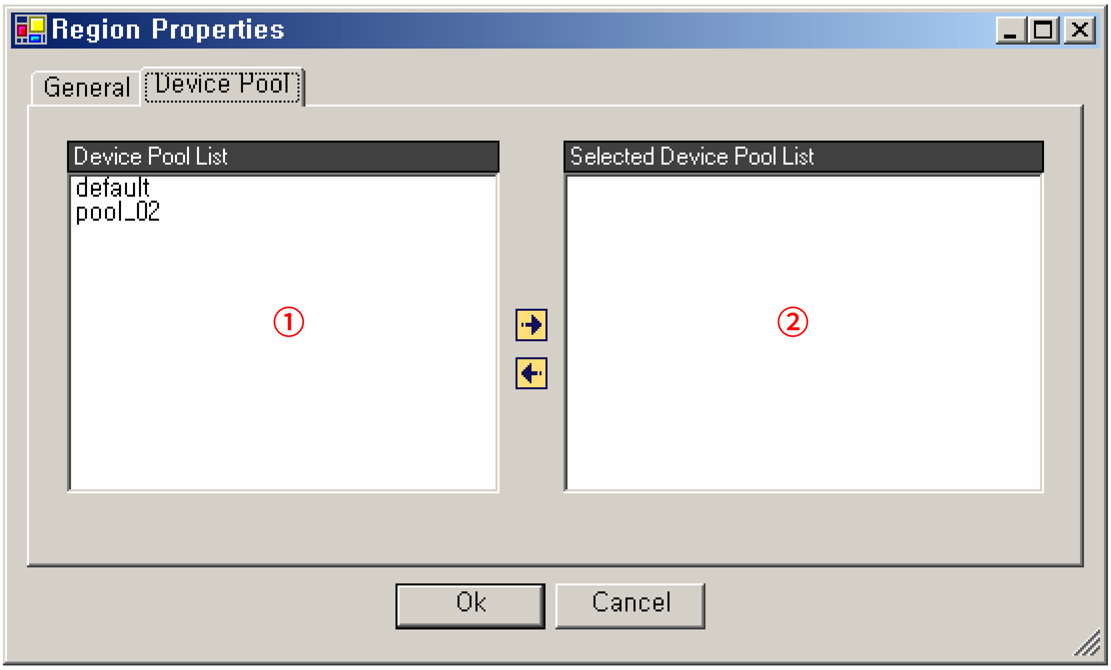

Figure 2-38 Region – Device Pool Properties

Table 2-24 Region – Description of Device Pool Properties

Field	Description
1	This is the list of Device Pool which is not applied to the present Region.
2	This is the list of Device Pool to be applied to the present Region

Configuring Location

Location Configuration is to set the maximum bandwidth within a particular location and it is used for Call Admission Control.

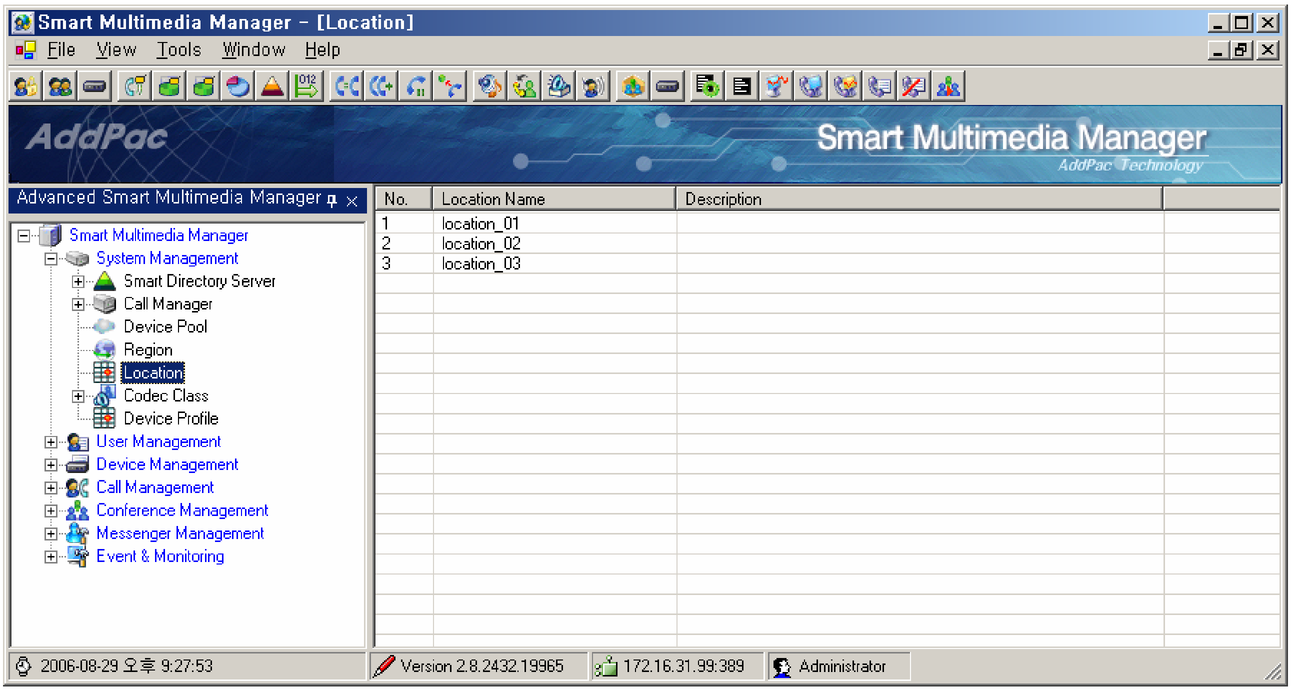

Figure 2-39Location Screen

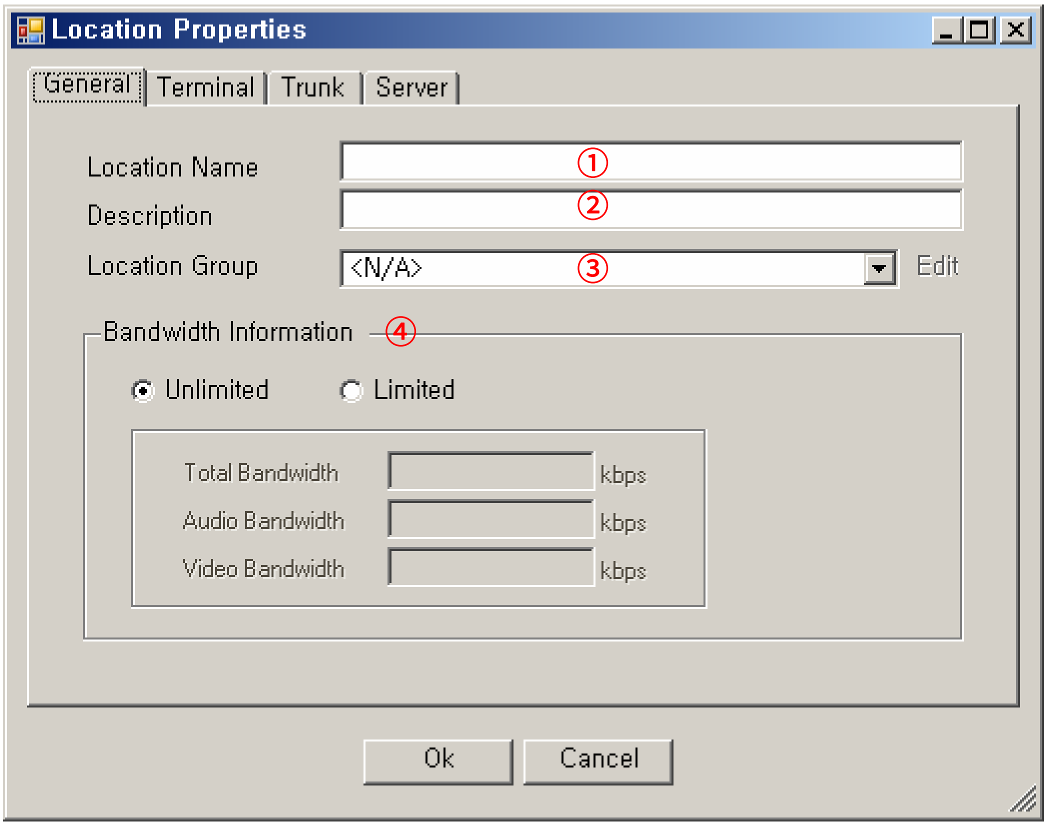

Figure 2-40　Location – General Properties

Table 2-25 Location – General Properties

Field	Description
1	Register a name of Location
2	Register a description of Location
3	Select the Location Group which includes the Location
4	Set the limit on the bandwidth of Device for related Location Unlimited : no limit Limited : bandwidth is limited (register Total, Audio, Video bandwidth)

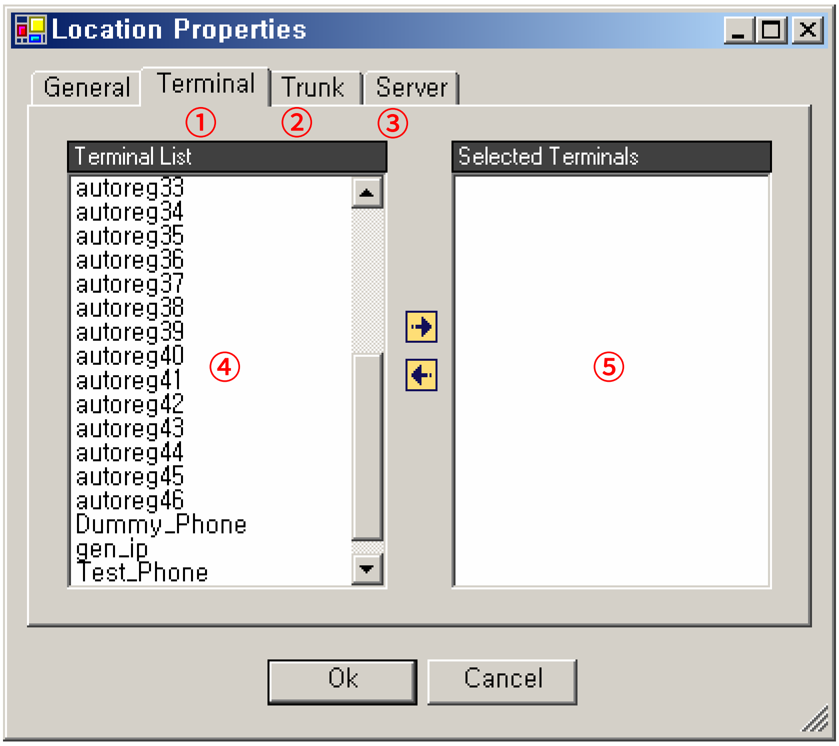

Figure 2-41 Location Properties

Table 2-26 Location Properties Description

Field	Description
1	This is the list of Terminal to which Location to be applied
2	This is the list of Trunk to which Location to be applied
3	This is the list of Server to which Location to be applied
4	This is the list of Device(Terminal, Trunk, Server) to which Location is not applied
5	This is the list of Device (Terminal, Trunk, Server) to which the present Location to be applied.

Configuring Location Group

Location Group organizes many locations into one group and sets the maximum bandwidth within the group.

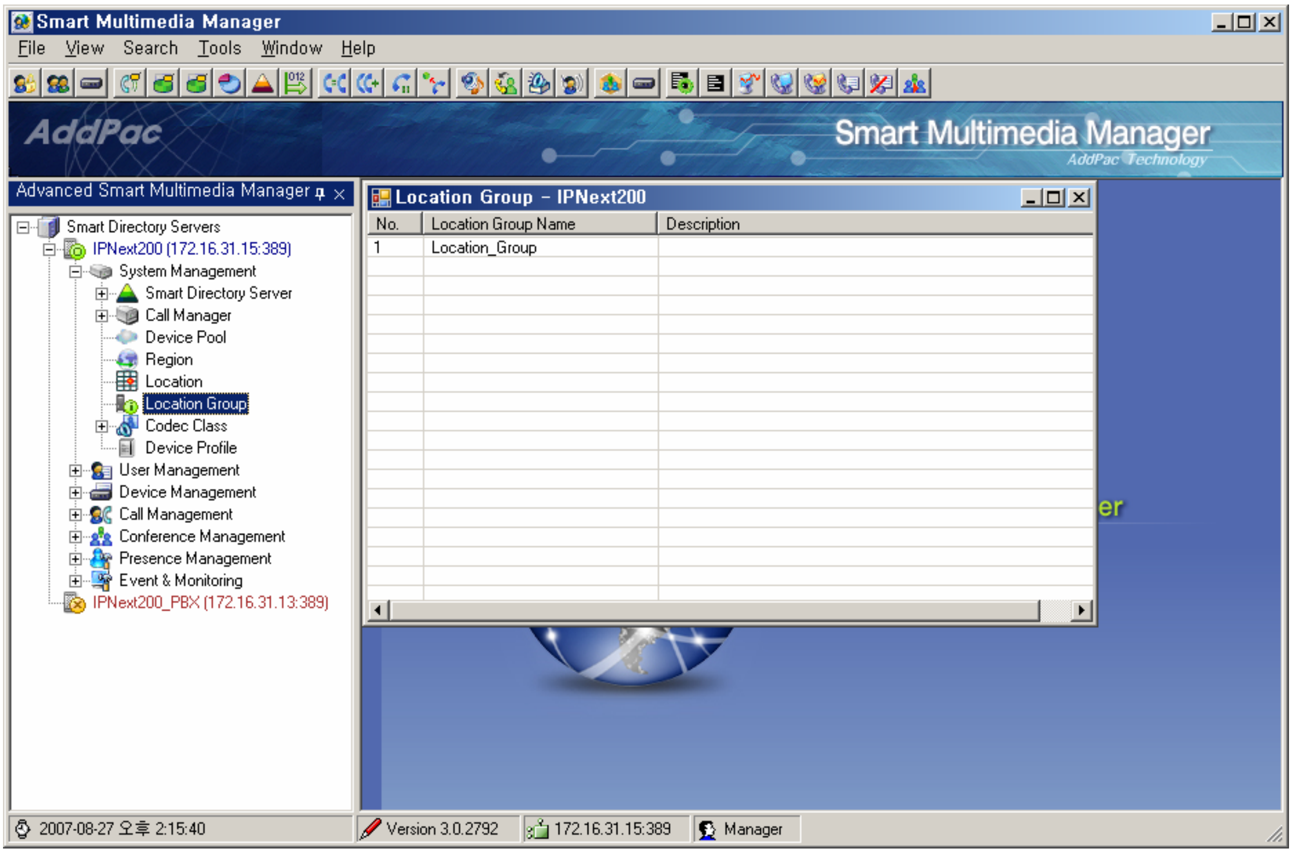

Figure 2-42 Location Group

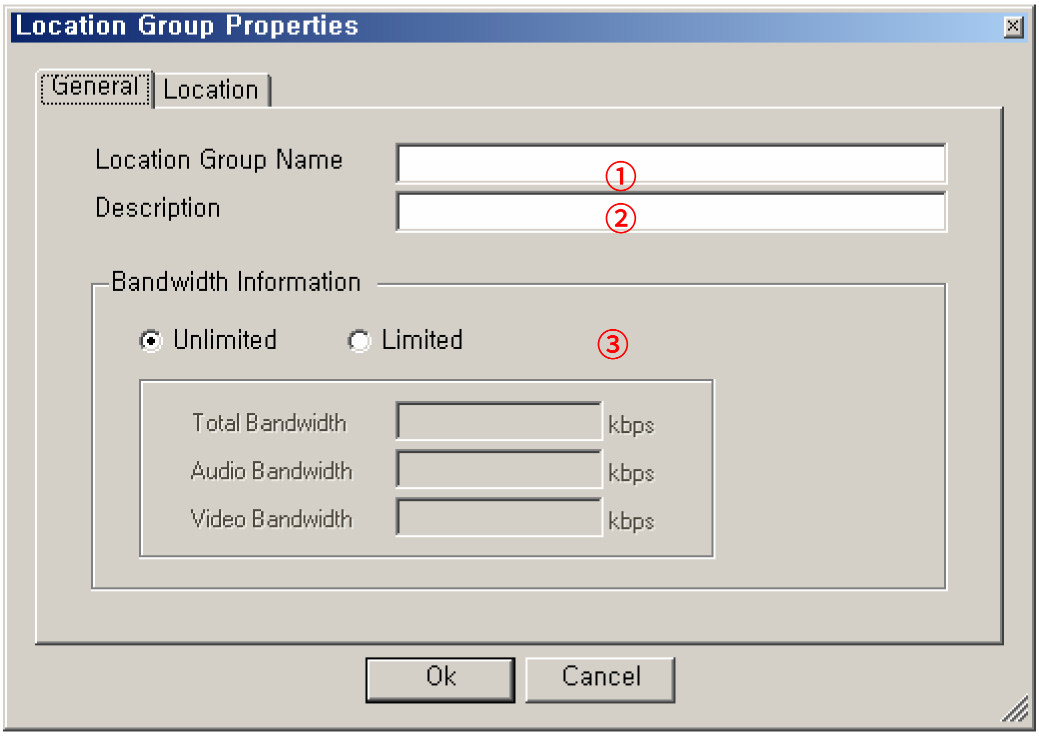

Figure 2-43 Location Group – General Properties

Table 2-27 Location Group – Description of General Properties

No.	Description
1	Register a location group name
2	Register a description of the location group
3	Set a limit to the bandwidths of the devices in the location group Unlimited: Set no limit to the bandwidth Limited :Set a limit to the bandwidth (register Total, Audio, Video bandwidth)

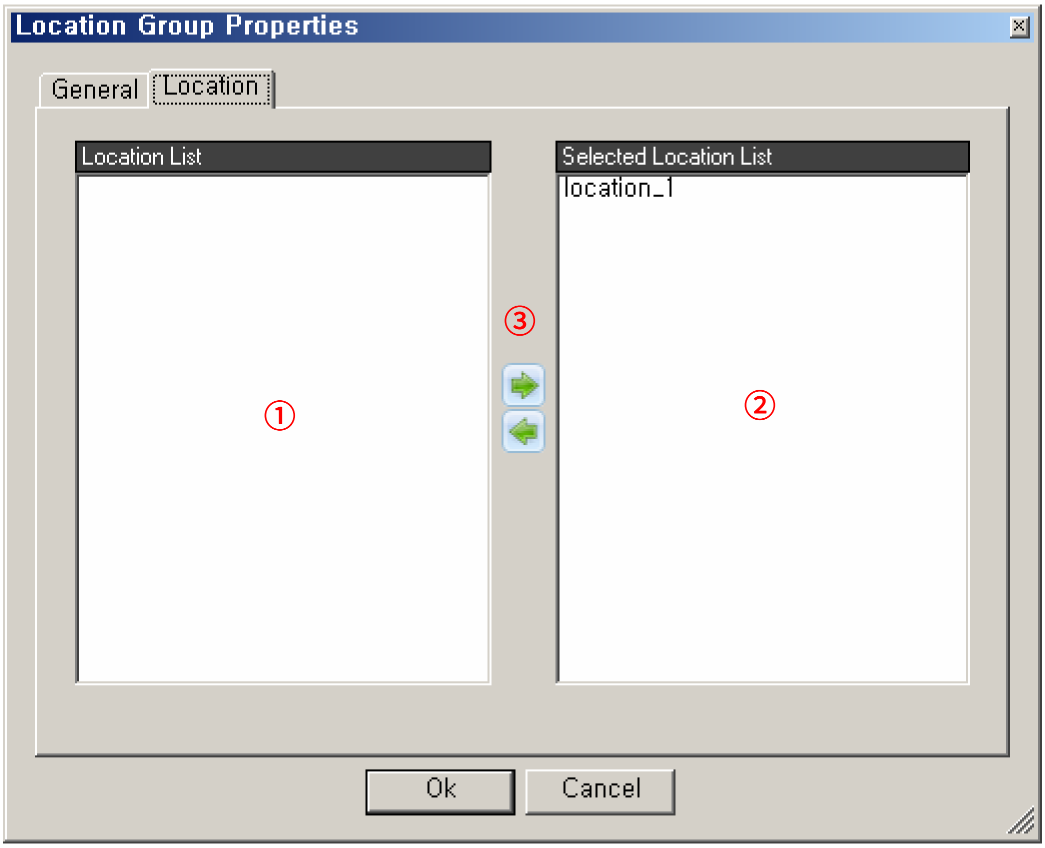

Figure 2-44 Location Group - Location Properties

Table 2-28 Location Group – Description of Location Properties

No.	Description
1	The location list to be applied with location group.
2	The location group listed applied to the location group
3	Add or delete location information to be applied to the location group

Configuring Codec Class

Codec Class is a function to group the list of supporting codec. After registering codec class (audio, video) in SMM, the configuration of codec can be applied, as the property of Region, which is applied to the calls between each group.

Audio Codec

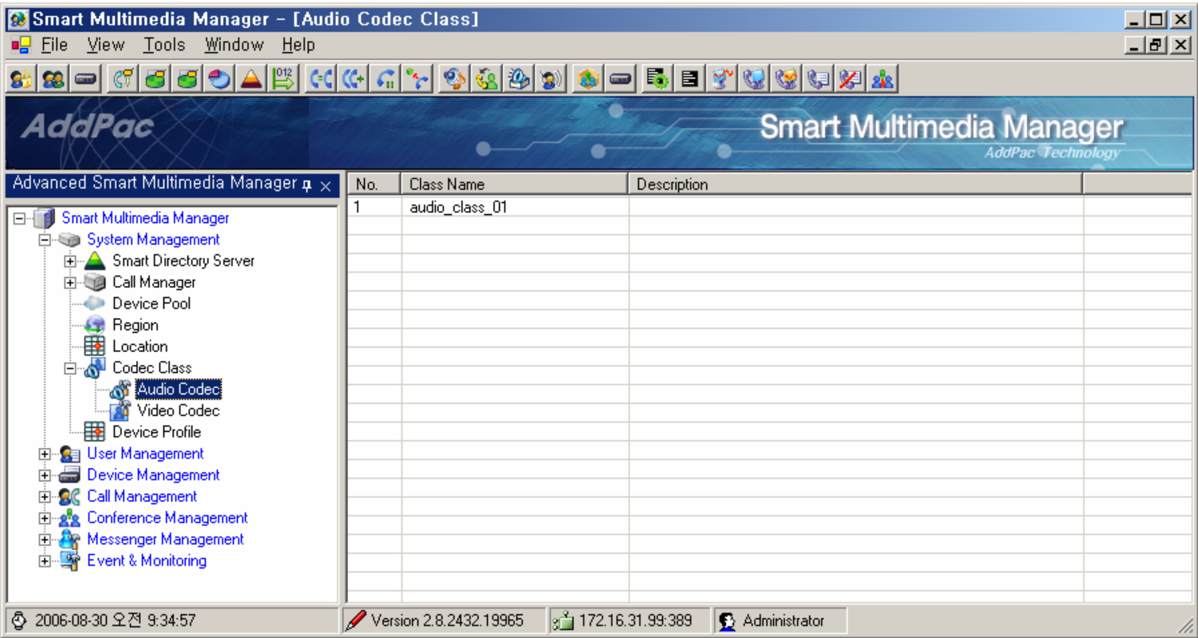

Figure 2-45 Codec Class – Audio Codec Screen

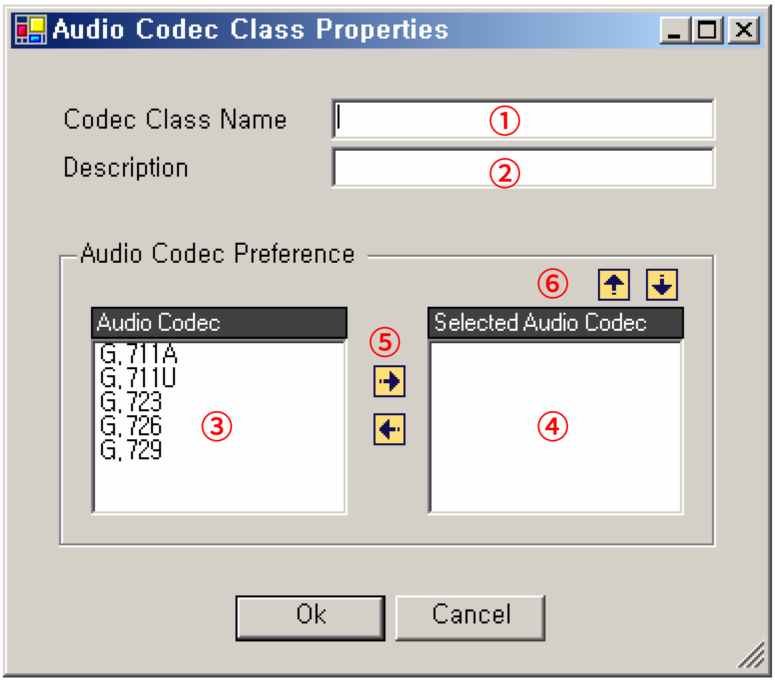

Figure 2-46 Audio Codec Class Properties

Table 2-29 Audio Codec Class Properties Description

Field	Description
1	Register a name of Audio Codec Class
2	Register a description of Audio Codec Class
3	This is the list of Audio Codec
4	This is the list of Audio Codec assigned to Codec Class
5	Delete or register codec to Codec class
6	Adjust the priority level of codec assigned to Codec Class

Setting Video Codec

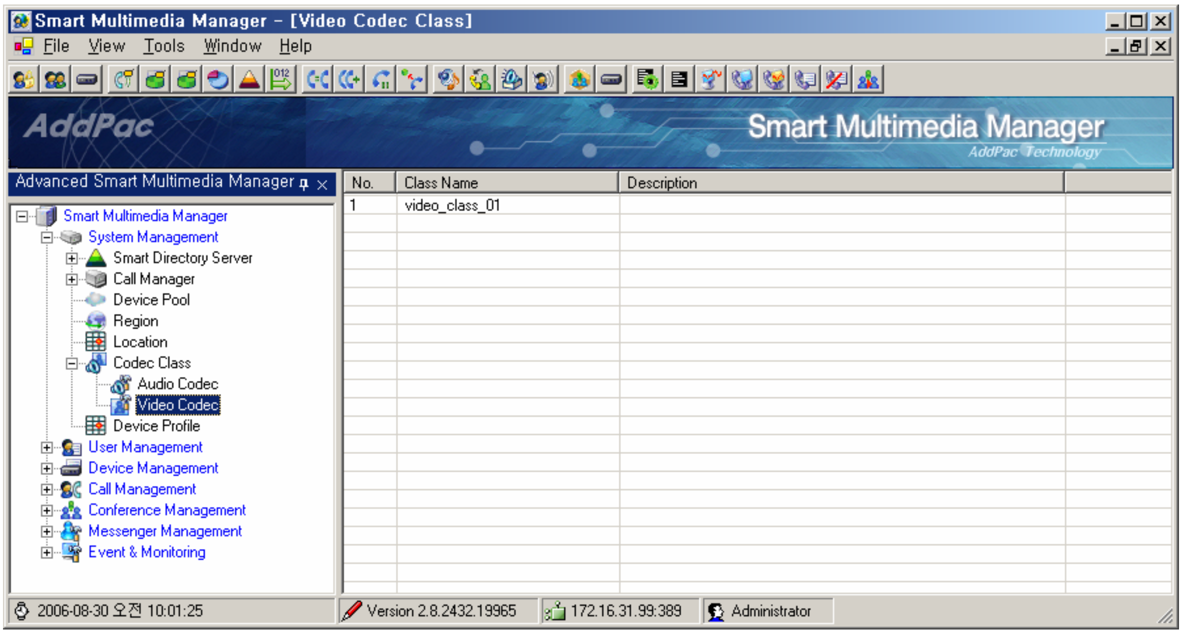

Figure 2-47 Codec Class – Video Codec Screen

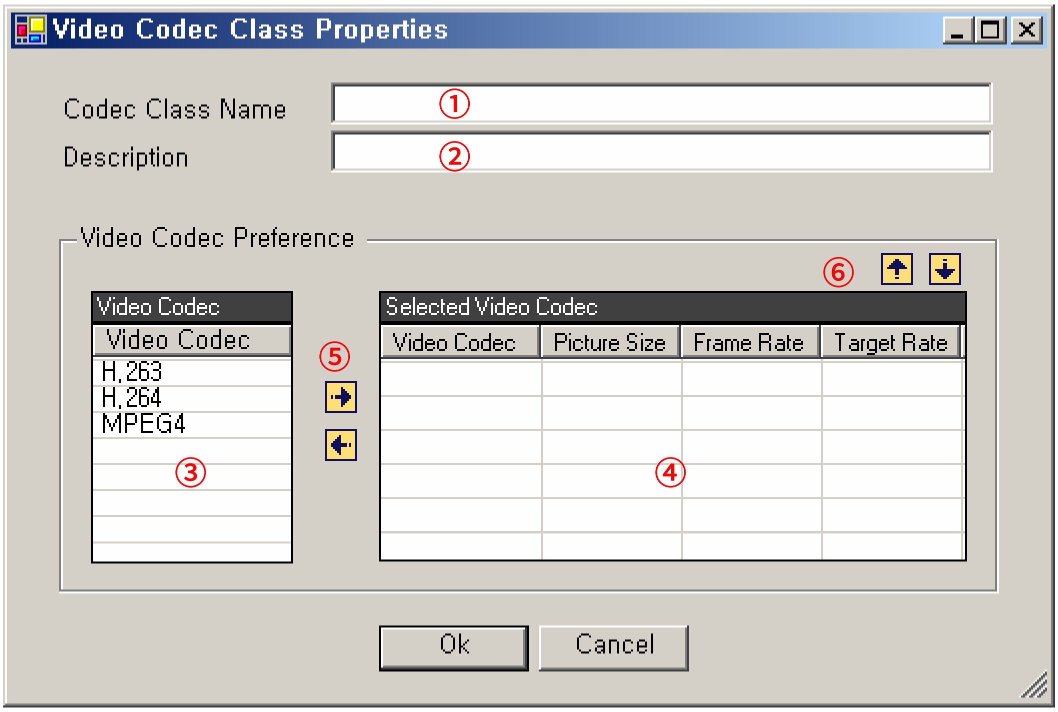

Figure 2-48 Video Codec Class Properties

Table 2-30 Video Codec Class Properties Description

Field	Description

1	Register a name Video Codec Class
2	Register a description of Video Codec Cclass
3	This is the list of Video Codec
4	This is the list of Video Codec assigned to Codec Class. Picture Size : this can be changed depending on a type of video codec (QCIF, CIF, 4CIF, QVGA, HVGA, VGA) Frame Rate : 9 ~ 30 fps Target Rate : 64K ~ 2048K
5	Delete or register a codec to Codec Class
6	Adjust the priority level of codec registered to Codec Class

Device Profile Configuration

Device Profile is used as a function to gather the common configuration of many terminals , such as the AddPac IP Phone, and to register them to a single profile. Device Profile can be selected and applied after registering Device Pool to SMM and at the time of registering the terminals.

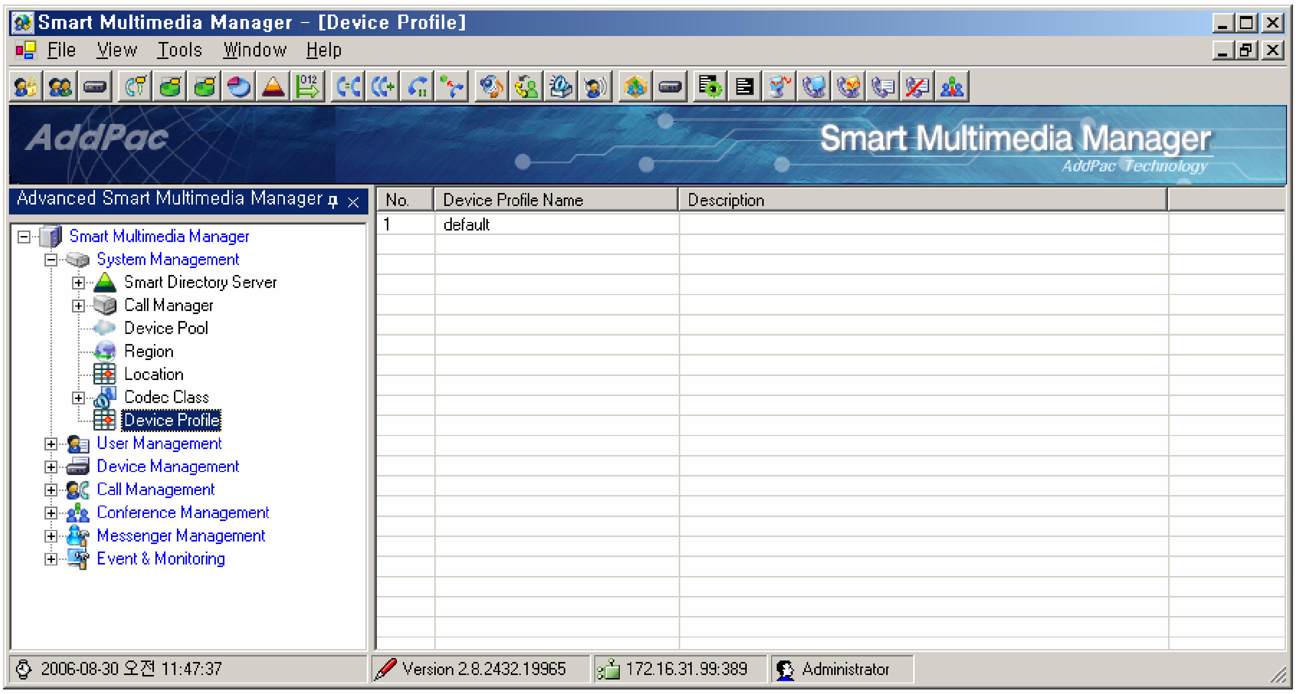

Figure 2-49 Device Profile Screen

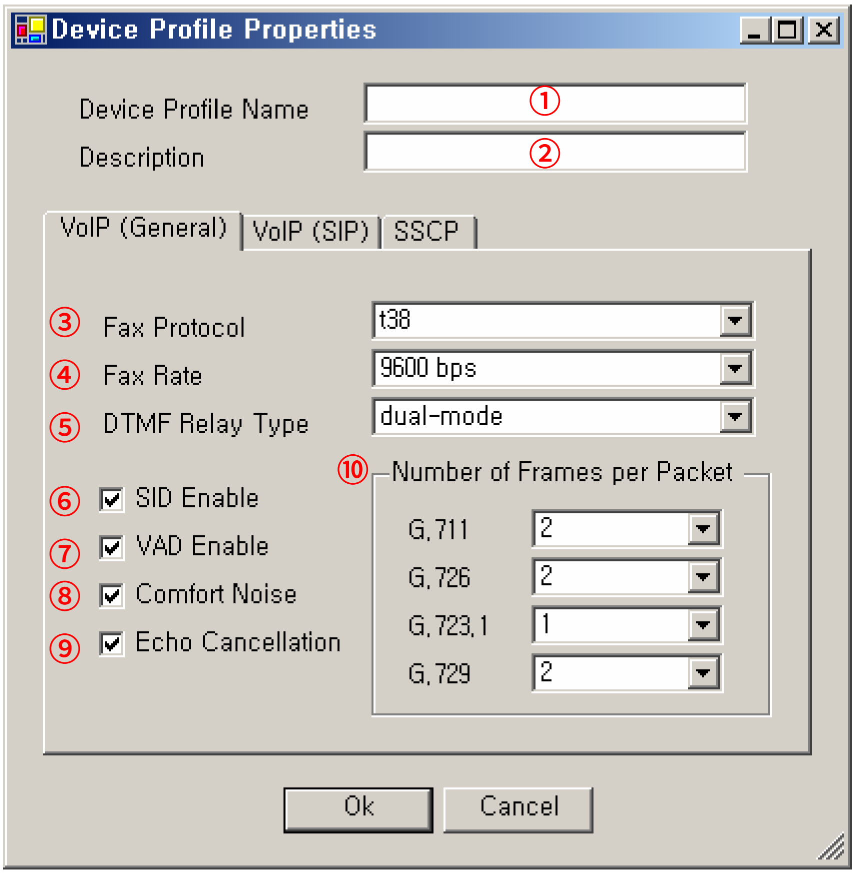

Figure 2-50 Device Profile - General Properties

Table 2-31 Device Profile - General Properties Description

Field	Description
1	Register a name of Device Profile
2	Register a description of Device Profile
3	Select VoIP Fax Protocol
4	Select VoIP Fax Rate
5	Select VoIP DTMF Relay
6	Set to determine whether to transmit a voice packet depending on the status of mute
7	Set to stop transmitting a voice packet when mute is enabled
8	Set to generate Comport Noise when mute is enabled
9	Set to cancel Echo
10	Determine a number of Frames per Packet for each codec

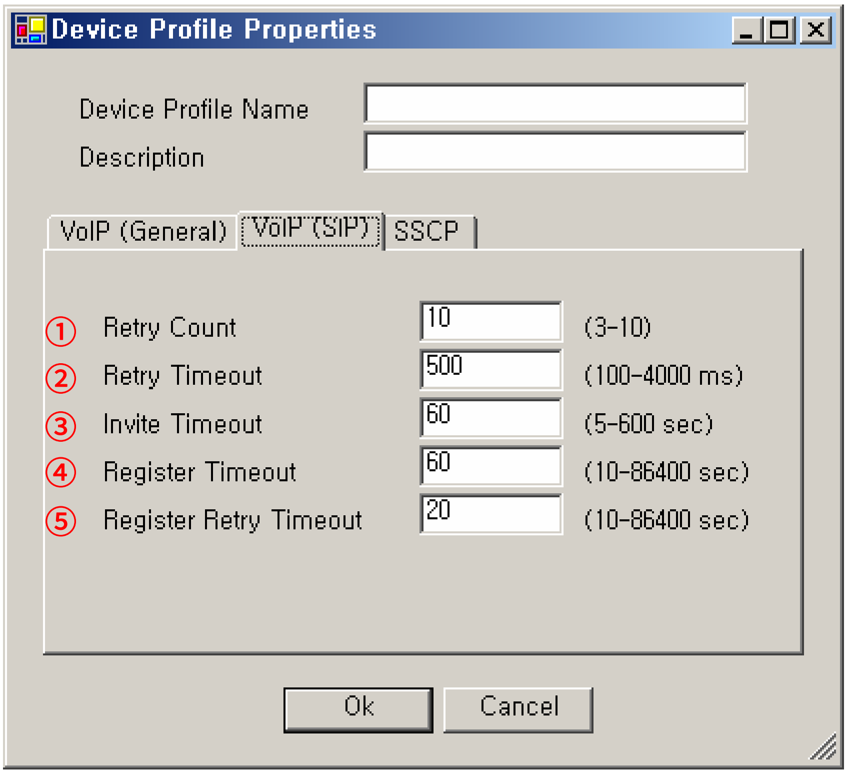

Figure 2-51 Device Profile – SIP Properties

Table 2-32 Device Profile – Description of SIP Properties

Field	Description
1	The number of frequency for retransmitting SIP messages
2	The expiration time for retransmitting SIP messages
3	The expiration time for SIP Invite
4	The expiration time for registering SIP terminals
5	The expiration time for retransmitting the message of registering SIP terminals

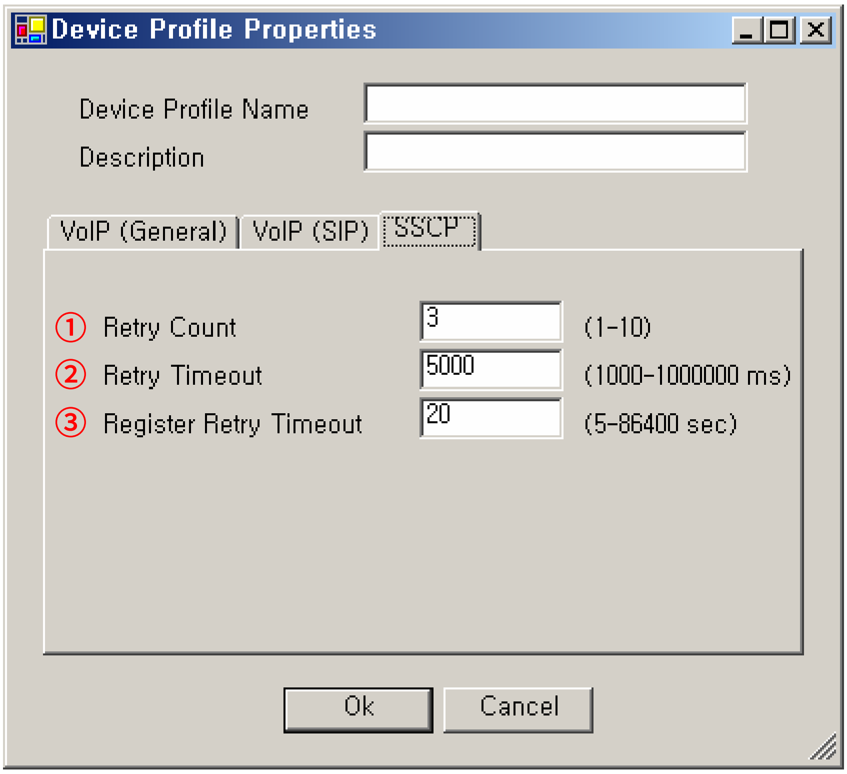

Figure 2-52 Device Profile – SSCP Properties

Table 2-33 Device Profile – Description of SSCP Properties

Field	Description
1	The number of frequency for retransmitting SSCP message
2	The expiration time for retransmitting SSCP message
3	The time for retransmitting SSCP message for registration even after it has filed to retransmit in the number of Retry Count that has been entered above

User Management

Administrator Configuration

This function is to administer and register an administrator account for operating IPNext PBX through SMM. Basically, an administrator's account is registered and the user can not delete the account neither its ID can not be changed. Only the password can be changed. The initial password for the administrator's account is 'router'. The user can delete, edit and register administrators by moving the cursor to the list screen of Administrators and pressing the right button of the mouse.

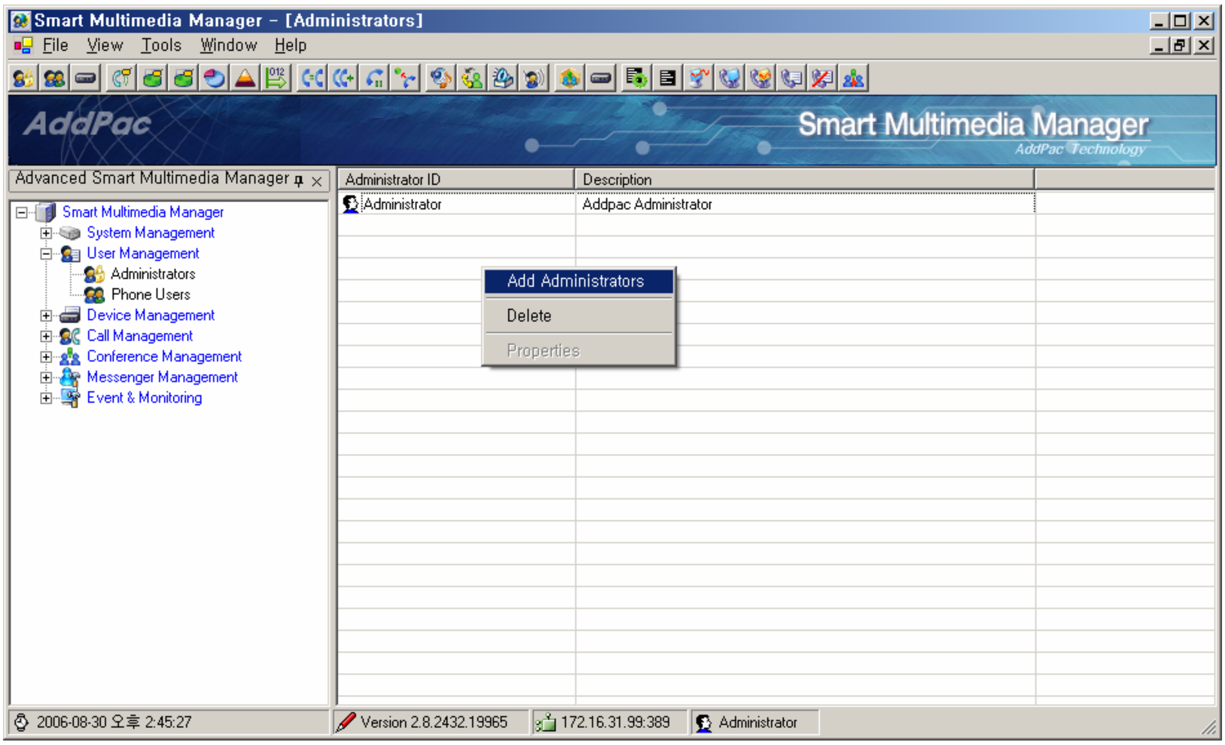

Figure 2-53 Administrator Screen

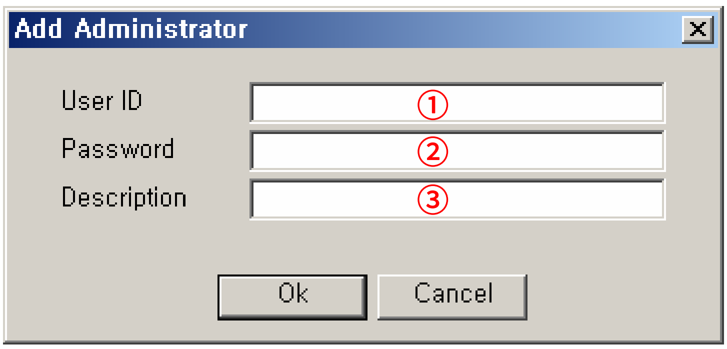

Figure 2-54 Administrator Properties Screen

Table 2-34 Description of Administrator Properties

Field	Description
1	Register an administrator's account ID
2	Register the administrator's password
3	Enter a description of administrator

Configuring Phone Users

This function enables you to register and administer the information on the users of the phones from IPNext PBX. The configuration of Phone Users provides the functions of configuring the basic information of the user, telephone number, Voice Mail and others.

This function provides classification of Phone Users by Organization unit makes registration easy as creating, moving and registering. After moving the cursor to the list screen for Phone Users and pressing the right button of the mouse, you can see the pop-up menu with options for deleting, registering and editing Organization.

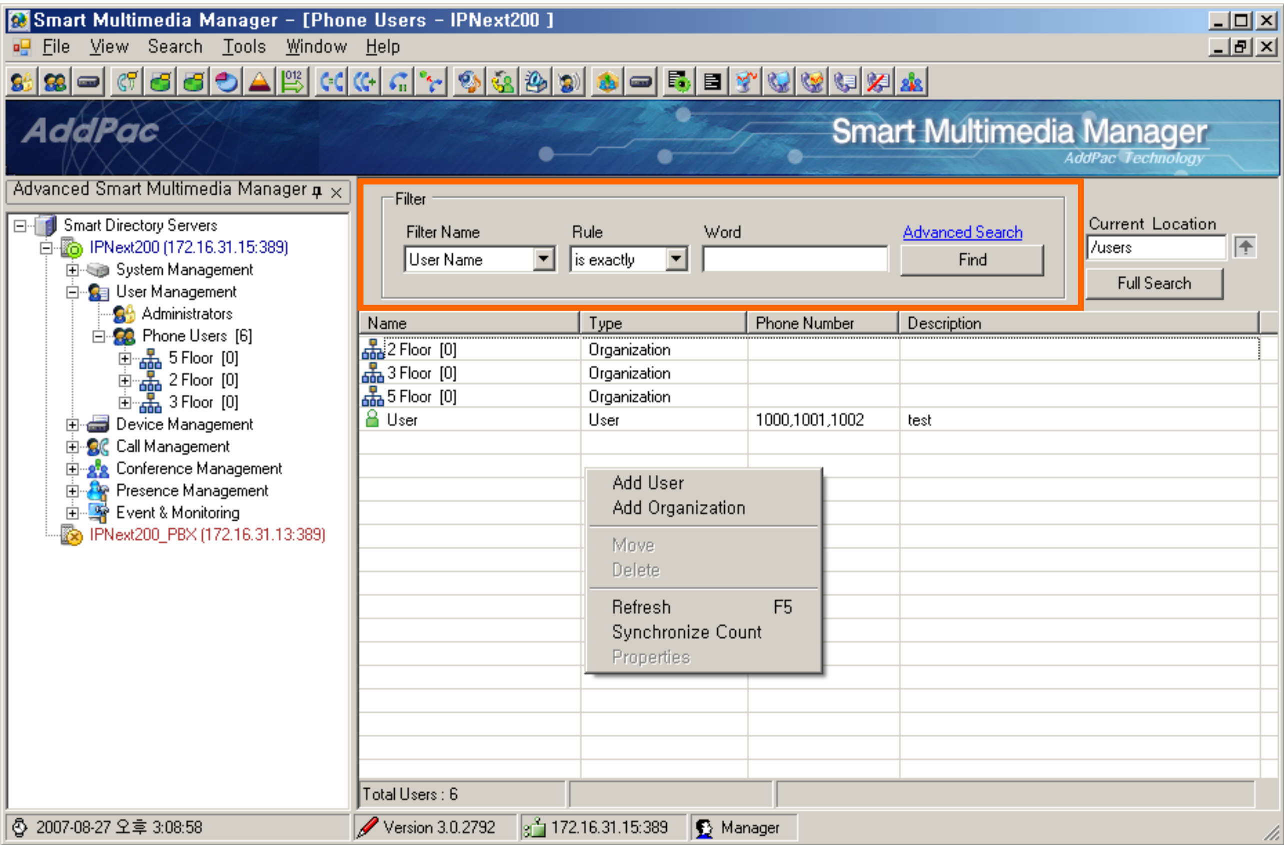

Figure 2-55 Phone Users 화면

The 'Filter' section in the above figure can search users by entering a 'Filter Name' (user name, ID, nick name, phone numbers). If you want to search all the users, press 'Full Search' or use Search > User of the main menu to search a user.

Table 2-35 Description of Phone User Menu

Menu Options	Description

Add User	Register a new user
Add Organization	Register the user's organization
Move	Mover the user or organization
Delete	Delete the user or organization
Refresh	Renew the user list
Synchronize Count	When the total number of users does not match with the number of users in user organization, this feature matches the number automatically
Properties	Execute properties

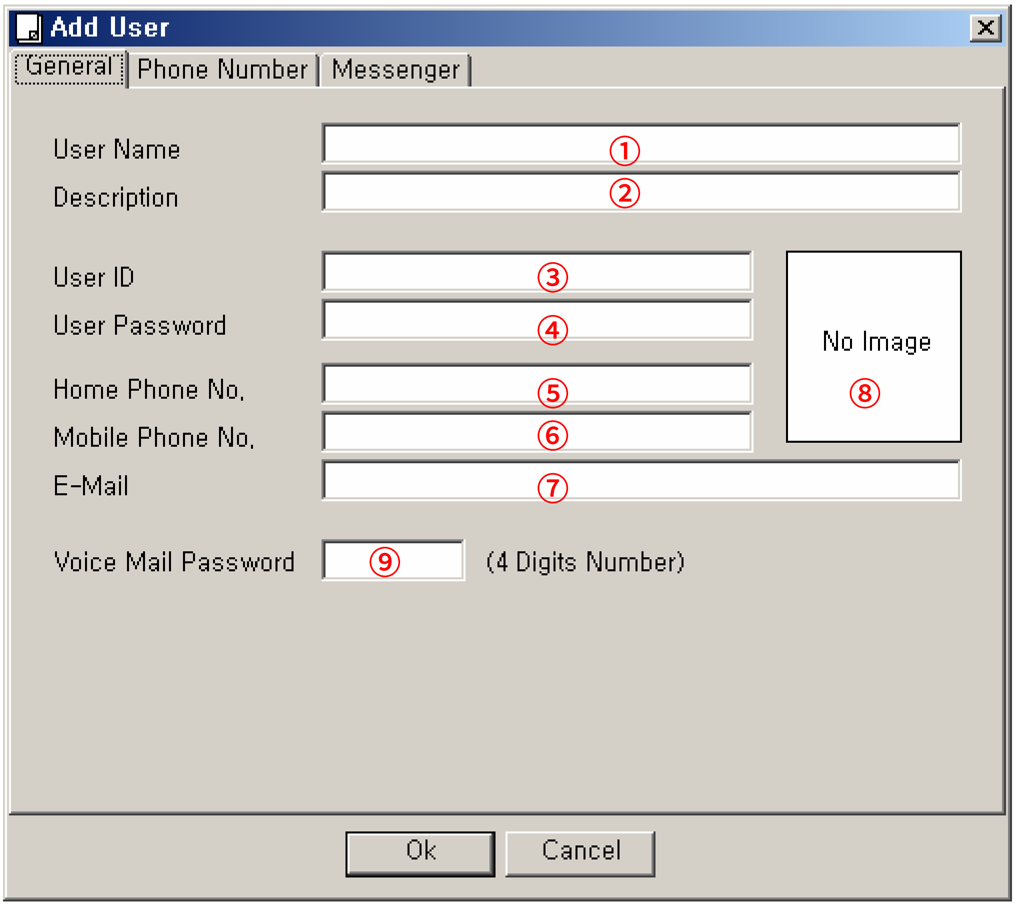

Figure 2-56 User – General Properties

Table 2-36 User – Description of Gernal Properties

No.	Description
1	Register a user's name
2	Register a description of the user
3	Register the user's ID

4	Register the user's password
5	Register the user's phone number
6	Register the user's mobile phone number
7	Register the user's e-mail
8	Register the user's picture
9	Register the user's password for Voice Mail (4 numeric digits)

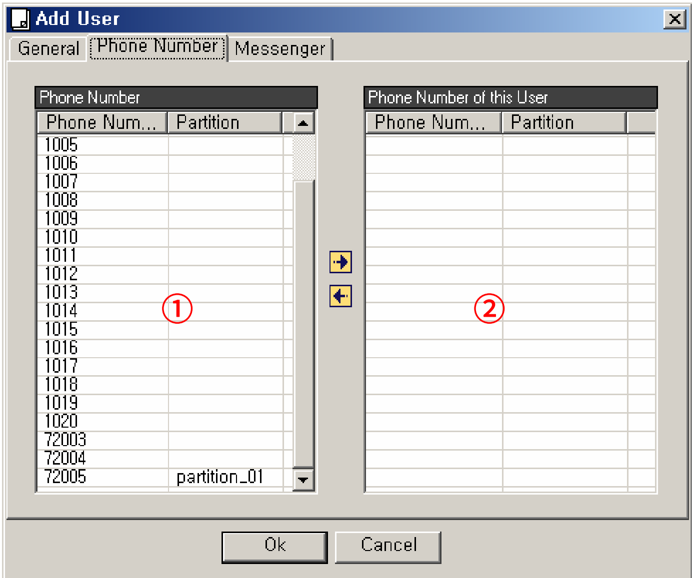

Figure 2-57 User – Phone Number Properties

Table 2-37 User – Phone Number Data Registration Description

Field	Description
1	This is the list of phone numbers which are not assigned (the list of phone numbers that are available to the user)
2	The assigned phone number to the user

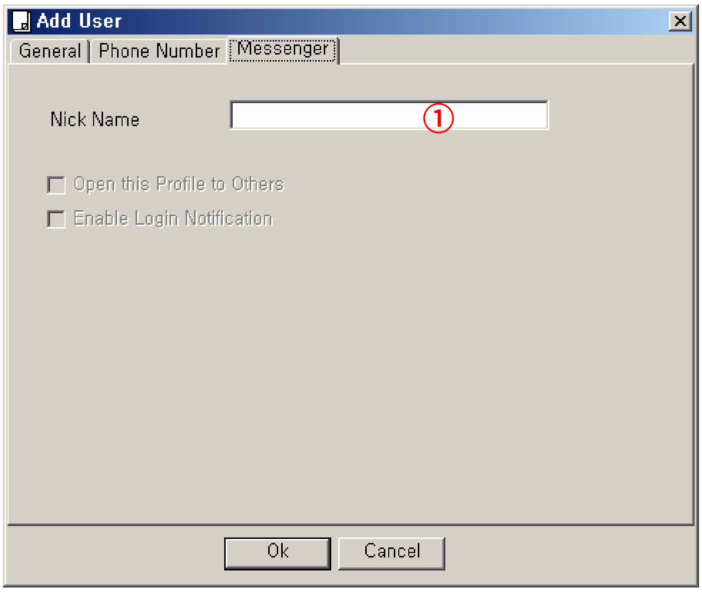

Figure 2-58 User – Registering Messenger

Table 2-38 User – Description of Registering Messenger

Field	Description
1	Enter a nick name to be displayed on the user's Smart Messenger

The following screen describes how to configure Organization for the user:

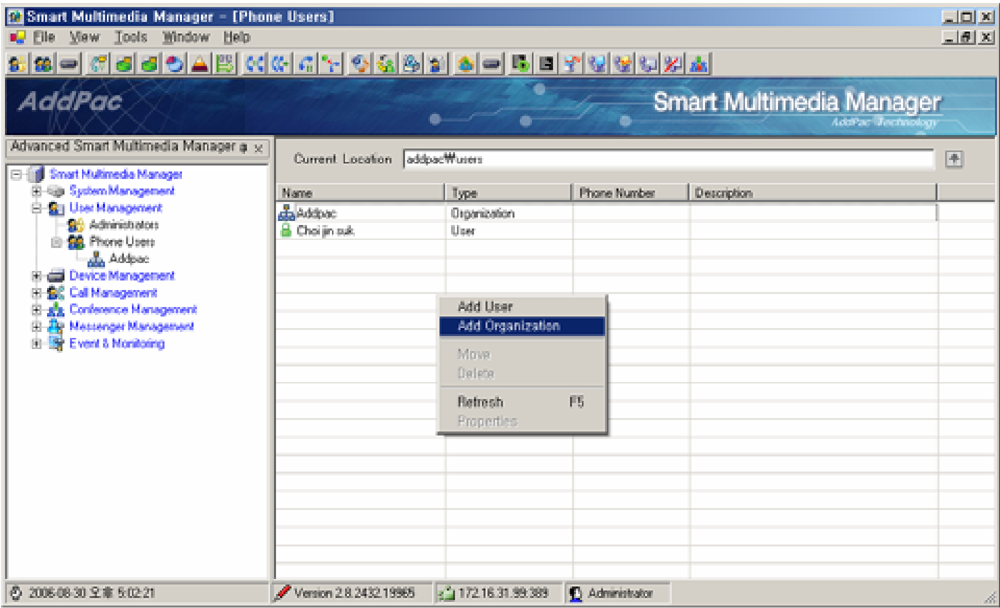

Figure 2-59 Phone Users - Registering Organization

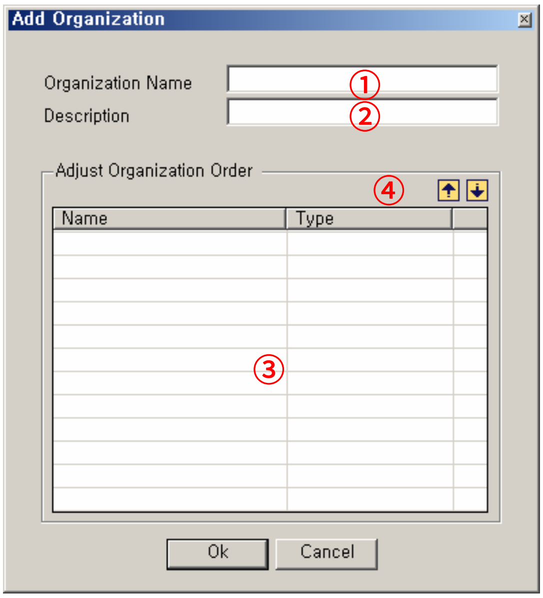

Figure 2-60 Registering Organization

Table 2-39 Description of Registering Organization

Field	Description
1	Register a name of the organization
2	Register a description of the organization
3	Configure the order to be displayed in Smart Messenger for the organization that the user belongs to or other organization on the list
4	Change the order of the users and the organization

Device Management

Terminal Configuration

Terminal Configuration is to register all the terminals (IP-Phone, Gateway) which can be operated through IPNext PBX.

There are 2 different type of terminal registration: Static Type for using static IP address and Dynamic Type: for using dynamic IP address

And also a method of authentication can be used and it supports registration without a separate authentication. Currently, IPNext PBX supports SSCP Registration, SIP Registration, H.323 Registration[2] for each terminal. Without Terminal Configuration, all the VoIP calls through IPNext PBX are limited. Terminal can be operated in the same way of classification as Phone Users and Organization. This type of organizational operation based on the tree structure makes Terminal operation of a large organization easy.

As it is shown in the Figure below, you can click Device Management in the tree menu on the left and select Terminals in the sub-category. You can move the cursor to the list menu on the right. Click the right button of the mouse to see the options for registering deleting and editing Terminal and Organization.

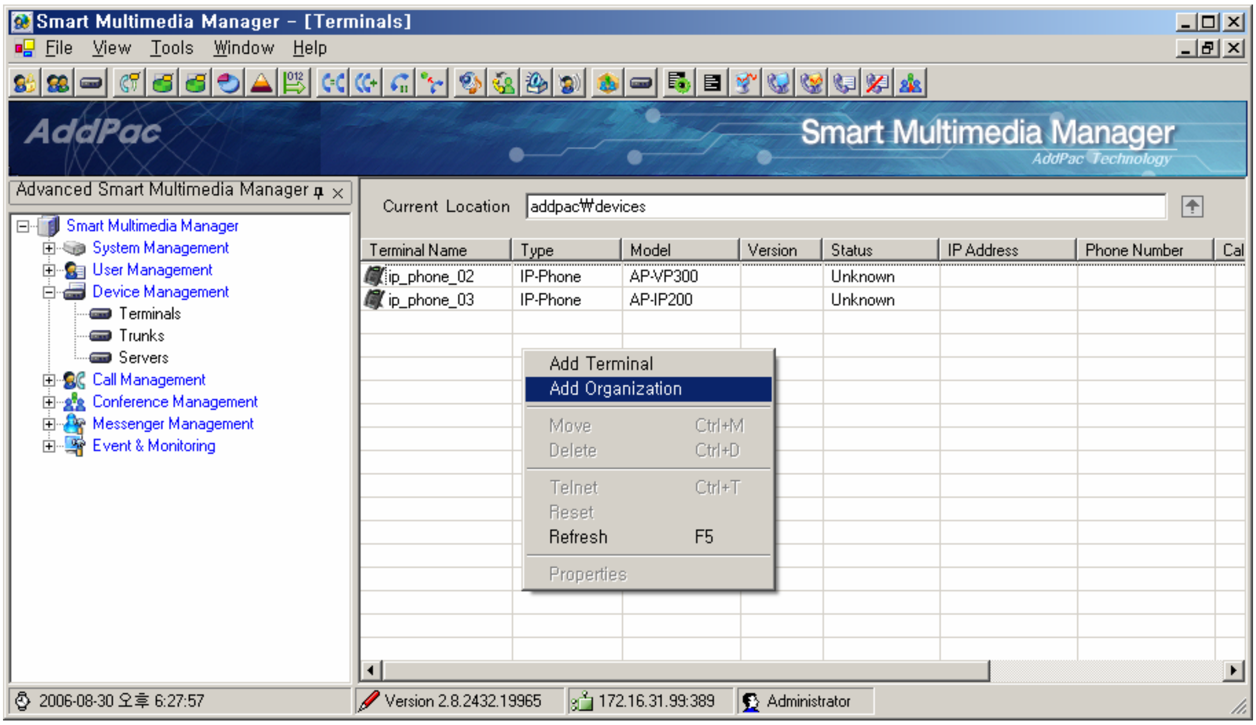

Figure 2-61 Device Management - Terminals Screen

Table 2-40 Terminal Menu Description

2 H.323 Terminal registration은 추후 지원 예정입니다.

Field	Description
Add Terminal	Register a Terminal(IP-Phone, Gateway)
Add Organization	Register an Organization
Move	Move the Terminal or Organization
Delete	Delete the Terminal or Organization
Telnet	Perform a remote access for the registered terminal
Reset	Reset the Terminal that supports SSCP
Refresh	Refresh the Terminal List
Properties	Perform Properties of the Terminal

The below Figure describes Properties for Organization of Terminals

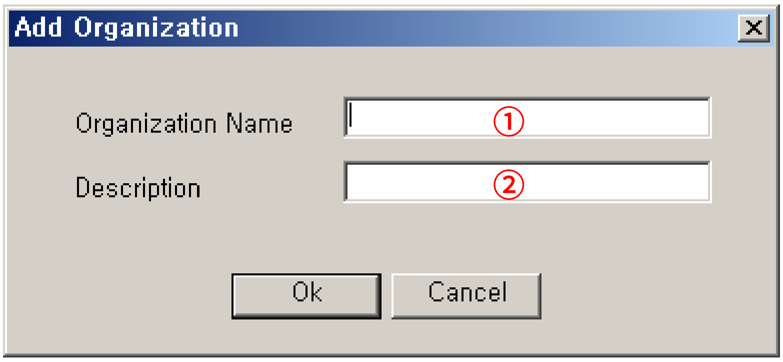

Figure 2-62 Terminal Organization Properties

Table 2-41 Terminal Organization Properties Description

Field	Description
1	Register a name of Terminal Organization
2	Register a description of the Organization

The figure below describes Properties of the Terminal:

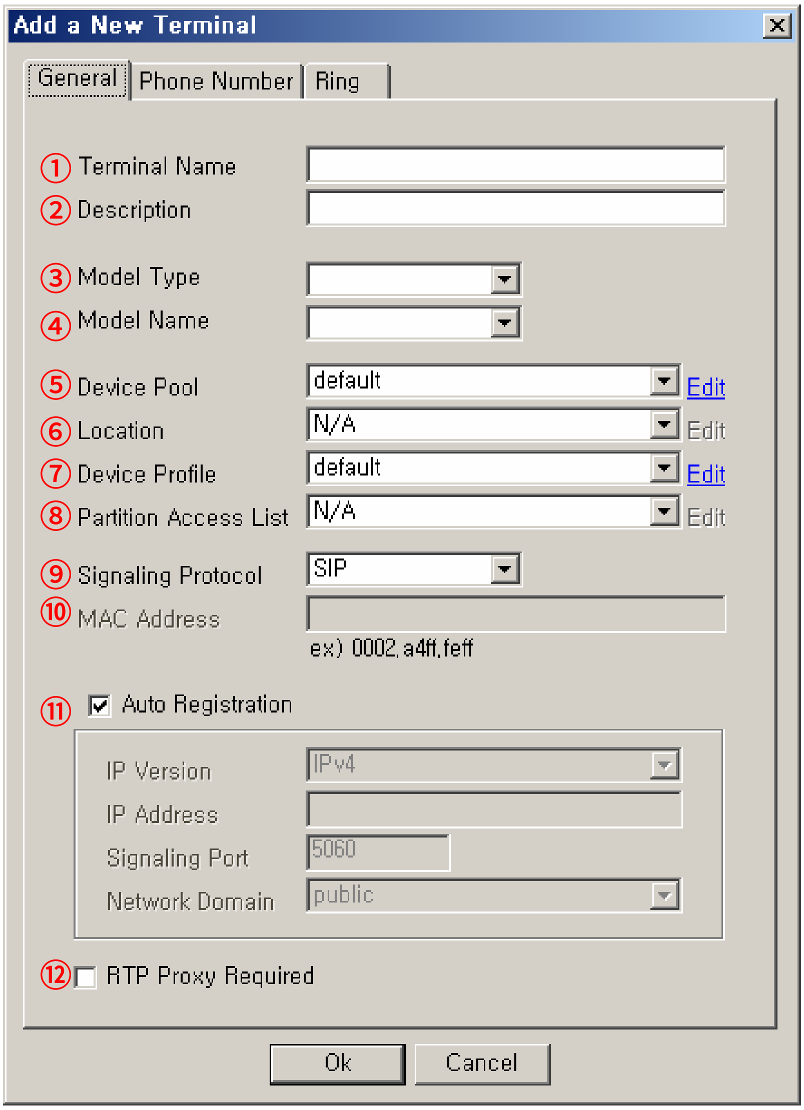

Figure 2-63 Terminal - General Properties Screen

Table 2-42 Terminal – Description of General Properties

FIELD	Description
1	Register a name of the Terminal
2	Register a Description of the Terminal
3	Select a type of product for the Terminal (IP-Phone, Gateway)
4	Select a product model name for the terminal (this is a sub-category of Model Type
5	Select Device Pool ('default' is the basic setting)
6	Select Location ('N/A' is the basic setting)
7	Select Device Profile ('default' is the basic setting)
8	This is to configure Partition Access List which applies to this Device. When an inbound call from the Device is processed, the outbound call can be processed only for the terminal (or Trunk) of which number belongs to the Partition (or Pattern). When Partition Access List is set to 'N/A' for no configuration, any number with the property of Partition or the

	Pattern (any object not configured in Partition) can be routed.
9	Select VoIP Signaling Protocol which is to be used by the Terminal.
10	Enter the SIP signaling port of the Device for Static Register For IP-Phone, the terminal is registered automatically as well as its MAC address.
11	Cancel Auto Registration for Static Register, because Auto Registration doe not apply. Then select IP Version of Terminal, IP address, Signaling Port, Network Domain of the call manager. For Dynamic Register, select Auto Registration, because it applies.
12	This is a forced setup to use RTP Proxy while the call is on line.

The template of Phone Number presents a screen to assign the telephone numbers to each port of Terminal. Depending on Device Model, a number of ports can be created automatically.

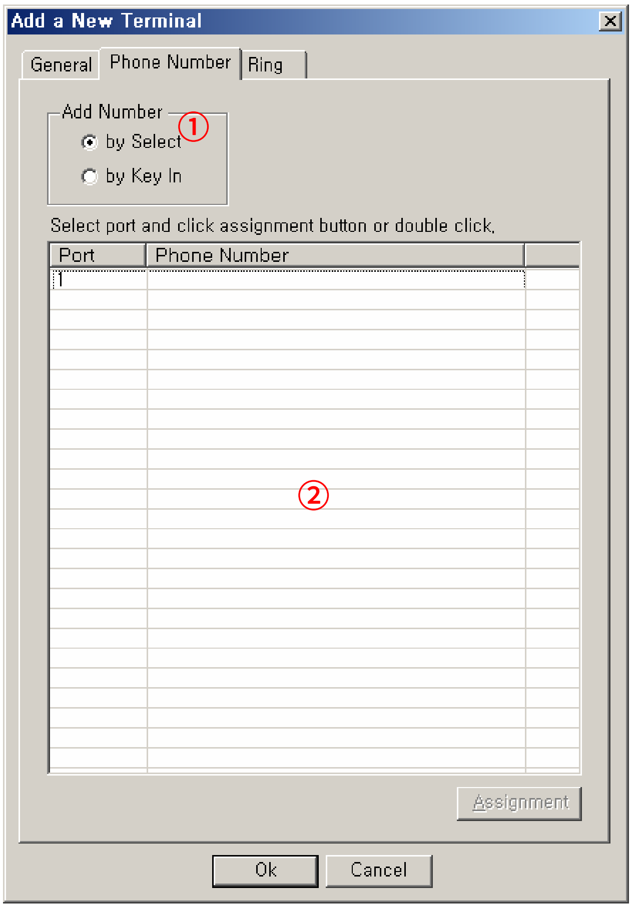

Figure 2-64 Terminal – Phone Number Properties

Table 2-43 Terminal – Description of Phone Number Properties

Field	Description
1	Choose whether to add number by selecting a telephone number or keying in directly
2	The telephone numbers are created for the applicable port of Terminal. Then assign the telephone number which

	corresponds to the selected port

The following Figure shows the screen to select a telephone number that corresponds to the port.

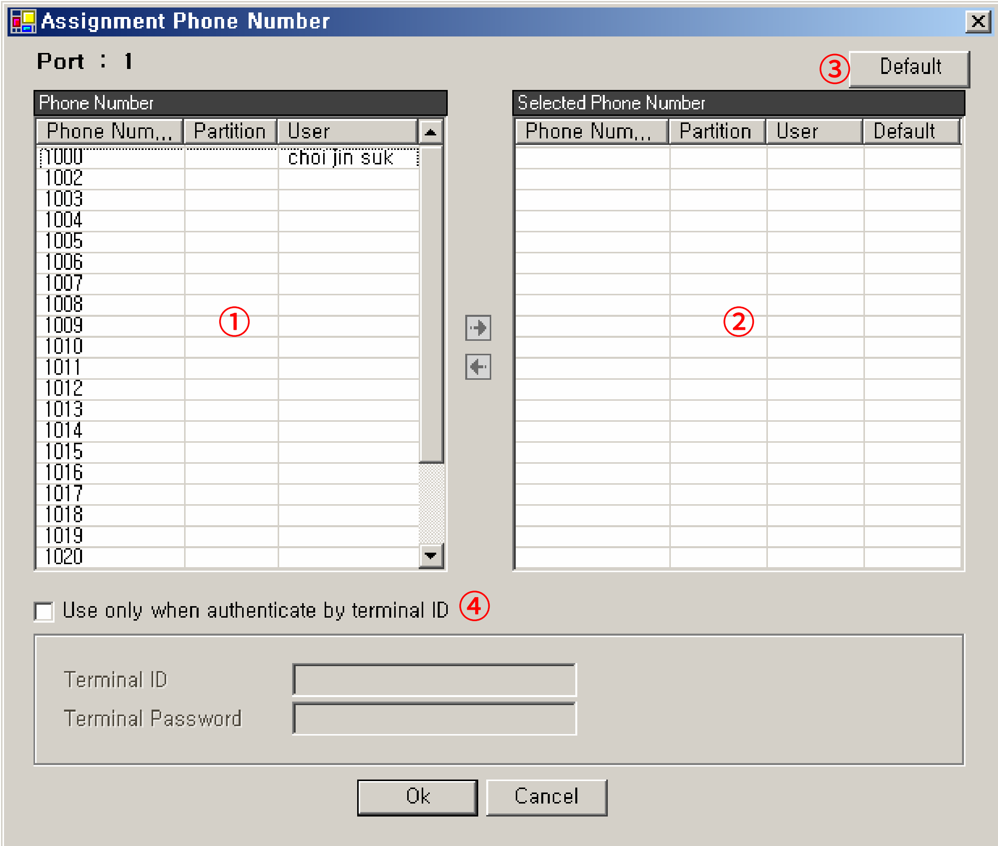

Figure 2-65 Terminal – Selecting a Phone Number

Table 2-44 Terminal – Description of Phone Number Selection

Field	Description
1	The list of phone numbers which can be assigned to each port.
2	The list of phone numbers which have assigned to each port.
3	Select one phone numbers to be transmitted for the outbound call from the corresponding Terminal.
4	Enter an ID and a password which will be used for authentication when the corresponding terminal is registered to the IPNext PBX by SIP Register

The following screen describes how the user registers the telephone number directly to the corresponding port:

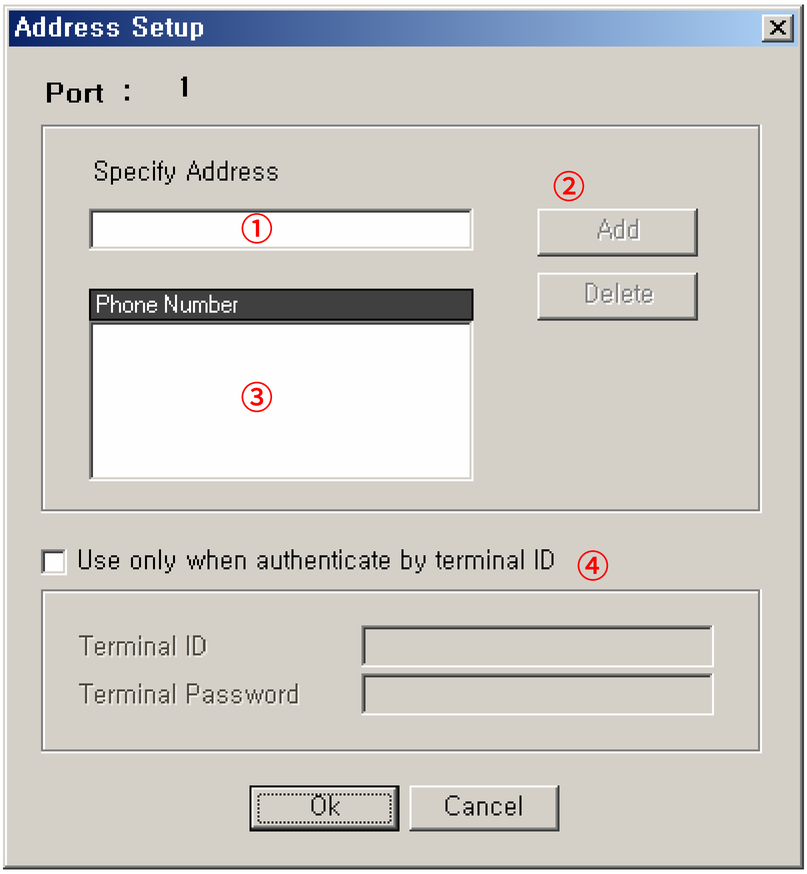

Figure 2-66 Terminal – Direct Entry of Phone Number

Table 2-45 Terminal –Description for Direct Entry of Phone Number

Field	Description
1	Enter a telephone number to be assigned to the corresponding port
2	Register the entered telephone number to or delete the entered telephone number from the list
3	This is the list of telephone numbers that the user has entered.
4	Enter an ID and a password which will be used for authentication when the corresponding terminal is registered to the IPNext PBX by SIP Register. Without the entry, it does not authenticate.

The template for Ring is available to be configured only after IP-Phone has been selected for Device Type.

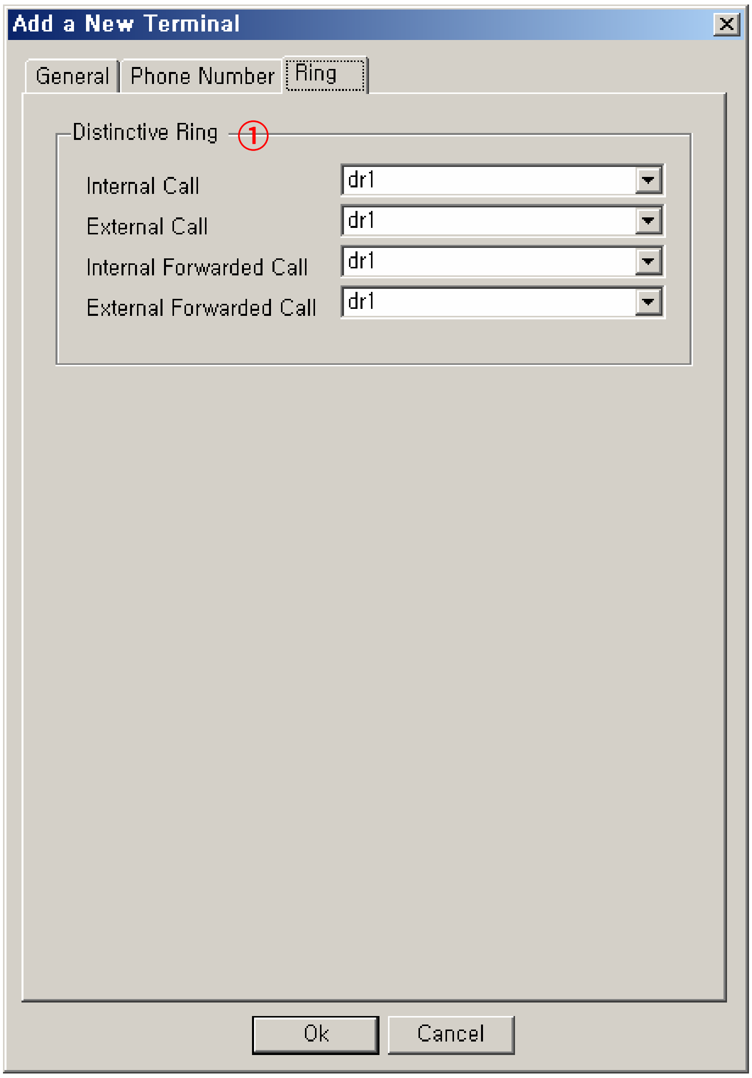

Figure 2-67 Terminal – Ring Setup

Table 2-46 Terminal – Ring Setup

Field	Description
1	Configure a bell sound of the terminal basing on a type of call - Internal Call : Set a bell sound for the call between extensions. - External Call : Set a ball sound for the incoming call from outside - Internal Forwarded Call : Set a bell tone for forwarding the incoming call from another extension - External Forwarded Call : Set a bell tone for forwarding the incoming call from outside

Trunks

Trunk Gateway Configuration

Trunk Gateway is an equipment that takes an outgoing call (either VoIP or PSTN) sent from the inside user (caller).
The IPNext PBX which can be registered to interoperate with Trunk Gateway. The IPNext PBX has a direct access to VoIP or PSTN network, but it can be connected, through the other external device, such as Trunk Gateway. This external device is to be configured and registered to the Call Manager imbedded in the IPNext PBX with Trunk Gateway accordingly.

And when the inside user is to be connected with an external terminal by dialing a number, the one needs to configure Routing Pattern. For instance, when you call outside from you office, you need to dial a certain number (9 is very commonly used). This is to configure that number.
The following figure shows the screen for Trunk Gateway performing Properties.

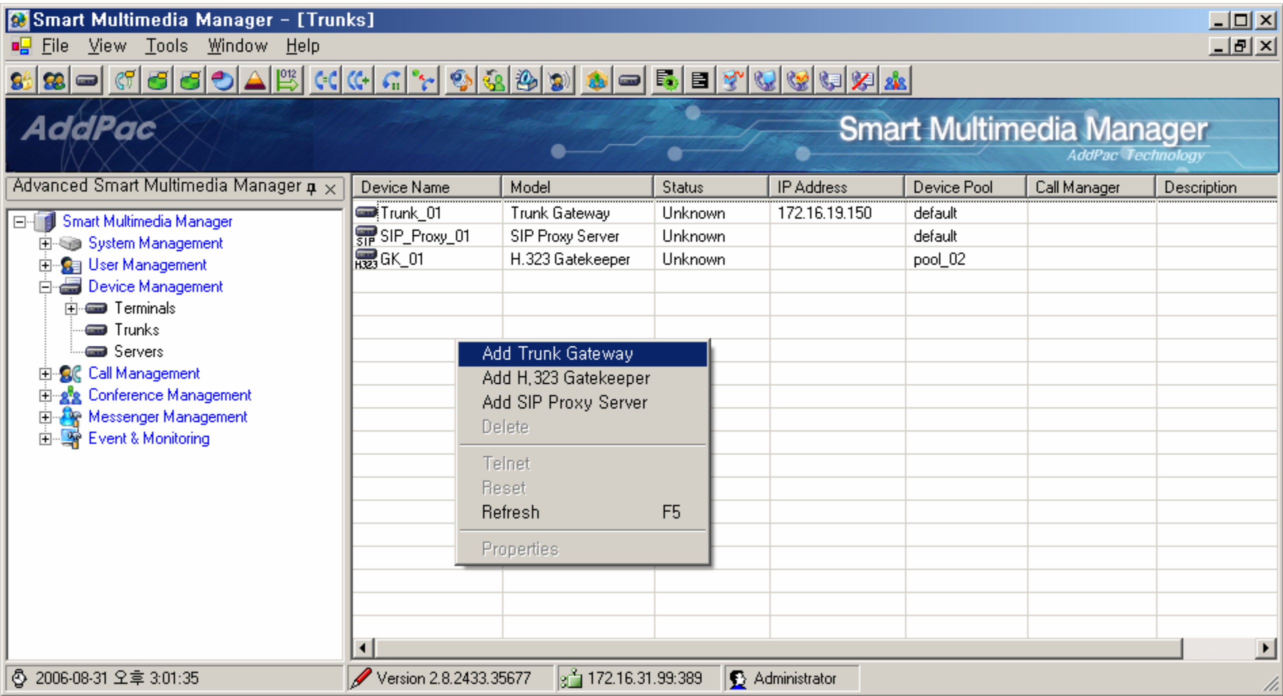

Figure 2-68 Trunks – Trunk Gateway Screen

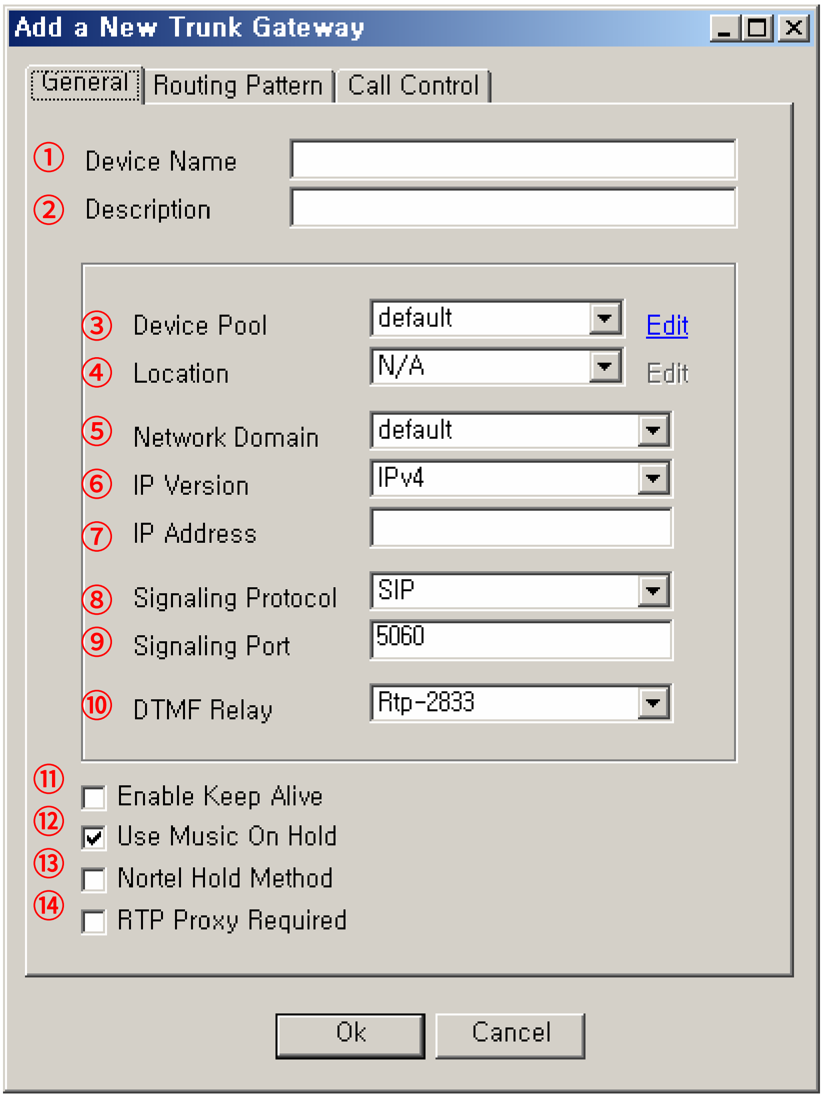

Figure 2-69 Trunk Gateway - General Properties Screen

Table 2-47 Trunk Gateway – Description of General Properties

Field	Description
1	Register a name of Trunk Gateway
2	Register a description of Trunk Gateway
3	Select Device Pool (basically set to default)
4	Select Location (basically no setting)
5	Select Network Domain of Call Manager
6	Select IP version
7	Register IP address
8	Select VoIP protocol
9	Enter a corresponding port to a protocol of VoIP
10	Select a type of DTMF-Relay to be transmitted to Trunk

	- Rtp-2833 : Transmit DTMF basin on RFC-2833 - Inband : Transmit Inband DTMF tone through RTP - Out-of-band : Transmit DTMF by using SIP INFO, H.245 Signal
11	Check Trunk Gateway status by using SIP PING periodically
12	Configure MOH to be provided for hold requested from a terminal
13	Configure MOH to be provided for interoperating with Nortel Softswitch
14	This is a forced setup to use RTP Proxy while the call is on line.

The following figure shows the list of Routing Pattern registered to Trunk Gateway:

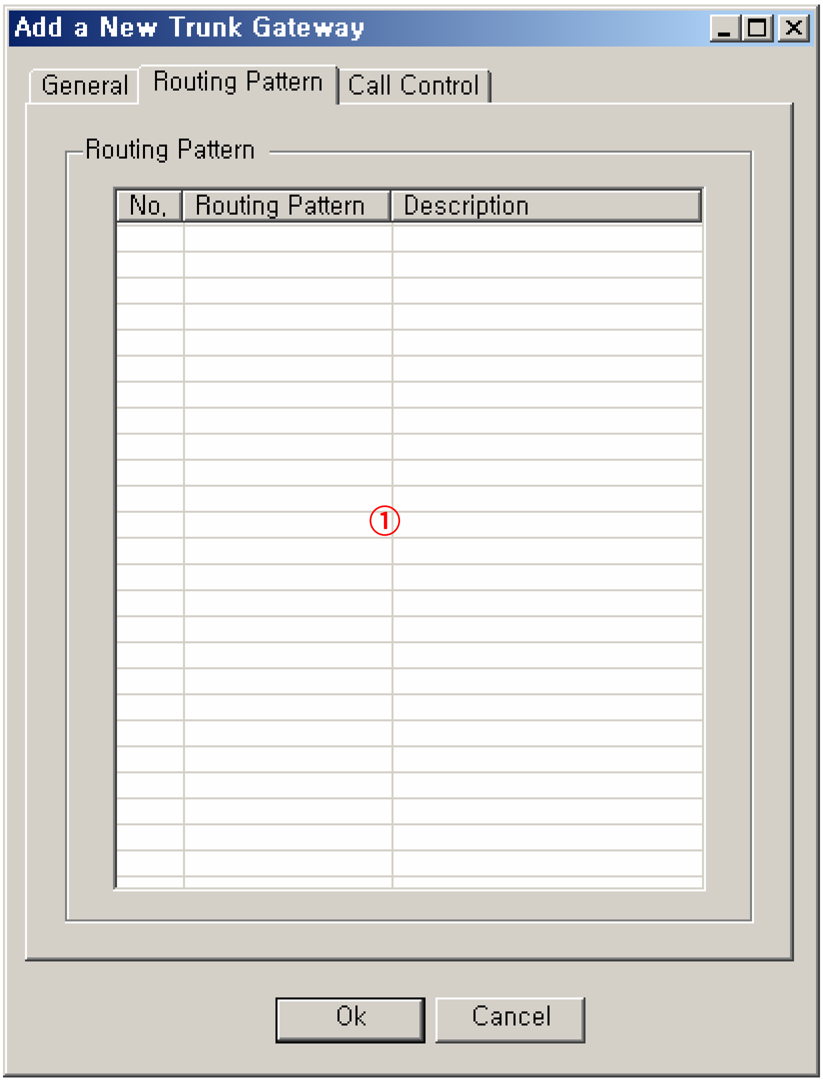

Figure 2-70 Trunk Gateway – Routing Pattern Properties

Table 2-48 Trunk Gateway - Routing Pattern Properties Description

Field	Description
1	Routing Pattern that belongs to Trunk Gateway can be executed and the user can see or edit Properties Pattern by selecting Routing Pattern

The following figure shows the screen for Call Control of Trunk Gateway:

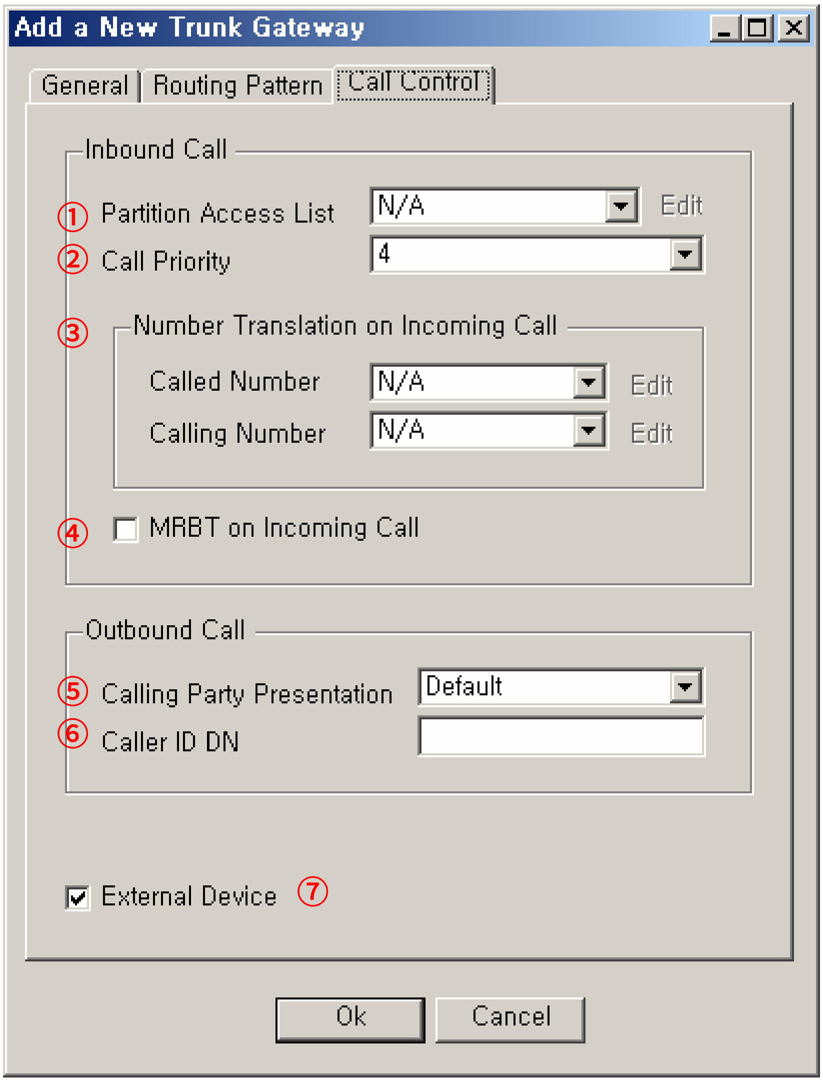

Figure 2-71 Trunk Gateway – Call Control Properties Screen

Table 2-49 Trunk Gateway - Call Control Properties

Field	Description
1	This is to configure Partition Access List which applies to this Device. When an inbound call from the Device is processed, the outbound call can be processed only for the terminal (or Trunk) of which number belongs to the Partition (or Pattern). When Partition Access List is set to 'N/A' for no configuration, any number with the property of Partition or the Pattern (any object not configured in Partition) can be routed.
2	Set the priority level for the Inbound call
3	Translation Rule is applied for the incoming and outgoing call to change the number for Inbound call
4	Choose an option whether to provide MRBT(Multi-media Ring Back Tone) or not for Inbound call
5	Choose an option whether to display the outgoing call for Outbound call Default: Fooling the Default of Call Manager to (Call Manager Cluster > Options > Calling Party Presentation)

	Allowed: Displaying a caller's telephone number to the other party Restricted: Not the caller's telephone number to the other party
6	This a forced setup to the outgoing call (for example: the general directory number)
7	Configure an option whether to connect to the outside is Trunk or not

H.323 Gatekeeper Configuration

This is a function to register Gatekeeper for connection to the outside.
For example, when you call outside from the inside of your company after registration of the telephone number the outside Gatekeeper, this is the function to make the call possible through the registered gatekeeper.

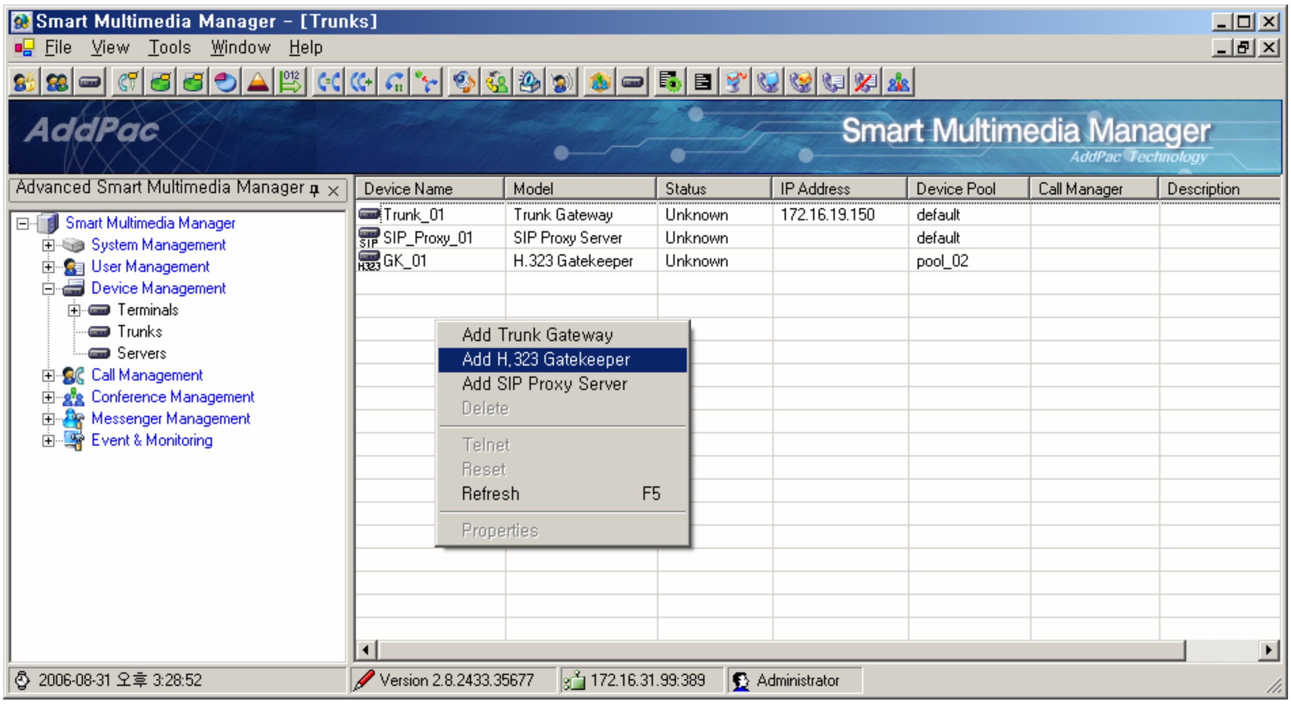

Figure 2-72 Trunks – H.323 Gatekeeper Screen

As the Figure above shows, you can select Add H.323 GatekeeperMenu in the following procedure:

- Move to Device Screen (Device Management > Trunks)
- Click the right button of the mouse to open the popup menu
- Select Add H.323 GatekeeperMenu

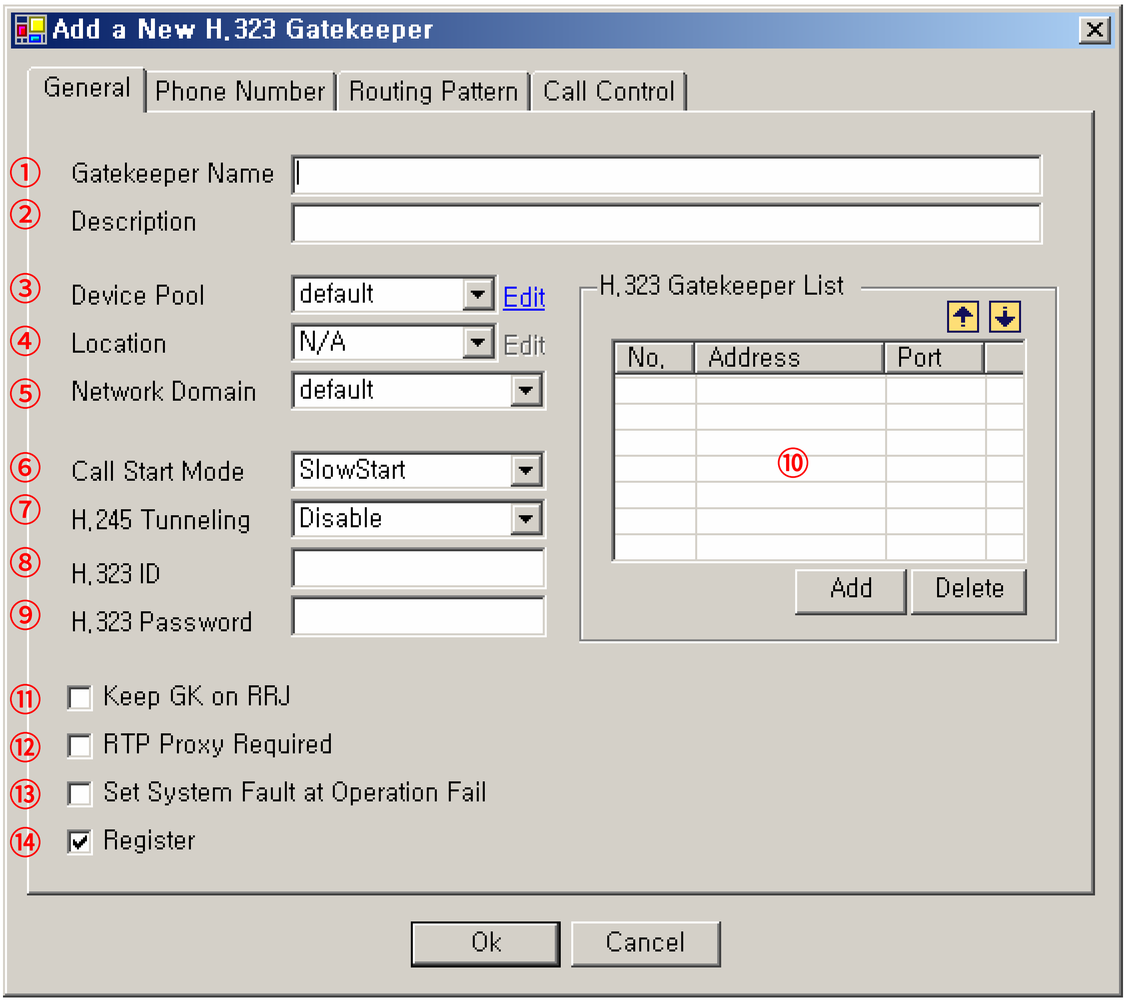

Figure 2-73 H.323 Gatekeeper - General Properties

Table 2-50 H.323 Gatekeeper – Description of General Properties

Field	Description
1	Register a name of Gatekeeper
2	Register a description of Gatekeeper
3	Select Device Pool
4	Select Location
5	Select Network Domain
6	Select a mode of Call Setup for H.323 (FastStart, SlowStart, PreferredSlow)
7	Enable or disable Tunneling mode of H.245
8	Register H.323 ID
9	Register H.323 Password
10	Register Gatekeeper IP address to the list then the priority level can be assigned
11	This is to configure to keep trying to register Gatekeeper repeatedly even when the registration is rejected.

12	This is a forced setup to use RTP Proxy while the call is on line.
13	Select an option whether to set Registration failure as the System Fault of Call Manager
14	Select an option whether Register a Phone Number for Gatekeeper registration

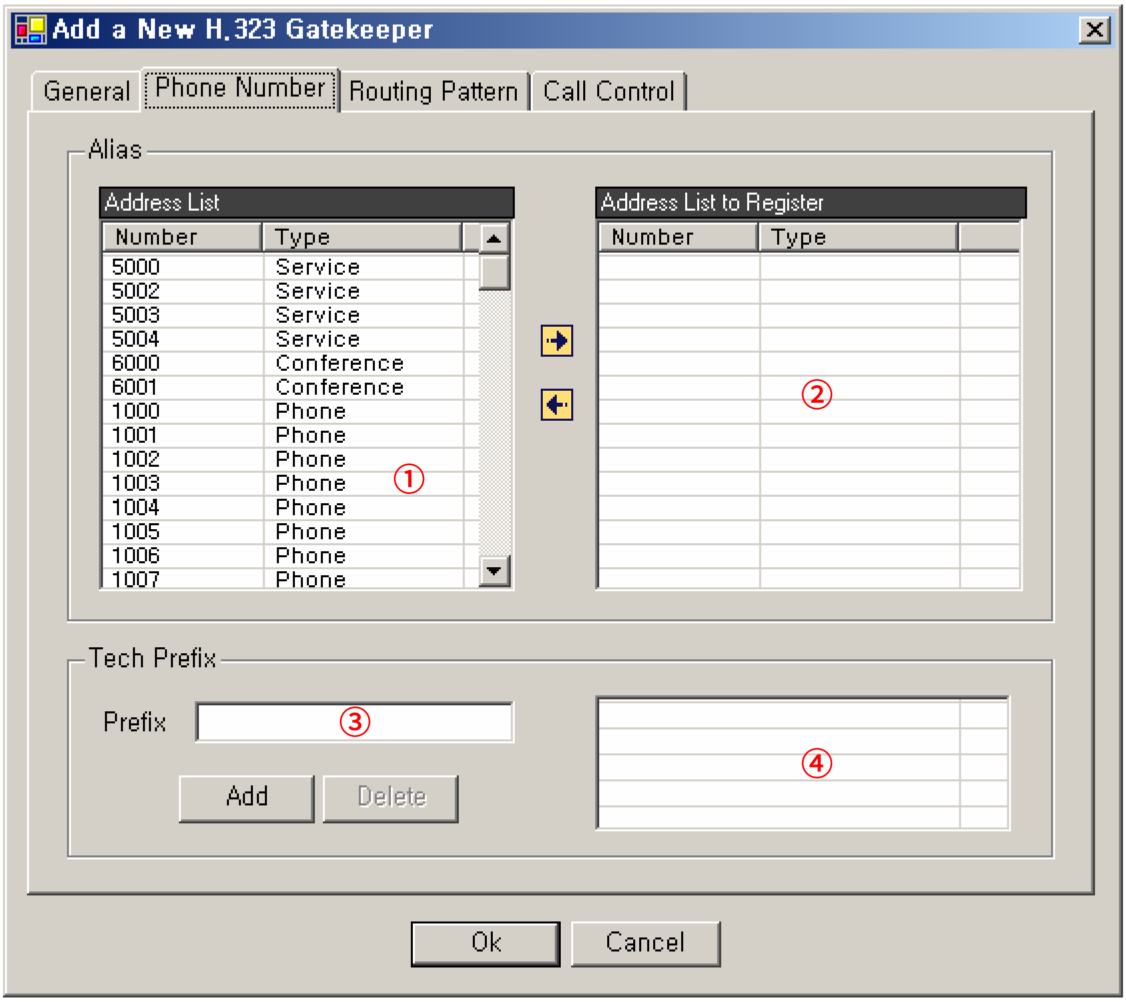

Figure 2-74 H.323 Gatekeeper – Phone Number Properties

Table 2-51 H.323 Gatekeeper – Description of Phone Number

Field	Description
1	This is the list of telephone numbers not registered to Gatekeeper
2	This is the list of the telephone numbers registered to Gatekeeper
3	Enter and register the Prefix for the telephone numbers for the registration of Gatekeeper
4	This is the list of the Prefix for the telephone numbers to be registered to Gatekeeper

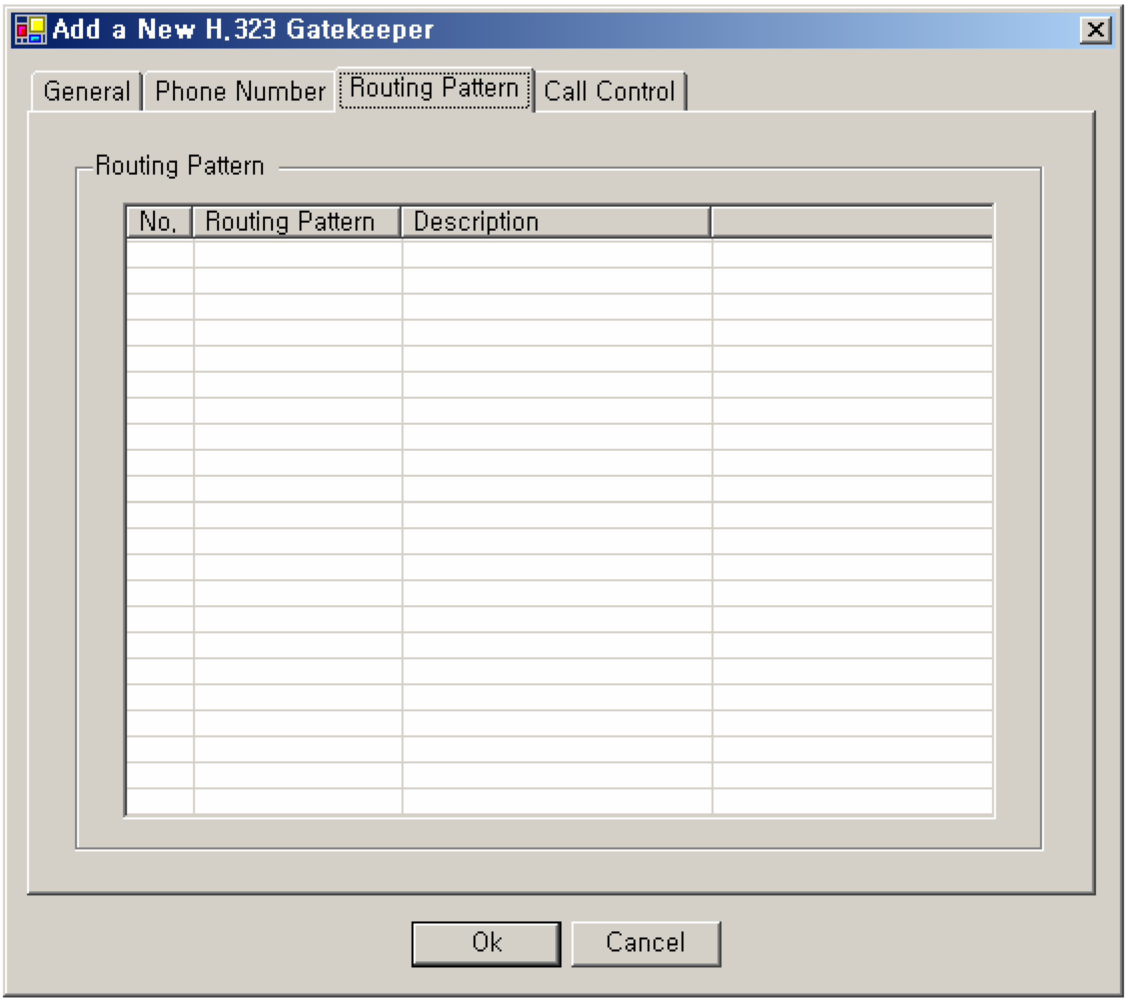

Figure 2-75 H.323 Gatekeeper – Routing Pattern Properties

Table 2-52 H.323 Gatekeeper – Description of Routing Pattern Properties

Field	Description
1	Routing Pattern that belongs to H.323 Gatekeeper can be executed and the user can see or edit Properties Pattern by selecting Routing Pattern

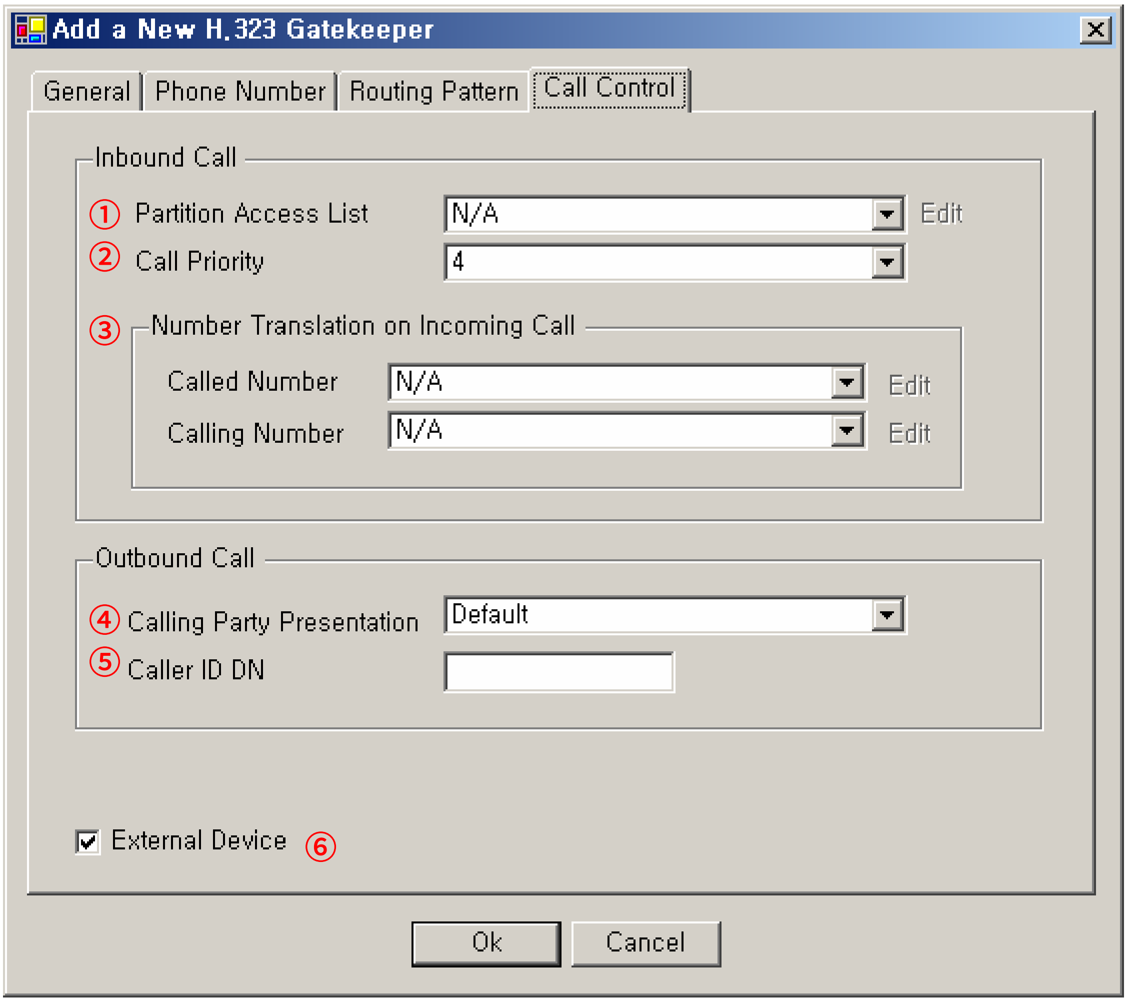

Figure 2-76 H.323 Gatekeeper – Call Control Properties

Table 2-53 H.323 Gatekeeper – Description of Call Control Properties

Field	Description
1	This is to configure Partition Access List which applies to this Device. When an inbound call from the Device is processed, the outbound call can be processed only for the terminal (or Trunk) of which number belongs to the Partition (or Pattern). When Partition Access List is set to 'N/A' for no configuration, any number with the property of Partition or the Pattern (any object not configured in Partition) can be routed.
2	Set the priority level for the Inbound call
3	Translation Rule is applied for the incoming and outgoing call to change the number for Inbound call
4	Choose an option whether to provide MRBT(Multi-media Ring Back Tone) for Inbound call or not

5	Choose an option whether to display the outgoing call for Outbound call Default: Fooling the Default of Call Manager to (Call Manager Cluster > Options > Calling Party Presentation) Allowed: Displaying a caller’s telephone number to the other party Restricted: Not the caller’s telephone number to the other party
6	This a forced setup to the outgoing call (for example: general directory number)

SIP Proxy Server Configuration

This is a function to register an external SIP Proxy Server for interoperation.

For instance, when you call outside through an external SIP Proxy Server to which a telephone number of your company is registered, this function allows you to make the call through the registered SIP Proxy Server.

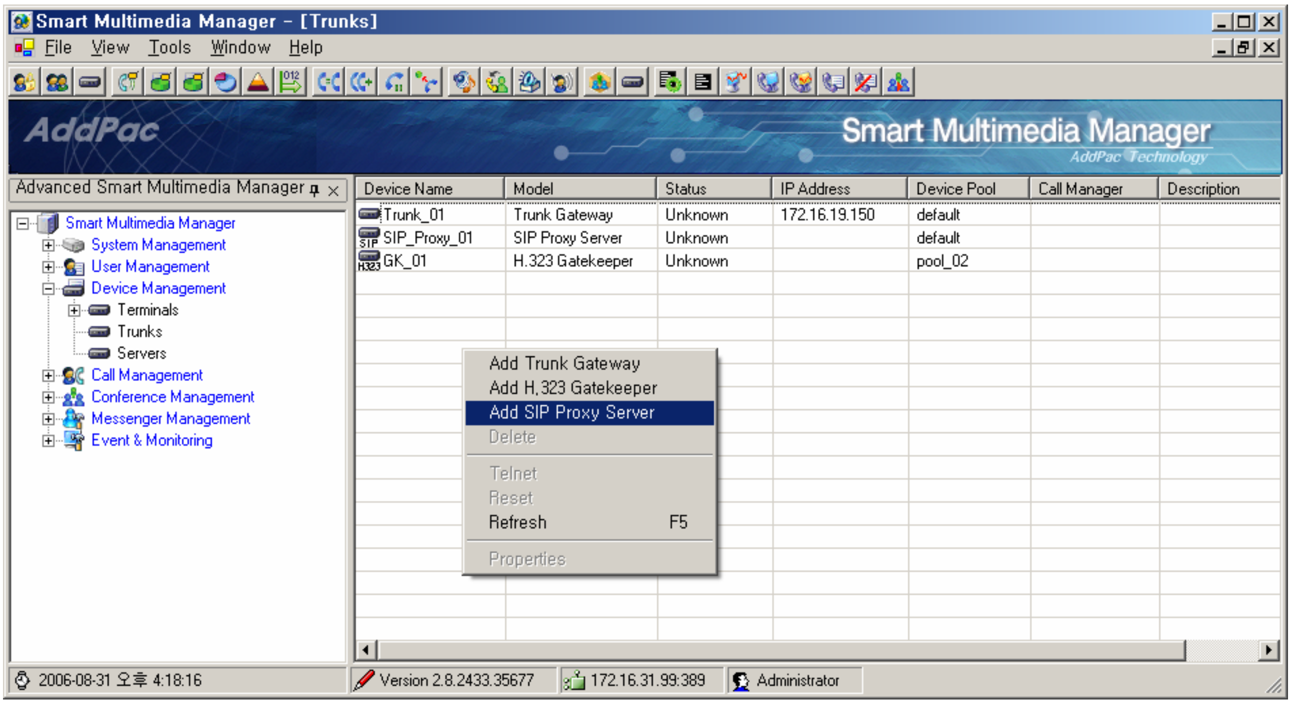

Figure 2-77 Trunks – Screen for SIP Proxy Server Registration

As the Figure above shows, you can select Add SIP Proxy ServerMenu in the following procedure:

- Move to Device Screen (Device Management > Trunks)
- Click the right button of the mouse to open the popup menu
- Select Add SIP Proxy ServerMenu

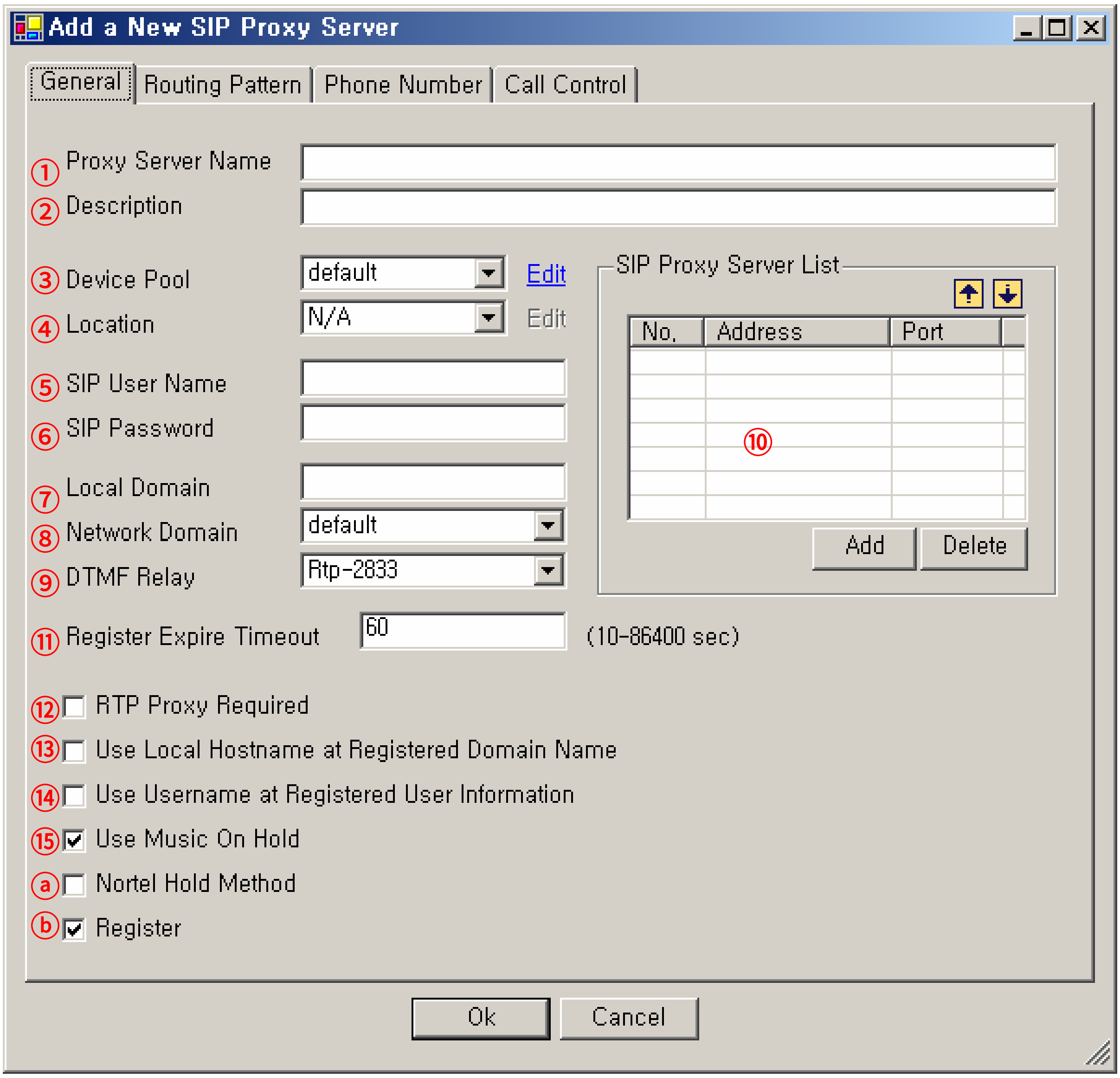

Figure 2-78 SIP Proxy Server – General Properties

Table 2-54 SIP Proxy Server – General Properties Description

Field	Description
1	Register a name of SIP Proxy Server
2	Register a description of SIP Proxy Server
3	Select Device Pool (basically set to default)
4	Select Location
5	Register a User Name for accessing to SIP Proxy Server
6	Register a password for accessing to SIP Proxy Server
7	Register Local domain
8	Register Network Domain
9	Select a type of DTMF-Relay to be transmitted to Trunk - Rtp-2833 : Transmit DTMF based on RFC-2833 standard

	- Inband : Transmit Inband DTMF tone through RTP - Out-of-band : Transmit DTMF by using H.245 Signal. SIP INFO
10	Set IP address and port of SIP Proxy Server to the list and the priority level can be placed
11	Set expiration time for registering SIP Proxy Server
12	This is a forced setup to use RTP Proxy while the call is on line
13	Use local hostname instead of domain name for Proxy Server to be registered To : 7000@local hostname From : 7000@local hostname
14	PRegister username instead of the number e164 for the information to be registered to Proxy Server To : jschoi@172.16.32.40 From : jschoi@172.16.32.40 Contact : jschoi@172.16.32.70
15	This is the configuration to provide MOH for ‘hold’ requested from the terminal
A	This is the configuration to provide MOH while it is connected to Nortel softswitch
B	Configure an option whether to register Phone Number to Proxy Server or not

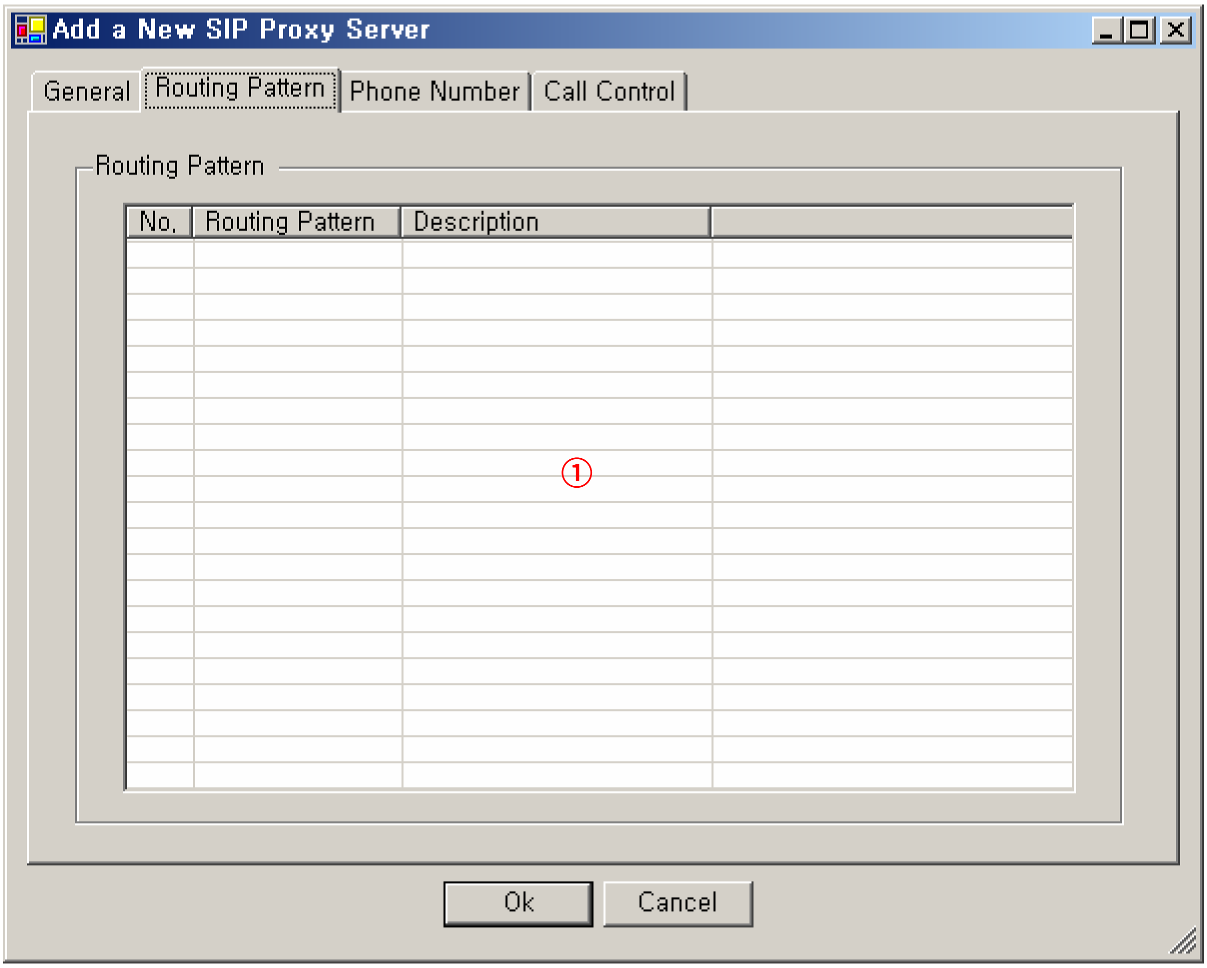

Figure 2-79 SIP Proxy Server – Routing Pattern Properties

Table 2-55 SIP Proxy Server – Description of Routing Pattern Properties

Field	Description
1	Execute Routing Pattern which applies to SIP Proxy Server. Select Routing Pattern then Properties can be shown or edited

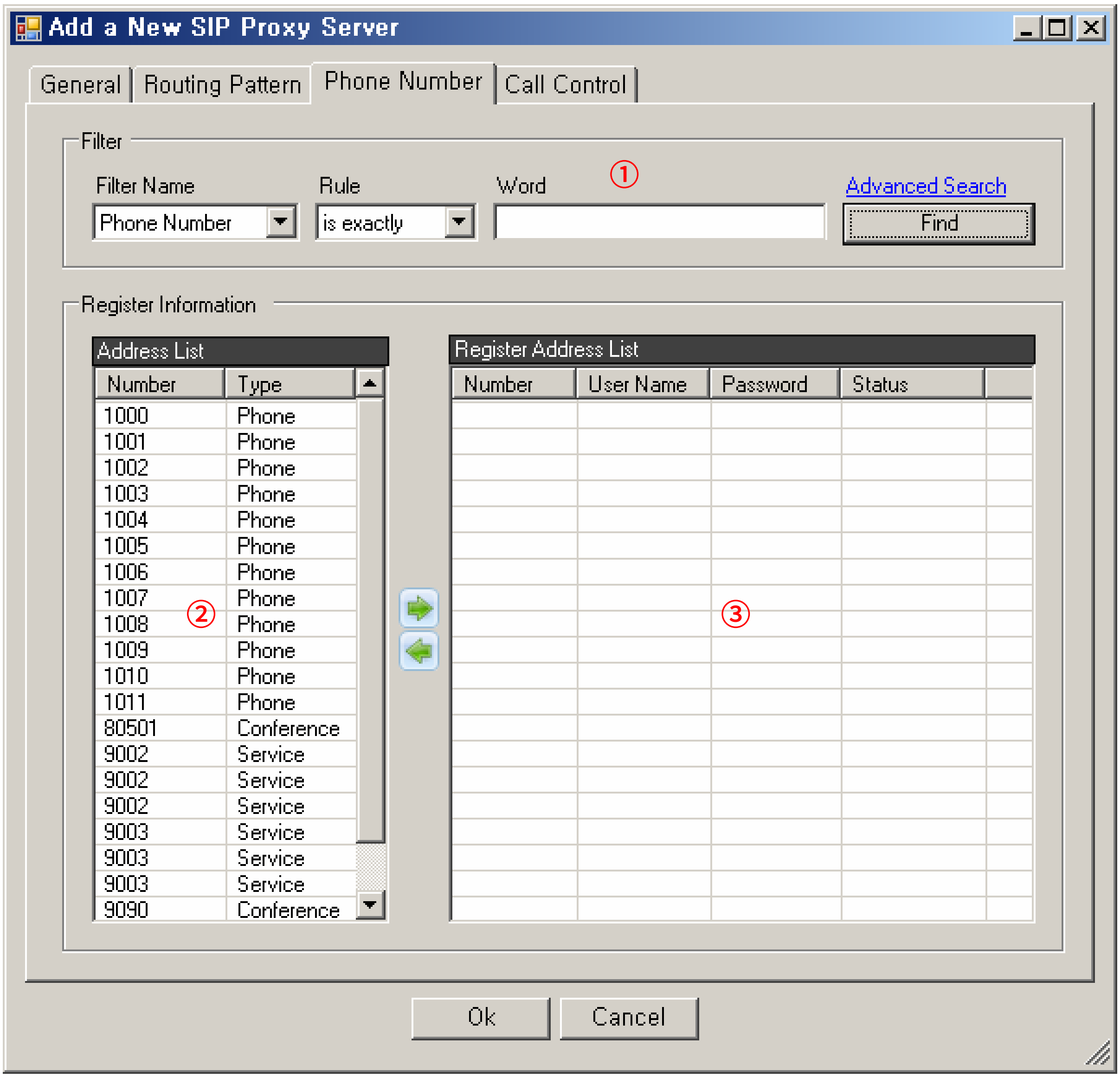

Figure 2-80 SIP Proxy Server – Phone Number Properties

Table 2-56 H.323 Gatekeeper – Phone Number Properties Description

Field	Description
1	Search the phone number list to be registered to SIP Proxy Server
2	This is the list of telephone numbers that are not registered to SIP Proxy Server

3	This is the list of the telephone numbers to be registered to SIP Proxy Server ● User Name: Register a user's name for authentication of the telephone number ● Password: Register a password for authentication of the telephone number ● Status: Registration status of the telephone number

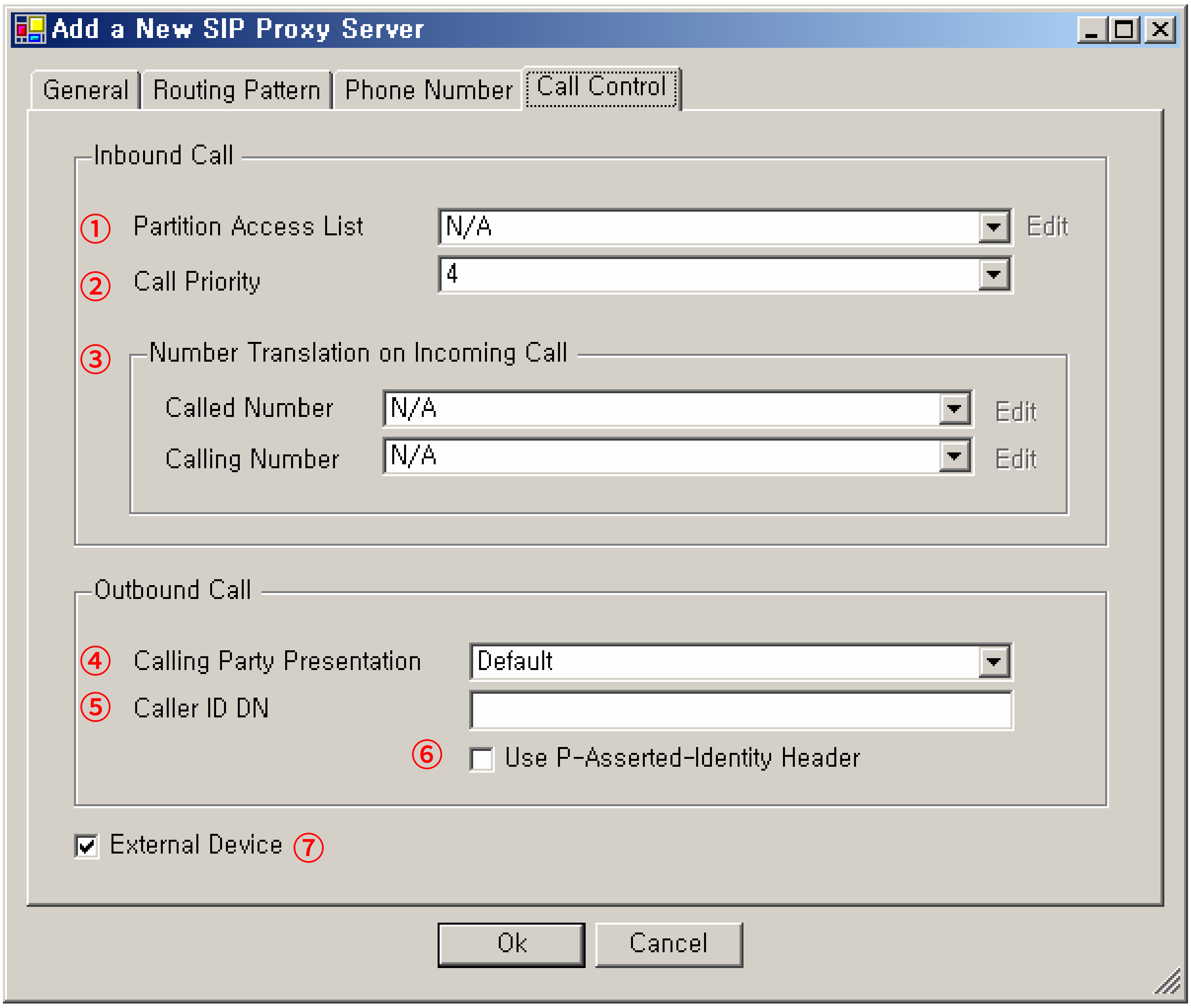

Figure 2-81 SIP Proxy Server – Call Control Properties

Table 2-57 SIP Proxy Server – Call Control Properties Description

Field	Description
1	Configure Partition Access List which applies to this Proxy Server. When an inbound call from the server is processed, the outbound call can be processed only for the terminal (or Trunk) of which number belongs to the Partition (or Pattern).

	When Partition Access List is set to 'N/A' for no configuration, the call is to be routed to any number with the property of Partition or the Pattern (any object not configured in Partition).
2	Set the priority level for the Inbound call
3	Translation Rule is applied for the incoming and outgoing call to change the number for Inbound call
4	Choose an option whether to display the outgoing call for Outbound call Default: Fooling the Default of Call Manager to (Call Manager Cluster > Options > Calling Party Presentation) Allowed: Displaying a caller's telephone number to the other party Restricted: Not the caller's telephone number to the other party
5	This a forced setup to the outgoing call (for example: general directory number)
6	Select an option whether to use P-Asserted-Identity Header or not
7	Select an option whether Trunk is to connect with an external device or not

Servers

MCU Server Configuration

As MCU Server is connected to IPNext PBX, it is to register all the other MCU servers (MCUServer), which can be used for the remote application.

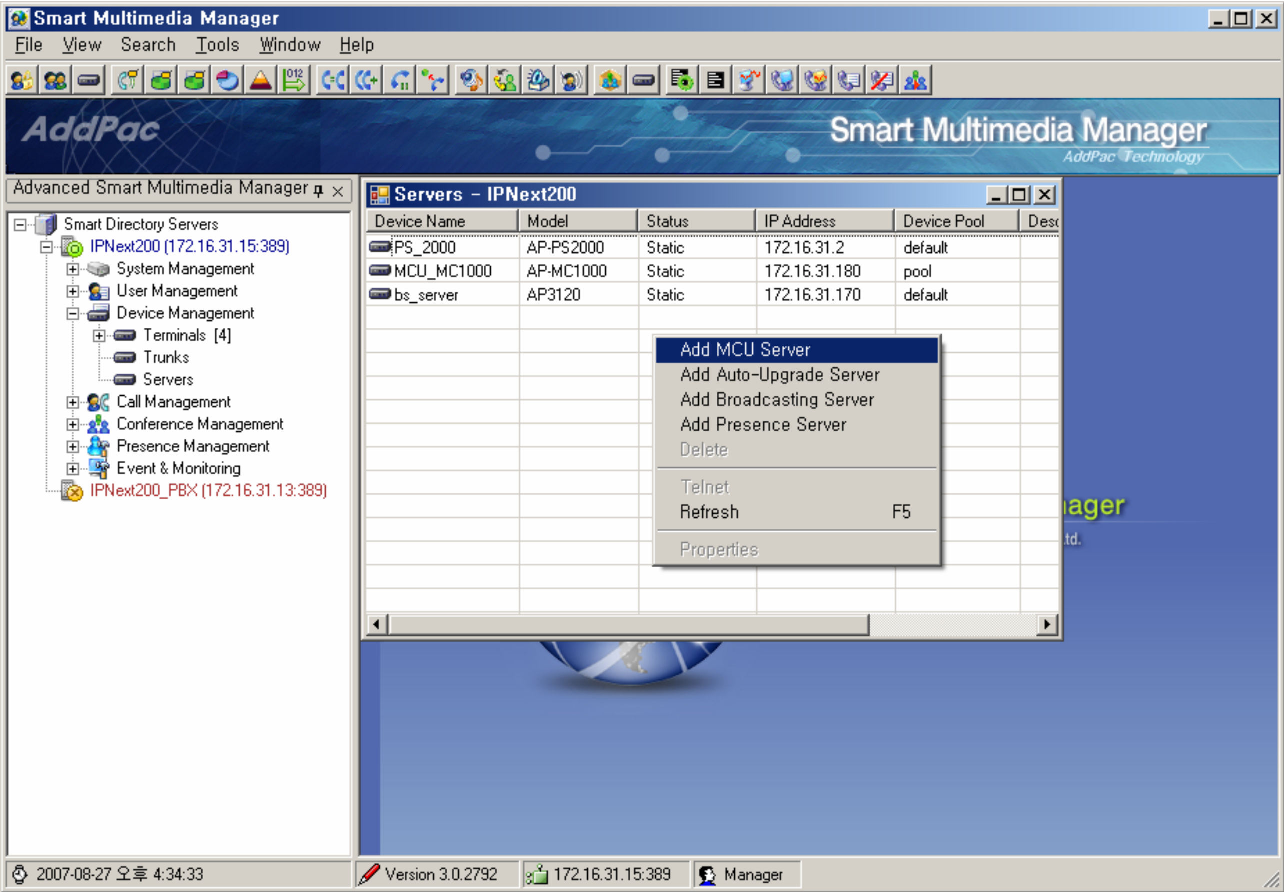

Figure 2-82 Servers – MCU Server Registration

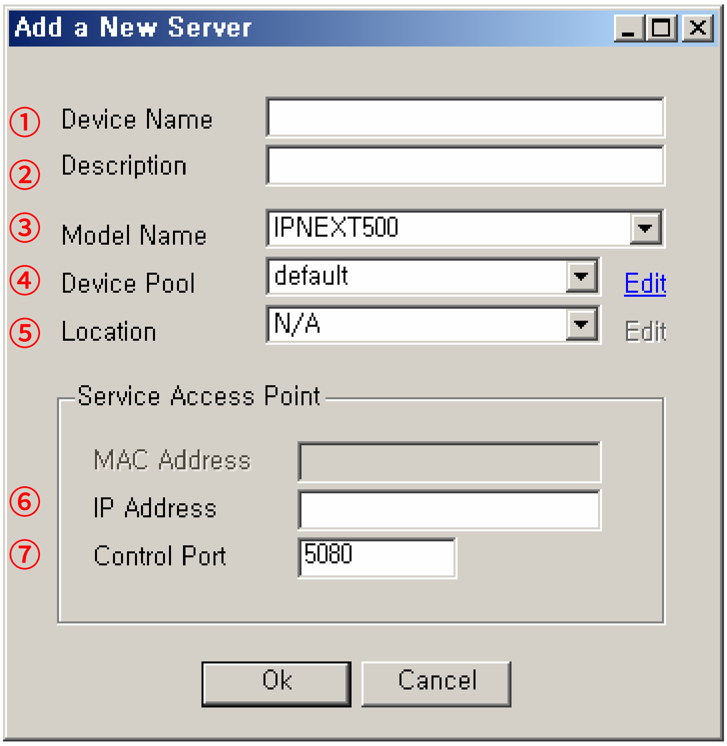

Figure 2-83 Servers – MCU Server Properties

Table 2-58 Servers – Description of MCU Server Properties

Field	Description
1	Register a name of the server
2	Register a description of the server
3	Select the server model
4	Select Device Pool
5	Select Location
6	Register IP address of the server
7	Register a control port to communicate with server and IPNext PBX

Auto Upgrade Server Configuration

Auto Upgrade Server is a function to register the servers for upgrading the firmware and programs applied to all the terminals or the messenger which are registered to IPNext PBX.

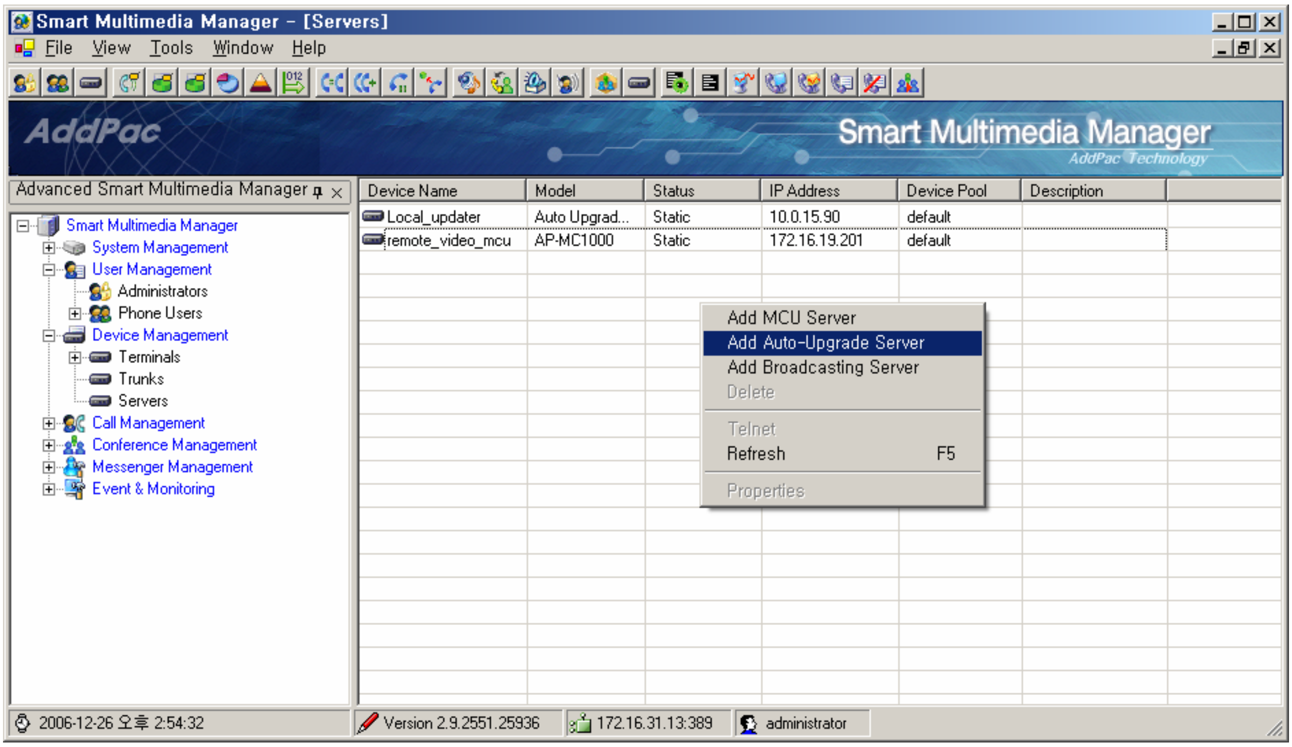

Figure 2-84 Servers – Auto Upgrade Server Registration

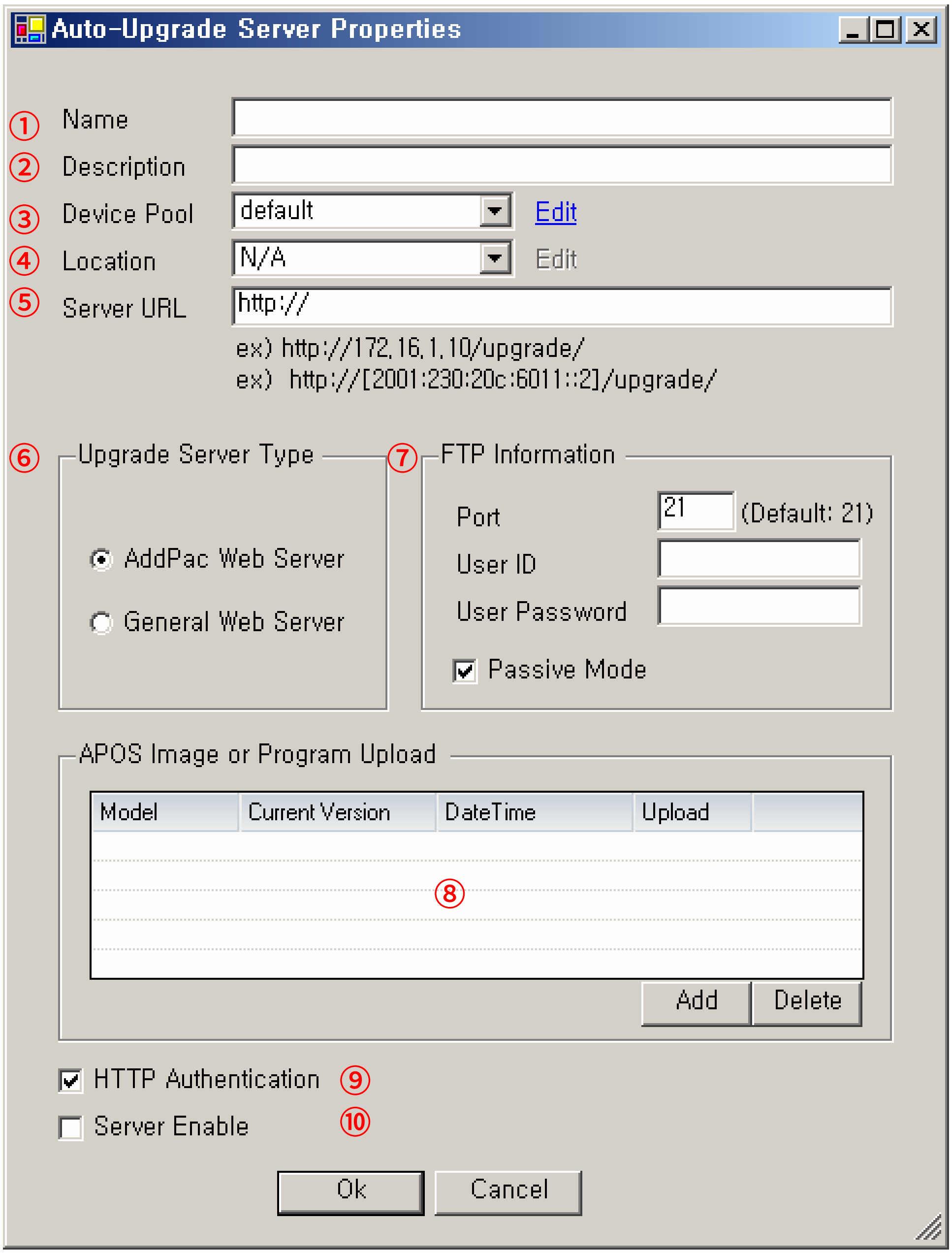

Figure 2-85 Auto Upgrade Server Properties

Table 2-59 Auto Upgrade Server Properties Description

Field	Description
1	Register a server name
2	Register a description of the sever
3	Select a Device Pool
4	Select a Location
5	Set URL of the server to be upgraded Ex) http://172.16.1.10/upgrade/ http://[2001:230:20c:6011::2]/upgrade/

6	Select a type of server to be upgraded (AddPac Webserver or a general webserver)
7	Enter FTP access information of the server to be upgraded (FTP access port, account, password, passive mode)
8	Register the product model to be upgraded, then upload the firmware by Auto Upgrade Server In this list, the product models, the current program version and the latest time of upload are to be displayed.
9	Select an option whether use HTTP authentication when to access Auto Upgrade Server
10	Select an option whether to enble Ato Upgrde Serve or not

In order to register the product model, you can have the following window displayed for the registration when you click 'Add' button from the template of 'APOS Image or Program Uploaded'

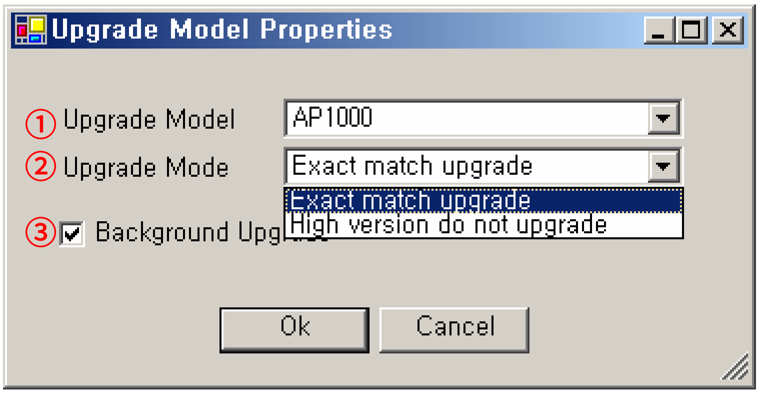

Figure 2-86 Upgrade Model Properties

Table 2-60 Description of Auto Upgrade Server Properties

Field	Description
1	Select the product model to be upgraded
2	Select Upgrade Mode Exact match upgrade: Upgrade all the terminals in operation to the current version High version do not upgrade: Upgrade all the terminals in operation to the previous version
3	Select an option whether to perform the upgrade after logging in terminals or not

After the registration is completed as shown below Figure, you can click 'Upload' button to perform registration of a firmware in another window.

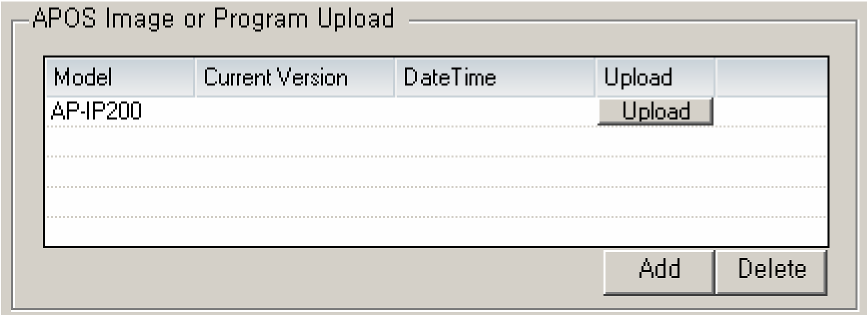

Figure 2-87 Registering a Product Model to Auto Upgrade Server

When the window is displayed, you can click ‘Open’ to add the firmware that matches to the product model and add it to the list. Then the version of the program is displayed. If the wrong file is registered, you (the administrator) need to make a correction. Now you can click ‘Upload’ on the template below, the register file is uploaded to Auto Upgrade Server. After the job is finished, the window closes.

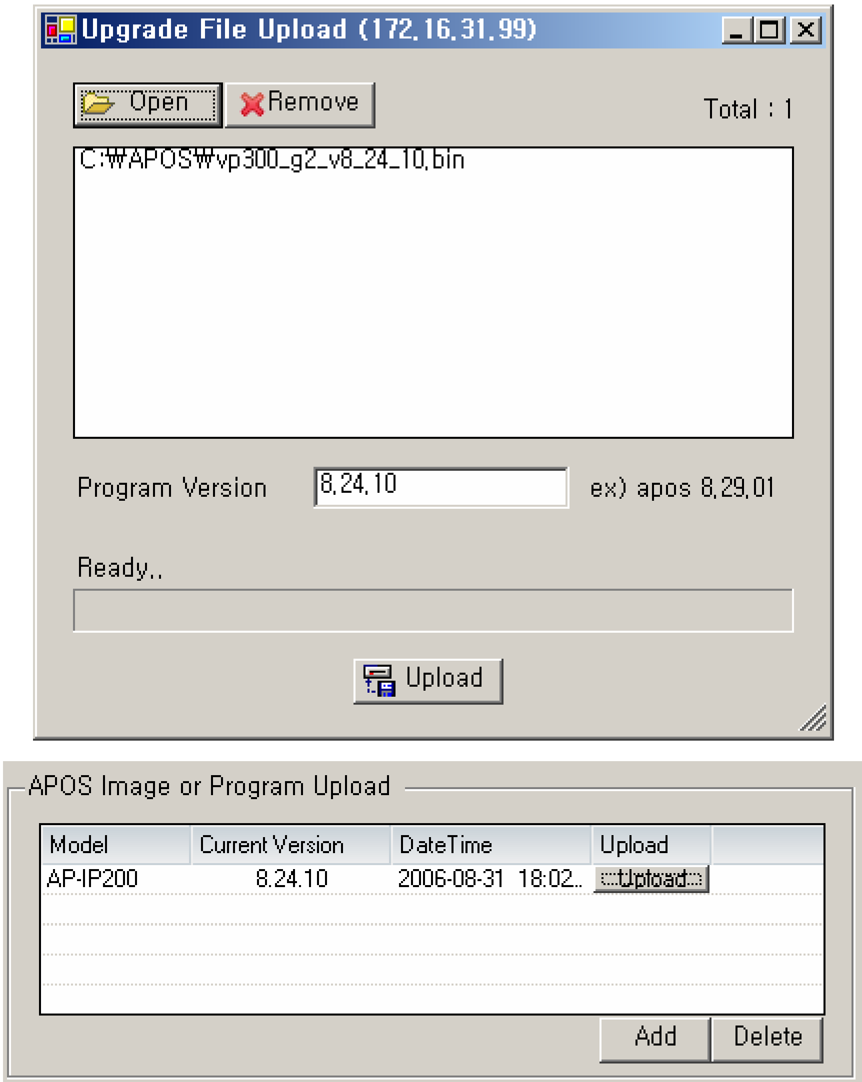

Figure 2-88 Upload of Firmware and Program

After the registration of the product model is finished, you may click 'OK' button on Auto Upgrade ServerProperties, then Auto Upgrade Server Configuration is completed.

Configuring Broadcasting Server

Broadcasting Server is to perform Virtual Audience function of Conference.

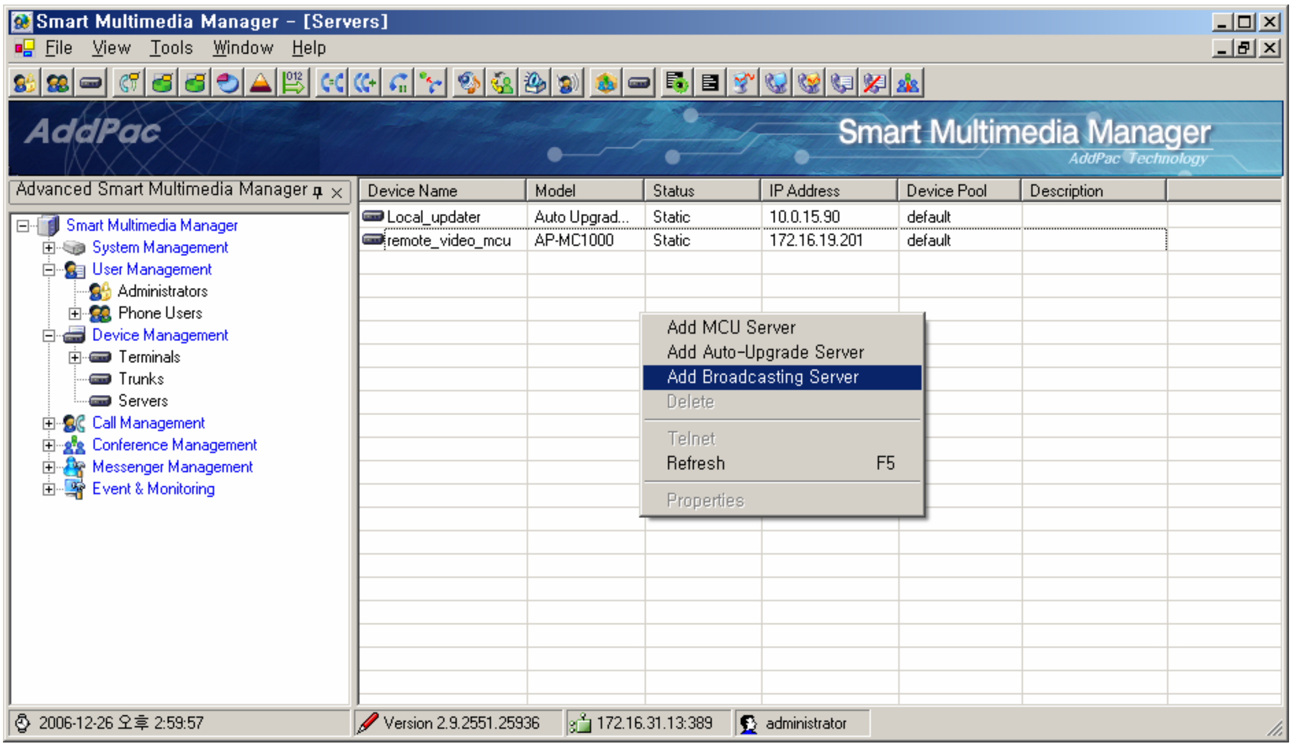

Figure 2-89 Servers – Broadcasting Server Registration

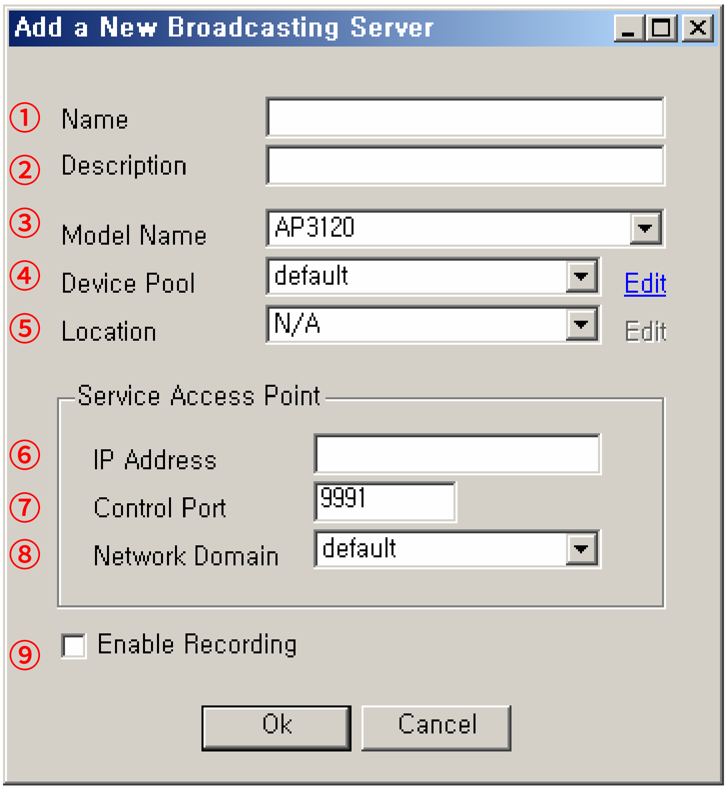

Figure 2-90 Servers – Broadcasting Server Properties

Table 2-61 Servers – Description of Broadcasting Server Properties

Field	Description
1	Register a name of the broadcasting server
2	Register a description of the broadcasting server
3	Select a product mode of the broadcasting server
4	Select Device Pool
5	Select Location
6	Register an IP address of the broadcasting server
7	Register a control port to communicate Server and IPNext PBX
8	Select a Network Domain
9	Record video or audio when Broadcasting Server interoperated with MCU

Configuring Presence Server

Presence Server는 Presence 서비스를 위한 서버 장비로서 IP-PBX와 연동되어 MS Window 기반의 PC 플랫폼 환경에서 운영되는 Smart Messenger 프로그램을 통해 사용자 Presence 서비스를 제공합니다. 사용자 Presence기능은 사용자가 On-line 상태에 있는지, 외출 중인지, 통화중인지 등을 표시하는 기능으로 Presence Server는 각 End-Point 에 있는 단말로부터 수집된 정보를 그룹별 또는 전체 IP 단말의 Smart Messenger 프로그램으로 브로드캐스팅하는 기능을 수행합니다 Presence Server provides user presence service, as an external server, through Smart Messenger program which is MS Window-based PC platform and interoperates with IPNext PBX. Presence Server provides the following functions:

- Indicating the user status: whether user is on-line, out of office or busy on line
- Collecting the information from the terminals located in each end point.
- Broadcasting the information to each group or the entire IP terminals of Smart Messenger program

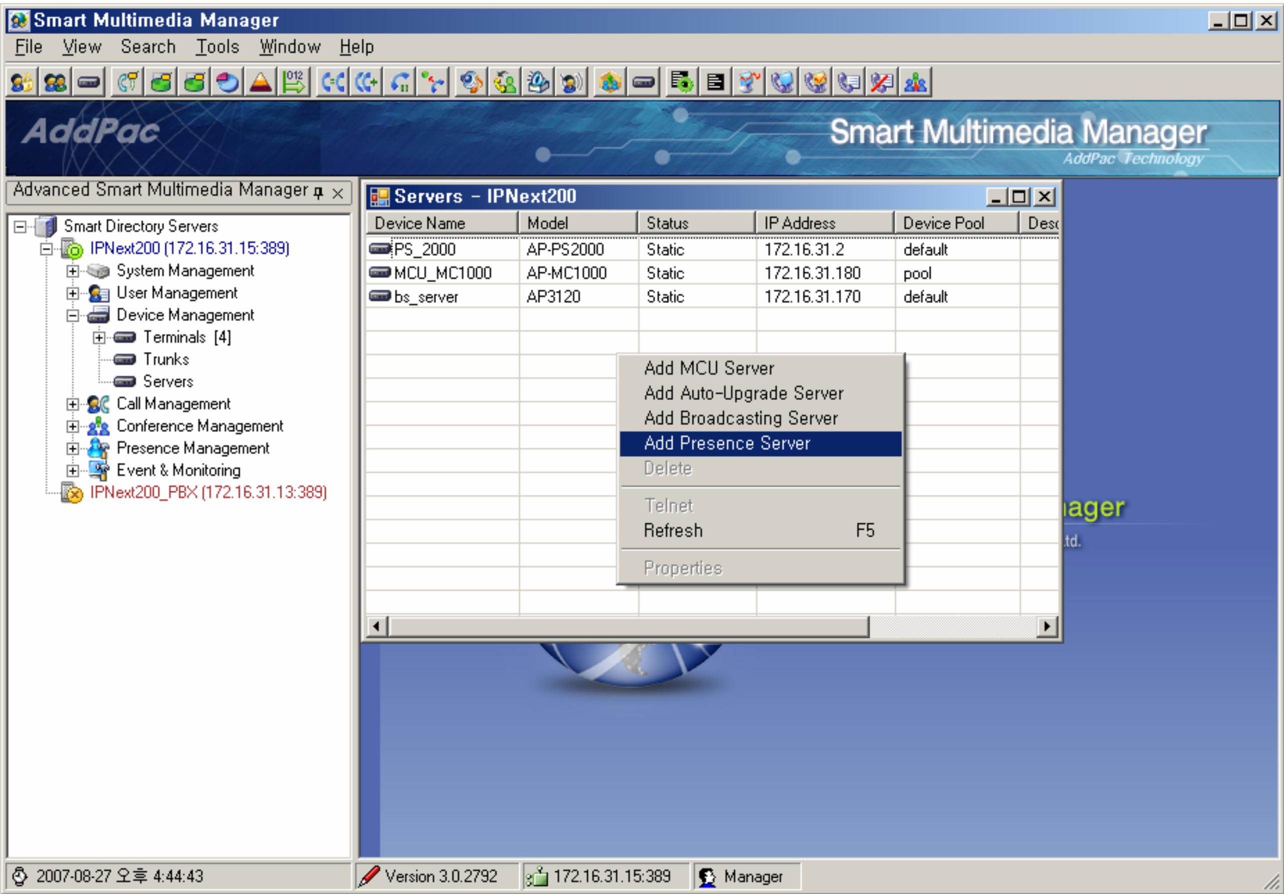

Figure 2-91 Servers – Presence Server Registration

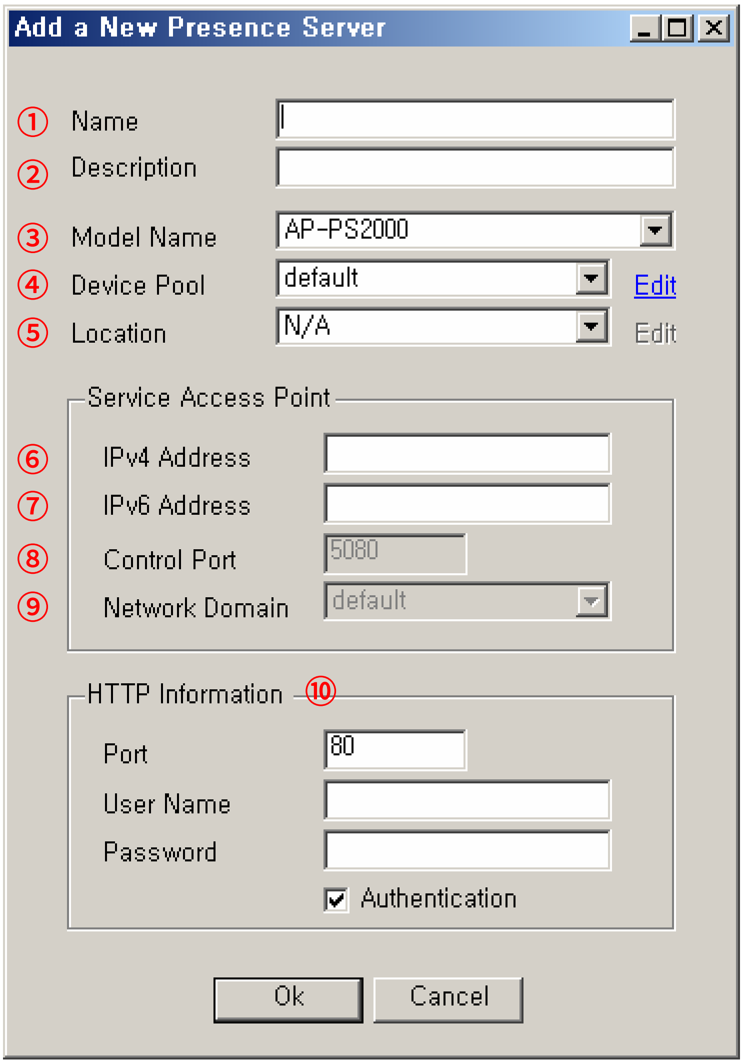

Figure 2-92 Presence Server Properties

Table 2-62 Description of Presence Server Properties

No	Description
1	Register a name of Presence Server
2	Register a description of Presence Server
3	Select a product model
4	Select Device Pool
5	Select Location
6	Presence 서버 Register IPv4 address of Presence Server
7	Register IPv6 address of Presence Server
8	Register a control port for communication between IPNext PBX and the server (not used in this figure)
9	Select a network domain (not used in this figure)
10	Set up HTTP access information of Presence Server - Port : port number - User name : user ID for authentication

- Password : password for authentication
- Authentication: Autheticate

Call Management – Dial Plan

Phone Number Configuration

Phone Number is to configure the telephone number which is used by a user and registered to IPNext PBX. A single telephone number or a rangemany telephone numbers can be created under the same condition (such as Partition, Partition Access List, Pickup Group, Park Group).

The telephone number must be unique in the same Partition. The same number can be used in other Partitions and this is appropriate when Centrex is configured.

Phone Number performs as Call Management > DialPlan > Phone Number on the left tree Menu

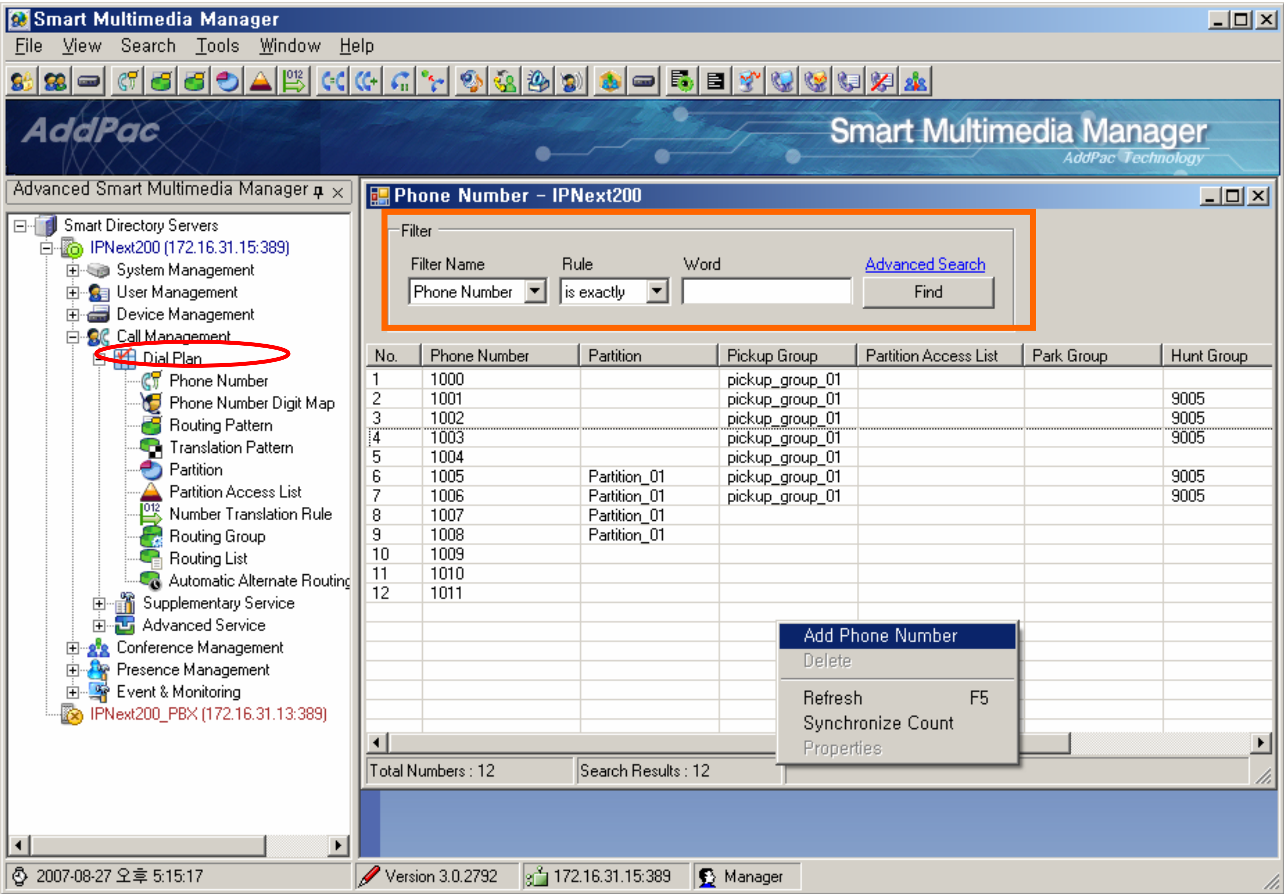

Figure 2-93 Phone Number Screen

When the number is configured, the details can be categorized as Inbound Configuration, Outbound Configuration and Forwarding Configuration.

When IPNext PBX processes a call, the call is sent out from a terminal or an external Trunk for the Inbound process and the call is received by the terminal or external Trunk for the Outbound process.

During the configuration of the telephone number, the numbers in the Inbound category are to be applied to the Inbound process and the numbers in the Outbound category are to be applied to the Outbound process. During the Outbound process of the internal terminals, many different call features are applied in the order of Call Waiting, Call Forwarding Busy, Hunting (when the individual Call Waiting, CFB are set to be disregarded), Voice Mail

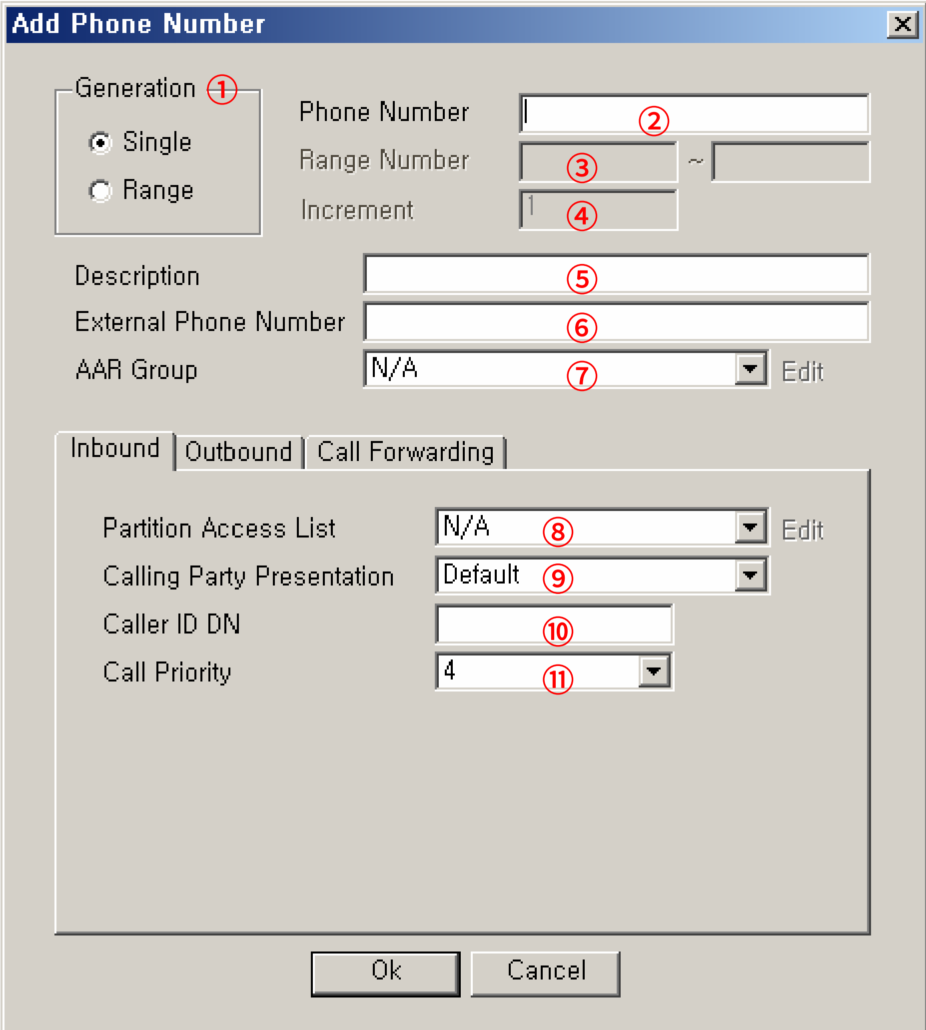

Figure 2-94 Phone Number Properties-Inbound

Table 2-63 Description of Phone Number Properties

Field	Description
1	Select an option whether to register a single or range of many telephone numbers
2	Register a single telephone number
3	Register a range of telephone numbers
4	(Range) Set an interval of creating each number when the range for the numbers are created.
5	Register a description of the telephone number
6	Register external PSTN phone number to the directory. AAR 이 적용 시에 사용되며 본 전화번호로 착 AAR reroutes the incoming call of the phone number to External Phone Number.

7	Select Automatic Alternate Routing (AAR) Group
8	Set up the Partition to be applied to this number. When the terminal sends out the call and process and the call is processed from the Terminal as an Inbound call, the routing is possible only when the Partition is included in the Partition list of the terminal or Trunk When the Partition is set to 'N/A' for no configuration, the access is allowed to the all. In other words, the access is possible from any terminal or Trunk with Partition Access List.
9	Choose an option whether to display the outgoing call for Outbound call Default: Fooling the Default of Call Manager to (Call Manager Cluster > Options > Calling Party Presentation) Allowed: Displaying a caller's telephone number to the other party Restricted: Not the caller's telephone number to the other party
10	This a forced setup to the outgoing call (for example: general directory number)
11	Set the priority level for the Inbound call

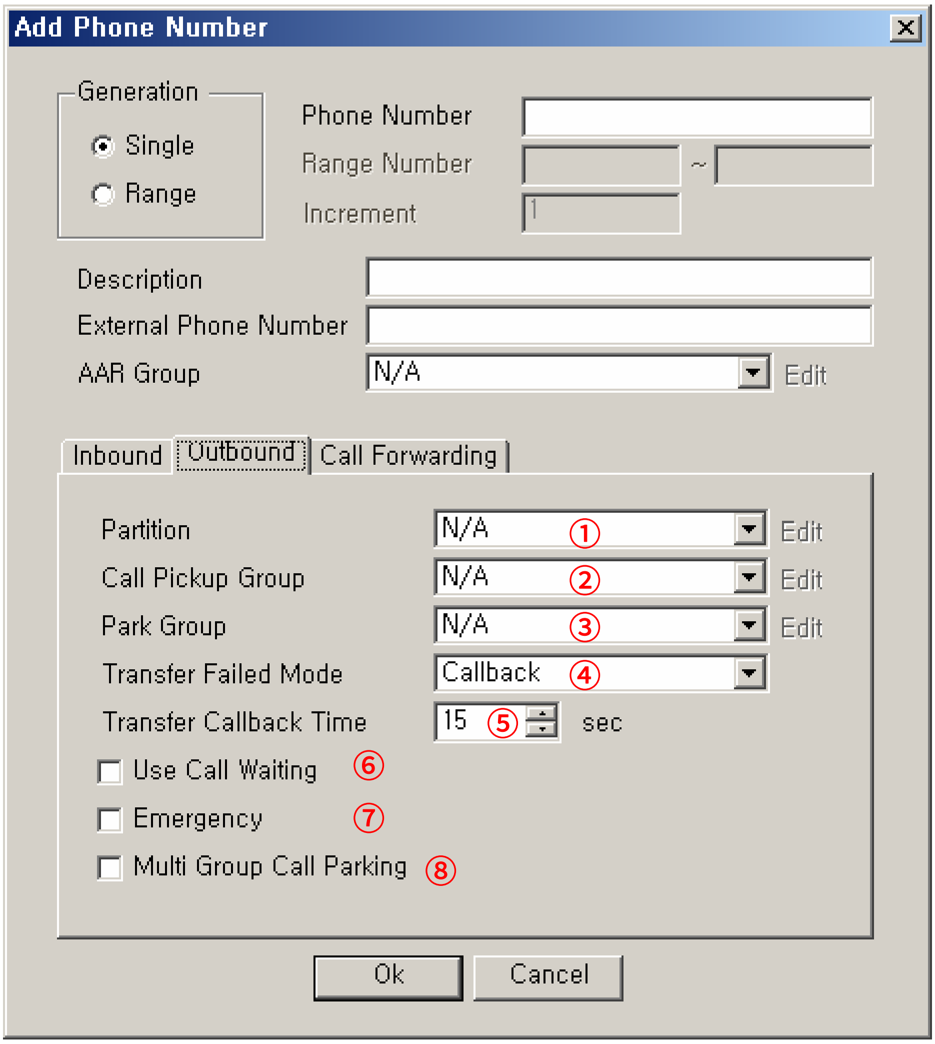

Figure 2-95 Phone Number – Outbound Properties

Table 2-64 Phone Number – Description of Outbound Properties

No.	Description
1	Set up the Partition to be applied to this number. When the terminal sends out the call and process and the call is processed from the Terminal as an Inbound call, the routing is possible only when the Partition is included in the Partition list of the terminal or Trunk When the Partition is set to 'N/A' for no configuration, the access is allowed to the all. In other words, the access is possible from any terminal or Trunk with the Partition Access List.
2	Set the phone numbers to be used for Pick Group When Call Pickup Group is set to 'N/A', the phone number does not belong to any group.
3	Set the phone number to be used for Park Group (optional). When Park Group is set to 'N/A', the phone number does not belong to any group.
4	When Call Transfer fails. set the reactions to the failure: - Announcement: Play announcement for the failure and terminate the call

	- Callback : Reconnect to the caller who requested for the transfer
5	The time for waiting till call transfer is connected. After Transfer Call Back Time, the transfer is treated as failed and processed depending on Transfer Failed Mode.
6	Enable/ disable Call Waiting When the telephone number is on another line, Call Waiting provides an audio alert to the phone user that the incoming call is arrived. When the user presses Hook Flash to place an hold to the call which is presently on the line and connects to the new phone call. When Call Waiting is enabled, Calling Waiting takes precedence over Call Forwarding Busy.
7	Emergency takes the highest priority over all calls.
8	Select an option whenther to enable or disable Multi Group Call Parking function (select the option to be enabled). In other words, this is an option whether to provision Call Parking function to the corresponding telephone number for other Park Groups

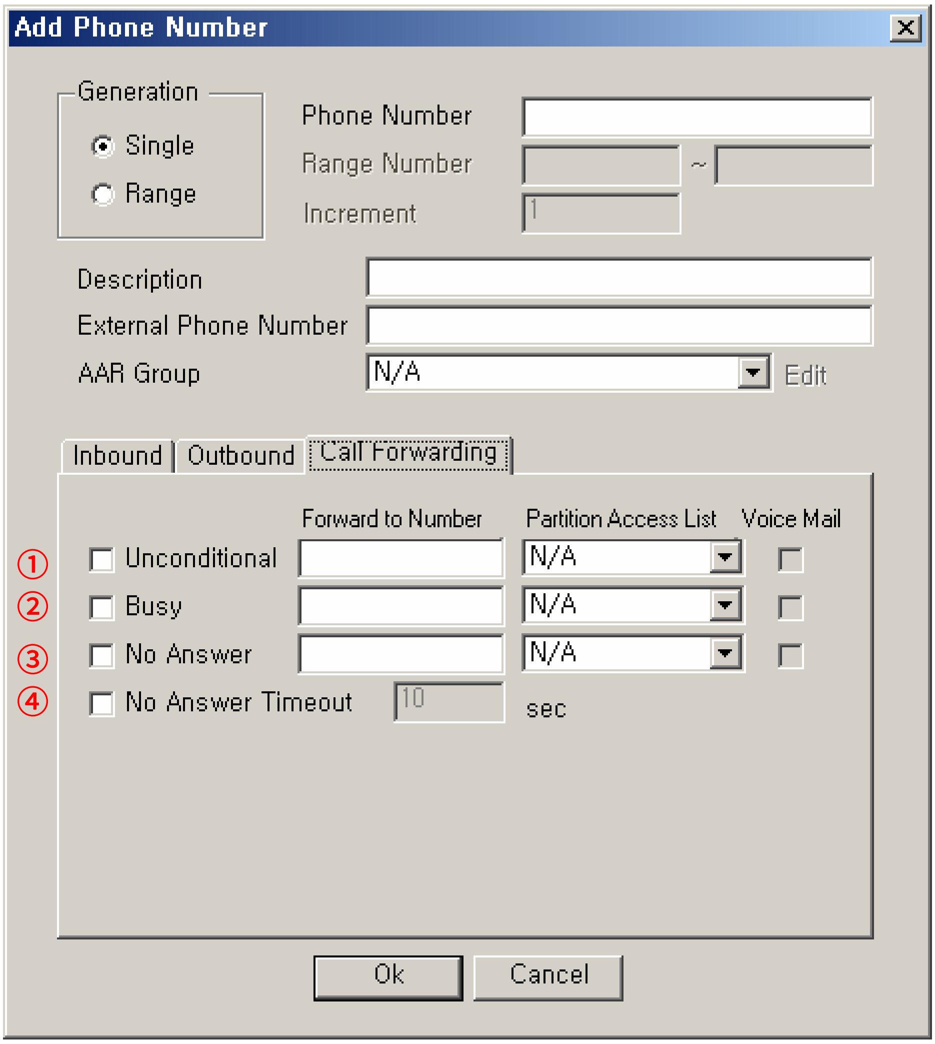

그림 2-96 Phone Number – Call Forwarding Properties

표 2-65 Phone Number – Description of Call Forwarding Properties

Field	Description
1	This is to configure Call Forwarding Unconditional. Enter the number to be forwarded and select an option whether to connect to Voice Mail, Partition Access List.
2	This is to configure a call to be forwarded when the line is busy. Enter the number to be forwarded and select an option whether to connect to Voice Mail, Partition Access List.
3	This is to configure a call to be forwarded when there is no answer. Enter the number to be forwarded and select an option whether to connect to Voice Mail, Partition Access List.
4	This is to set a time for no answer

Digit-Mapping Phone Numbers

This setting is not mandatory but optional. Digit-map is used by a large-sized business for number search of many phone numbers, but a small-sized business may not need one. This feature is to bundle the phone numbers with consistent pattern and sets them to one Pattern Digit Map for fast routing. The Digit Map must include all the phone numbers of the pattern.

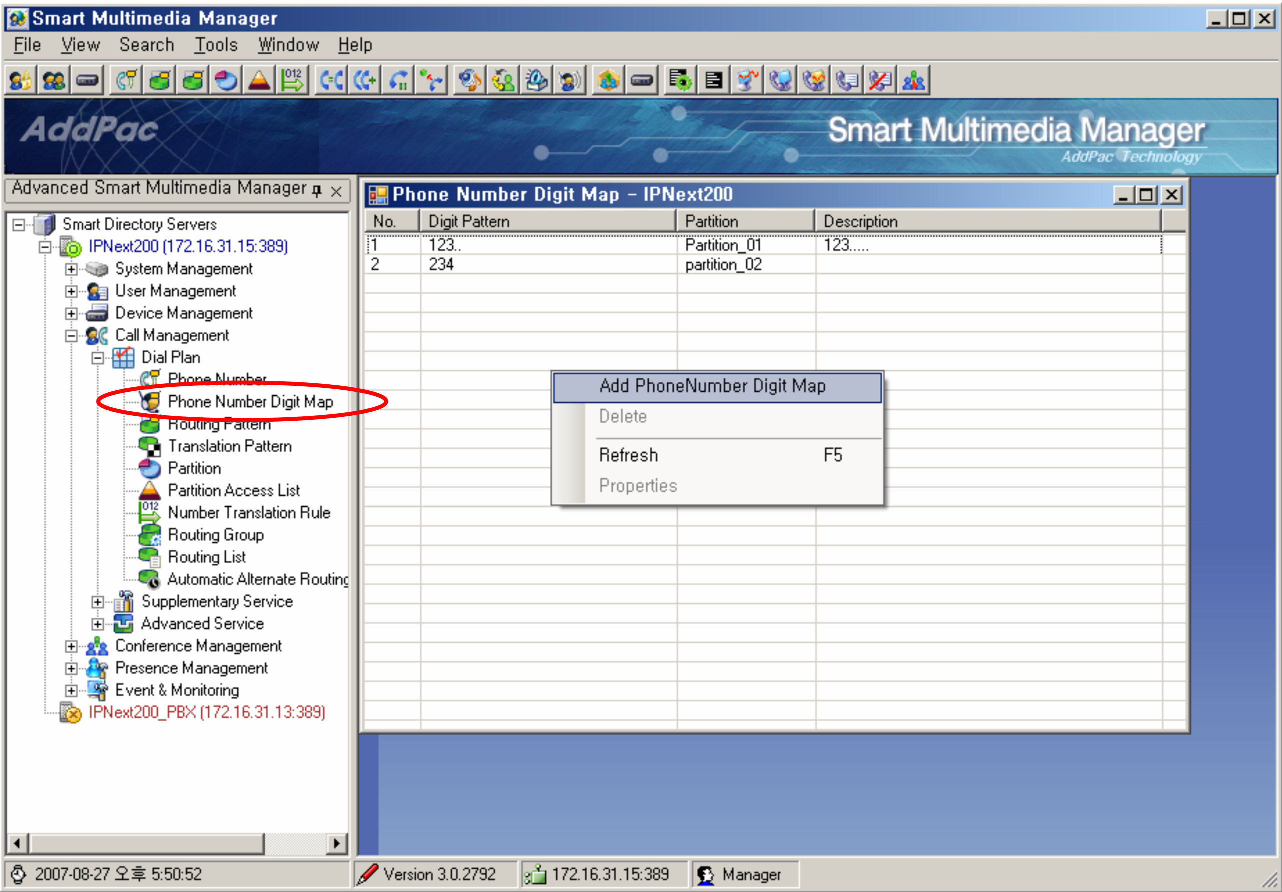

Figure 2-97 Phone Number Digit Map

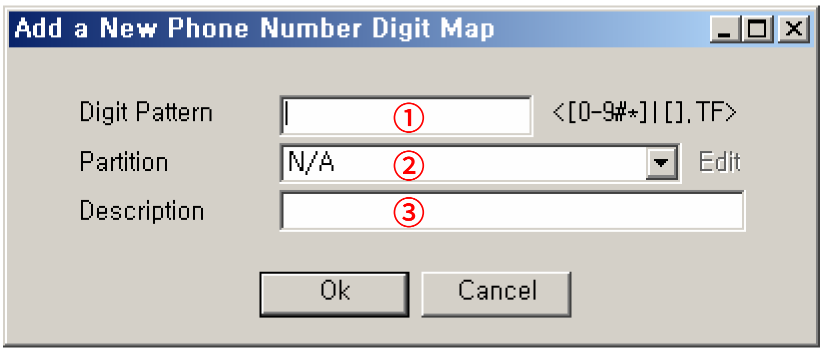

Figure 2-98 Add a New Phone Number Digit Map

Table 2-66 Description for Registering Phone Number Digit Map

No.	Description
1	Enter a pattern of many phone numbers
2	Choose a partition of the phone numbers in a pattern
3	Enter a description

Configuring Routing Pattern

Routing Plan is to configure Dial Plan. Dial Plan connects Trunk for Pattern with a particular number which is received from the equipment sending an outgoing call. In other words, the configuration is to be applied for the outbound call process. Routing Pattern can be carried out from the left tree Menu, by locating Call Management > DialPlan > Routing Pattern.

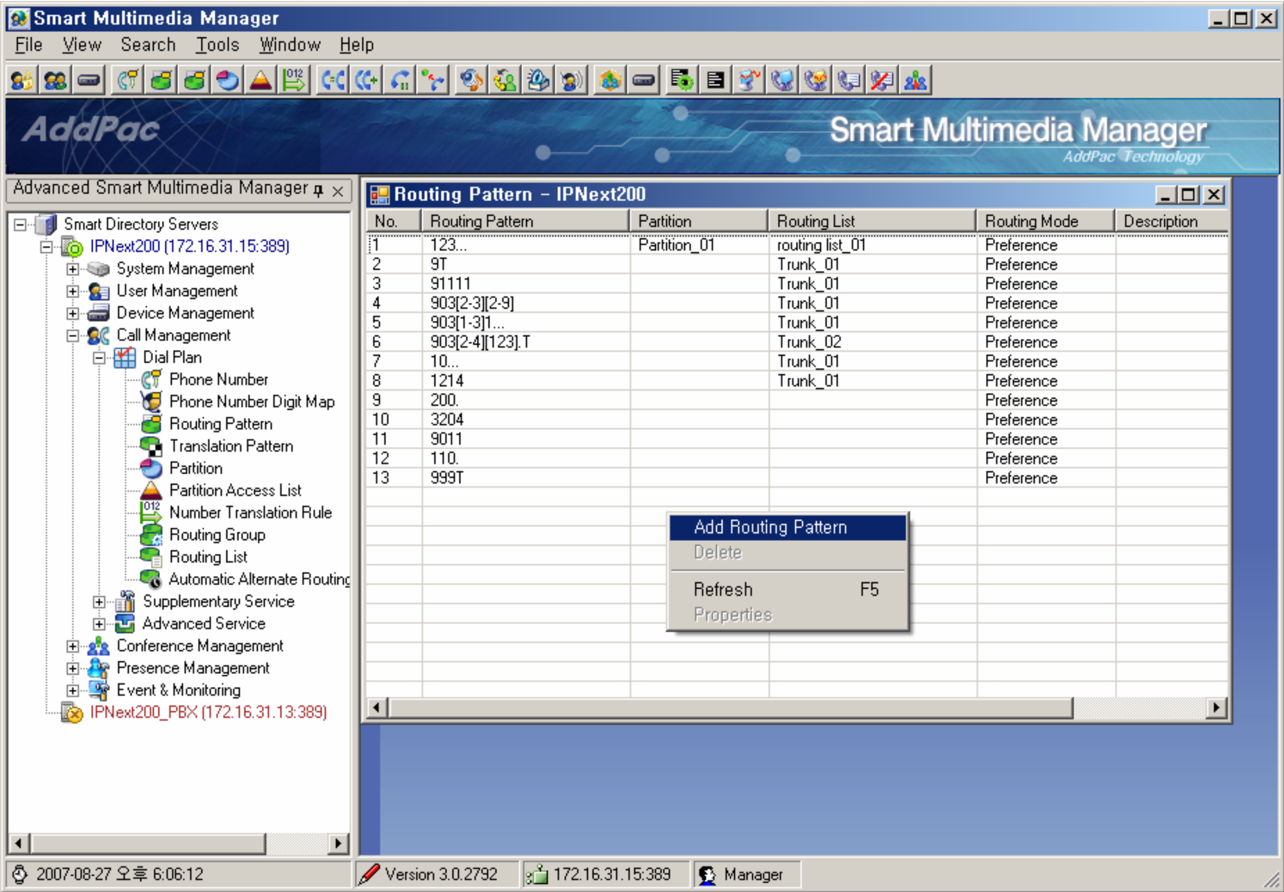

Figure 2-99 Routing Pattern

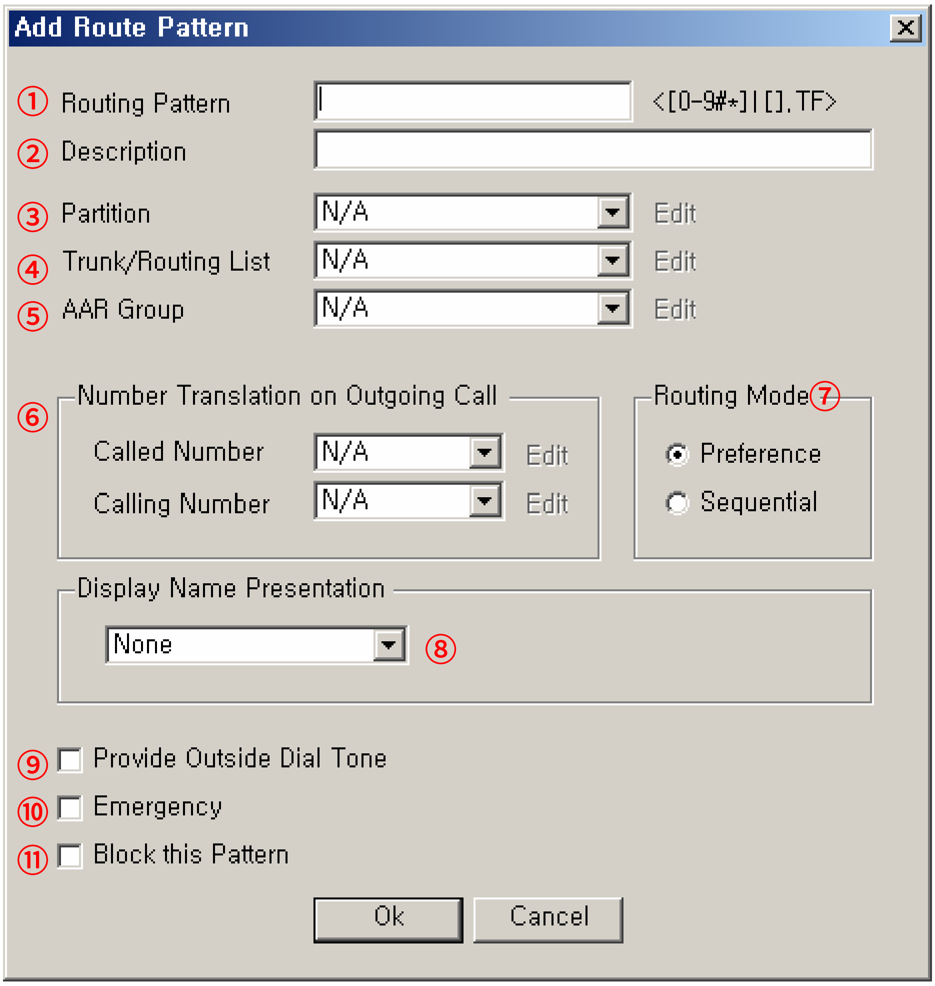

Figure 2-100 Routing Pattern Properties

Table 2-67 Description of Routing Pattern Properties

Field	Description
1	Configure the Pattern to be routed for a particular number. All the cases of this Pattern such as ‘9 for’ Longest Match Address, ‘[1-9]” for the Range Pattern, ‘9T’ for the Patter of Timeout. This configuration conforms to destination-pattern which is supported by AddPac Gateway.
2	Register a description of Routing Pattern.
3	Configure the Partition to be applied to this Routing Pattern. When the inbound call is processed from the Trunk or the terminal sending the outgoing call, routing is possible by this Routing Pattern, only when this Partition is included in the Partition Access List of the Trunk or the terminal. Setting the Partition to ‘N/A’ means allowing the access to the all. In other words, the access is possible from any terminal or Trunk with the Partition Access List.
4	Select Routing List with a higher priority or Routing Pattern that applies to the Trunk equipment. This Routing Pattern can be applied directly and simply to the Trunk equipment. To set the priority level of many Trunk equipments, Routing List can be applied.

5	Select AAR Group
6	Apply Translation Rule for the incoming and outgoing number for the number conversion of the outbound call.
7	Select Routing Mode of Routing List. Preference is the mode to select the equipment with a high priority in the order. When Setup Message is sent to the Device with a high priority and no response is received in a certain time and number of retry. Setup Message is transmitted to the Device with the next high priority. Sequential does not matter with the priority order and it selects one at a time.
8	
9	This is an option to select whether to provide a virtual Dial Tone
10	Set the highest priority to the Outbound call.
11	Block Outbound Routing for this pattern.

Translation Pattern Configuration

Translation Pattern is applied when a particular number, which is sent from a terminal, matches with a Pattern. When Translation Pattern is applied, Number Translation in Calling Party Number, Called Party Number and (or) Property Translation (Dial tone, call process for emergency, call limitations). The translated number can be applied again for the Inbound call process recursively.

In other words, the Partition configuration, which belongs to the Outbound property in the property of Translation Pattern, is limited by Partition Access List during the process of the Inbound call. After a change of number takes a place, the configuration of Partition Access List, with the Pattern of the Inbound property, is applied. Then the terminal or Trunk receiving the incoming call is selected as a procedure of the Outbound process.

Translation Patter can be performed from the left of tree Menu by locating Call Management > DialPlan > Translation Pattern.

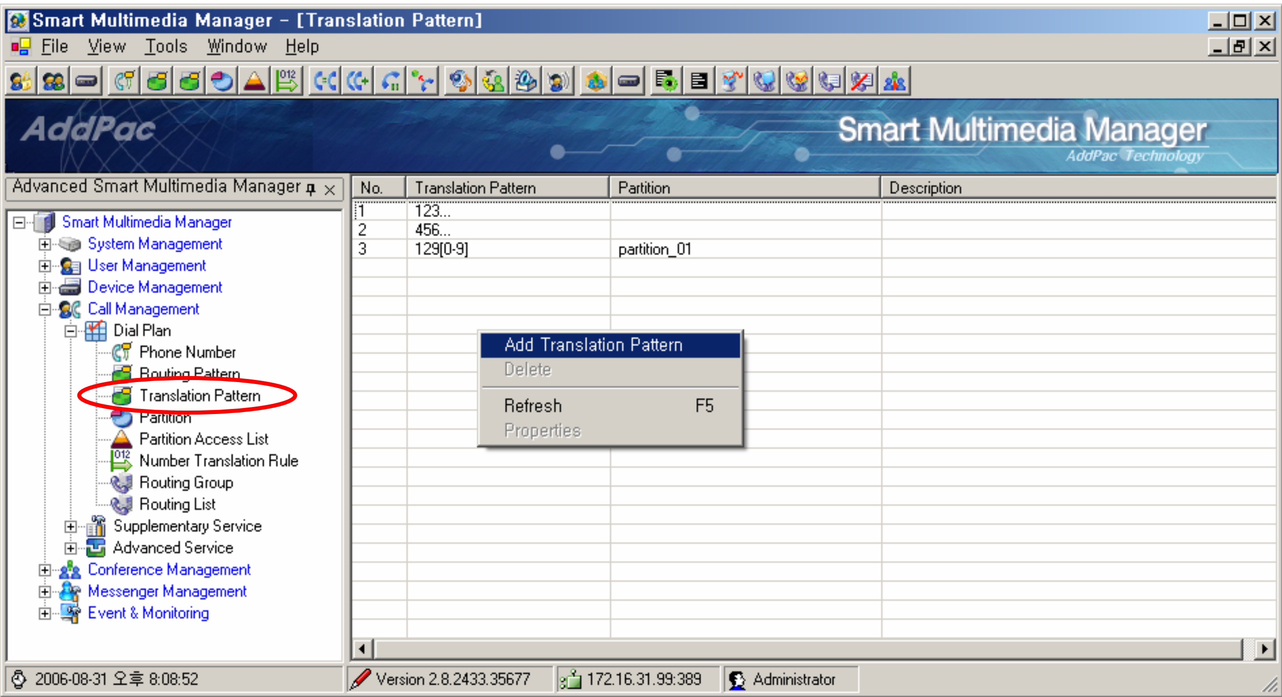

Figure 2-101 Translation Pattern Screen

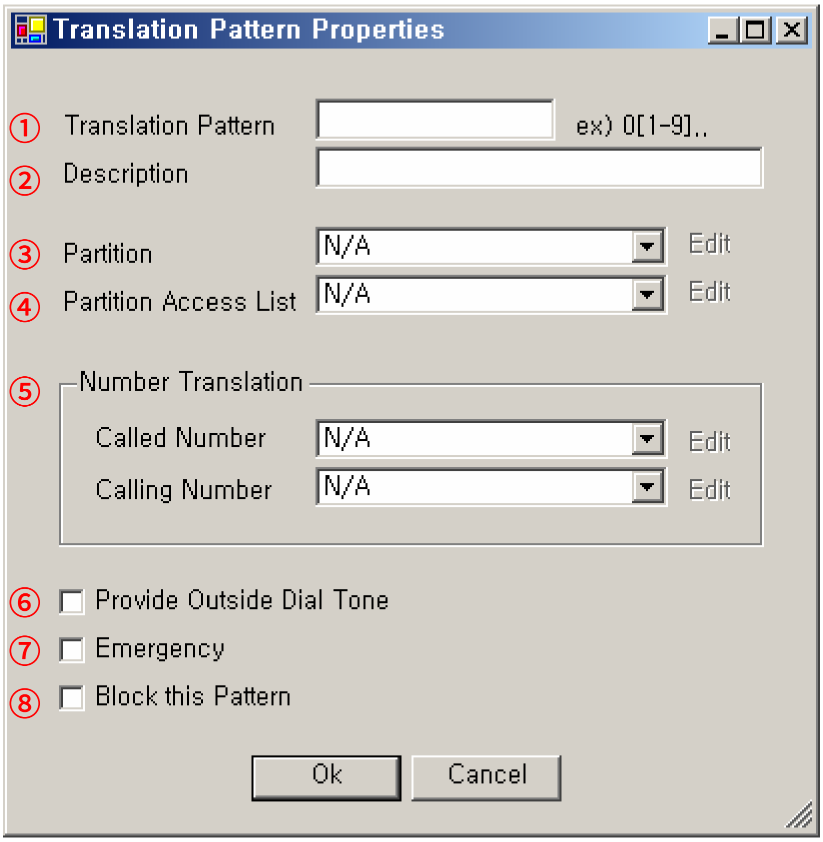

Figure 2-102 Translation Pattern Properties

Table 2-68 Translation Pattern Properties Description

Field	Description
1	Register a Translation Pattern. All the cases of this Pattern such as '9 for' Longest Match Address, '[1-9]" for the Range Pattern, '9T' for the Patter of Timeout. This configuration conforms to destination-pattern which is supported by AddPac Gateway.
2	Register a description of Translation Pattern.
3	Configure the Partition to be applied to this pattern. When the inbound call is processed from the Trunk or the terminal sending the outgoing call, this Patern can be applied, only when this Partition is included in the Partition Access List of the Trunk or the terminal. Setting the Partition to 'N/A' means allowing the access to the all. In other words, the access is possible from any terminal or Trunk with the Partition Access List.
4	Configure the Partition Access List to be applied to this Pattern. When this Pattern is applied and the inbound call is processed recursively, the outbound process is possible for only the terminal (or Trunk) using the number (or Pattern) belongs to the Partition in Partition Access list When the Partition Access List is set to 'N/A' for no configuration, routing is possible for any number with a property of Partition or Pattern (the object not configured with the Partition).
5	Translation Rule to be applied for the incoming and outgoing number for Number Translation.

6	Select whether to provide a virtual Dial Tone
7	Set the highest priority for the Pattern
8	Block Routing for the Pattern

Partition Configuration

Call Routing in Call Manager means that processing the inbound call from a terminal or Trunk to be outbound to the desired terminal or Trunk. So the call can be made and connected.

When Routing Plan is simple for a small company, Partition and Partition Access List can be used without configuration. However, Partition and Partition Access List play some important roles in routing, for the complex Routing Plan, provisioning and operating Centrix.

When an inbound call is processed, the Patition with routing possibility can be determined by the Partition Access List which is applied to the terminal or trunk for the inbound call.

The Partition in the left tree Menu can performed by locating Call Management > DialPlan > Partition.

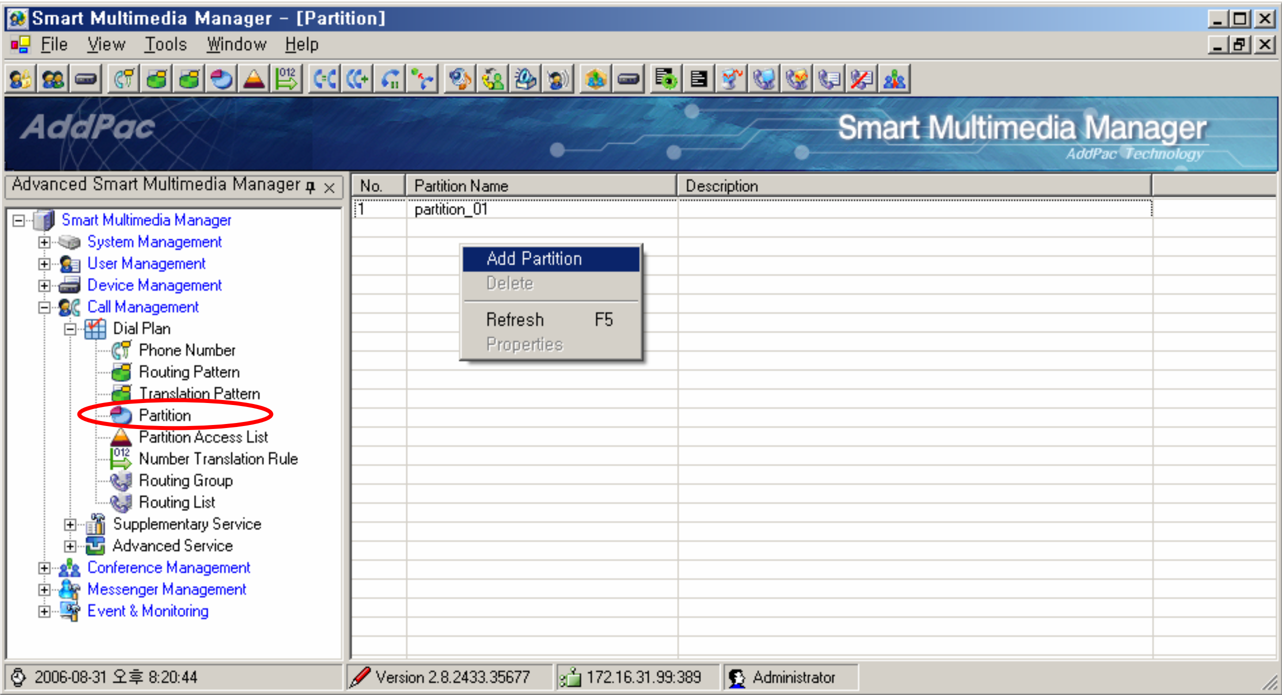

Figure 2-103 Partition Screen

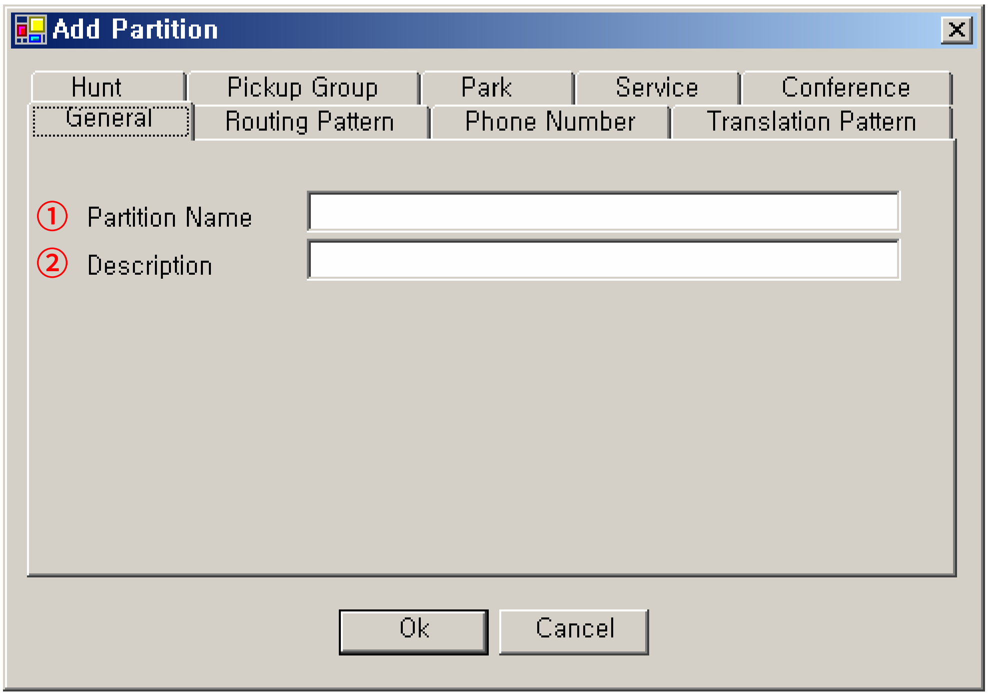

Figure 2-104 Partition – General Properties

Table 2-69 Partition – Description of General Properties

Field	Description
1	Register a name of Partition
2	Register a description of Partition

Figure 2-105 Partition Properties

Table 2-70 Description of Partition Properties

Field	Description
1	This is the list of Routing Pattern without registration of Partition.
2	This is the list of Routing Pattern sets with the present Partition.
3	This is the template to set the present Partition to the Phone Number with the same function described in the Figure above.
4	This is the template to set the present Partition to Translation Pattern with the same function described in the Figure above.
5	This is the template to set the present Partition to Hunt Group with the same Figure.
6	This is the template to set the present Partition to Pickup Group with the same function described in the Figure above.
7	This is the template to set the present Partition to Park with the same function described in the Figure above.
8	This is the template to set the present Partition to the Service with the same function described in the Figure above.
9	This is the template to set the present Partition to Conference with the same function described in the Figure above.
10	This is the template to set the present Partition to Phone Number Digit Map with the same function described in the Figure above.

Partition Access List Configuration

Call routing in Call Manager is to process the inbound call sent from a Trunk or terminal and to process the outbound call to the desired Trunk or Terminal. In this way, the call is made and connected.
Partition Access List is applied to this call when the inbound call is processed. When the inbound call is processed, a Partition, which can be routed, is determined by the Partition Access List. This Partition List is applied to the terminal or Trunk for processing the inbound call.

Partition Access List can be carried out from the tree menu on the left by locating Call Management > DialPlan > Partition Access List

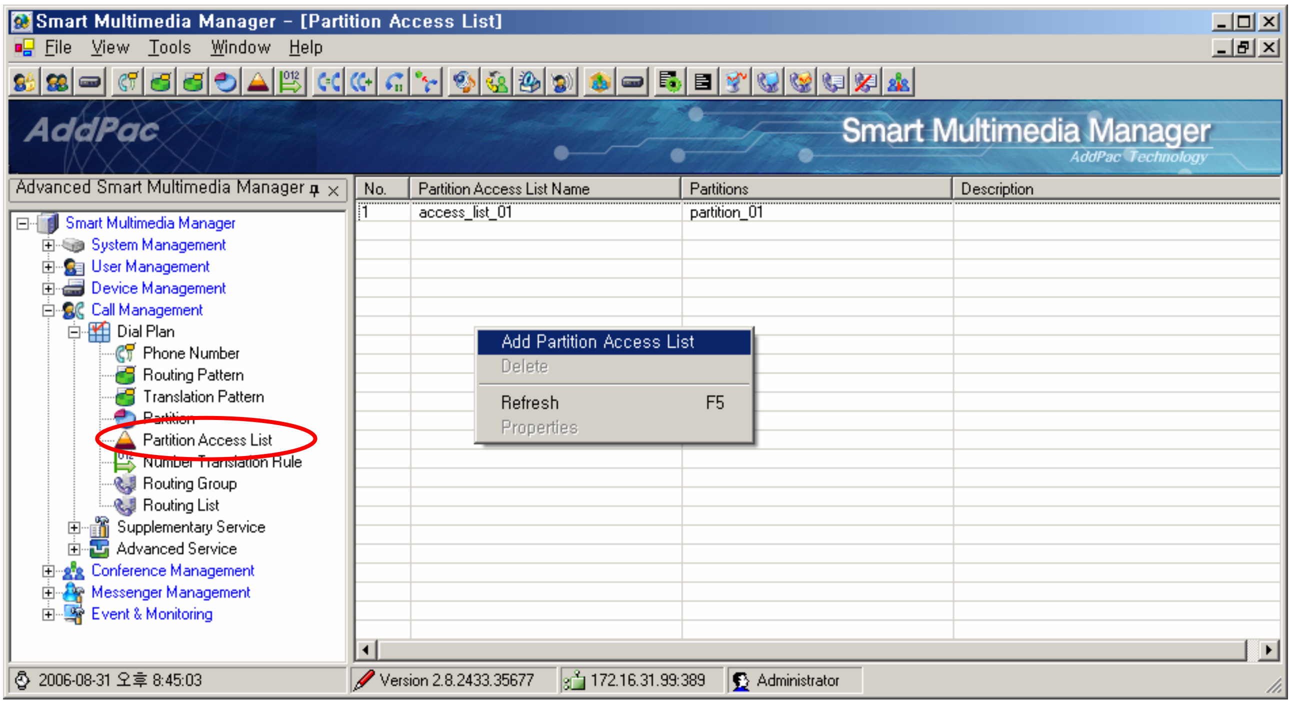

Figure 2-106 Partition Access List Screen

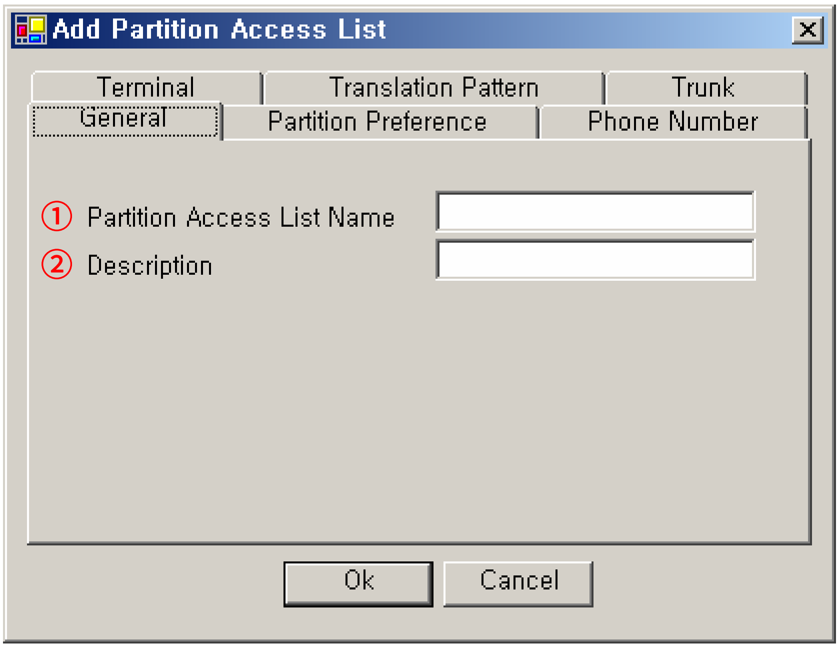

Figure 2-107 Partition Access List Properties

Table 2-71 Partition Access List - General Properties Description

Field	Description
1	Register a name of Partition Access List
2	Register a description of the Partition Access List

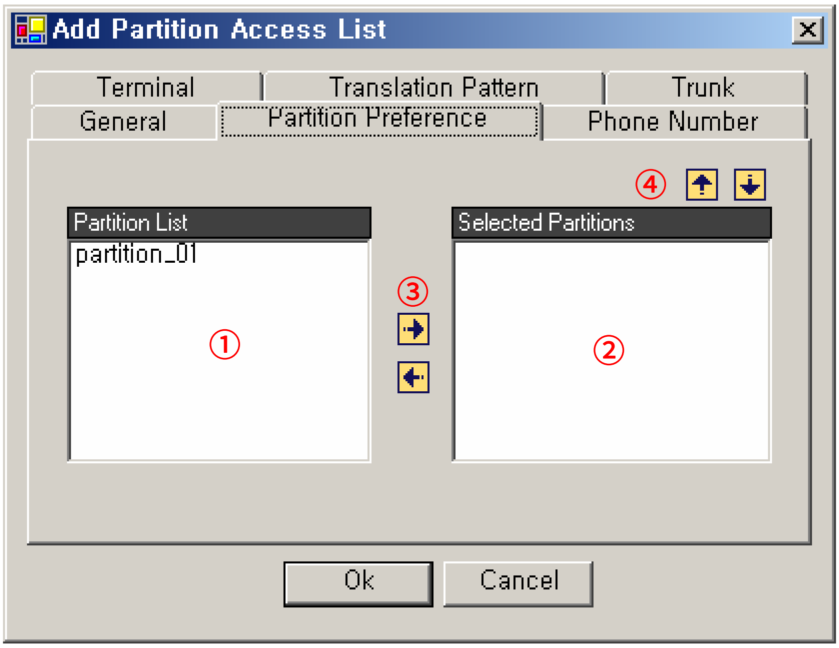

Figure 2-108 Partition Access List – Partition Preference Properties

Table 2-72 Partition Access List – Partition Preference Properties Description

Field	Description
1	This is the list of partitions which are not included in Partition Access List.
2	This is the list of partition which is included in Partition Access List.
3	Delete from or register to the partition list.
4	Adjust the priority level of the selected Partition

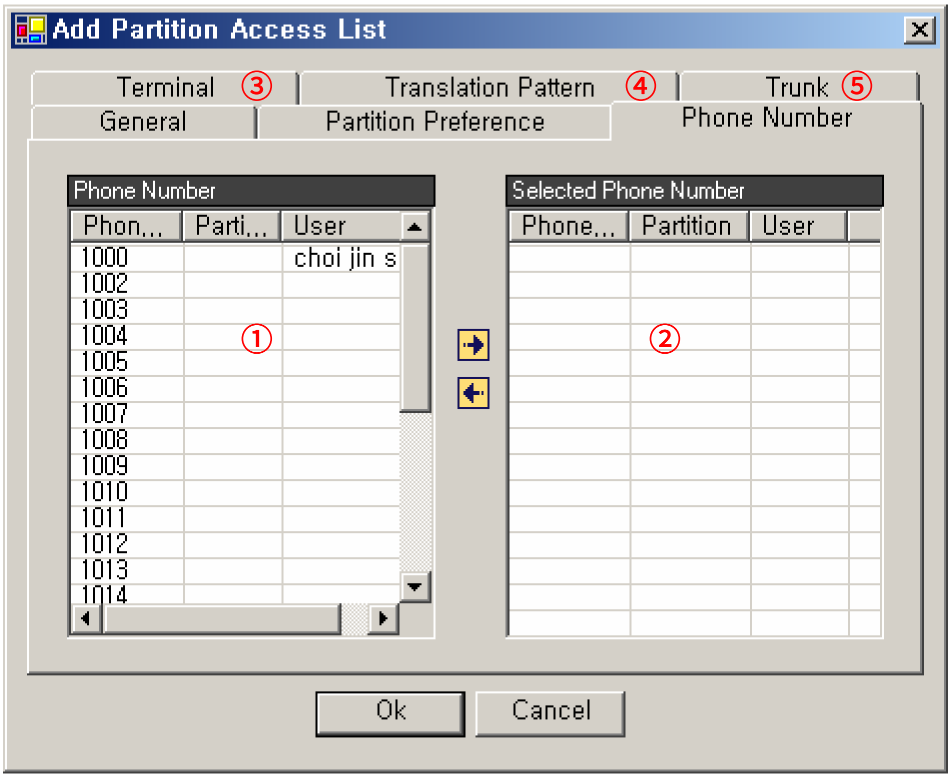

Figure 2-109 Partition Access List Properties

Table 2-73 Description of Partition Access List Properties

Field	Description
1	This is the list of the phone numbers which are not included in the present Partition Access List.
2	This is the list of the phone number which is included in the present Partition Access List.
3	This template is to configure the present Partition Access List to the terminal with the same function described in the Figure above.
4	This is the template to configure the present Partition Access List to Translation Pattern with the same function as the Figure above.
5	This template is to configure the present Partition Access List with the same function described in the Figure above.

Configuring Number Translation Rule

Number Translation is a function to change the called (or calling) party number to a different number for particular call. Translation rule can be applied to both of inbound call and outbound call of called party number and calling party number.

In a case of applying the Number Translation to the inbound call, the incoming called (or called) party number is changed according to a translation rule. Then this incoming called party number can be used to select the outbound called party. In the other case of applying the Number Translation to the outbound call, the outgoing called party (or calling) number is changed according to a Translation Rule. Then this outgoing called party number can be used.

The Number Translation is needed for the case of changing the Private Number to the Public Number (or changing the Public Number to the Private Number) or the number connection such as the Number Expansion. The Number Translation has more range of changes than the Number Expansion.

In order to perform the Number Translation, the Translation Rule Set must be created and this Translation Rule Set is to be applied to the Device or Routing Pattern and Translation Pattern.

When the Number Translation is applied to the Device, the number can be changed for the incoming call. For the outgoing call, the number can not be changed.

In case Route Address is applied, the number can be changed for the outgoing call from the corresponding Router Device.

Number Translation Rule can performed from the tree menu on the left by locating Call Management > DialPlan > Number Translation Rule.

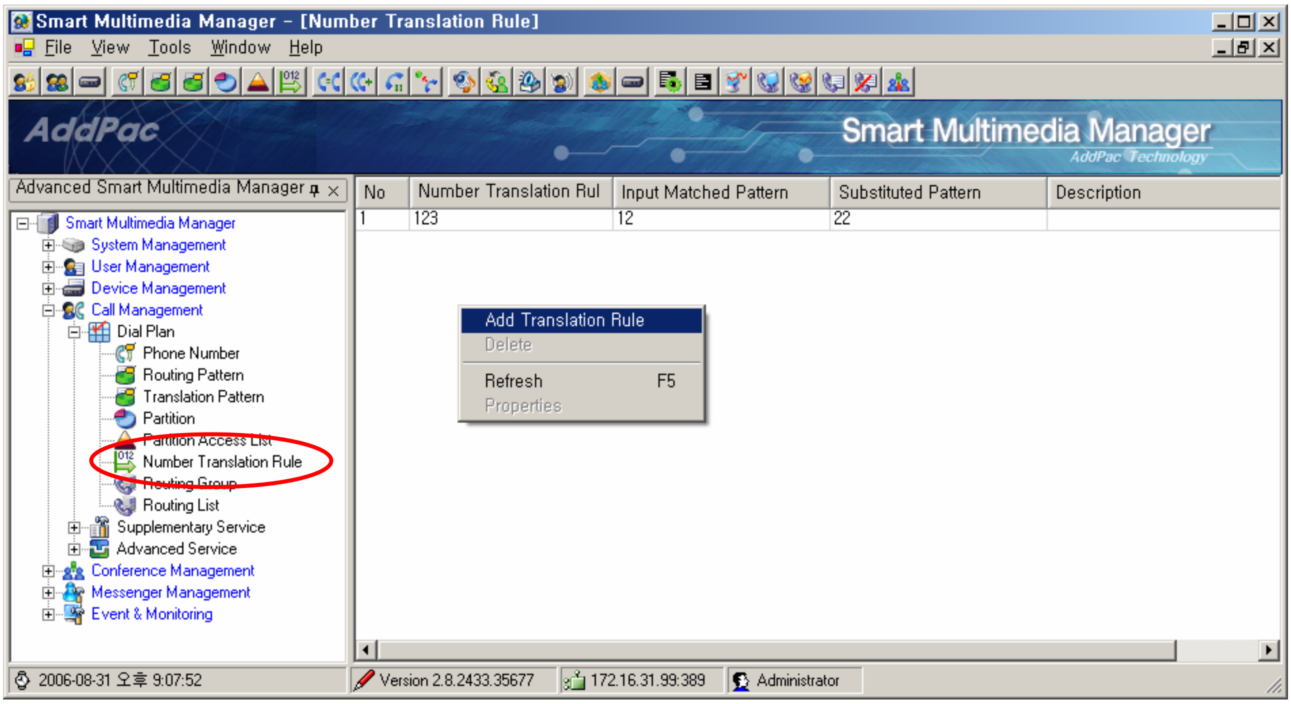

Figure 2-110 Number Translation Rule Screen

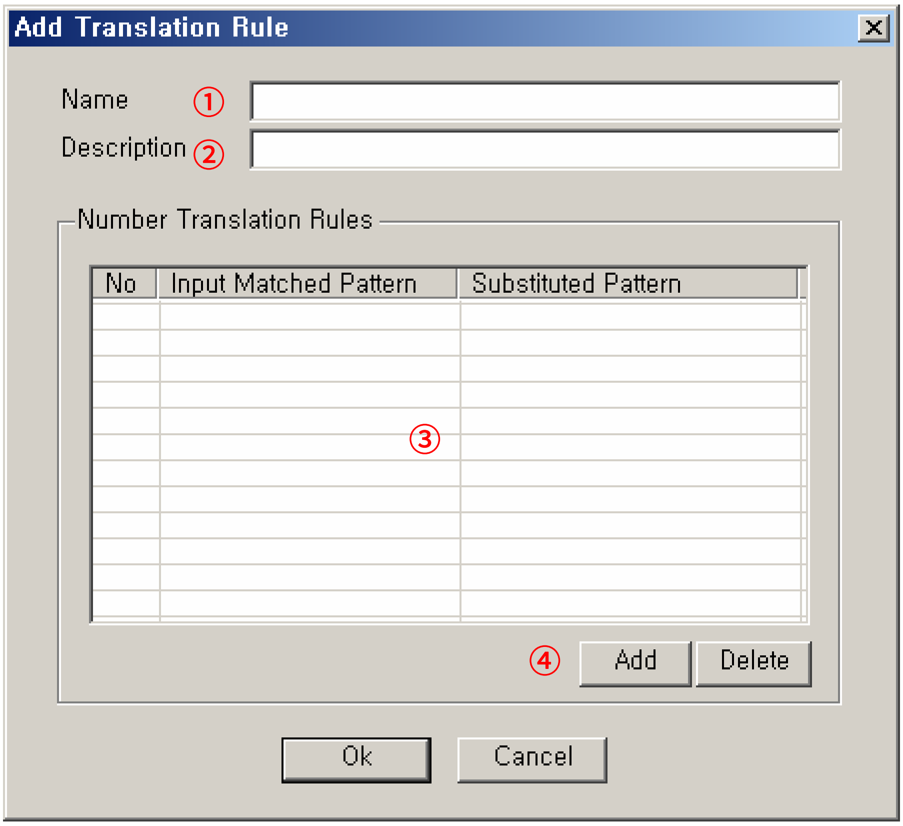

Figure 2-111 Number Translation Rule Properties

Table 2-74 Description of Number Translation Rule Properties

Field	Description
1	Register a name of Number Translation Rule
2	Register a description of Number Translation Rule
3	This is the list of Substituted Pattern (the Patter of the number to be changed by Input Number) and Input Matched Pattern (the Patter of the Input Number to be used for the Number Translation) for the Rule
4	Delete or register the Rule.

Routing Group Configuration

Routing List is the group of the trunk devices with a priority in order, which is to configure the trunk devices as a group and to register them in the following description

Routing Group can be carried out from the tree menu on the left by locating Call Management > DialPlan > Routing Group.

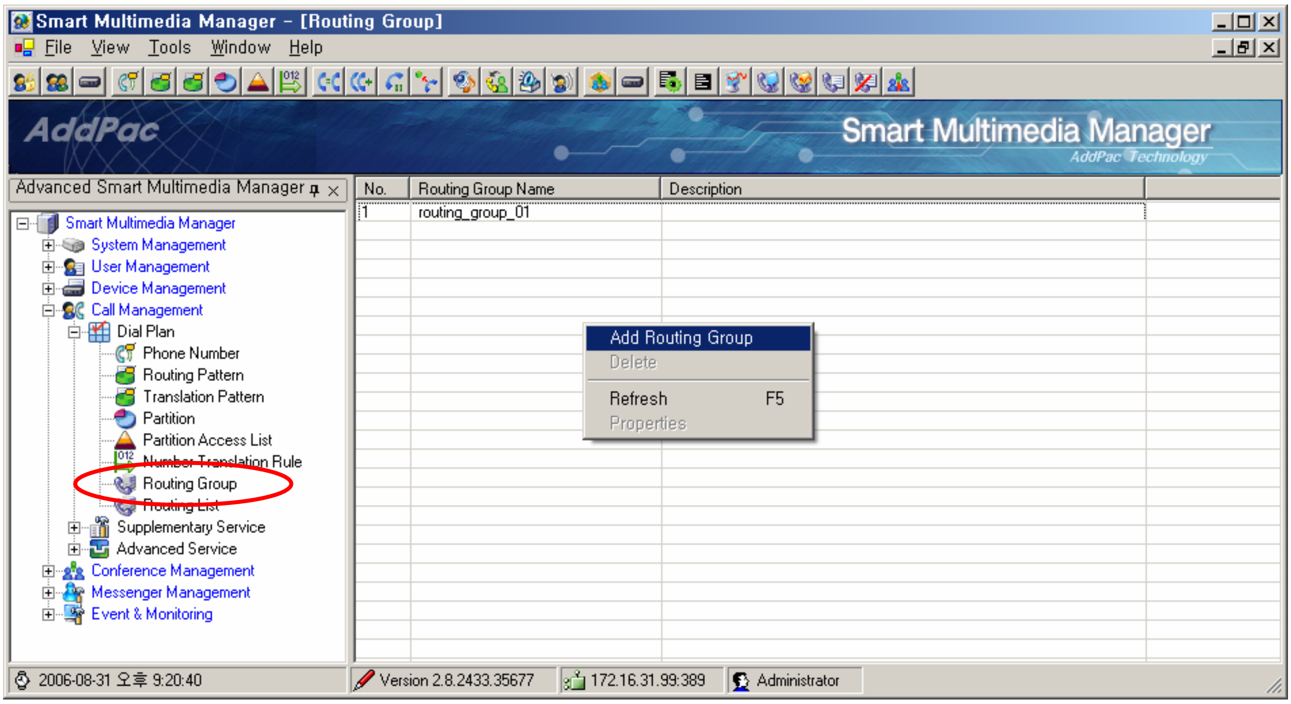

Figure 2-112 Routing Group Screen

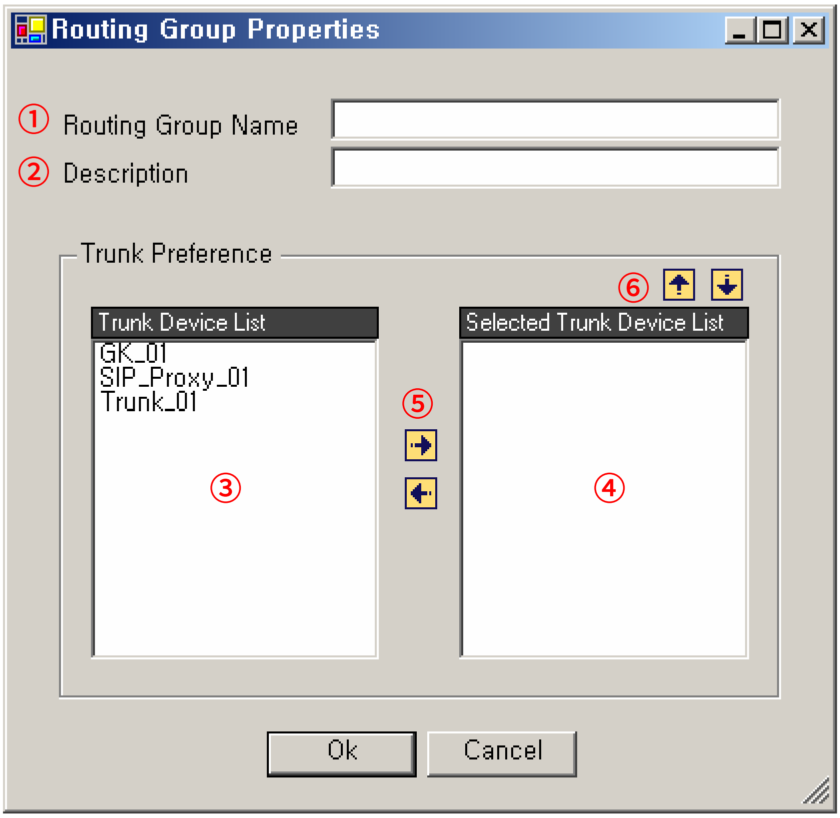

Figure 2-113 Routing Group Properties

Table 2-75 Routing Group Properties Description

Field	Description
1	Register a name of Routing Group
2	Register a description of Routing Group
3	This is the list of the Trunk devices which are not included in the present group.
4	This is the list of the Trunk devices which are registered to the present group.
5	Delete or register the Trunk device from or to Group
6	Adjust the priority level for the Trunk devices registered to the present Group.

Routing List Configuration

Routing List can be applied to Routing Pattern as Routing Group with a priority order.

Routing List can be performed as Call Management > DialPlan > Routing List on the tree menu on the left side of the screen.

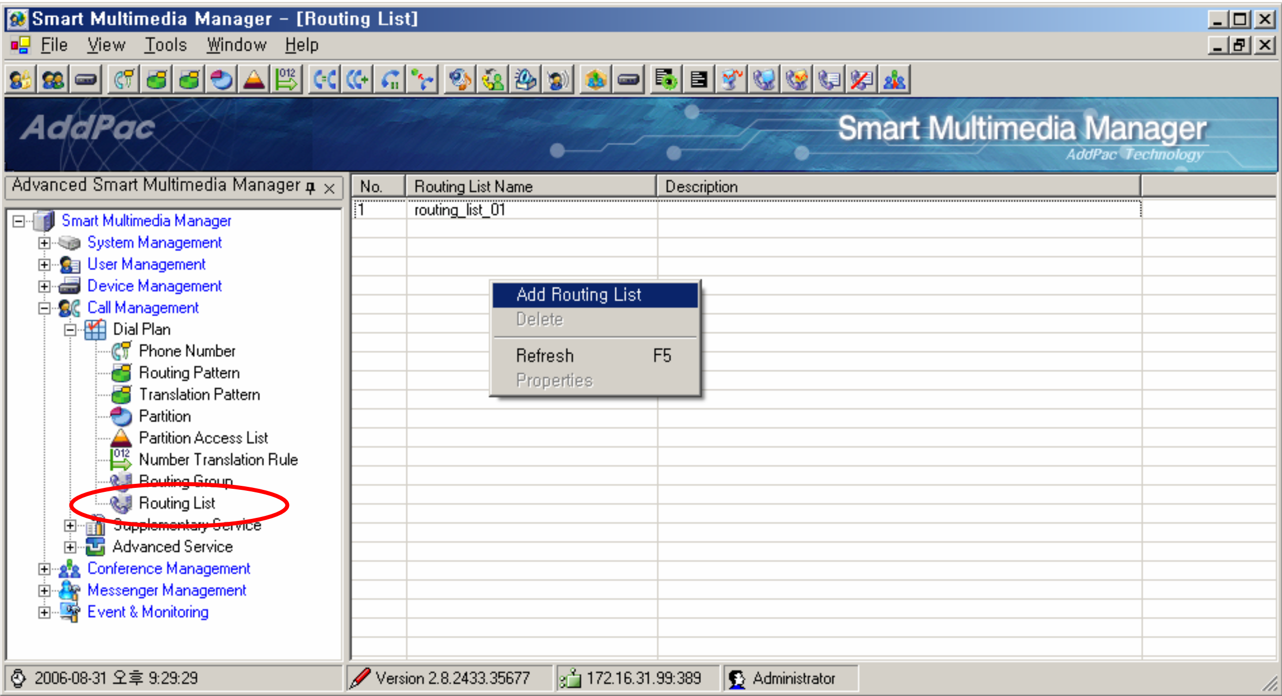

Figure 2-114 Routing List Screen

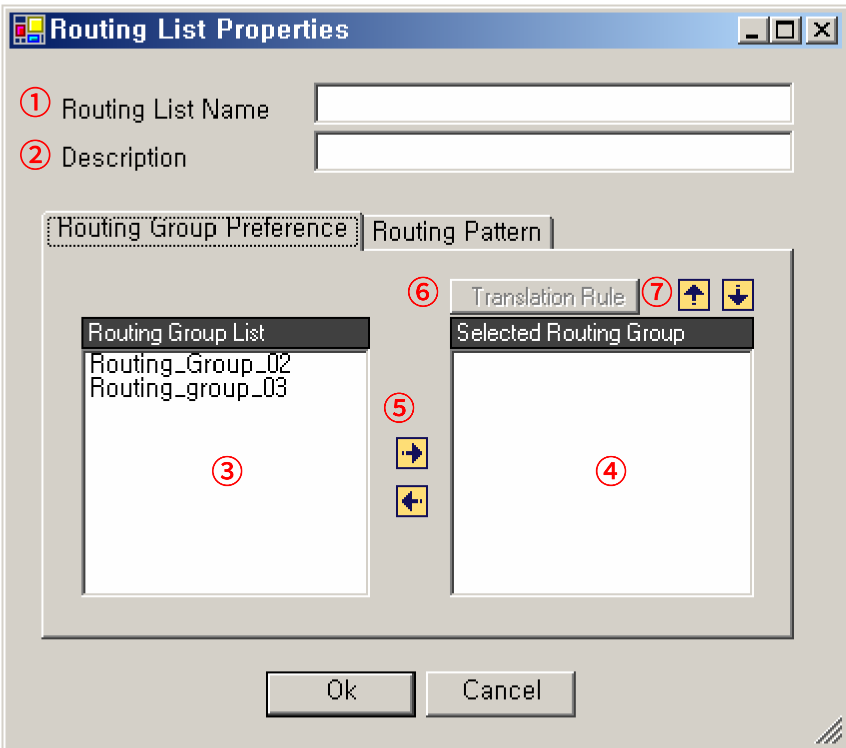

Figure 2-115 Routing List Properties

Table 2-76 Routing List Properties Description

Field	Description
1	Register a name of Routing List
2	Register a description of the Routing List
3	This is the list of Routing Group which is not included in the present Routing List.
4	This is the list of Routing Group which is registered to the present Routing List.
5	Register or delete the Routing Group from or to the Routing List
6	Set the Translation Rule for the Routing Group which is registered to the present Routing List.
7	Adjust a priority level for the Routing Group which is registered to the present Routing List.

Configuring AAR (Automatic Alternate Routing) Group

Automatic Alternate Routing (AAR) reroutes to PSTN or another path to other networks automatically, in case of a troublesome VoIP communication caused by lacking in bandwidth or network failure. The information of AAR Group can be set in Phone Number, Routing Pattern. The Phone Number and Routing Pattern, with the same attributes applied for routing, are configured to the same AAR Group.

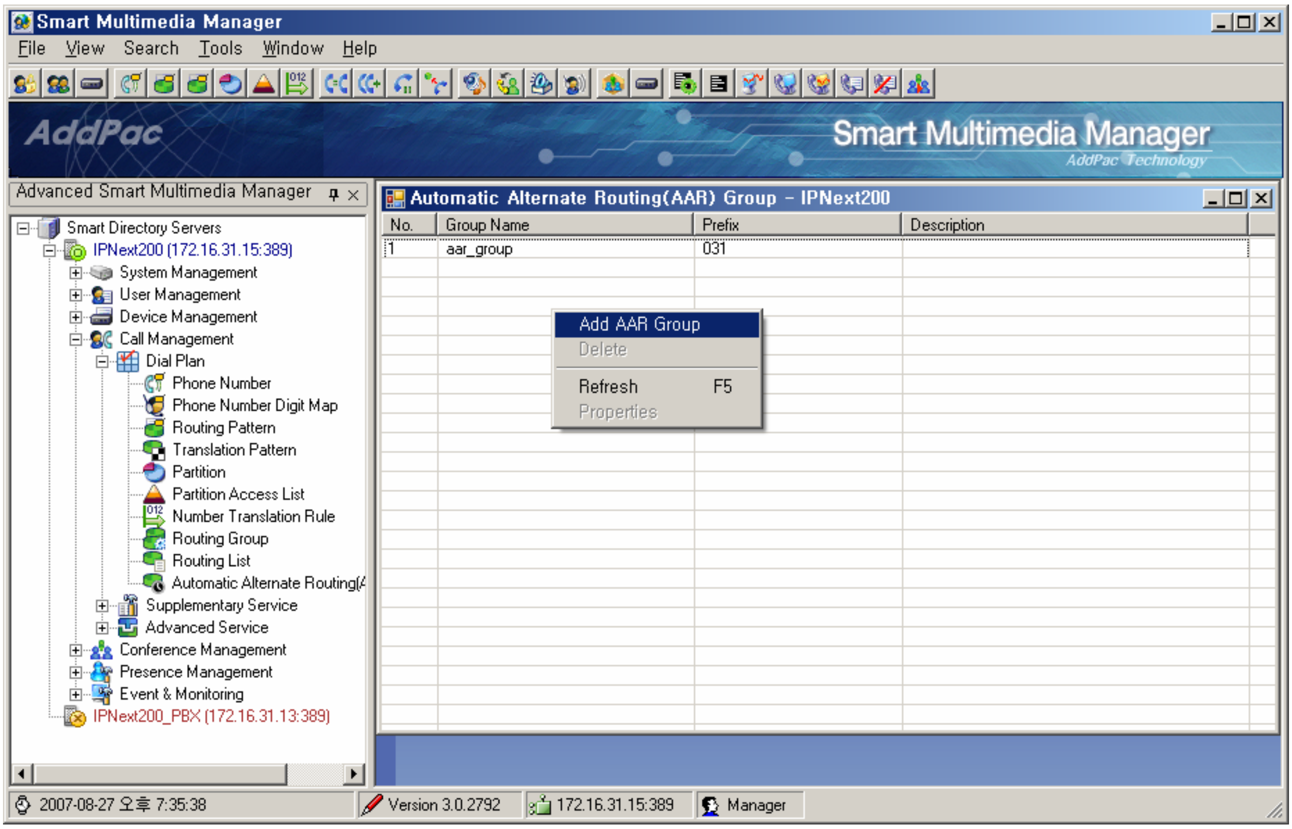

Figure 2-116 AAR Group

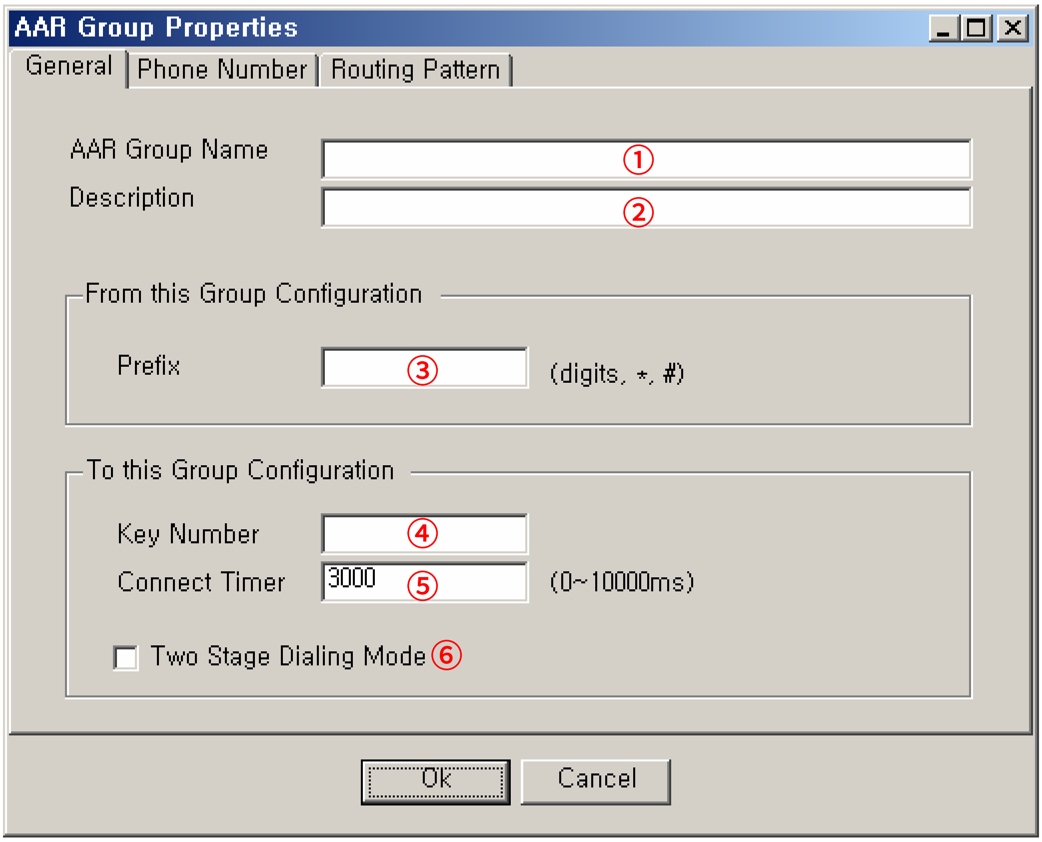

Figure 2-117 AAR Group – General Properties

Table 2-77 AAR Group – Description of General Properties

No.	Description
1	Enter a new AAR group name
2	Enter a description of AAP group
3	Enter a prefix, so rerouting can be executed by AAR. This AAR Group is applied to outbound
4	Enter a key number for the path to be replaced for routing. This AAR Group is applied to inbound.
5	Set up a timer after the 1st stage dialing of AAR two stage dialing. Start the 2nd stage dialing after a connect time value of the 1st stage dialing is passed.
6	Connect to two stage dialing for AAR

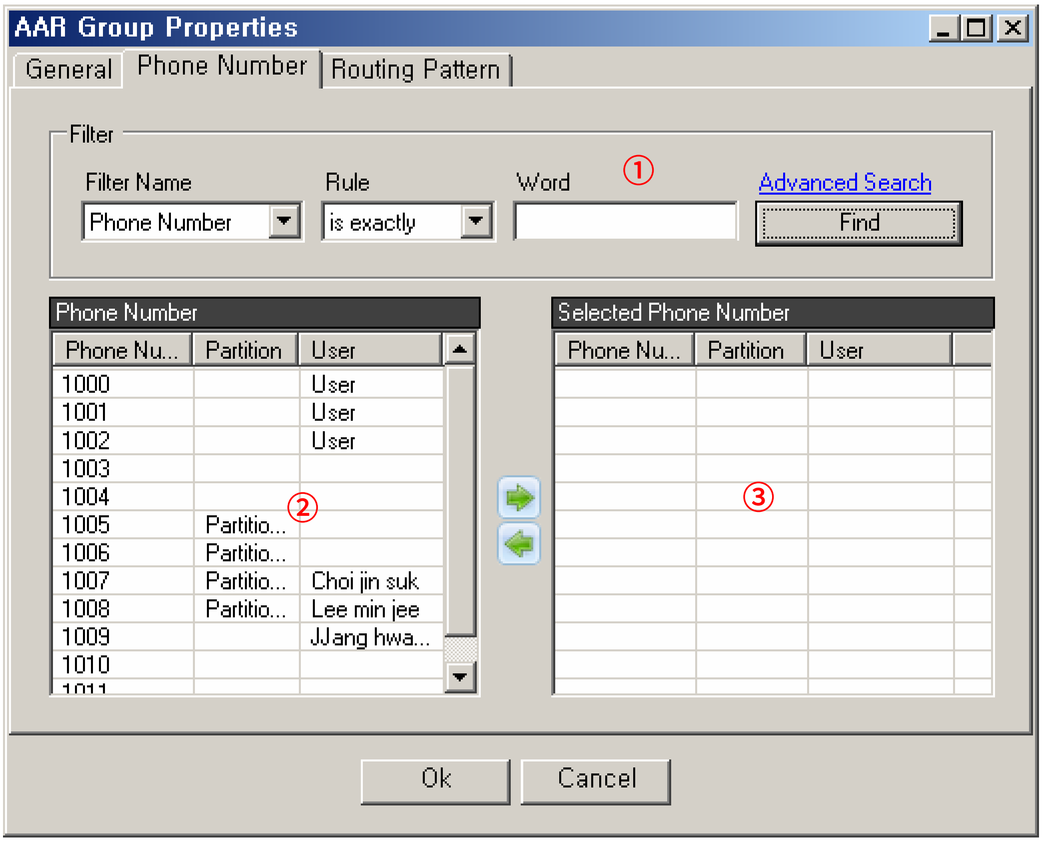

Figure 2-118 AAR Group – Phone Number Properties

Table 2-78 AAR Group – Description of Phone Number Properties

No.	Description
1	The list phone numbers, to be configured to AAR Group, can be searched by the conditions
2	The list of phone numbers to be configured to AAR Group
3	The list of phone numbers, which has been applied to AAR Group.

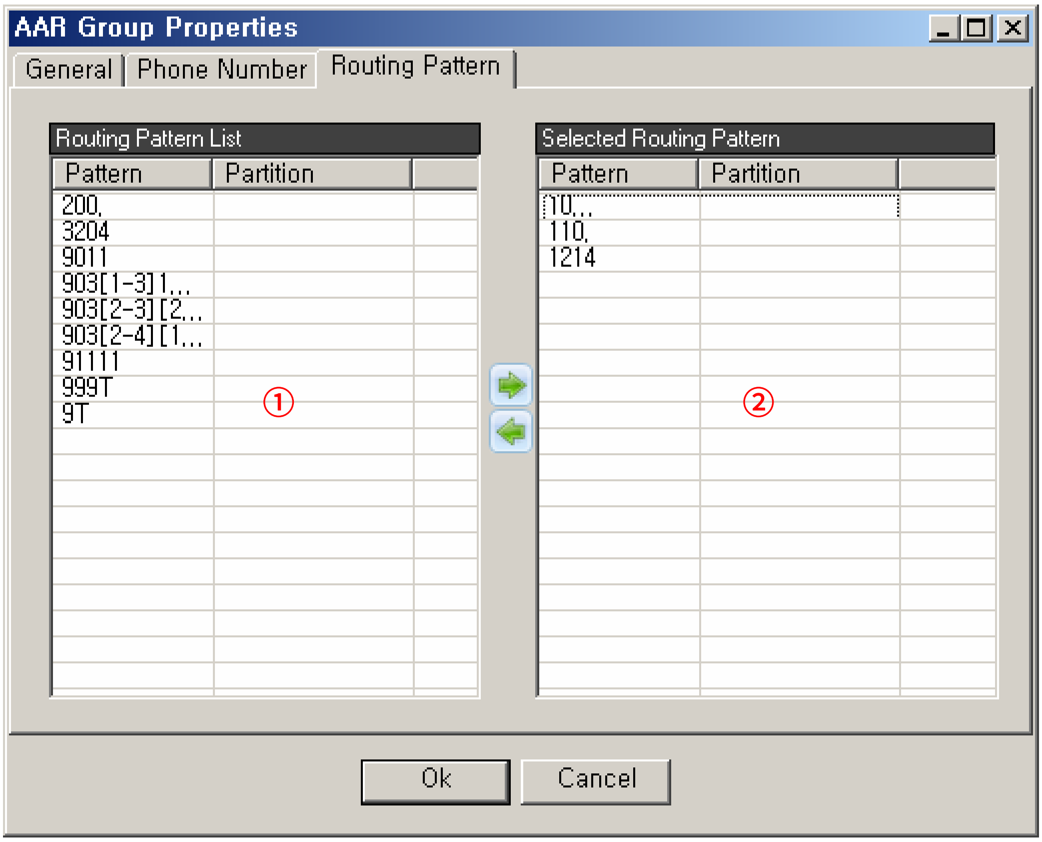

Figure 2-119 AAR Group – Routing Pattern Properties

Table 2-79 AAR Group – Description of Routing Pattern Properties

No.	Description
1	The Routing Pattern List to be configured to AAR Group
2	The Routing Pattern List, which has been applied to AAR Group

Call Management – Supplementary Service

Hunt Group Configuration

Hunt Group works as the general directory number. A call is transferred to one of the members of Hunt Group when the call is sent to the general directory number. When the member's line is busy or is not available to take the call, the call can be transferred again to the second member in Hunt Group.

Hunt Group can be performed as Call Management > Supplementary > Hunt Group from the tree menu on the left side of the screen.

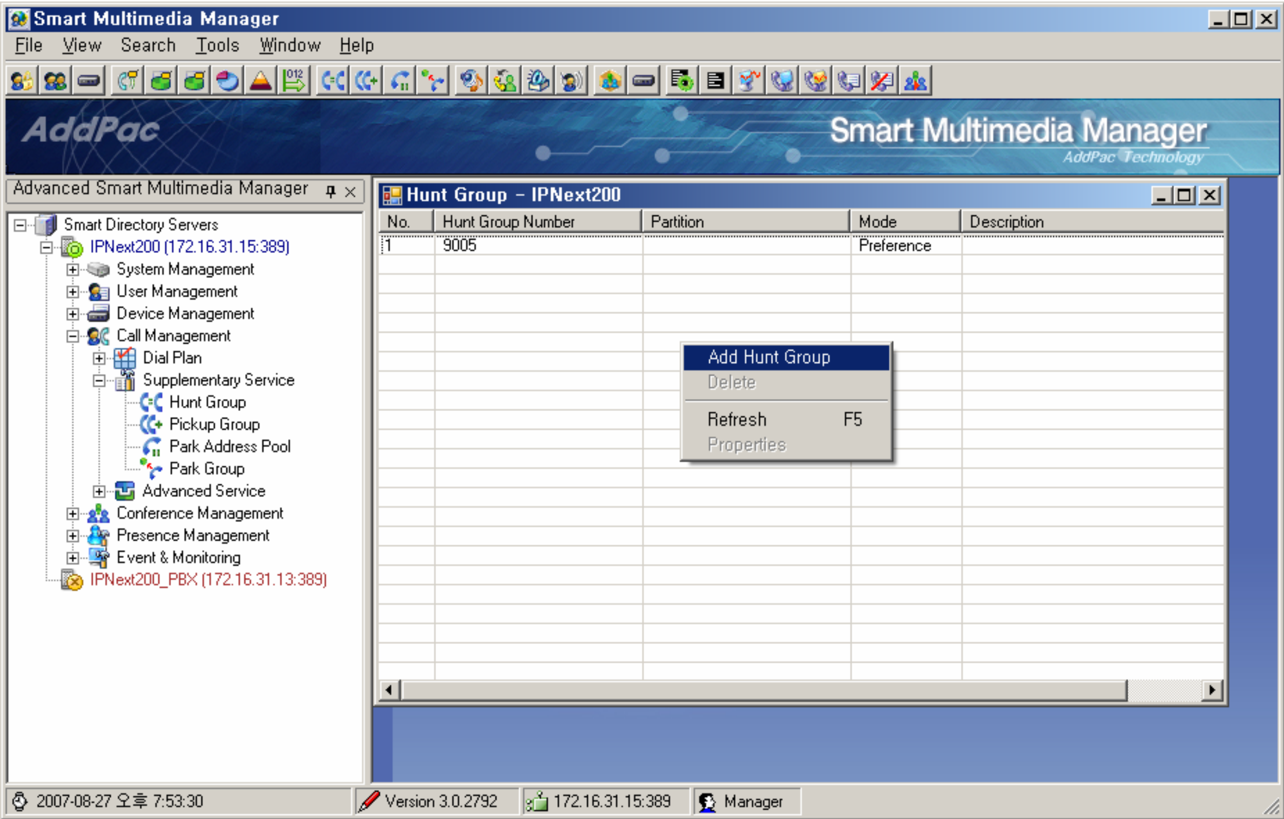

Figure 2-120 Hunt Group Screen

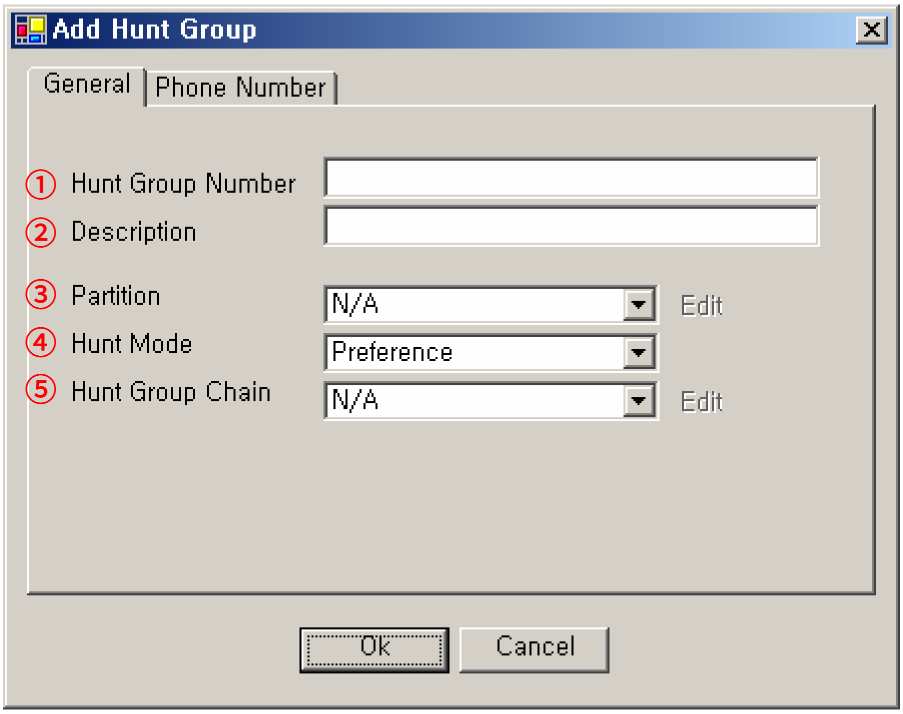

Figure 2-121 Hunt Group Properties

Table 2-80 Routing List Properties Description

Field	Description
1	Register a number of Hunt Group
2	Register a description of Hunt Group
3	Configure the Partition to be applied to the Hunt Group. Routing the number of Hunt Group is possible when the Partition Access List of the Trunk or terminal includes the Partition during the inbound call process of the terminal or Trunk sending the call. When the Partition is set to 'N/A for no configuration, this means the Partition is allowing the access to the all. In other wordsw, any terminal or Trunk with any Partition Access List can access.
4	Select an option for Hunt Mode: - Preference: Calling in the priority order which is set - Simultaneous: Calling simultaneously regardless of the priority order - Random: Calling randomly regardless of the priority order
5	Another Hunt Group can be assigned when Hunt Group is connected.

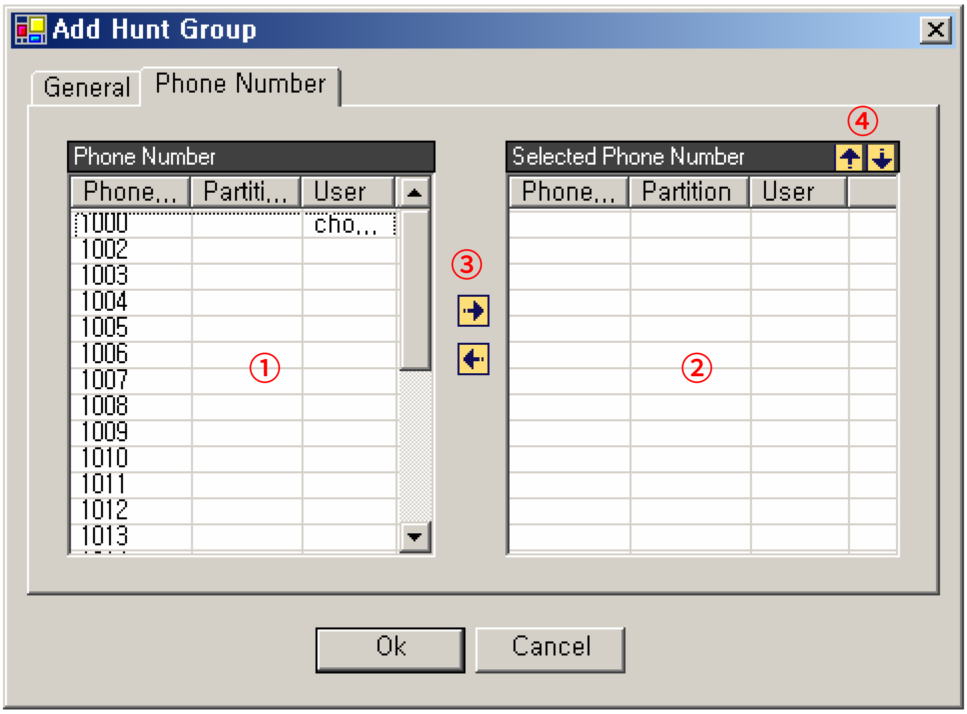

Figure 2-122 Hunt Group – Phone Number Properties

Table 2-81 Hunt Group – Description of Phone Number Properties

Field	Description
1	This is the list of the phone numbers which is not included in Hunt Group
2	This is the list of the phone numbers included in Hunt Group
3	Delete or register the phone number from or to the Hunt Group
4	Adjust the priority level of the phone numbers registered to Hunt Group

Configuring Pickup Group

Pickup allows the phone user to redirect a call that is ringing on another phone to the phone user's own phone so that the phone user can answer the call. To configure Pickup, a Pickup Group must be created first, then the phone number to be configured and assigned to the Pckup Group

The Pickup can be performed by the input of '**' of the user's phone, then the pick up is possible under the same name of the Pickup Group. Inputting a Pickup Group number of another group enables the phone user to answer a call ringing on a phone in another group.

Pickup Group can be carried out as Call Management > Supplementary > Pickup Group on the tree menu on the left side of the screen.

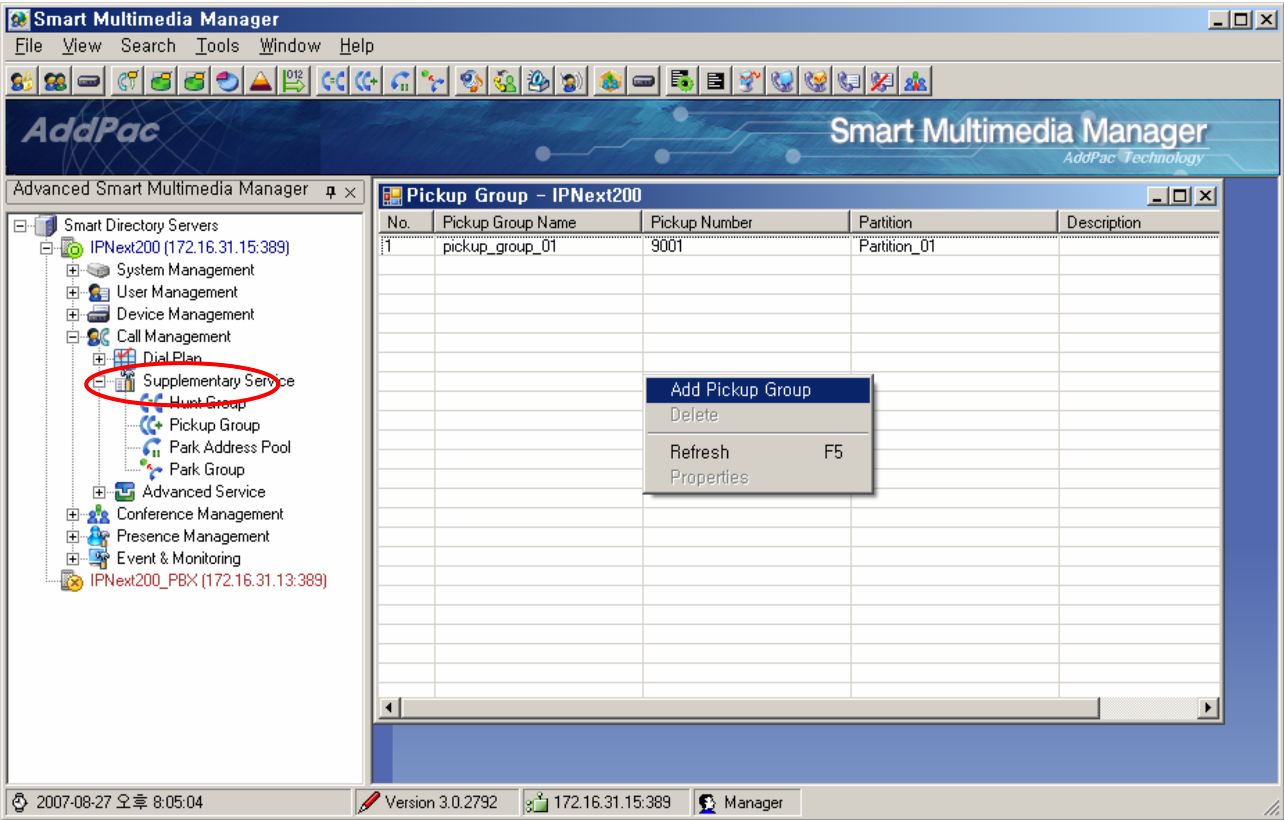

Figure 2-123 Pickup Group Screen

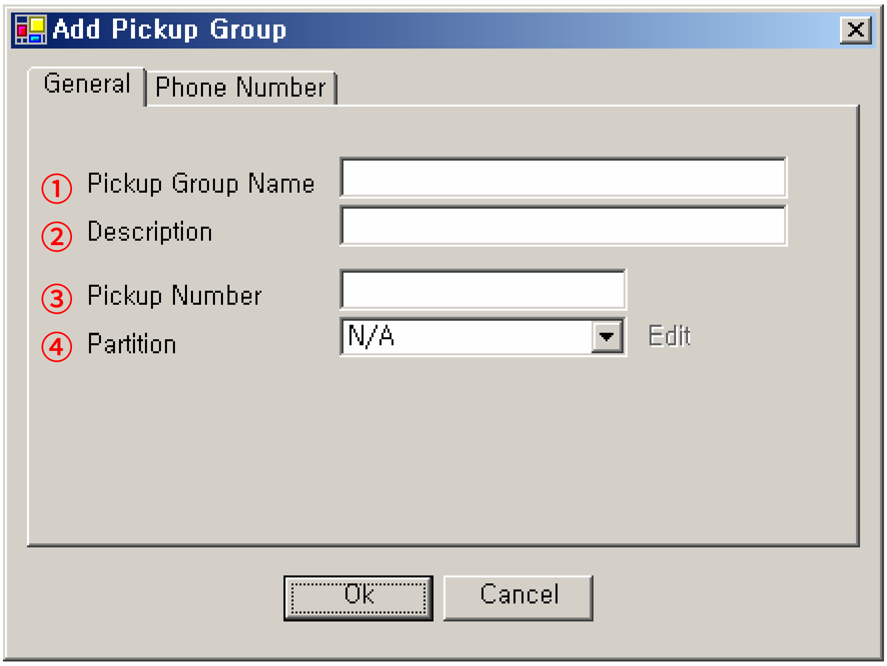

Figure 2-124 Pickup Group – General Properties

Table 2-82 Pickup Group – Description of General Properties

Field	Description
1	Register a name of a Pickup Group
2	Register a description of a Pickup Group
3	Register the Pickup Number. This number can be used to answer a call ringing in another group.
4	Select an option for the Partition to be applied to the Pickup Number. The pickup is possible for the terminal with the Partition Access List including the Partition when the phone user wants to answer a call by using the Pickup Number. When the Partition is set to 'N/A' for no selection, it allows the access to the all. In other words, any terminal with Partition Access List can access.

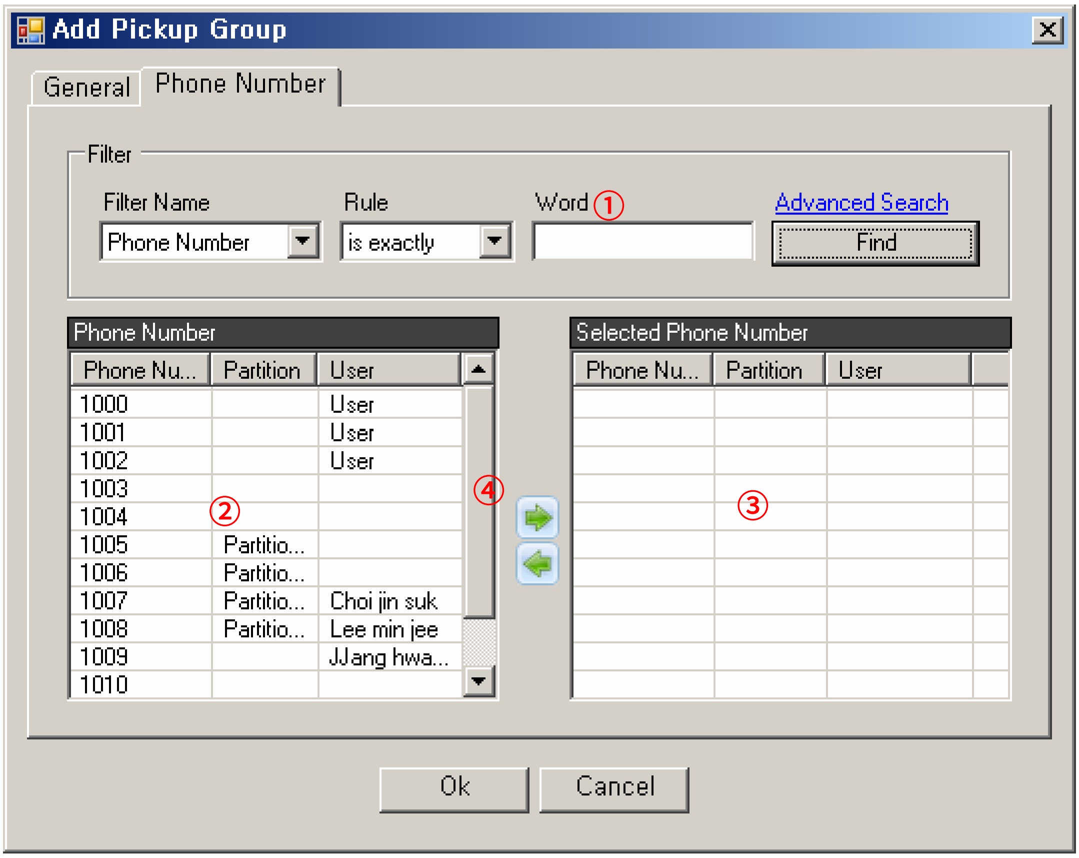

Figure 2-125 Pickup Group – Phone Number Properties

Table 2-83 Pickup Group – Description of Phone Number Properties

Field	Description
1	Search the telephone number to be registered to Pickup Group
2	This is the list of the phone numbers not included in the present Pickup Group.
3	This is the list of the phone numbers included in the present Pickup Group.
4	Delete or register the phone number from or to Pickup Group

Park Address Pool Configuration

Park is a function allowing the phone user to park (temporary store) a call and then retrieve the call by using another Phone. Park can be useful if the phone user wants to transfer a call from the phone to a phone in another location which is not definite. The number can be used from 0 to 9 and assigning a range of the numbers is possible by using the Brackets ([]). The Brackets indicate a Range. The Range is the character sequence placed between the Brackets and only the number from 0 to 9 can be used.

For example, 10 Park numbers can be created by setting 800[0~9], ranging from 8000 to 8009.

To enable Park function for use, push the Hook Flash Button, then press '*71' then IPNext PBX shows a Park number.

When the phone user wants to move from one location to another or notify the corresponding Park number to the other user and connect the call, the phone user can be connected to the other user ,who is parked, by dialing to the corresponding Park number.

Park Address Pool can be carried out as Call Management > Supplementary > Park Address Pool on the tree menu on the left of the screen.

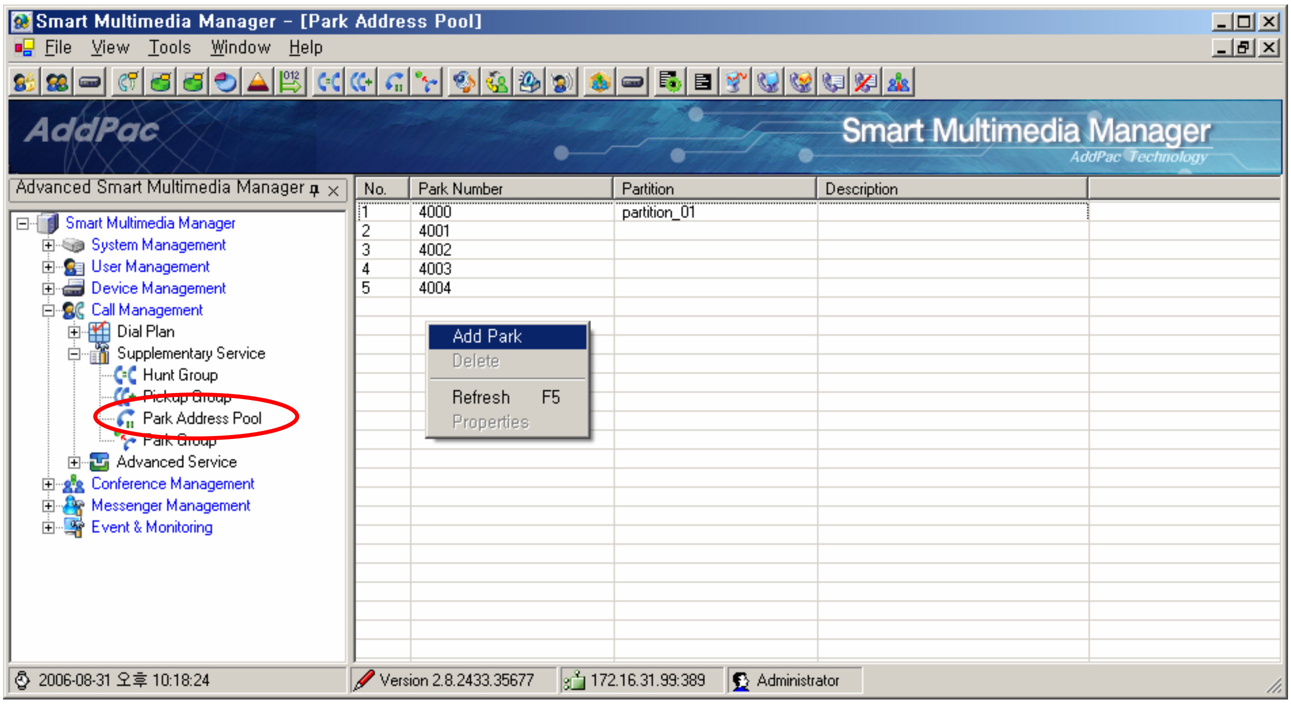

Figure 2-126 Park Address Pool Configuration

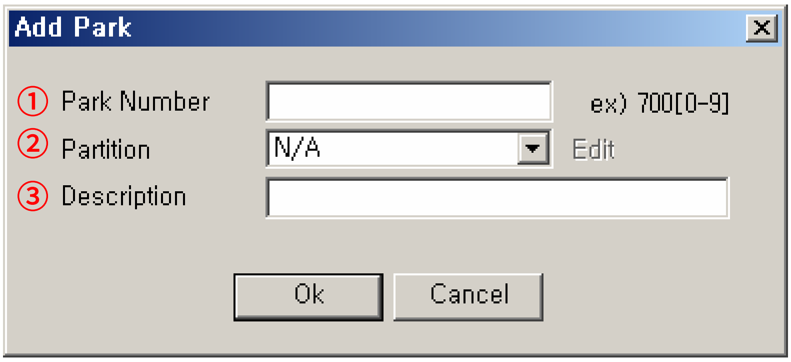

Figure 2-127 Park Address Pool Properties

Table2-84 Description of Park Address Pool Properties

Field	Description
1	Register a Park Number. One number can be assigned or many numbers can be assigned by using the bracket [ ]. For instance, when 800[09] is set, 10 park numbers have been assigned from 8000~8009.
2	Select a Partition (optional)
3	Register a description of the Park Address Pool.

Park Group Configuration

This is a function allowing the phone user to park a particular call to a logical group by presuming any terminal may belong to any logical group (such as technical support, receptionist group and etc.).
For example, while a receptionist is busy on the line for the call made on the general directory number, the call can be parked by assigning to one of the groups such as sales group or technical support group.

Park Group can be carried out as Call Management > Supplementary > Park Group from the tree menu on the left.

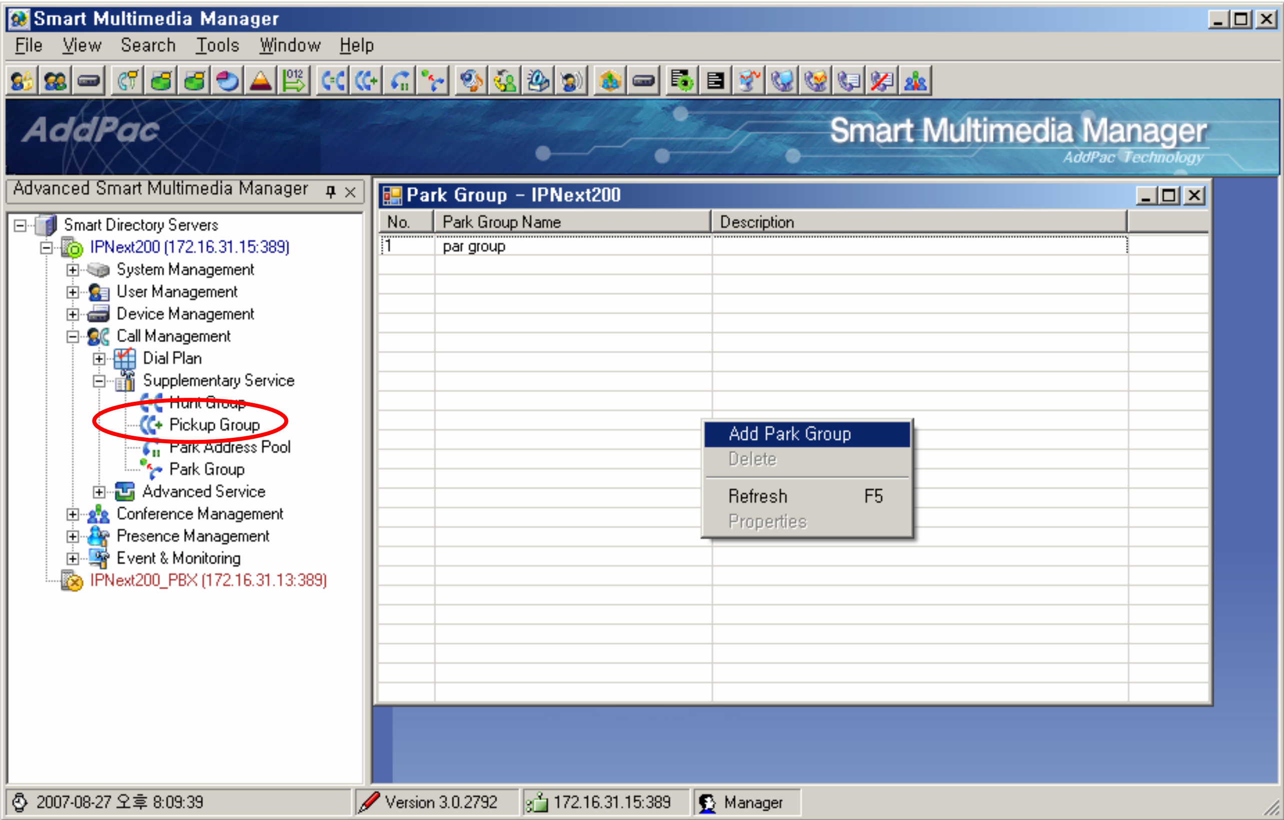

Figure 2-128 Park Group Screen

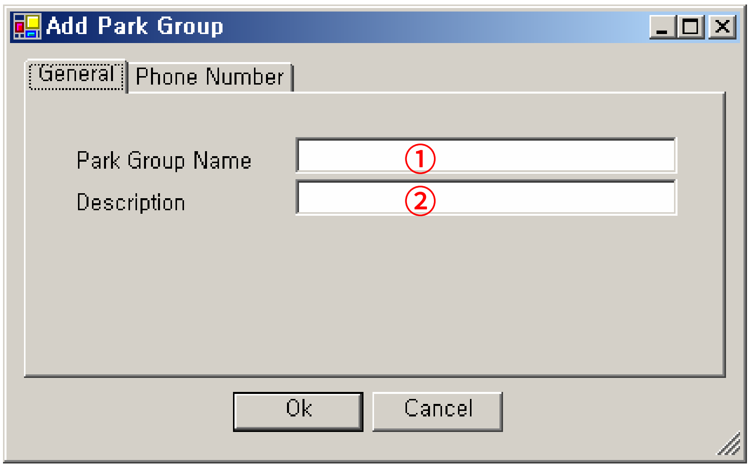

Figure 2-129 Park Group – General Properties

Table2-85 Park Group – Description of General Properties

Field	Description
1	Register a name of a Park Group
2	Register a description of the Park Group

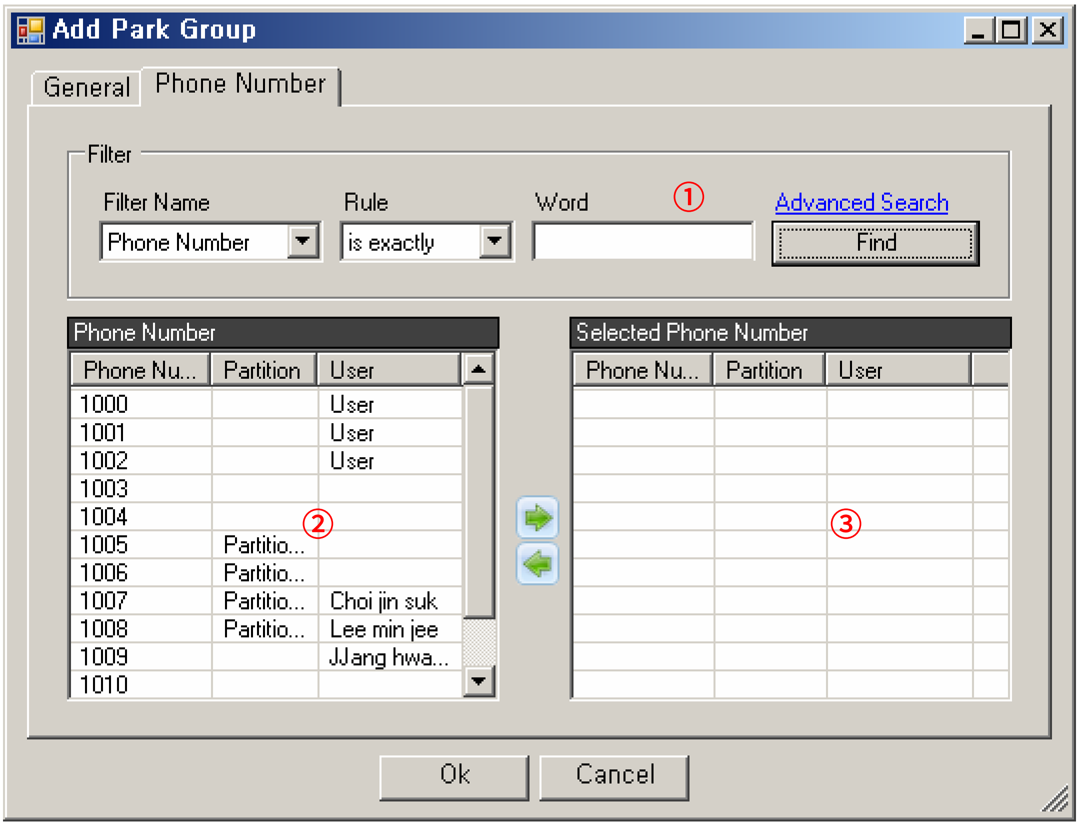

Figure 2-130 Park Group – Phone Number Properties

Table 2-86 Park Group – Phone Number Properties Description

Field	Description
1	Search the phone number, to be registered to Park Group, by conditions
2	This is the list of the phone numbers not included in the present Park Group.
3	This is the list of the phone numbers included in the present Park Group.

Call Management – Advanced Service

Music & Announcement Service Configuration

This configuration is to enable/ disable MoH (Music on Hold), dial tone and other voice announcement services.

The service can be performed as all Management > Advanced Service > Music & Announcement Service which is shown in the below figure.

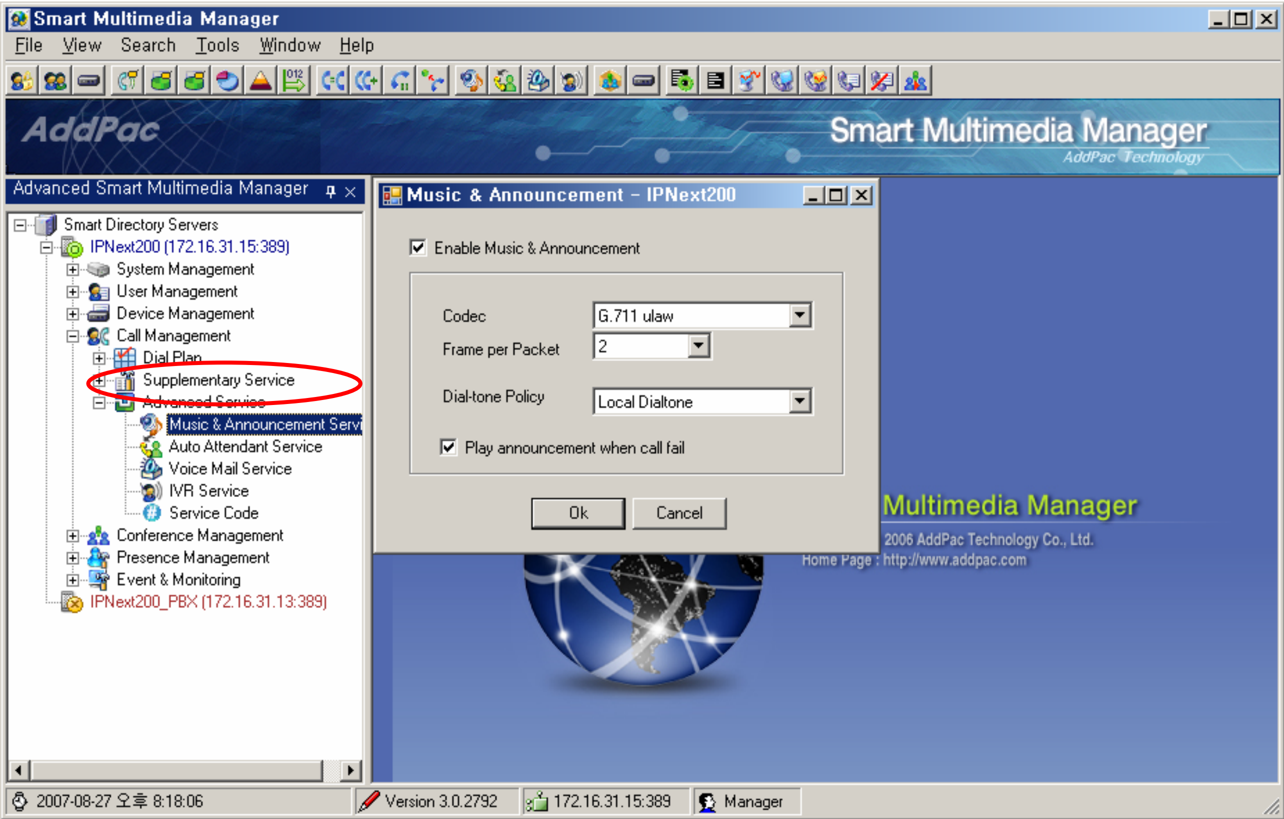

Figure 2-131 Music & Announcement Service Screen

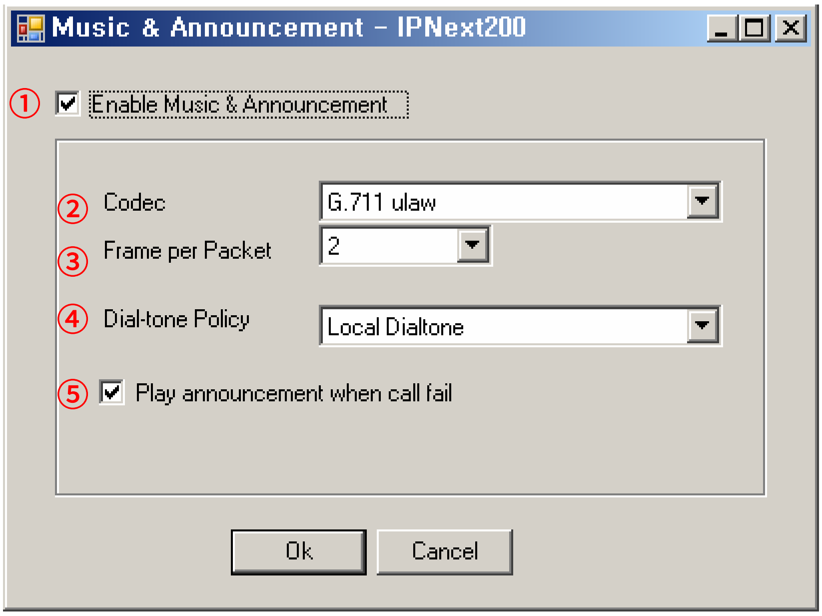

Figure 2-132 Music & Announcement Service Properties

Table 2-87 Description of Music & Announcement Service Properties

Field	Description
1	Enable/ Disable Music Announcement service. When the service is connected to IPNext PBX, the configured codec can provide a Dial-tone.
2	Select a type of codec matching with VoIP setting (G.711 A/Ulaw, G.729)
3	Select a number for Frame/ Packet.
4	Select a policy for the dial-tone which applied to IP Phone system supporting SSCP. - Local Dial tone: A dial tone provided from IP Phone while the phone is hooked off. - Remote Dial tone: A dial-ton provided from the Call Manager while the phone is hooked off - Local + Remote Dial tone: When the IP phone is hooked off, a dial-tone is provided from the IP Phone first, then the dial-tone is replaced with the one provided from the Call Manager.
5	When a call is failed, play announcement

Configuring Auto Attendant Service

This is a type of service used for the settings of the general office. When a call is made to a company, the caller listens to the company introduction through IVR service, then the caller may enter the corresponding extension to make a call. If the caller doe not know the extension, he/she can press ‘0’ to be connected to a receptionist. The IVR service provided in this example is the simple one which can be provided at the default setting. For more sophisticated IVR service, IVR Editor can be used to edit a scenario.

The service can be performed as Call Management > Advanced Service > Auto Attendant Service shown in the Figure below.

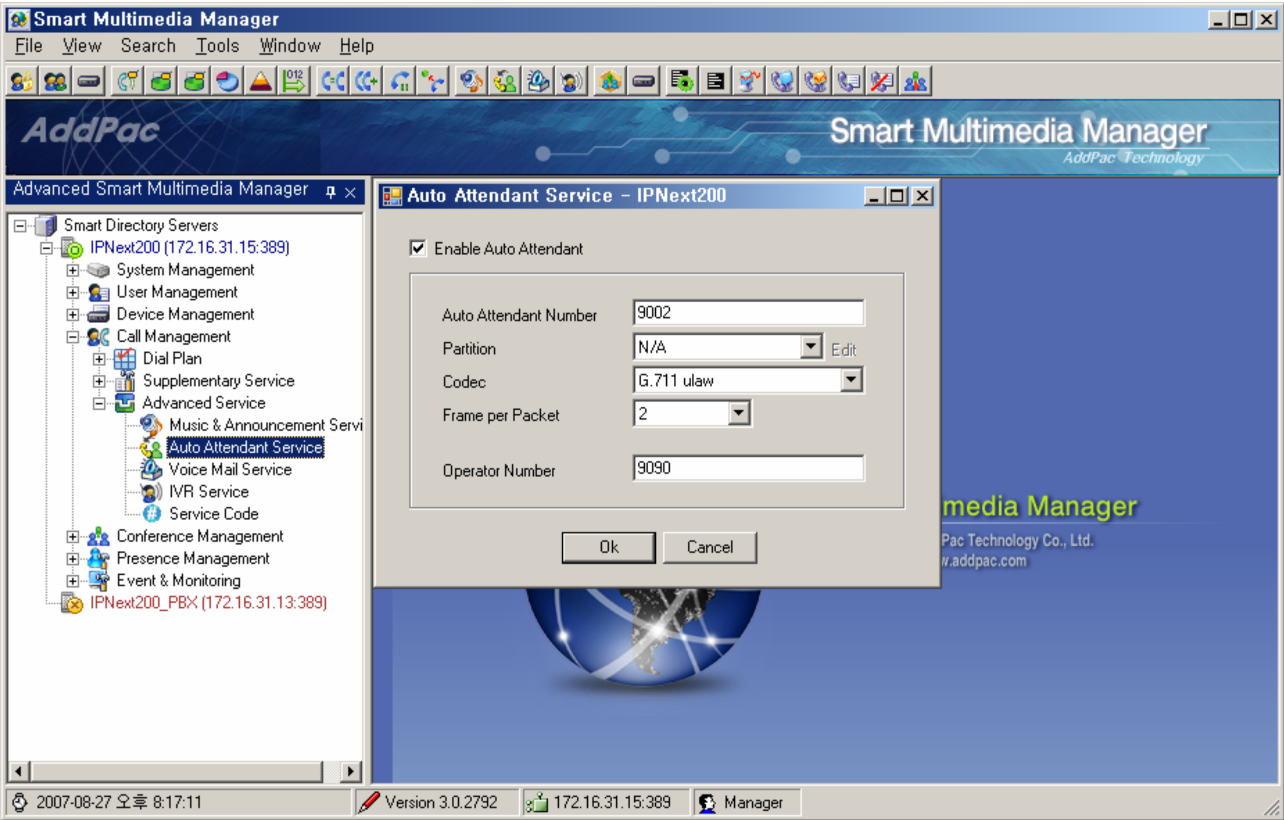

Figure 2-133 Auto Attendant Service Screen

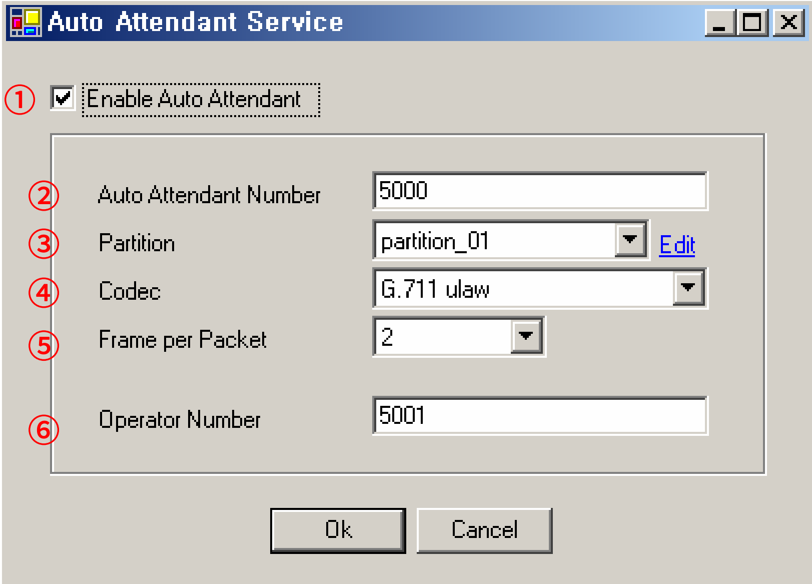

Figure 2-134 Auto Attendant Service Properties

Table 2-88 Description of Auto Attendant Service Properties

Field	Description
1	Enable/disable Auto Attendant service.
2	Set up a number for the Auto Attendant service.
3	Select a Partition (optional)
4	Select a type of audio codec to be used for the Auto Attendant service. The supported codecs are G.711A/Ulaw and G.729
5	Select a number of Frame per Packet
6	Set up an extension number to be connected when '0' is pressed from the Auto Attendant service.

Configuring Voice Mail Service

Voice Mail service allows the phone user, who is registered to the IPNext PBX, to listen to the voice mails which are left during the user's absence or while the user's line is busy.

The service can be carried out as Call Management > Advanced Service > Voice Mail Service shown in the Figure below.

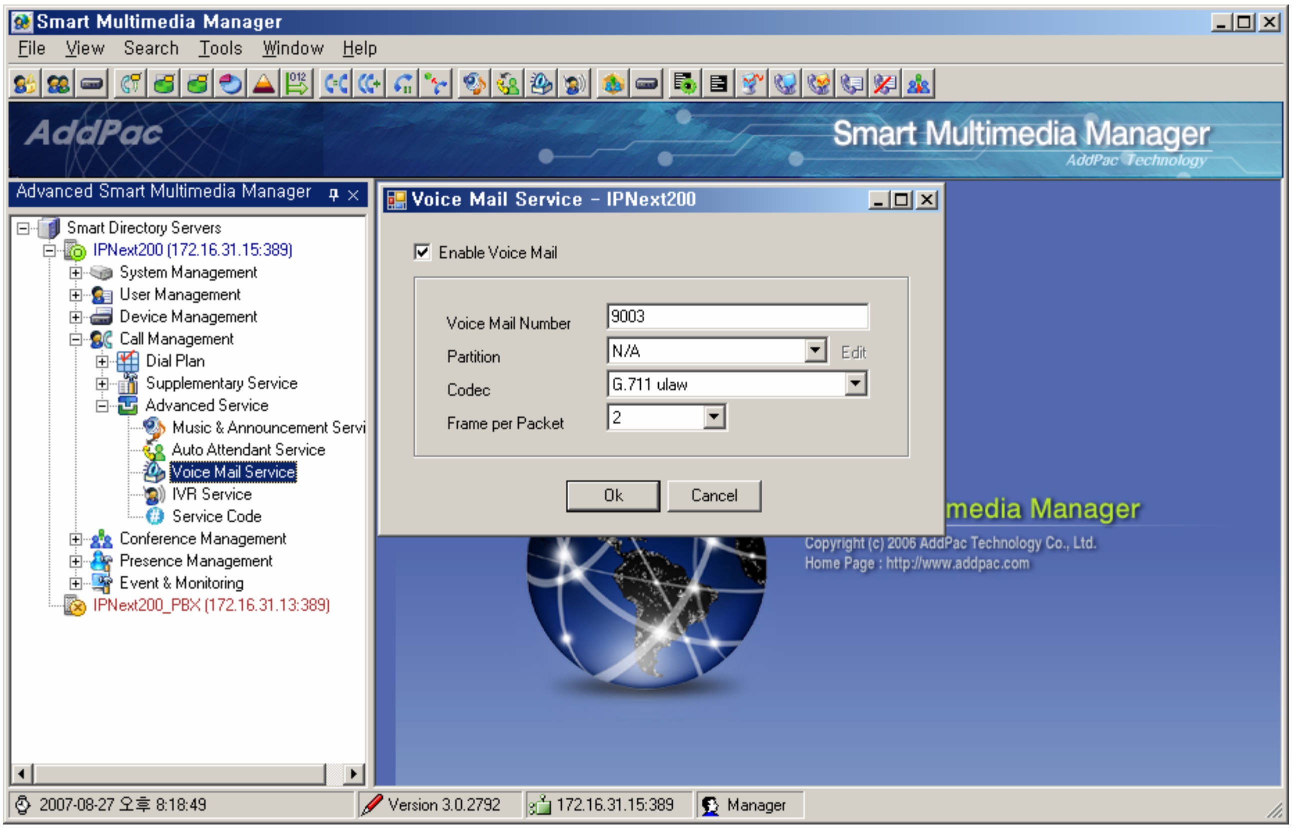

Figure 2-135 Voice Mail Service Screen

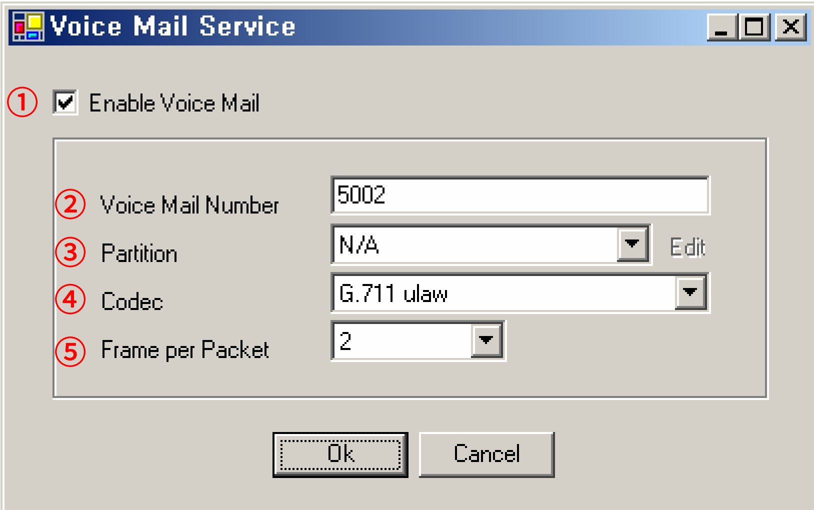

Figure 2-136 Voice Mail Service Properties

Table 2-89 Description of Voice Mail Service Properties

Field	Description
1	Enable/disable Voice Mail service.
2	Set up a phone number for the Voice Mail Service.
3	Select a Partition (optional).
4	Select a type of audio codec to be used for the Voice Mail service. The supporting codecs are G.711A/Ulaw and G.729
5	Set up a number of Frame/Packet.

Configuring IVR Service

Configuration of IVR(Interactive Voice Response) Service is completed when an administrator makes a scenario by using Smart IVR Editor and registers to IVR Service of SMM. For more detailed description of Smart IVR Editor, please refer to the manual of Smart IVR Editor.

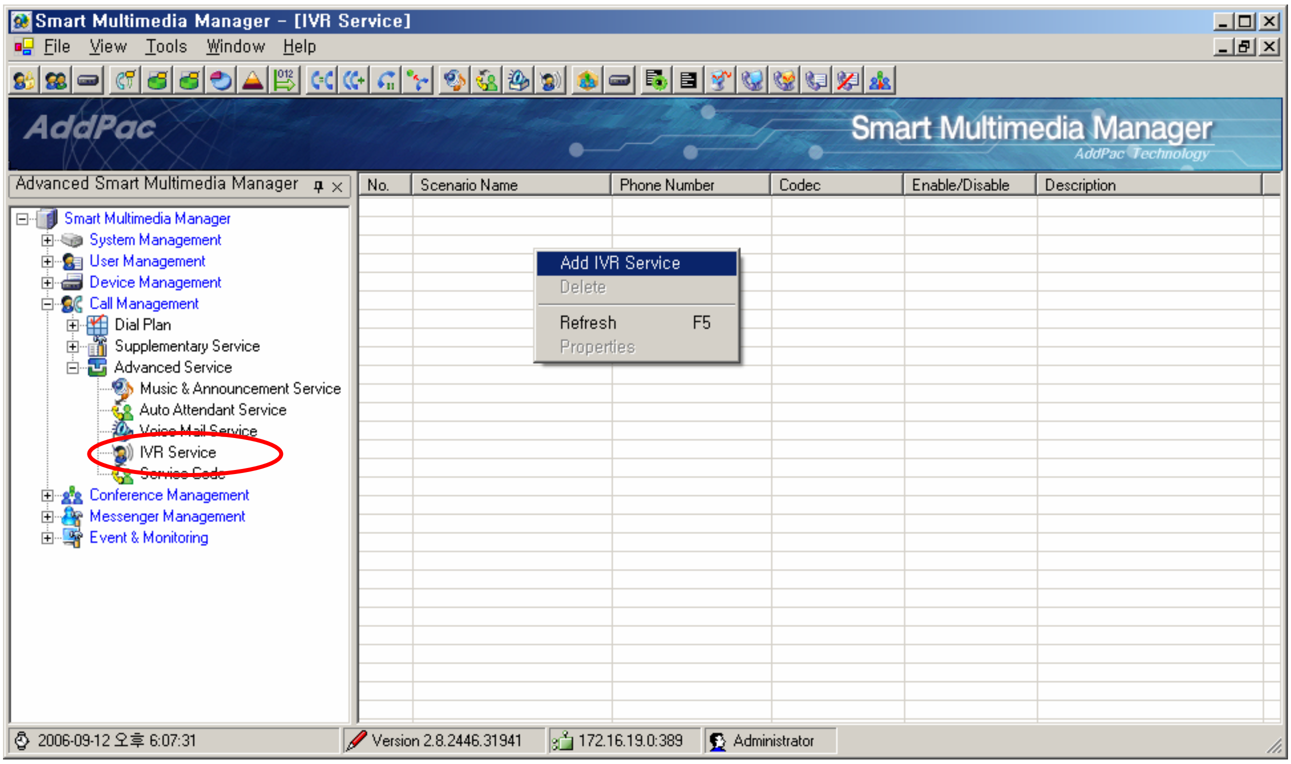

Figure 2-137 IVR Service Screen

The Figure 2-117 above is the screen to perform Advanced Service > IVR Service. By pressing the right button of the mouse, a new registration can be made.

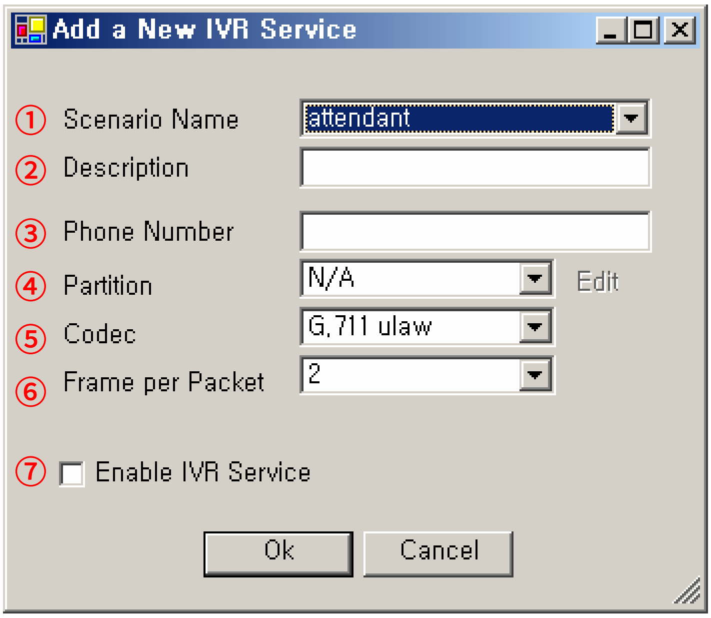

Figure 2-138 IVR Service Properties

Table 2-90 IVR Service Properties

Field	Description
1	Select a name of scenario. After the scenario is created by Smart IVR Editor, then IPNext PBX is configured and only the registered scenario list can be selected,
2	Register a description of IVR Service
3	Set up a number of IVR Service
4	Select a Partition (optional)
5	Select a type of audio codec
6	Set up a number of Frame/ Packet.
7	Select an option whether to enable/ disable IVR Service

Service Code Configuration

Service Code Configuration can assign 2 digit numbers at maximum starting from ‘#’ or ‘*’, as a function to set up the supplementary service numbers, which are supported from IPNext PBX. This configuration is to enable/ disable the supplementary service and more useful to the general terminals which are not supported by SSCP than the IP phone using Softkey supported by SSCP.

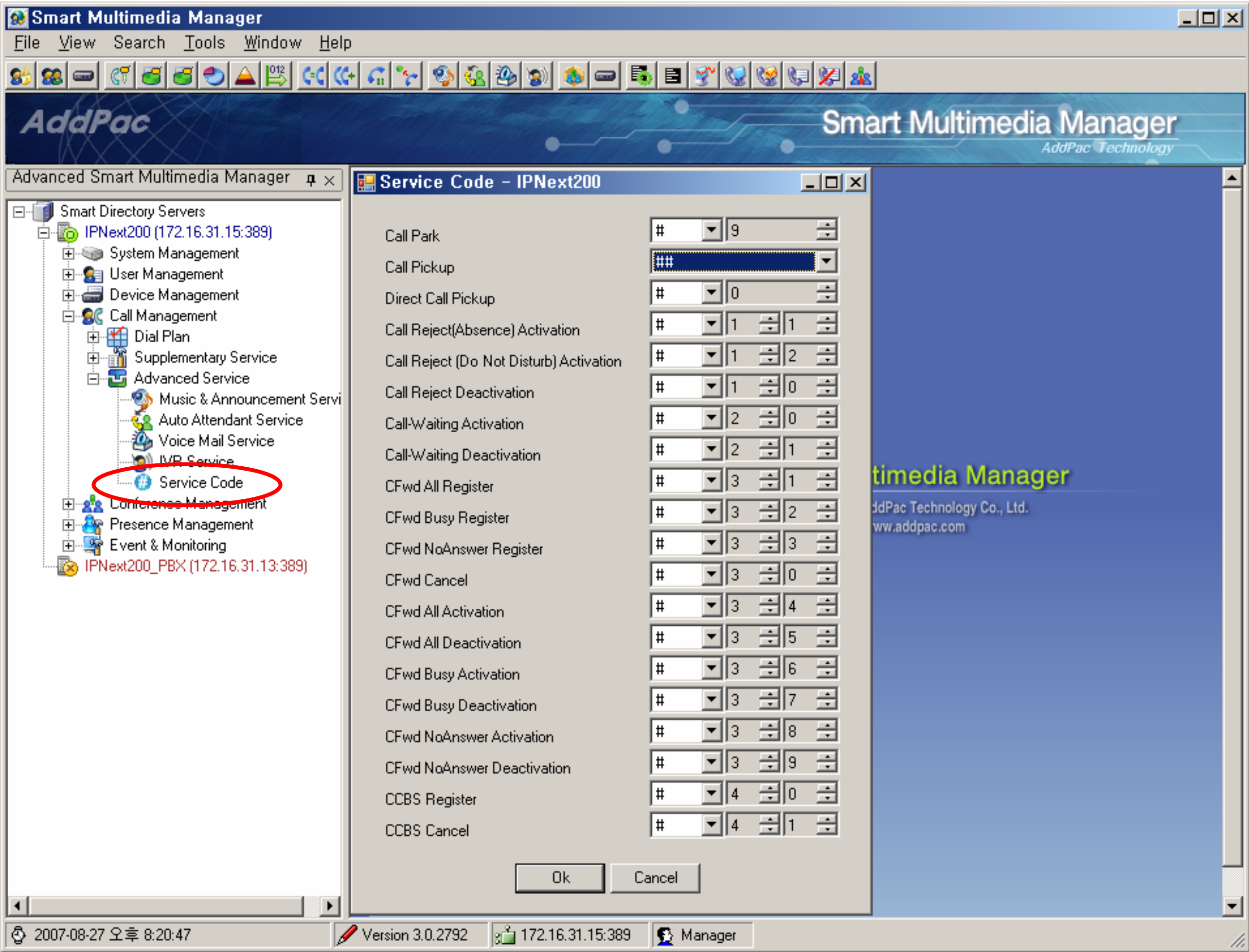

Figure 2-139 Service Code Performing Screen

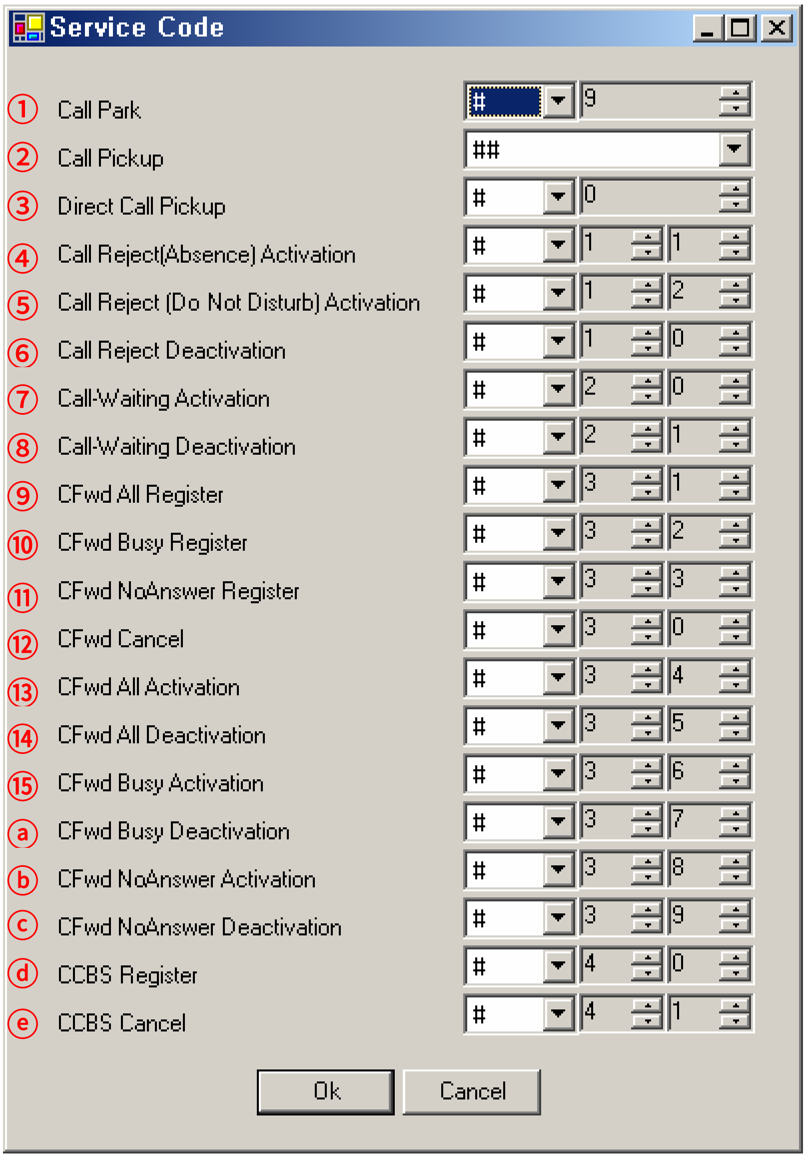

Figure 2-140 Service Code Properties

Table 2-91 Service Code Properties Description

Field	Description
1	Set up the code for Call Park
2	Set up the code for Call Pickup
3	Set up a call to the code which does not belong to the same group for Call Pickup
4	Set up the code for the phone user’s absence
5	Set up the code for rejecting the incoming call
6	Deactivating the code from Call Reject (Absence and Do not Distub)
7	Set up the code to activate Call Waiting
8	Set up the code to deactivate Call Waiting
9	Set up the code to register the forwarding number unconditionally
10	Set up the code to register the forwarding number for the busy line

11	Set up the code to register the forwarding number during the phone user's absence
12	Cancel the code for Call Forwarding
13	Set up the code to activate Call Forwarding unconditionally
14	Set up the code to deactivate forwarding unconditionally
15	Set up the code for Call Forwarding when the line is busy
A	Set up the code to deactivate Call Forwarding when the busy line
B	Set up the code to activate Call Forwarding when the phone user is absent
C	Set up the code to deactivate Call Forwarding for absence
D	Set up CCBS[3]
E	Cancel CCBS

[3] **CCBS**(Termination of Calls to Busy Subscriber) - If A makes a call to B and B is busy on the line, CCBS allows A to connect B after B is finished with the call without calling back

Conference Management

MCU Operation and Configuration

MCU(Multipoint Conferencing Unit) Module is device connecting conference calls.
The type of conference call can be categorized as voice and video conference calls depending on a type of media. IPNext PBX is equipped with a module[4] for voice conference call and an additional device[5] is required for video conferencing call.

The following kinds of conference call can be supported depending on its methods:

- **Ad Hoc Conference Call**
  This is a kind of conference call used when the conference call is not expected.
  Let's assume that a user A's extension number is 1001 and another user B's number is 1002 and the other user C's number is 2001.
  A makes a phone call to B by dialing 1002. While A and B talks to each other on the phone, they found that they need to call to C. At this time, A or B can invite C to the conference call by pressing '+2001+', then C can join in for the 3-Party conference call.
  Certainly, one of these 3 users can add one more user to make the 4-Party conference call by pressing '+extension number+'.

- **Dial-Out Conference Call**
  Dial-Out Conference Call connects all the phone users who are assigned with a phone number of a shared phone line. By dialing the number, the users can join in the conference. The conference number can be created by SMM for Dial-Out Conference, then a list of users, who can participate in the corresponding conference number, is to be configured.
- For instance, the conference number of "#1500" is set for each member of the sales team 1 and each member's telephone number is mapped, any member from the sales team 1 can dial '#1500' to have a conference call with the mapped members.
  In case the number of the users are limited to 3-party, a conference can be established only when other 2 parties respond to the conference call with an exception of the user making a request.

- **Ad Hoc Dial-Out Conference Call**
  Ad Hoc conference allows the phone user to initiate a conference by calling each participant. Add Hoc Dial-Out Conference Call creates a conference number which can be available all the time by using SMM. To start a conference, the phone user may enter the conference number, then enter '*'

4 There
5 There are MCU-specific product such as the AddPac MC1000 and MCU imbedded in terminal product such as VC2000.

to divide the phone user's number to be called, then enter '#' to complete after entering the user's telephone number (ex: Conference Number Phone User 1*Phone User 2*Phone User3 #)

- **Meet Me Conference Call**
  Meet Me Conference Call allows the phone user to call a predetermined number a t a schedule time to host or join a conference. It is different from Dial-out Conference or Ad Hoc as it does not cancel out the conference session because of no participants and the user who wants to participate can join the conference within a scheduled time. The conference session can be closed by an administrator or Chair, the user can participate in the conference through a particular phone number or range of phone numbers. An access to Meet Me Conference is limited by a Partition of the Conference Number and the access can be granted by a password.

The MCU module of IPNext PBX enables 4-party Voice Conference for 2 sessions[6].
When a separate MCU device is added, besides MCU modules of IPNext PBX (ex. AP-MC1000), the MCU device needs to be configured as it is shown in the following Figure.

As it is shown in the figure below, a new MCU Server registered (Device Management > Servers > Add MCU Server) as to be added from the device management function.

6 For some specific product models

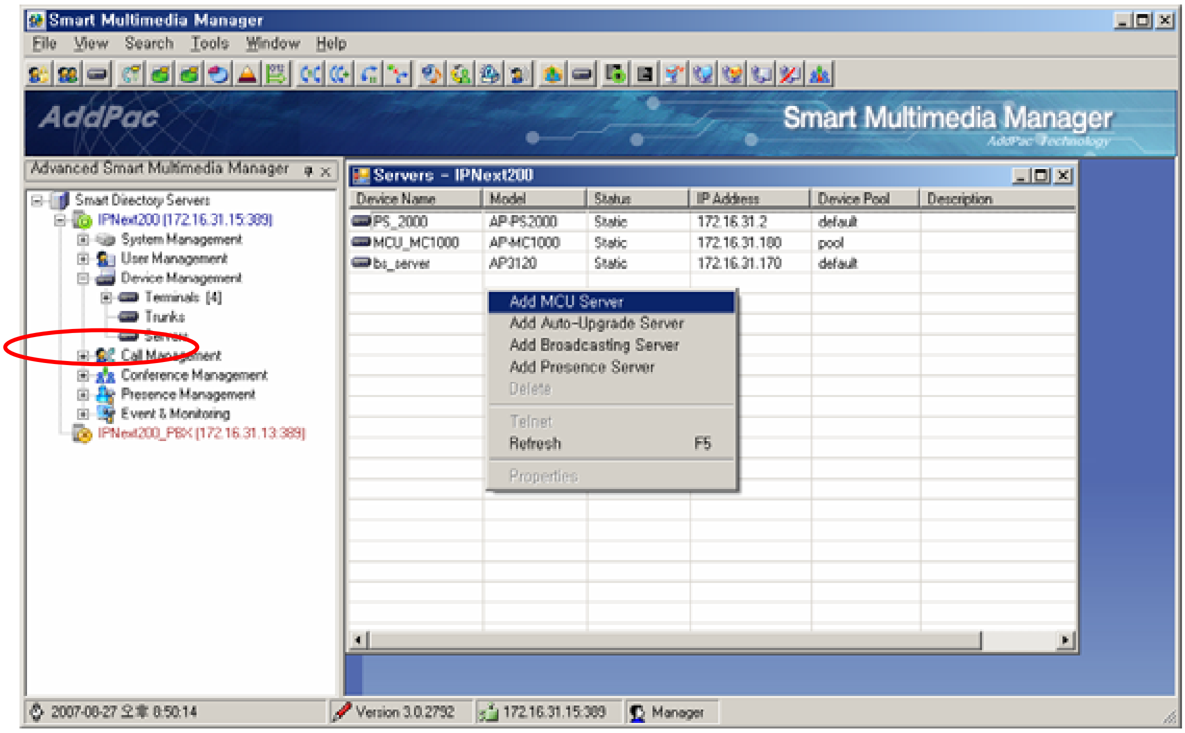

Figure 2-141 Registration Screen 1 for MCU Server

After registration of a MCU server in the section of Server, the registered MCU servers can be listed by selecting MCUs Menu (Conference Management > MCUs), as it is shown in the figure below.
By double-clicking the corresponding the MCU server or using the pop-up menu, Datata Registration can be performed.

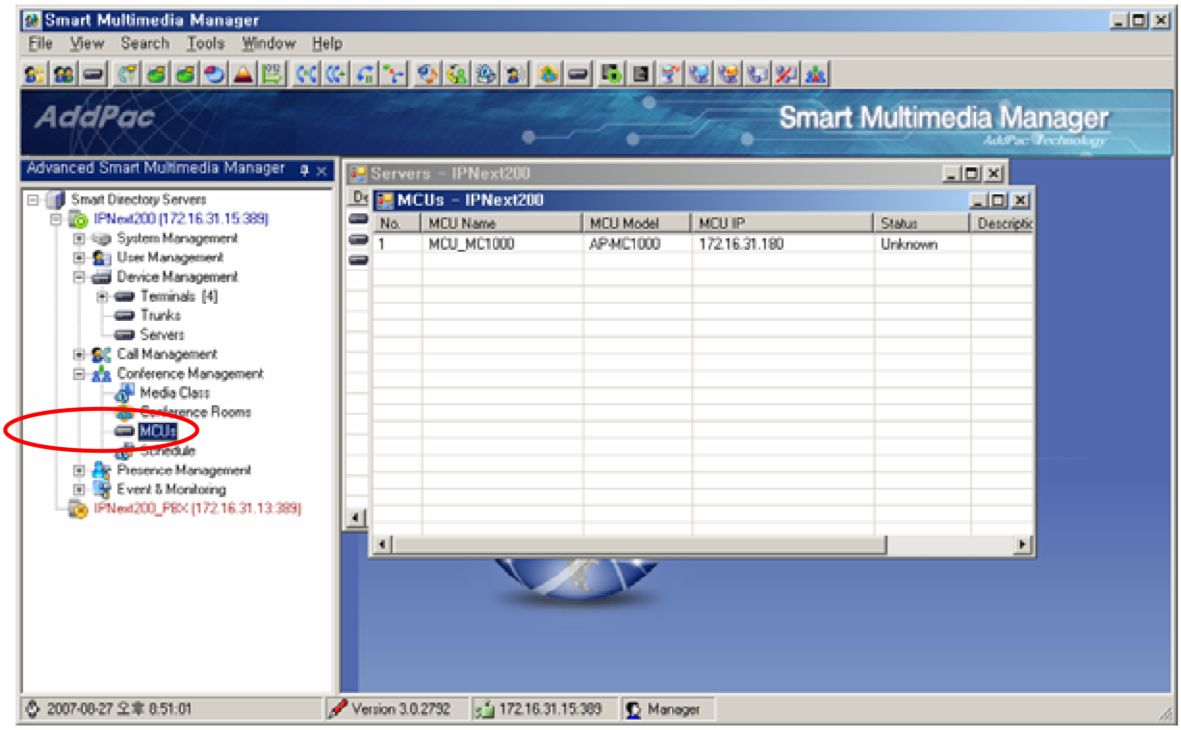

Figure 2-142 Registration Screen 2 for MCU Server

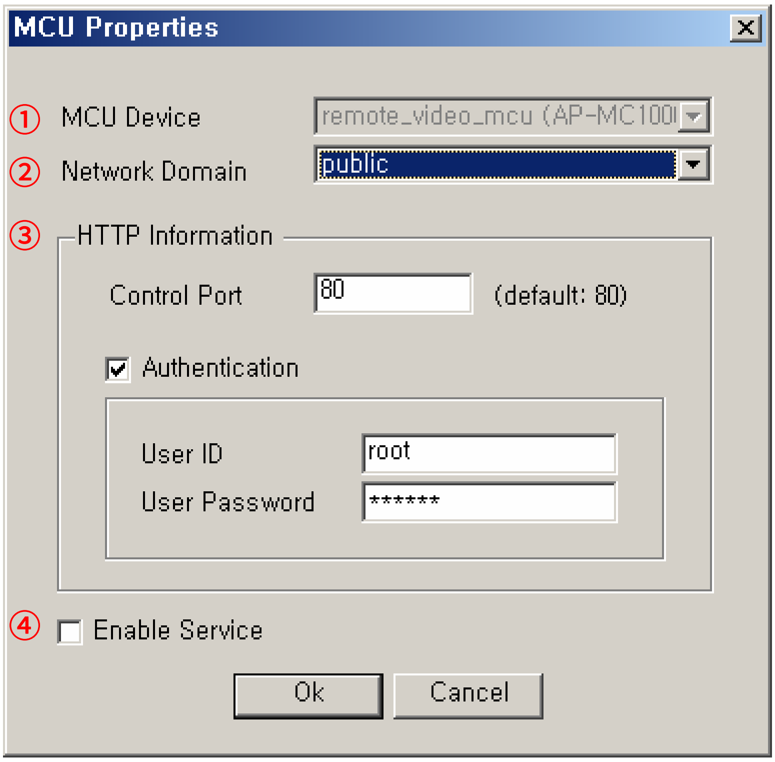

Figure 2-143 MCU Server Properties Screen

Table 2-92 MCU Server Properties Description

Field	Description
1	A name of MCU Server to be displayed (the name can not be changed from MCU Properties) The name can be changed from Properties of Server section (Device Management > Servers)
2	Select Network Domain to be used for connection with MCU server
3	Configure the account access information and HTTP access port to control MCU server
4	Enable/Disable the service of MCU Server

Configuring Media Class

Media Class is a profile which has been configured for audio and video codec. This Media Class can be selected and applied for configuring Conference Room. The default Media Class is applied for creating a conference room

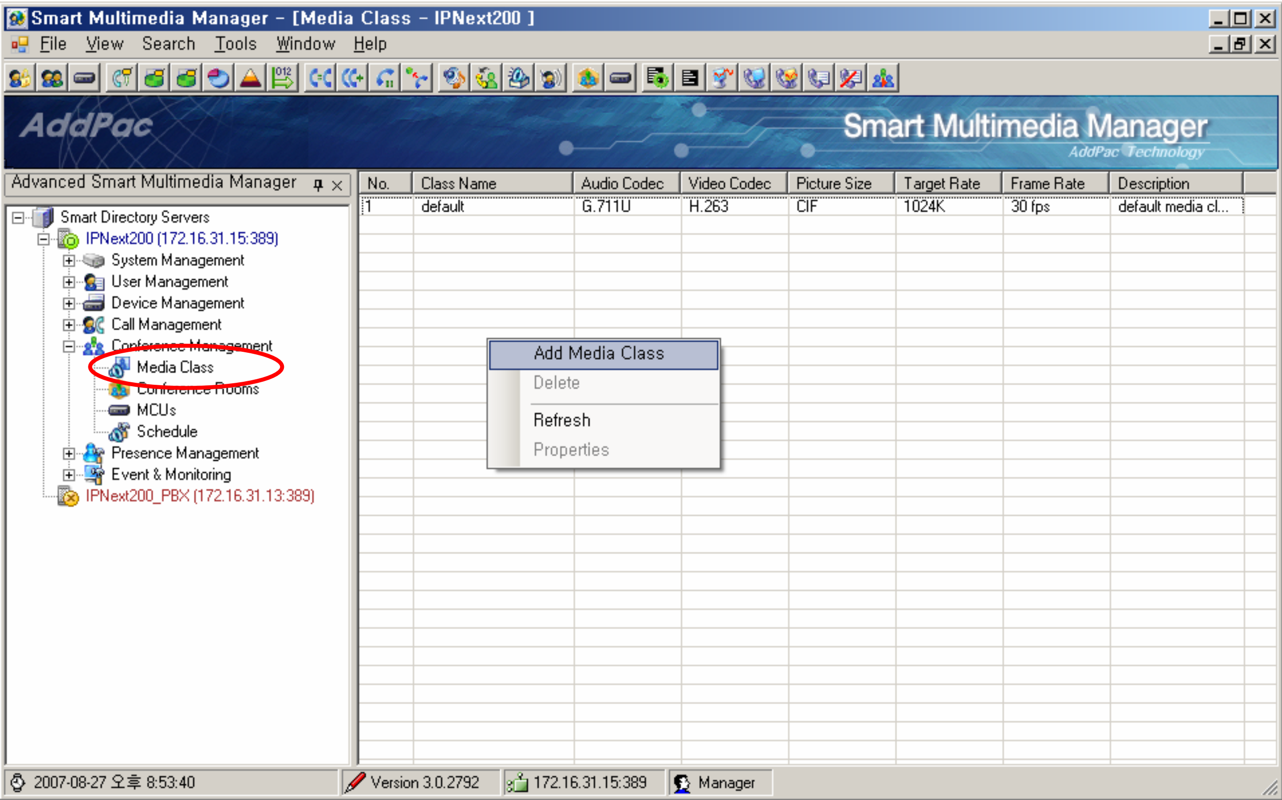

Figure 2-144 Executing Media Class

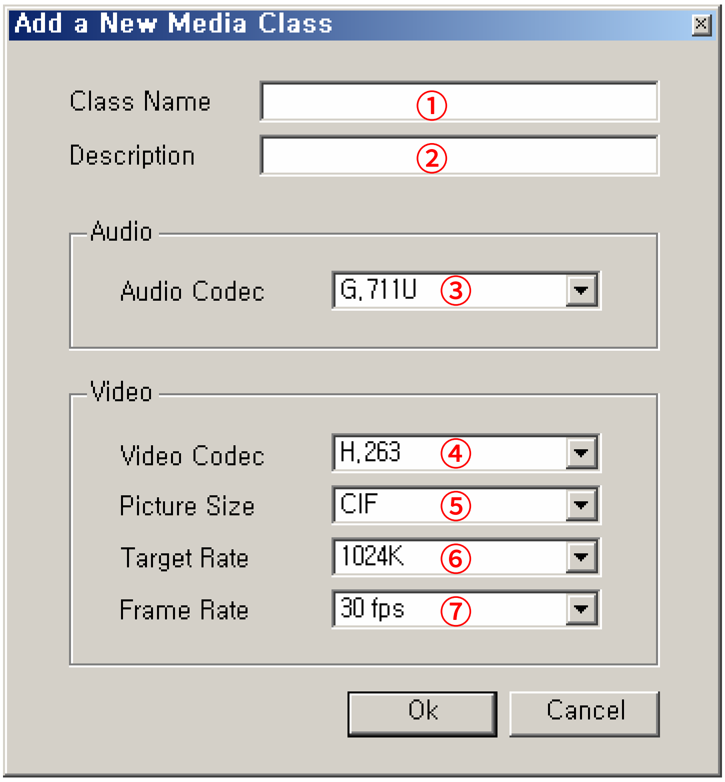

Figure 2-145 Media Class Properties

Table 2-93 Description of Media Class Properties

No.	Description
1	Enter a name of Media Class
2	Enter a description of Media Class
3	Select a type of Audio Codec
4	Select a type of Video Codec
5	Select a picture size for video
6	Select a bandwidth for video
7	Select a frame rate (a number of frames per second)

Conference Rooms Configuration

Conference Room is to register and manage the basic configuration of Ad Hoc Conference and other conference features such as Dial-out, Ad Hoc Dial-out and Meet Me.

Ad-Hoc Conference is basically configured in the default settings, so it does not needs to registered separately. To change the settings, the settings of Ad-Hoc Defaults need to be changed.

The following figure shows the screen of Conference Rooms and Ad-Hoc Defaults Properties

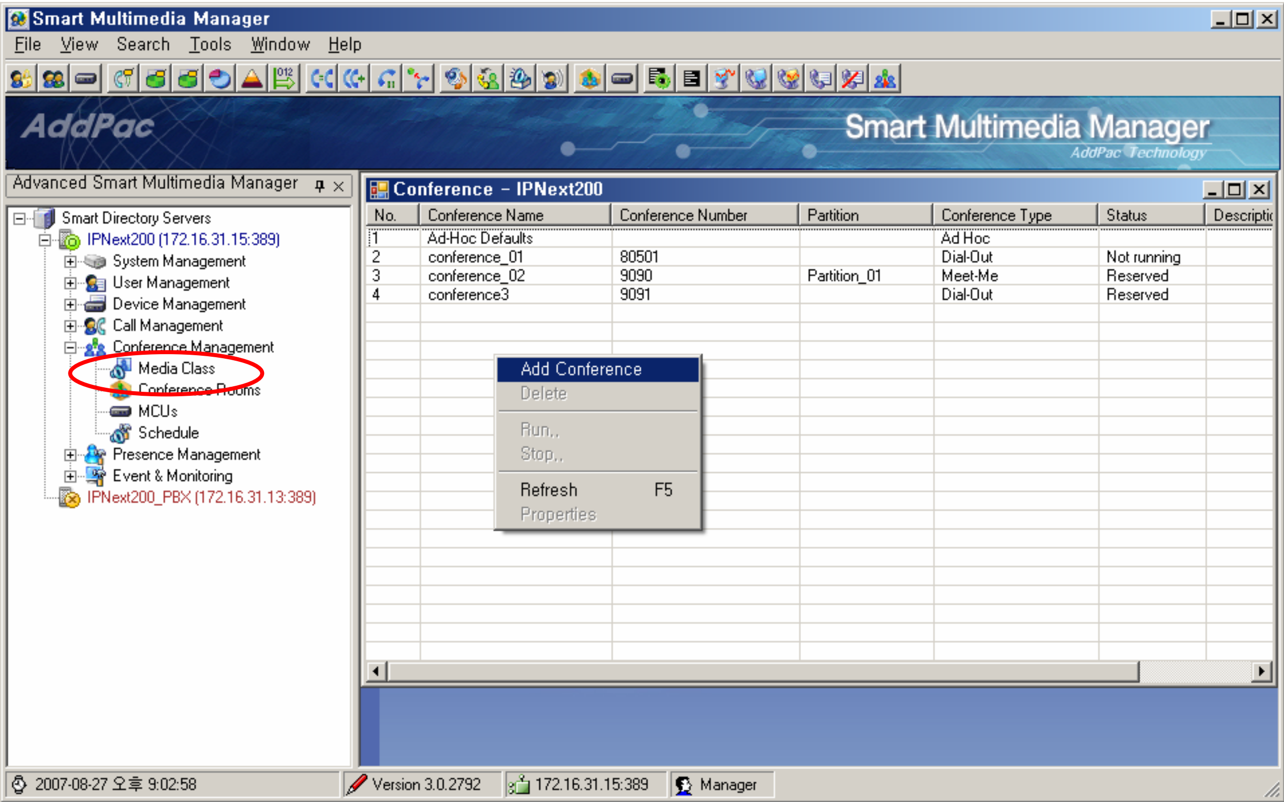

Figure 2-146 Setting screen for Conference Rooms

Configuring Ad-Hoc Defaults s

Settings of Ad-Hoc Defaults are already registered basically when Conference Rooms is performed. Then select, double-click and perform Properties from the pop-up menu

This Conference Room is the configuration used for Ad-Hoc Conference from the system, so it can not be deleted.

①
②

③

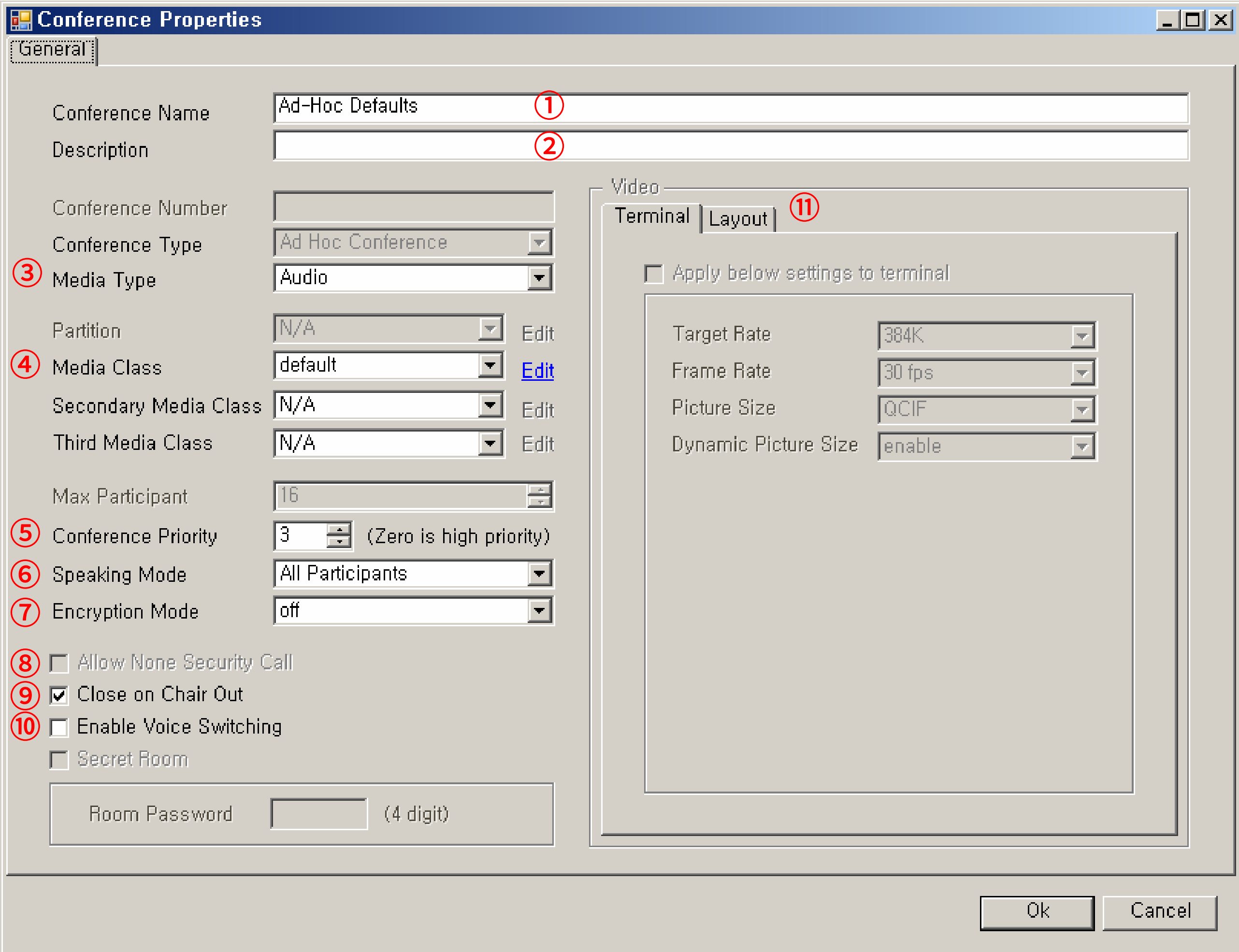

Figure 2-147 Ad-Hoc Defaults Properties

Table 2-94 Description of Registering MCU Server

Field	Description
1	Register a name of Conference.
2	Register a description of Conference
3	Select a type of media (Audio, Video + Audio) for Ad-Hoc Default. According to the type of media is selected, the settings of video can beenabled or disabled.
4	Select Media Class to be applied to Conference Room. Media Class can be set to 3 participants at maximum and 1 must be selected at least. Media Class takes a priority as the fist Media Class does not support, the second one does.
5	Set a priority for the Conference
6	Set a range of participants who can take the floor. - All Participants can speak - All Visible Participants : all the participants displayed on the layout can speak - Floor + Chair : Only the participants with the floor and Chair can speak - Floor Only : Only the participants with the floor can speak

7	Set encryption mode (AES, DES, 3-DES) for video and voice data of Conference
8	Allow the terminals which do not support encryption to participate into Conference
9	This is an option to select whether the conference call is to be terminated when Chair Man, who invites the Conference Call, terminates a call when he/she ends. When the option is selected, Conference Call ends when Chair Man terminated his/her call.
10	When the floor does not exist in Conference, Set the screen for the speaking participant to be stretched automatically
11	This is to configure the settings of video for the media type which is Video + Audio. - Terminal : These are the characteristics of video transmitted from the participating Terminals (optional) - Layout : Configuration of the layout for video

Configuring Dial-Out Conference

Add Conference is configured by a pop-up menu from Conference Rooms Screen.

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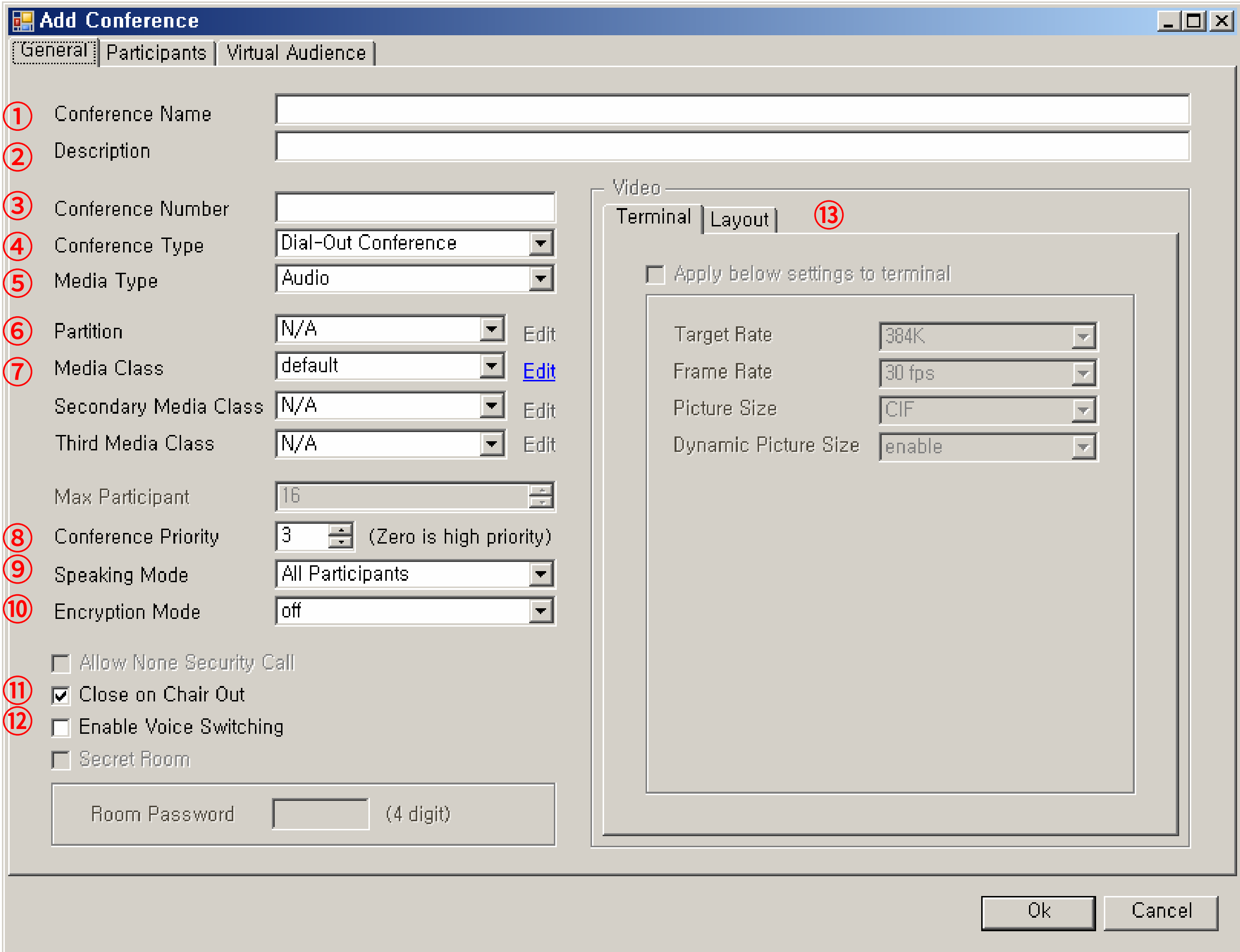

Figure 2-148Dial-Out Conference - General Properties Screen

Table 2-95 Dial-Out Conference – Description of General Properties

Field	Description
1	Register a name of Conference.
2	Register a description of Conference
3	Enter a number of Conference
4	Select a type of conference. In here, select Dial-Out Conference.
5	Select a type of media (Audio, Video + Audio) for Ad-Hoc Default. According to the type of media is selected, the settings of video can beenabled or disabled.
6	Select Partition (optional)
7	Select Media Class to be applied to Conference Room. Media Class can be set to 3 participants at maximum and 1 must be selected at least. Media Class takes a priority as the fist Media Class does not support, the second one does.
8	Set a priority for the Conference
9	Set a range of participants who can take the floor. - All Participants can speak

	- All Visible Participants : all the participants displayed on the layout can speak - Floor + Chair : Only the participants with the floor and Chair can speak - Floor Only : Only the participants with the floor can speak
10	Set encryption mode (AES, DES, 3-DES) for video and voice data of Conference
11	This is an option to select whether the conference call is to be terminated when Chair Man, who invites the Conference Call, terminates a call when he/she ends. When the option is selected, Conference Call ends when Chair Man terminated his/her call.
12	When the floor does not exist in Conference, Set the screen for the speaking participant to be stretched automatically
13	This is to configure the settings of video for the media type which is Video + Audio. - Terminal : These are the characteristics of video transmitted from the participating Terminals (optional) - Layout : Configuration of the layout for video

This is a description of video related settings when Media Type is selected to Video.
The following figure describes the characteristics of video transmitted from the Terminal that participates in a conference. Basically each Terminal has each video characteristic. Without the configuration below, each terminal is to be applied with the saved settings. This Conference Room is to follow the settings below after they are applied.

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④

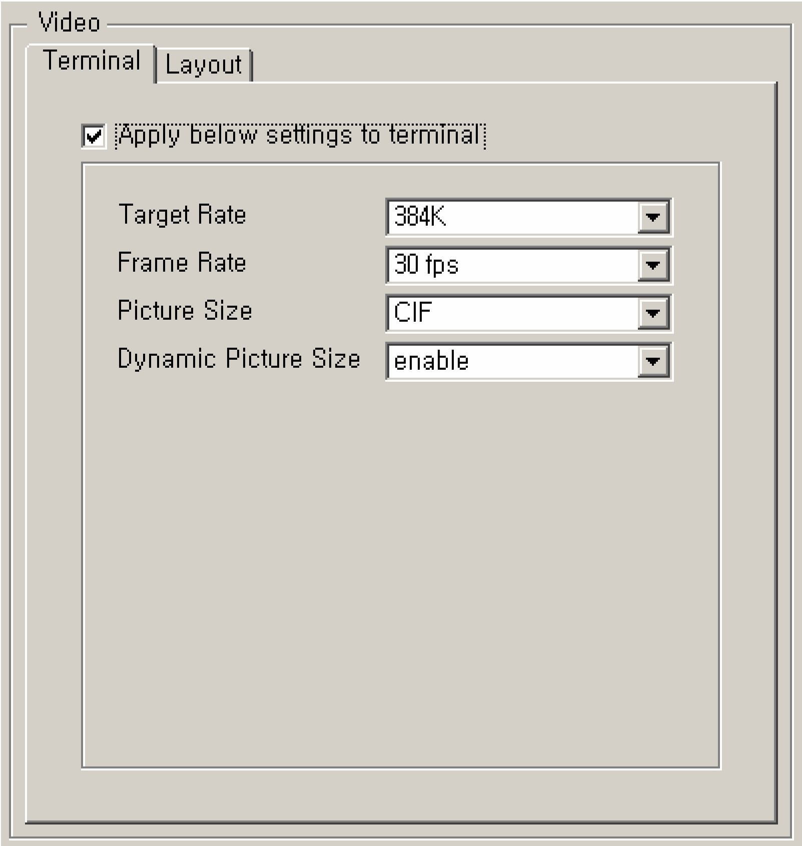
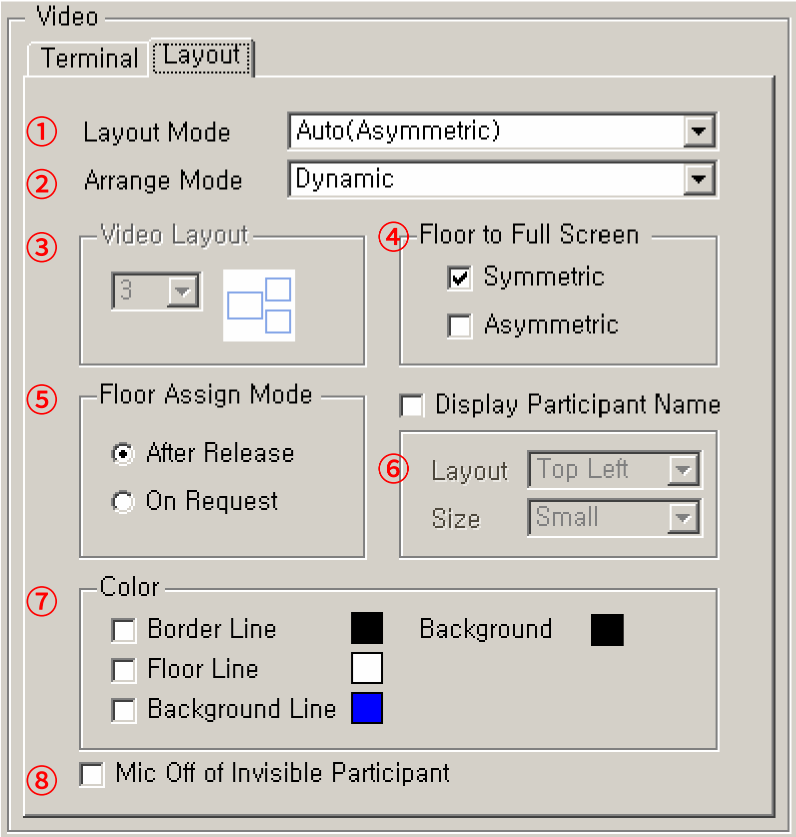


Figure 2-149 Dial-out Conference – Video (Terminal) Settings

Table 2-96 Dial-Out Conference – Description of Video (Terminal) Settings

Field	Description
1	Select an option whether to enable or disable the settings of the video data sent from a terminal to MCU.
2	Select a bandwidth of video.
3	Select a rate of frame (a number of frames per seconds)
4	Select a picture size of video
5	This is to configure whether to request a change of the picture size at maximum for a change of Layout with in the configured Picture Size.

Configuring Video Layout is shown in the following:



:

Figure 2-150 Dial-out Conference – Video(Layout) Settings

Table2-97 Dial-out Conference – Video(Layout) Settings

Field	Description
1	Select a Layout mode for Video - Auto (Symmetric) : Automatically assigned to the symmetric Layout. - Auto (Asymmetric) : Automatically assigned to the asymmetric layout. - Manual : The settings of the user-dependent Layout
2	Select a mode of Arrange for video - Dynamic : The location of the participant changes when he or she has the floor - Fixed : The location of the participant is fixed when he or she has the floor - Fixed(Center Reserved) : The location of the participant moves to the middle when he to she has the floor
3	Select a shape of Layout for video (when Layout Mode is set to Manual, this is possible)
4	This is an option whether to configure each symmetric or asymmetric layout to see the participant with the floor, in the full screen.
5	This is to configure a way to takes the floor. - After Release : The participant with the floor has to end first, so the other person can take

	the floor. - On Request : The participant takes the floor by asking for it.
6	Select an option whether to display a name of the participant and location and size of the display
7	Select an option whether to display Background, Border line, Floor line, Background line and a color.
8	Select an option to set whether to turn on or off the microphone for the participant who are displayed.

The following is the screen for configuring the Participant of a Conference:

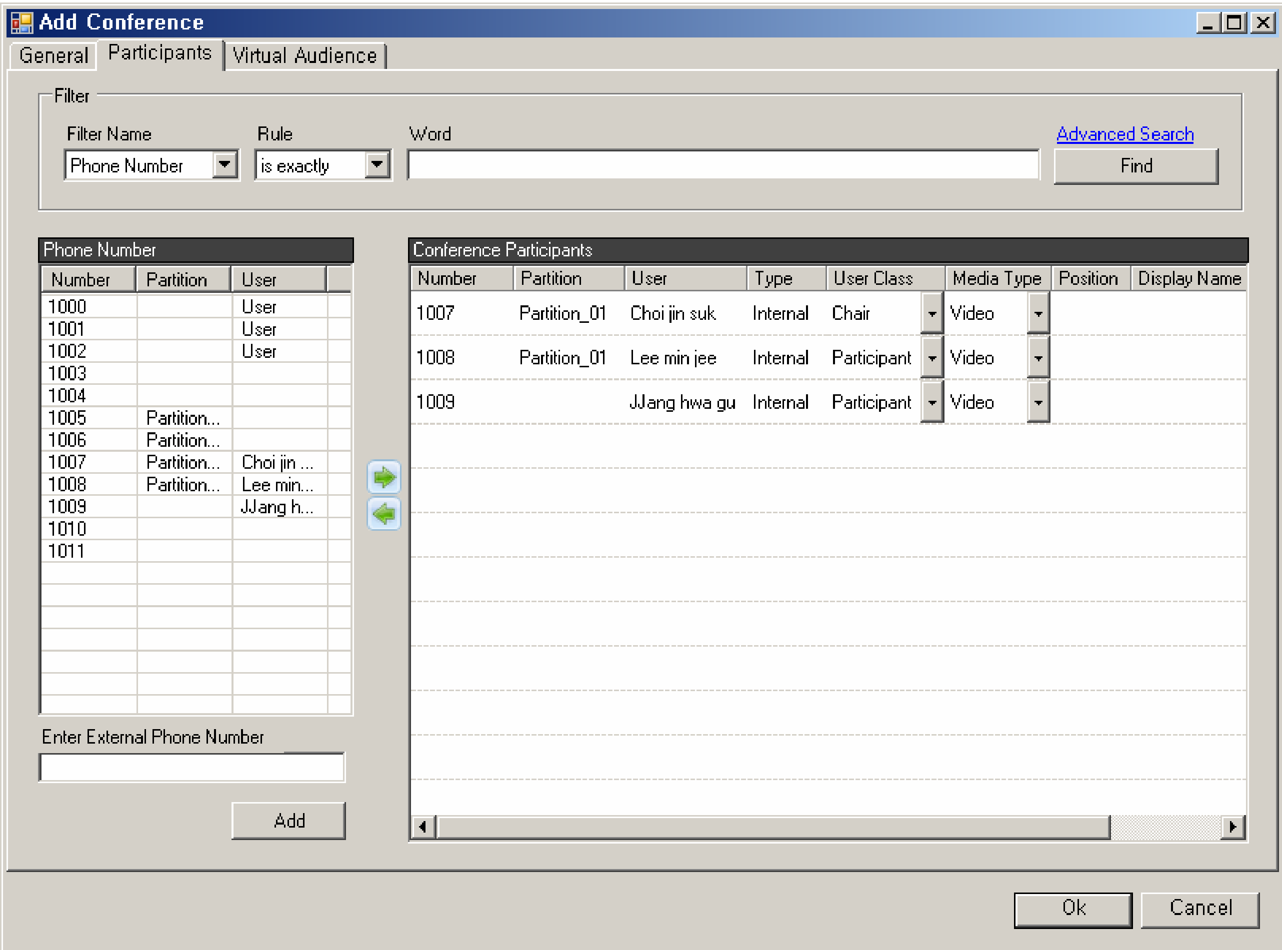


Figure 2-151 Dial-Out Conference – Registering Participant

Table 2-98 Dial-Out Conference – Description of ParticipantRegistration

Field	Description
1	This is the user's list of telephone numbers which is not registered to the present Conference Room.
2	

3	This is the list of participants' telephone numbers which are to registered to the present Conference Room Number – a participant's telephone number - Partition – a Partition information of a participant's telephone number - User – a participant's name - Type – Internal or External - User Class – the participant's authority in conference (Chair, Operator, Participant, Audience) - Media Type – Audio, Audio(Seat Reserved), Video - Position – Set a position of Layout for Video (only when Layout Mode is set to Manual, the setting is possible) - Display Name –a name to be displayed on OSD in case of Video (when no name is entered, the user's name is displayed)
4	Register an external phone number, which is not registered, as a Conference user.

The followings are to describe a function of Virtual Audience for Dial-out Conference.

Virtual Audience provides a function that allows many people only to listen to a conference, as to be connected with a broadcasting system. A broadcasting server can be registered to the conference as a virtual audience. The broadcasting server and terminals can be configured by the broadcasting system (MBMS)[7]. For more details of configuring the broadcasting terminals, you may refer to MBMS operation manual.

[7] The broadcasting system constitutes to the broadcast sever, management system, management GUI Tool and others for audio/video broadcasting.

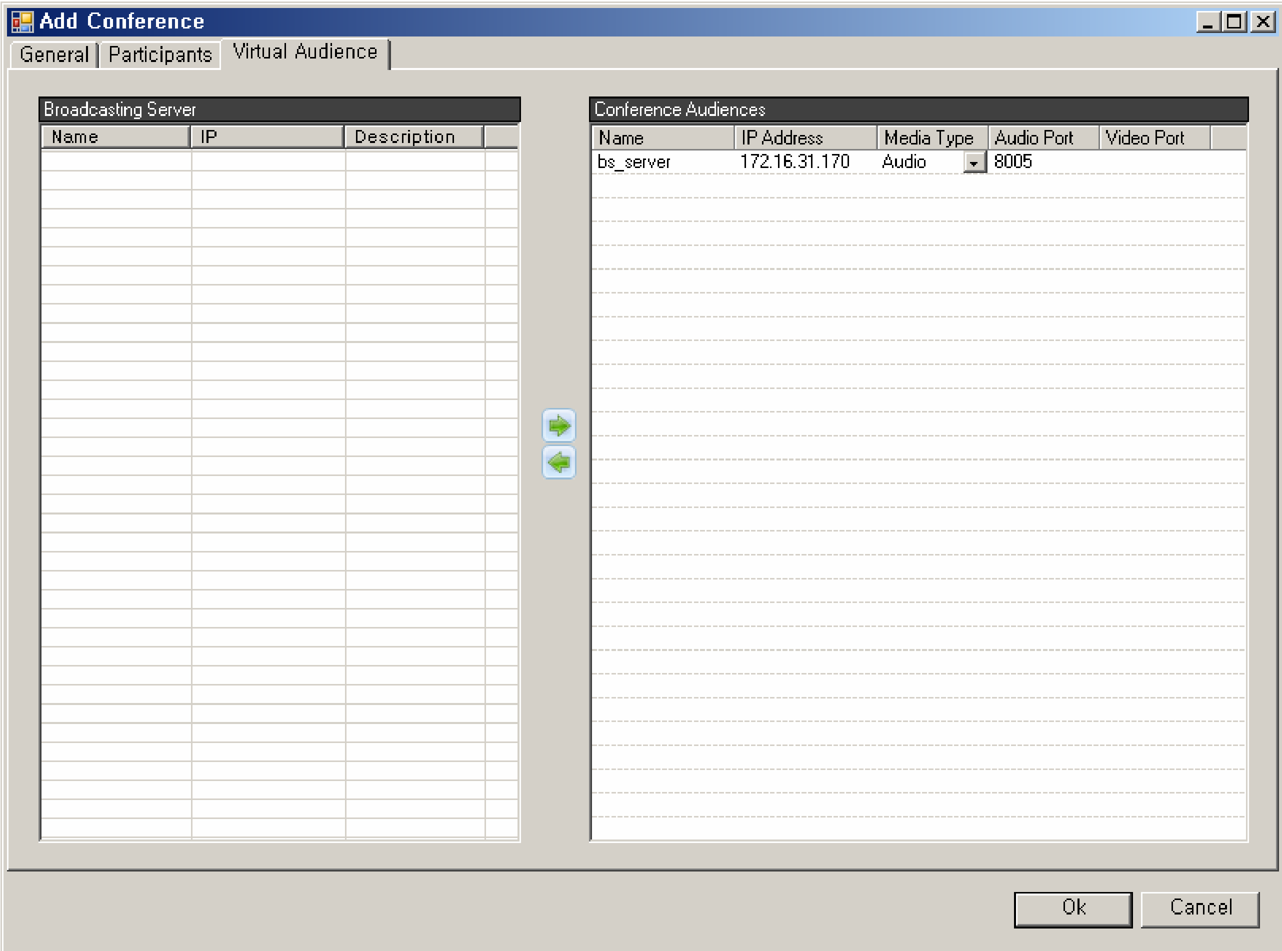


Figure 2-152 Dial-out Conference – Configuring Virtual Audience

Table 2-99 Dial-out Conference – Description of Virtual Audience Configuration Settings

Field	Description
1	This is the list of Broadcasting Server which is registered to Servers.
2	This is a list of the Broadcasting Server which are registered to Virtual Audience in Conference - Media Type : Audio or Video - ①Audio Port : Audio signal port③ ② - Video Port : Video signal port
3	Register or delete the Broadcasting Server to or from Virtual Audience

All the conference features can be controlled from the terminal except Dial-Out Conference. Dial-Out Conference can be started or stopped through SMM. You can see the conference list in progress through Active Conference Monitoring of Event & Monitoring section.

The following figure shows the screen for running and stopping Dial-Out Conference.

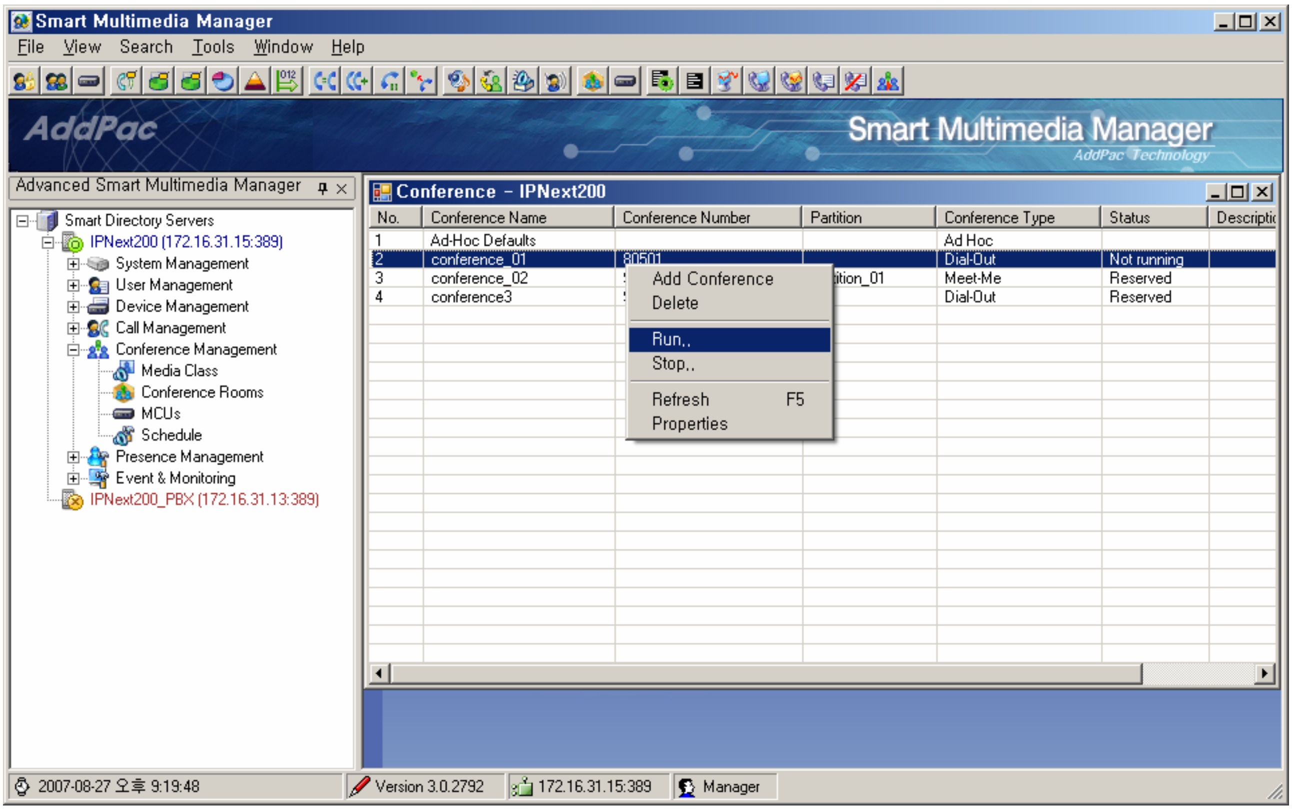


Figure 2-153 Running and Stopping Dial-Out Conference

Meet Me Conference Configuration

Add Conference can be carried out though a pop-up menu from Conference Rooms Screen.

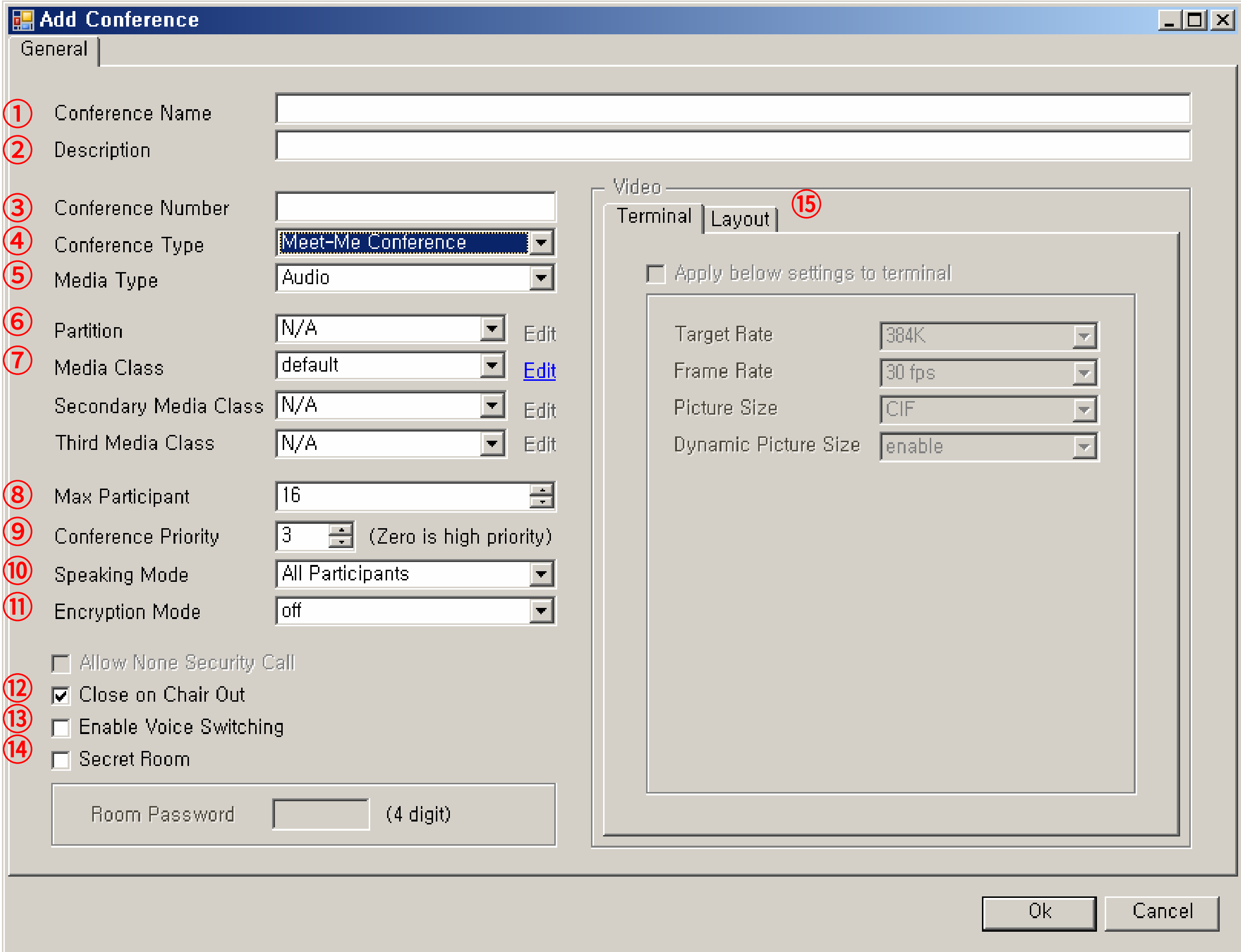


Figure 2-154 Meet-Me Conference Properties

Table 2-100 Meet-Me Conference Properties Description

Field	Description
1	Register a name of Conference
2	Register a description of Conference
3	Register a conference number.
4	Select a type of conference. In here select Meet-Me Conference.
5	Select a type of media (Audio, Video + Audio) for Ad-Hoc Default. According to the type of media is selected, the settings of video can be enabled or disabled.
6	Select Partition (optional), Select Media Class for Conference Room
7	Select Media Class to be applied to Conference Room. Media Class can be set to 3 participants at

	maximum and 1 must be selected at least. Media Class takes a priority as the fist Media Class does not support, the second one does.
	-
8	Set the maximum number of participants (1~16 participants)
19	Set a priority for the Conference
11	Set encryption mode (AES, DES, 3-DES) for video and voice data of the Conference
12	This is an option to select whether the conference call is to be terminated when Chair Man, who invites the Conference Call, terminates a call when he/she ends. When the option is selected, Conference Call ends when Chair Man terminated his/her call.
13	When the floor does not exist in Conference, Set the screen for the speaking participant to be stretched automatically
14	Select an option whether to open or close the room. When the room is closed, set a password.
15	This is to configure the settings of video for the media type which is Video + Audio. - Terminal : These are the characteristics of video transmitted from the participating Terminals (optional) - Layout : Configuration of the layout for video

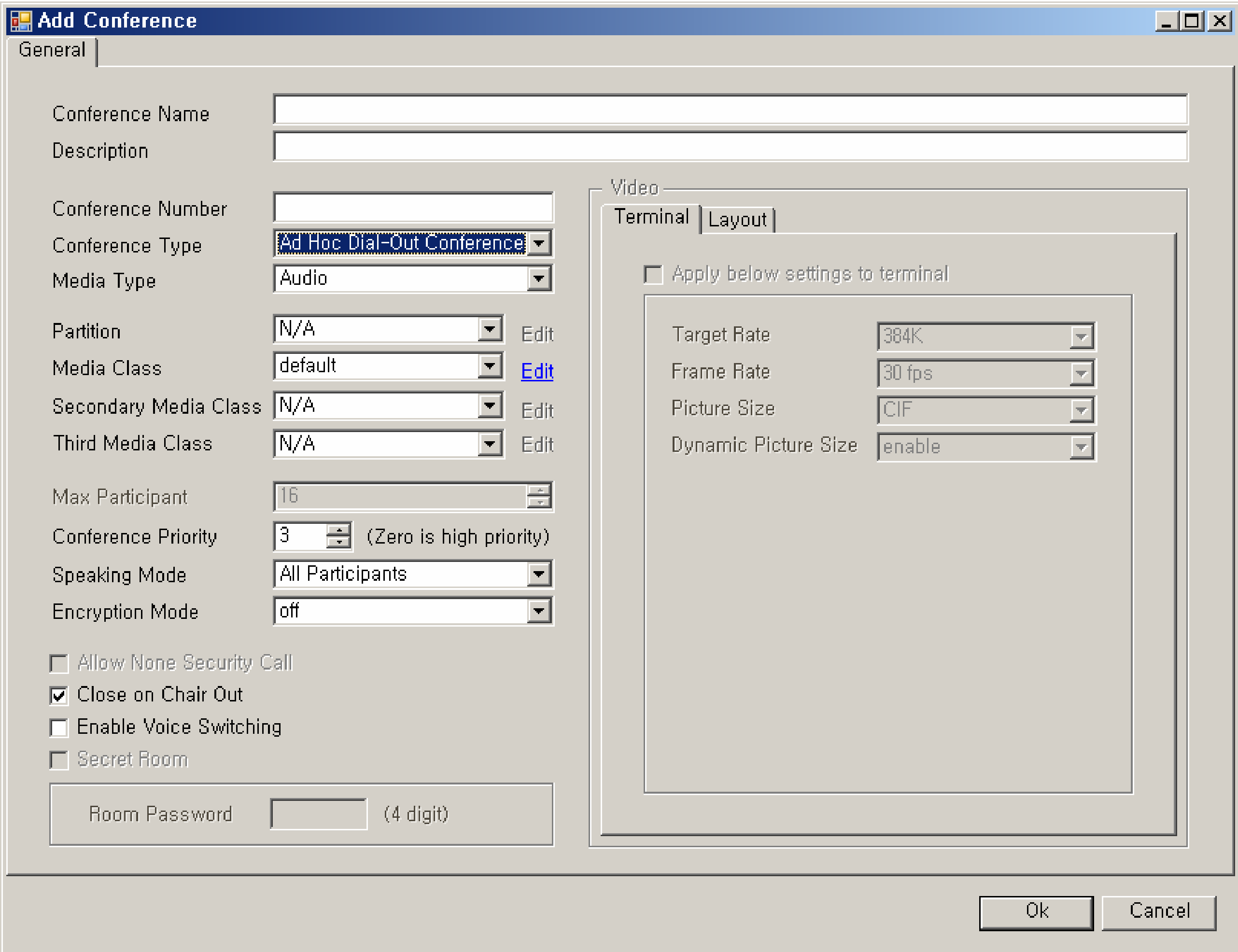


Figure 2-5 Ad-Hoc Dial-Out Conference Properties

Table 2-101 Meet-Me Conference Properties Description

Field	Description
1	Enter a name of Conference
2	Enter a description of Conference
3	Enter a Conference Number
4	Select a type of conference. In here, select Ad-Hoc Dial-Out Conference
5	Select a type of media (Audio, Video + Audio) for Ad-Hoc Default. According to the type of media is selected, the settings of video can be enabled or disabled.
6	Select Partition (optional), Select Media Class for Conference Room
7	Select Media Class to be applied to Conference Room. Media Class can be set to 3 participants at maximum and 1 must be selected at least. Media Class takes a priority as the fist Media Class does not support, the second one does.
8	Set a priority for the Conference

9	Set a range of participants who can take the floor. - All Participants can speak - All Visible Participants : all the participants displayed on the layout can speak - Floor + Chair : Only the participants with the floor and Chair can speak Floor Only : Only the participants with the floor can speak
10	Set encryption mode (AES, DES, 3-DES) for video and voice data of the Conference
11	This is an option to select whether the conference call is to be terminated when Chair Man, who invites the Conference Call, terminates a call when he/she ends. When the option is selected, Conference Call ends when Chair Man terminated his/her call.
12	When the floor does not exist in Conference, Set the screen for the speaking participant to be stretched automatically
13	This is to configure the settings of video for the media type which is Video + Audio. - Terminal : These are the characteristics of video transmitted from the participating Terminals (optional) - Layout : Configuration of the layout for video

Setting Conference Schedule

IP-PBX provides scheduling for carrying out Dial-Out Conference at a specified time. This is a useful feature to set conference on repetitive basis. The below figure shows how to set conference schedule:

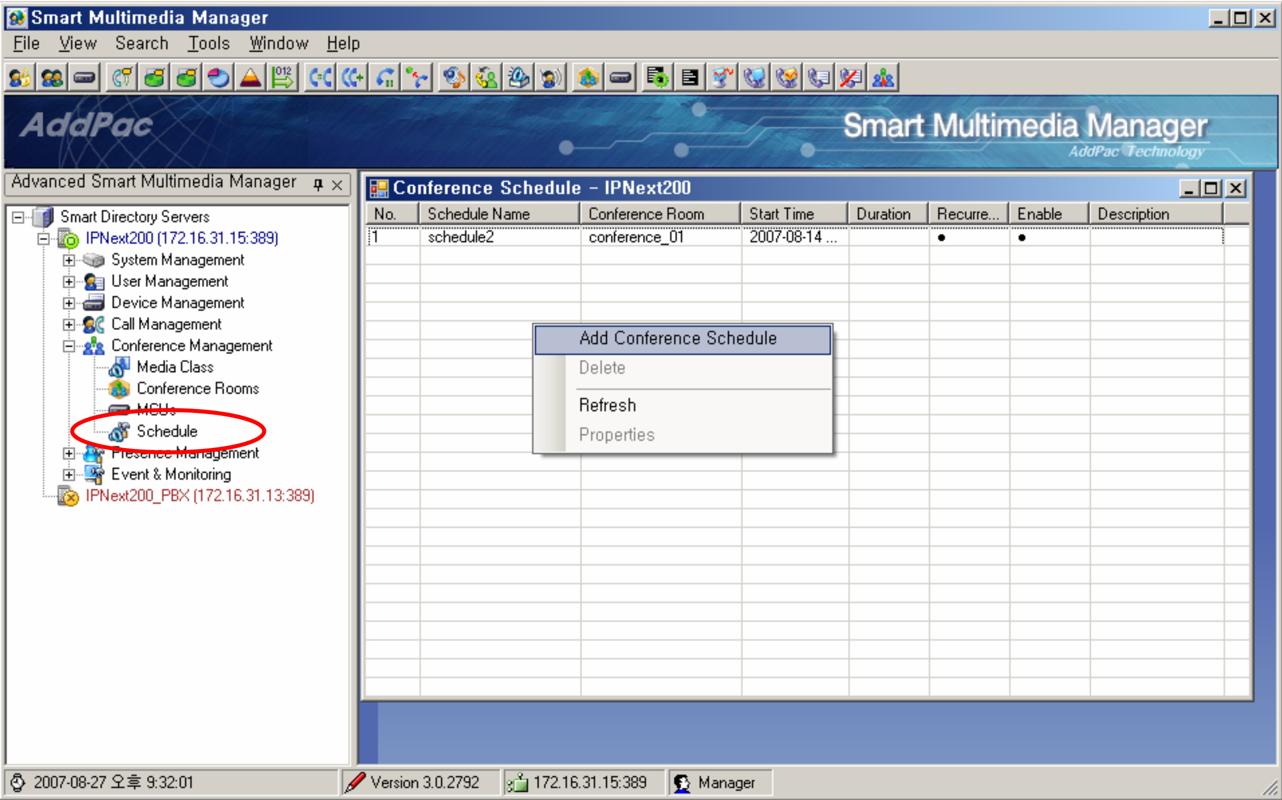


Figure 2-155 Setting Conference Schedule

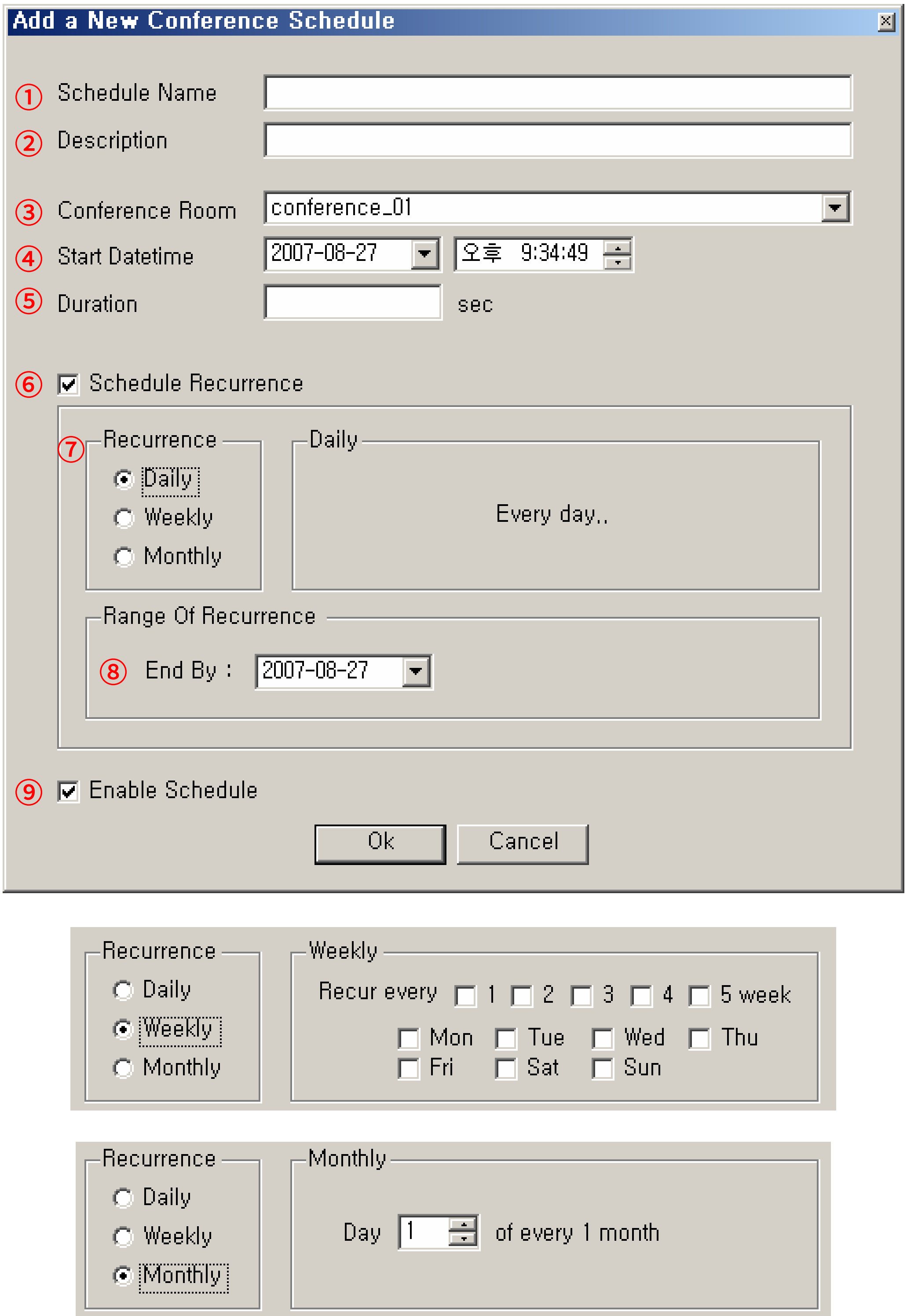


Figure 2-156 Conference Schedule Properties

Table 2-101 Conference Schedule Properties

No	Description
1	Enter a name of Conference Schedule

2	Enter a description of Conference Schedule
3	Select a conference room to be scheduled
4	Set a starting time of conference
5	Set a duration time of conference. The conference is terminated automatically when the duration time is set. When the duration time is not set, the conference is maintained until the participant ends the conference.
6	Select a recurrence of schedule
7	Select a frequency for recurrence of schedule - Daily : everyday - Weekly : Select a day and week - Monthly : Specify a date for recurrence on monthly basis
8	Select a last date of recurrence
9	Enable/ disable a schedule

Presence Management

Configuring Presence Server Preference

When Presence Server is configured in redundancy for providing presence service to the subscribers, SMM creates a priority template for the server to be managed. Preference is configured to Presence Group and applies to with the sever priority information.

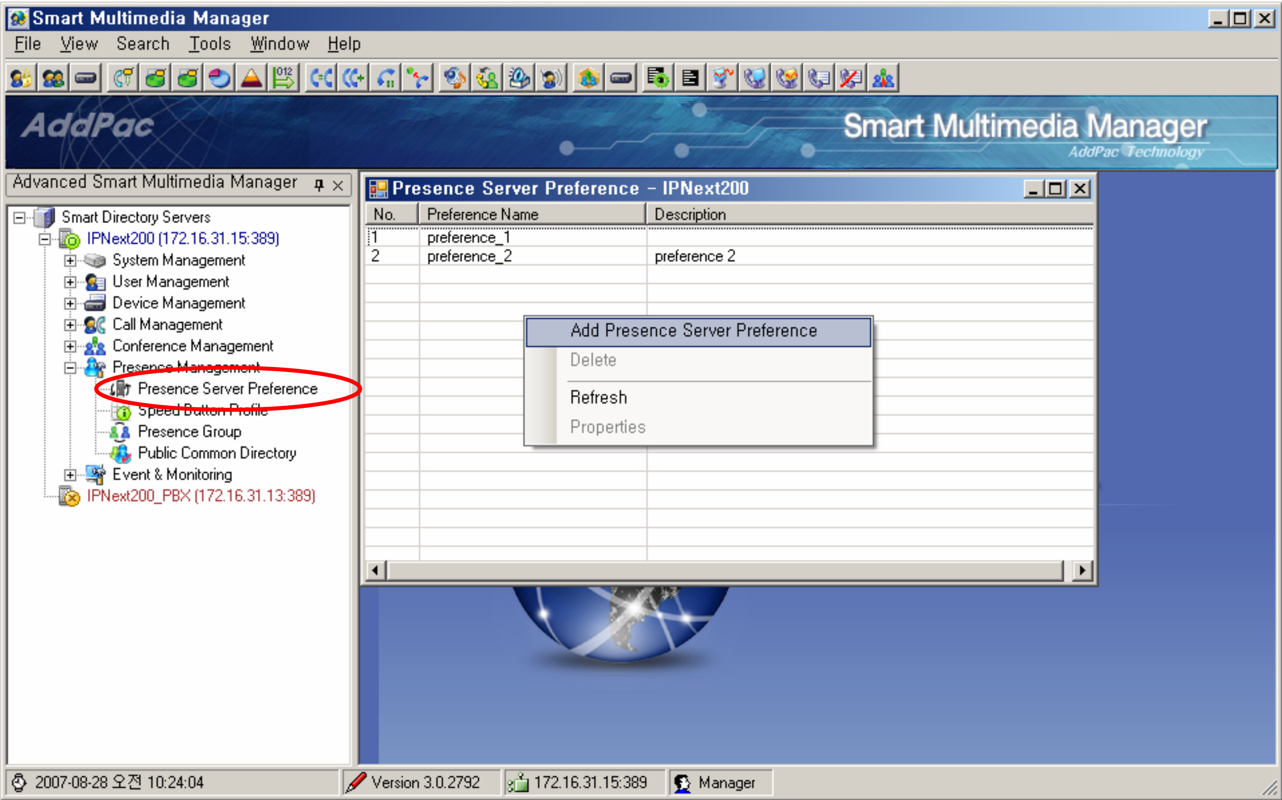


Figure 2-157 Configuring Presence Server Preference

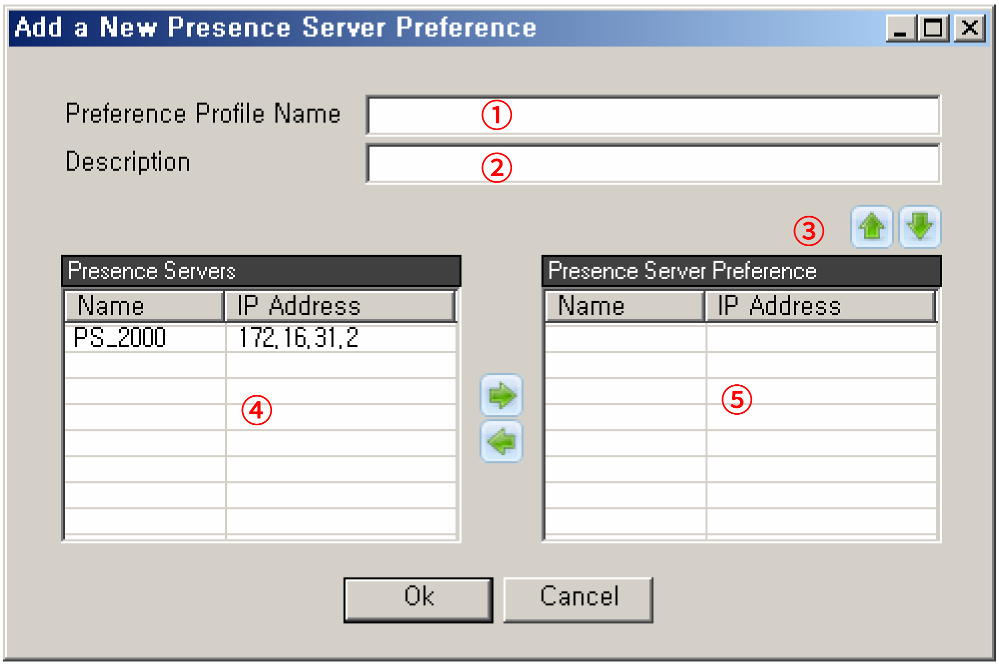


Figure 2-158Presence Server Preference

Table 2-102 Presence Server Preference Properties

No	Description
1	Create a Presence Profile Name
2	Enter a description
3	Adjust Presence Servers in priority
4	The presence server list which is not assigned to the profile
5	The presence server list which has been set to the profile.

Creating Speed Button Profile

Speed Button assigns a telephone number to a specified button and automatically makes a call to the phone number. It is the button information in both hardware and software aspects. Among the AddPac IP Phone product line, AP-IP300 and AP-VP250 support this speed button feature. This feature creates a profile and registers information to each button and the button information can be provided to the subscribers' terminals through Presence Server. Speed Button Profile can be created in Presence Group.

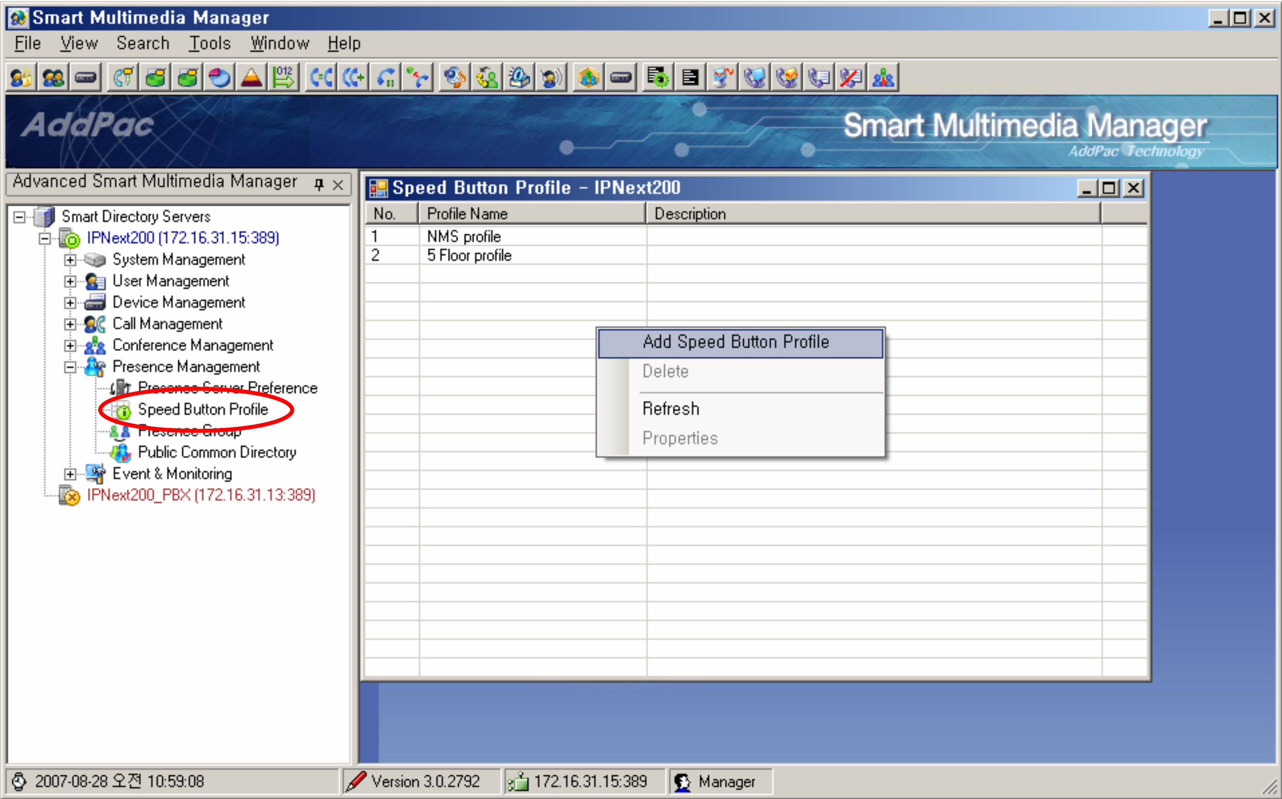


Figure 2-159 Speed Button Profile

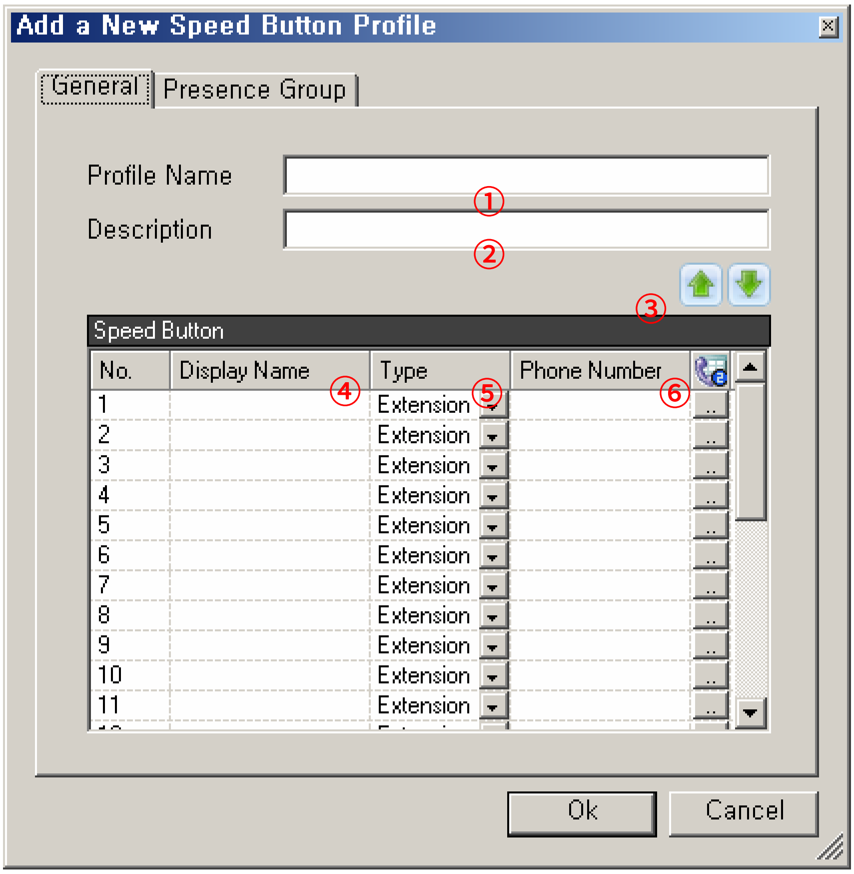


Figure 2-160 Speed Button Profile – General Properties

Table 2-103 Speed Button Profile – Description of General Properties

No.	Description
1	Enter a name of Speed Button Profile
2	Enter a description of Speed Button Profile
3	Move the settings up and down
4	Enter a name to be displayed for each Speed Button(1~25)
5	Select a type for each speed button (1~25) of phone number
6	Enter a telephone number for each speed button (1~25) (Pressing the right click, you can search and enter the telephone numbers which have been registered)

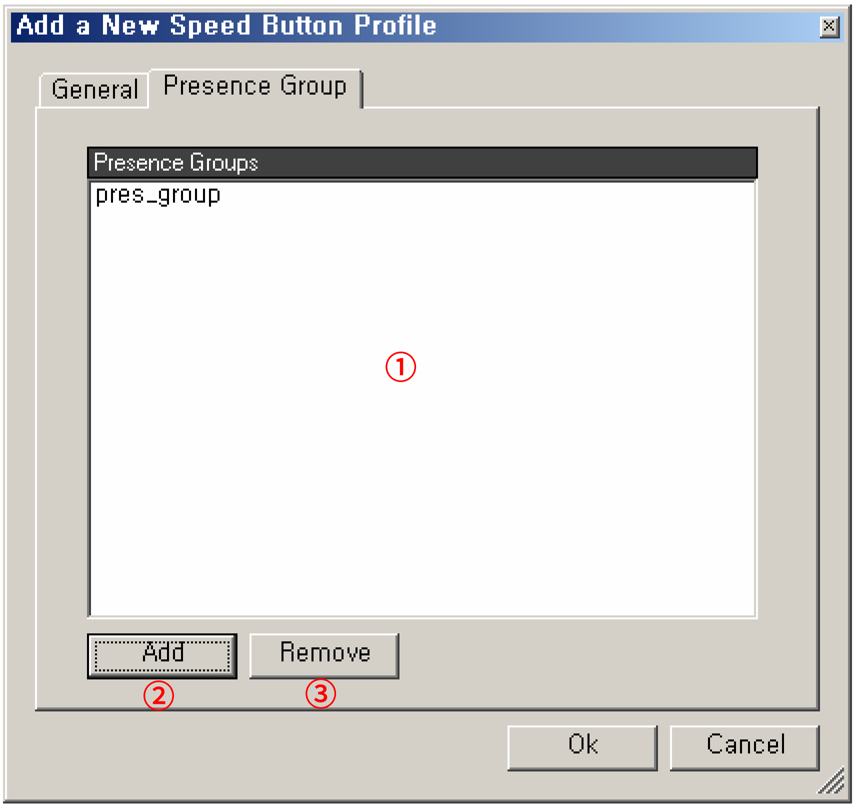


Figure 2-161 Speed Button Profile – Presence Group Properties

Table 2-104 Speed Button Profile – Description of Presence Group Properties

No.	Description
1	Display the presence group including the speed button
2	Add the presence group to be included in the speed button
5	Delete the presence group included in the speed button

Configuring Presence Group

Presence group is a unit to provide presence service and is able to group the settings of speed button and a range of information sharing between the users. One user's directory can configure many presence groups, share the speed button profiles and take presence message. The same profiles are applied for redundancy.

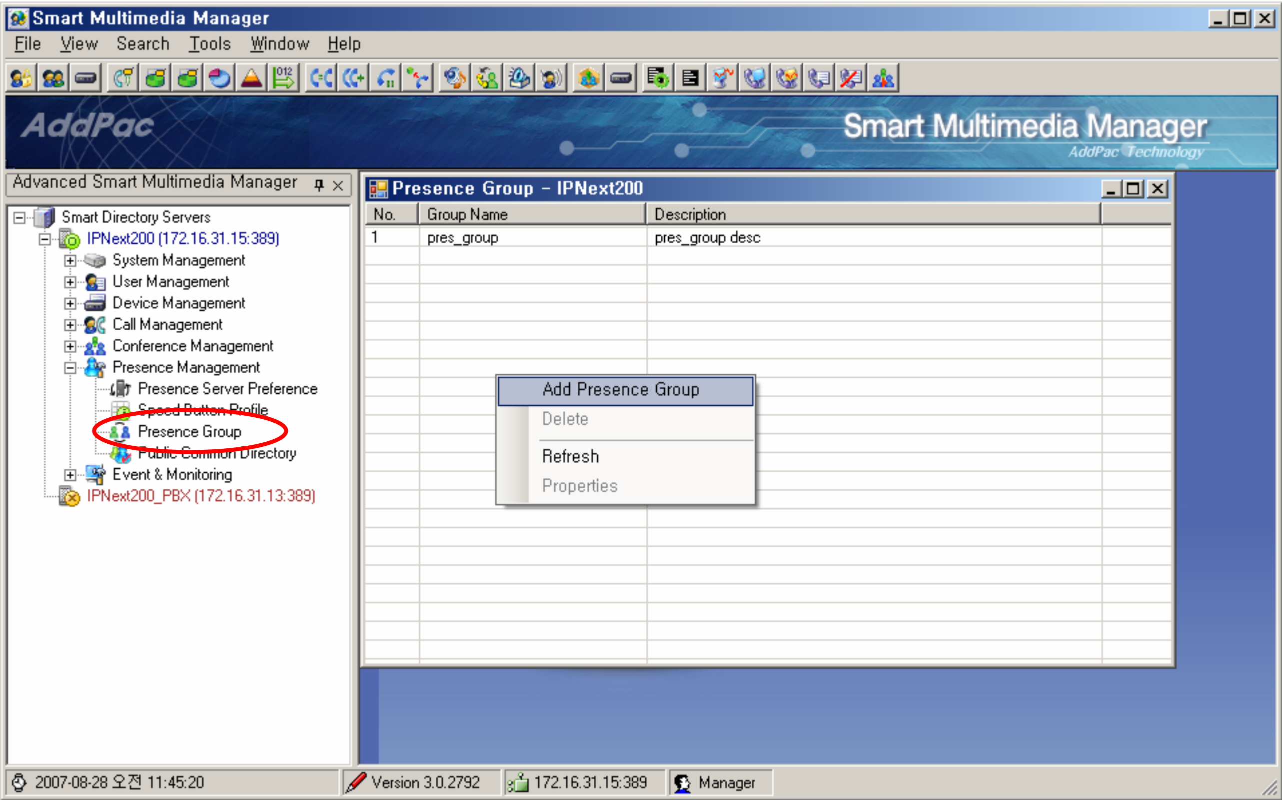


Figure 2-162 Configuring Presence Group

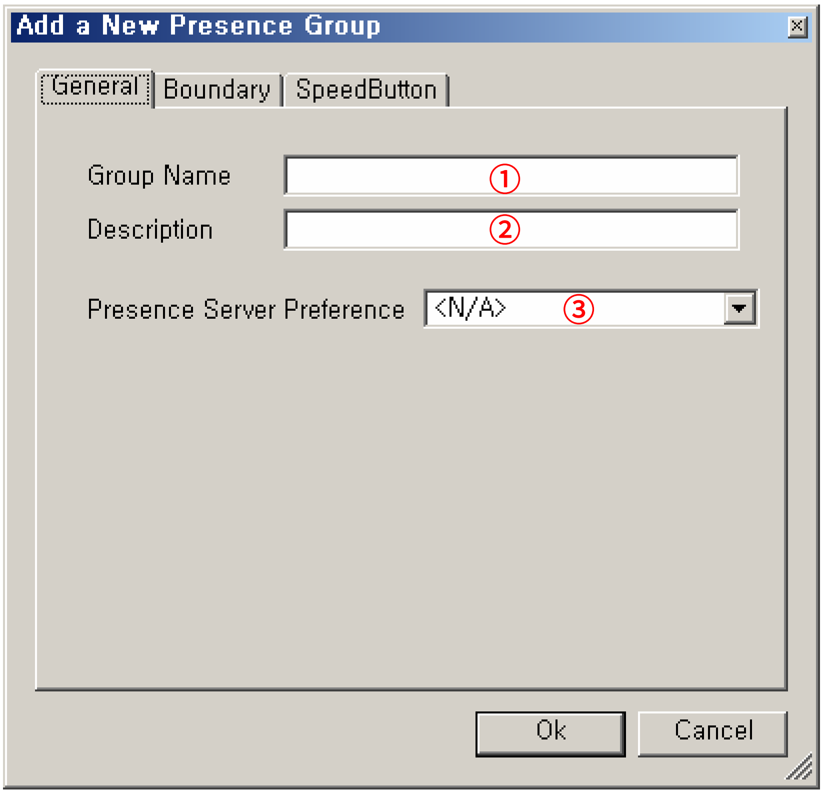


Figure 2-163 Presence Group – General Properties

Table 2-105 Presence Group – Description of General Properties

No	Description
1	Enter a name of presence group
2	Enter a name of presence group
3	Choose a presence sever preference (optional).

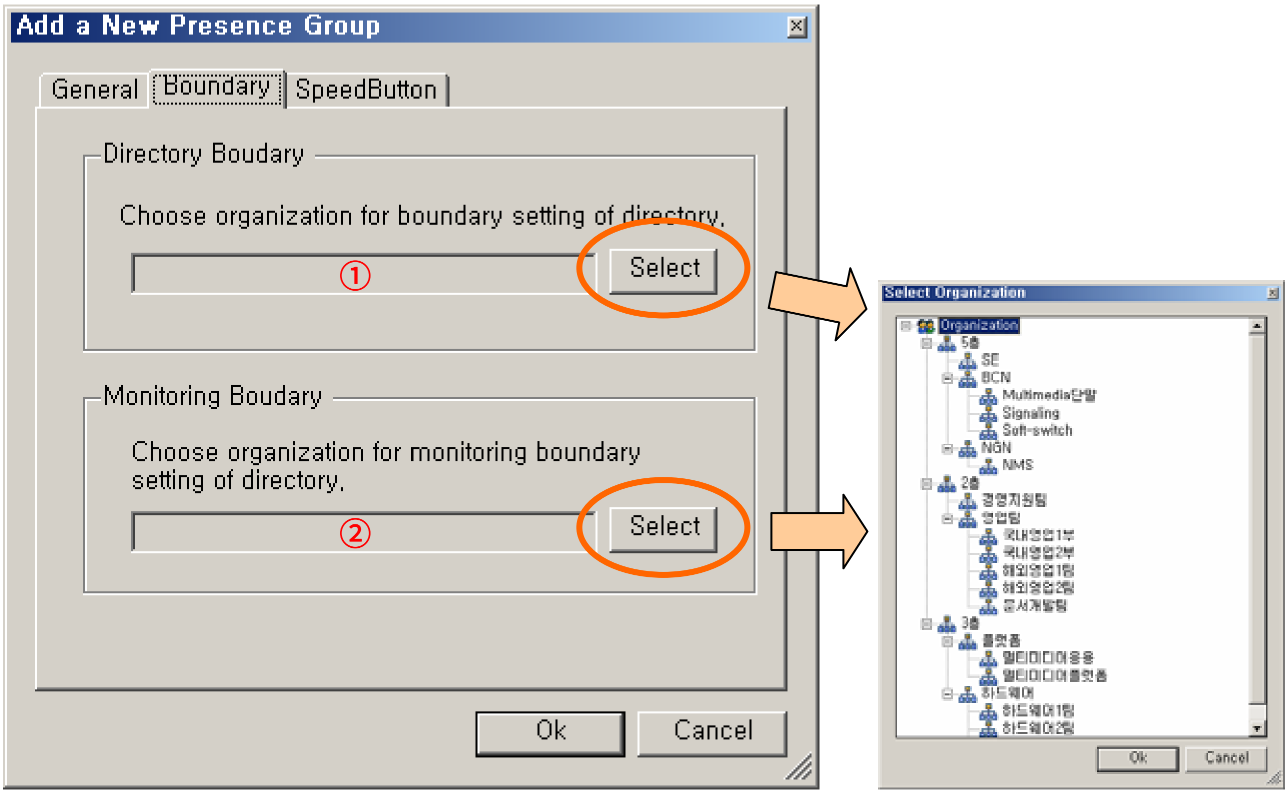


Figure 2-164 Presence Group – Boundary Properties

Table 2-106 Presence Group – Description of Boundary Properties

No.	Description
1	A boundary setting of directory which haven been applied to Presence Group. This is the directory information for Smart Messenger to download from Presence Server
2	Choose organization for monitoring boundary setting of directory. Only the organization in the lower case of the directory boundary setting can be selected

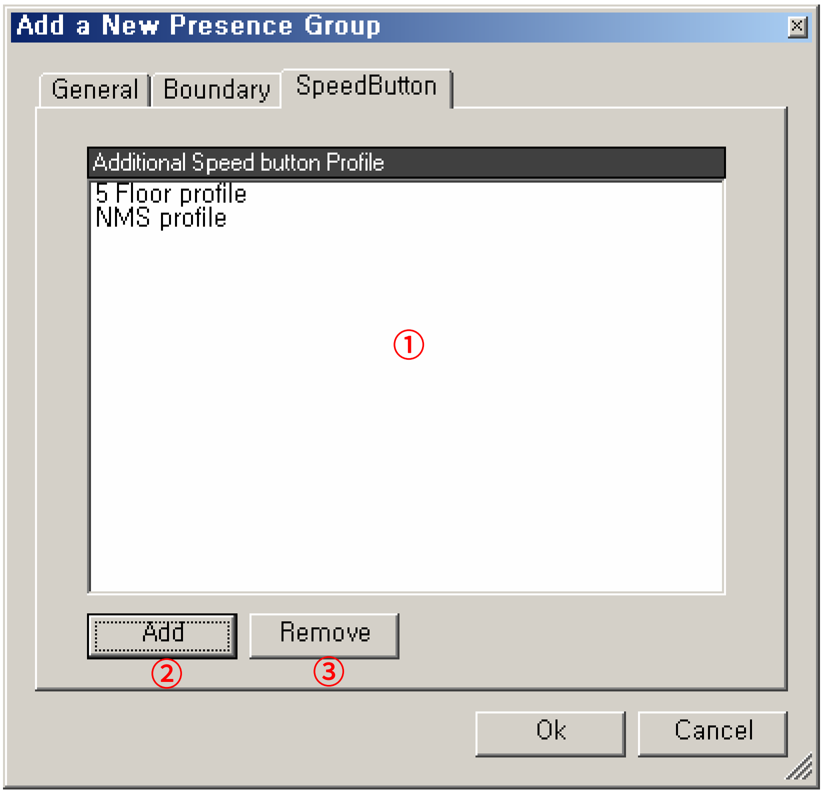


Figure 2-165 Presence Group – Speed Button Properties

Table 2-107 Presence Group – Description of Speed Button Properties

No.	Description
1	The Speed Button Profile included in Presence Group
2	Add Speed Button Profile to Presence Group
3	Delete Speed Button Profile from the group

Configuring Public Common Directory

Public Common Directory manages the common directory of Smart Messenger and registers organization and user information.

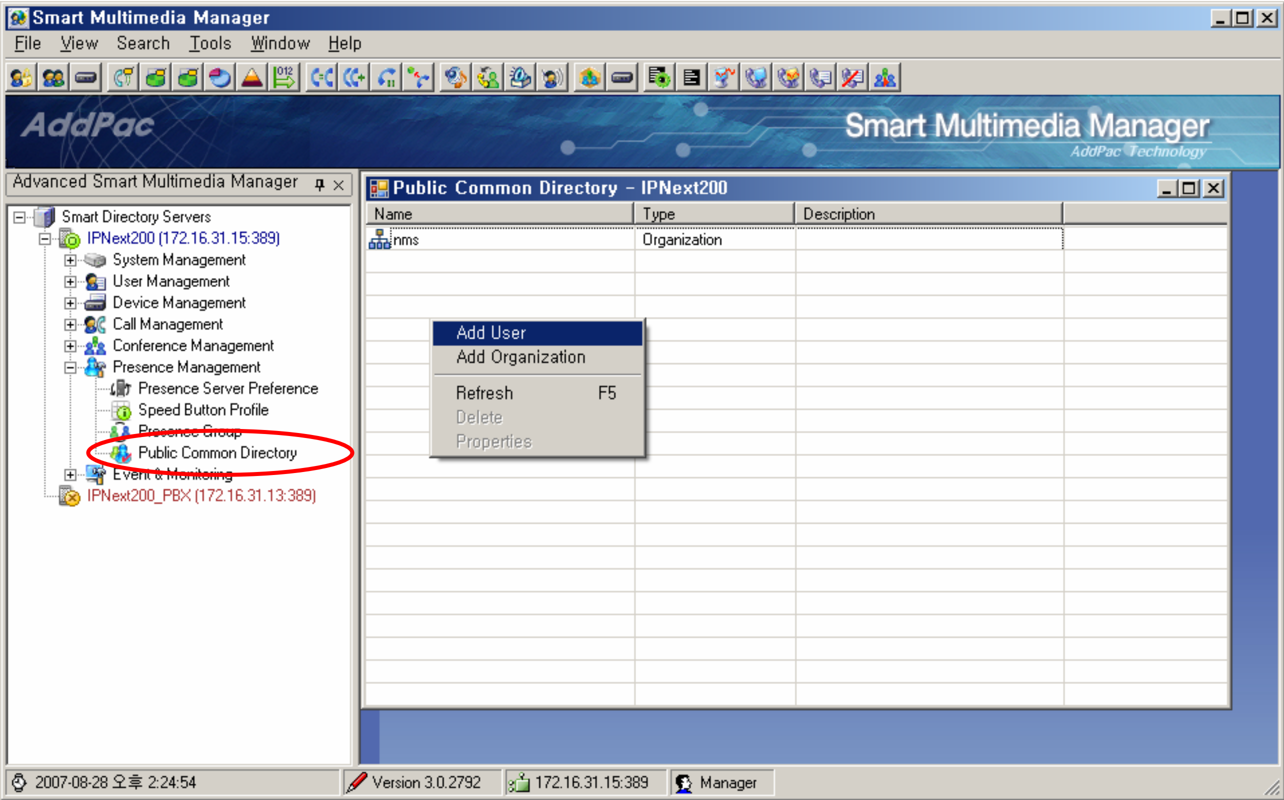


Figure 2-166 Configuring Public Common Directory

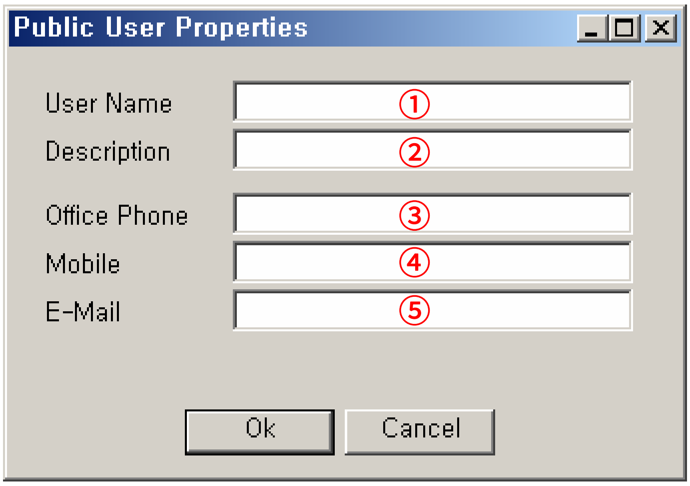


Figure 2-167 Public User Properties

Table 2-108 Description of Public User Properties

No.	Description
1	Enter a name of common user
2	Enter a description
3	Enter an Office phone number
4	Enter a mobile phone number
5	Enter an e-mail address

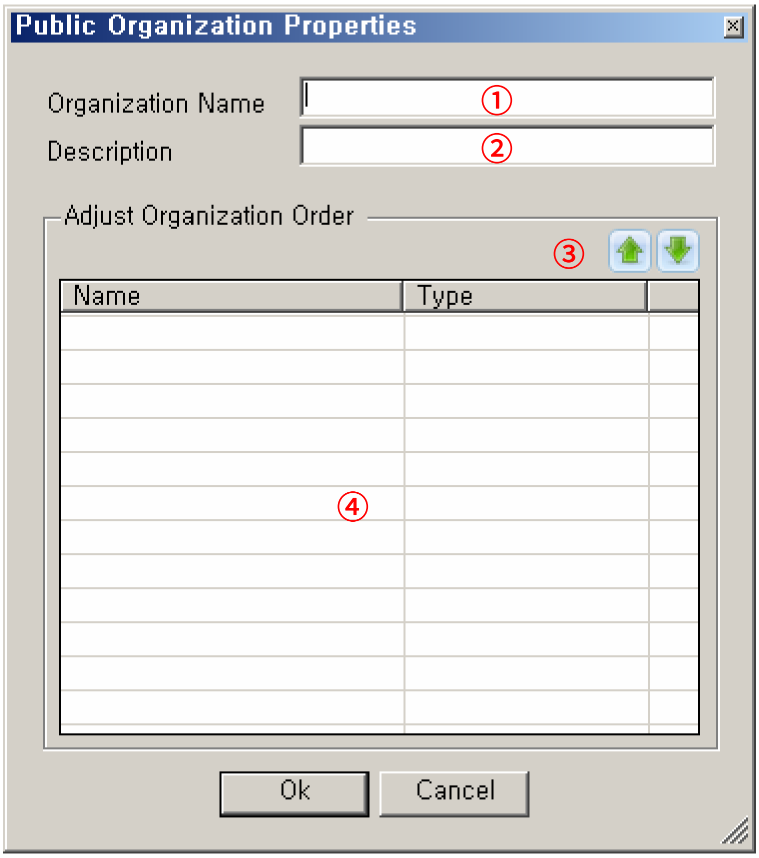


Figure 2-168 Public Organization Properties

Table 2-109 Public Organization properties

No.	Description
1	Enter a name of organization for the common user
2	Enter a description
3	Adjust organization in an order to be displayed in Smart Messenger
4	The user list in the organization of the common user

Chapter 3. Event Monitoring

Event Configuration

Event Monitoring information of SMM interoperates with log information of SEM (Smart Event Manager) which is generated from IPNext PBX System or takes transmission of Event directly from IPNext PBX System, so Event logging monitoring can be performed.

Event configuration can be processed in the following 2 ways:

1. Select the source that Event is to be transmitted to, such as SEM or Call Manager.
2. Configure Event Filter, Logging Filter for the subject (Call Manager) that produces Even and Event port settings

The following figure shows the screen of how Event Configuration can be carried out (Event & Monitoring > Event Configuration)

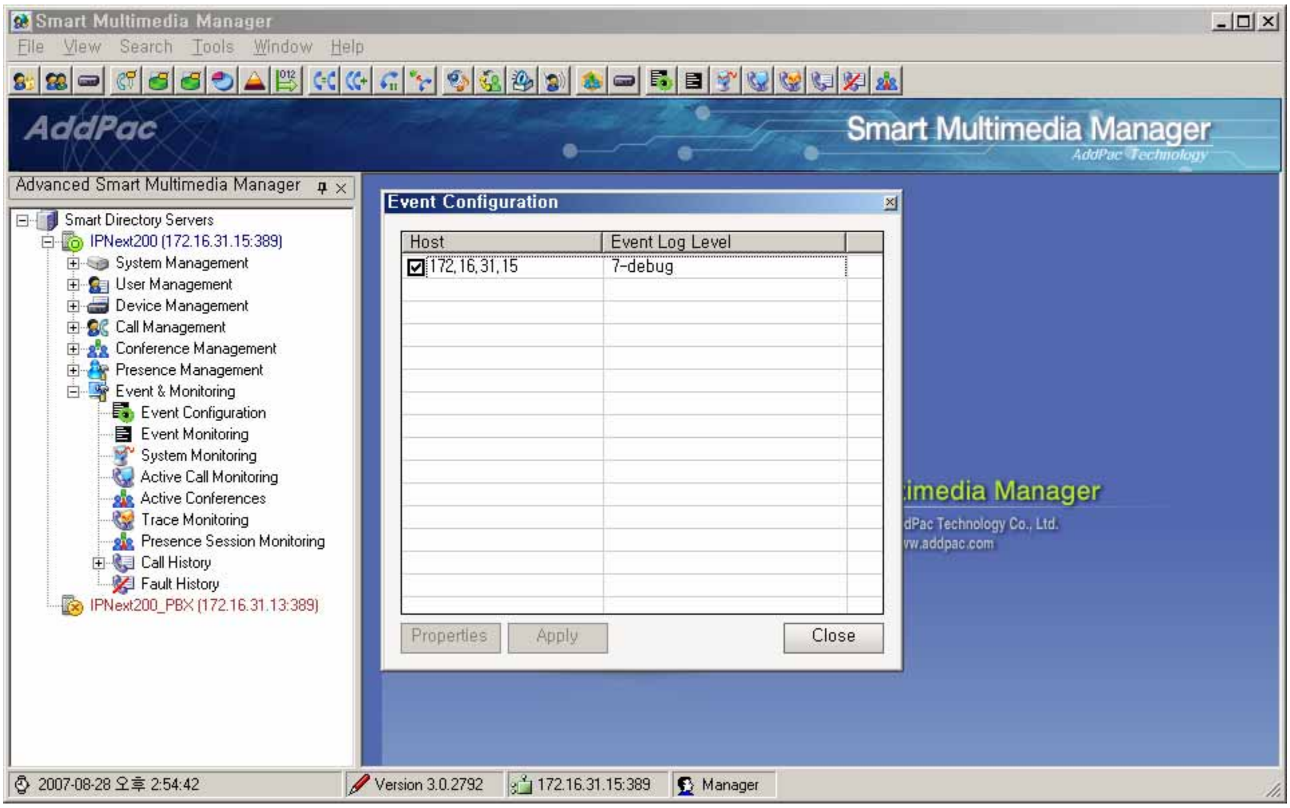


Figure 3-1 Event Configuration 설정 화면

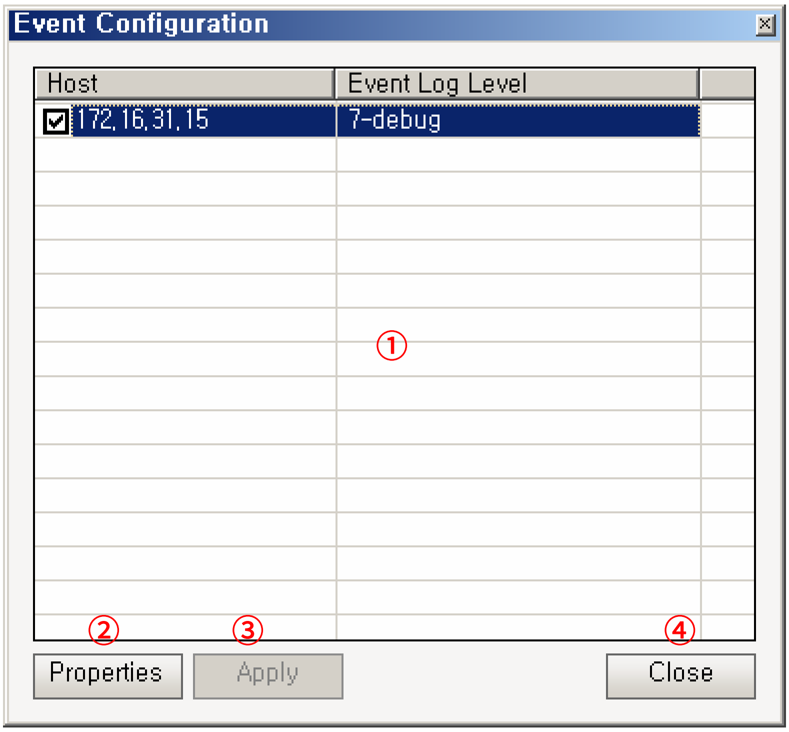


Figure 3-2 Event Configuration Properties

Table 3-1 Description of Event Configuration Properties

Field	Description
1	A list of Call Managers is displayed including the present Call Manager Cluster. You can receive Event information from the selected Call Manager. Then select property for configuring Event settings (logging level, Event Filter, Event Logging Filter..) for the Call Manager
2	Register the information for Event and Logging settings of the selected Call Manager
3	Apply Event Filter settings to SEM or the Call Manager basing on the Event Source from the main event configuration (Tools > Preferences > Event)
4	Close the window

The following is Properties Screen for Event Source. Select an IP address of Call Manager, then click the button of Properties, then Event Source Properties can be performed in the following:

Event Logging Filter setting is applied to Event Manager to be used as a function to configure and save Event Logging on category basis.

* When Event source is set to Call Manager, it is not displayed

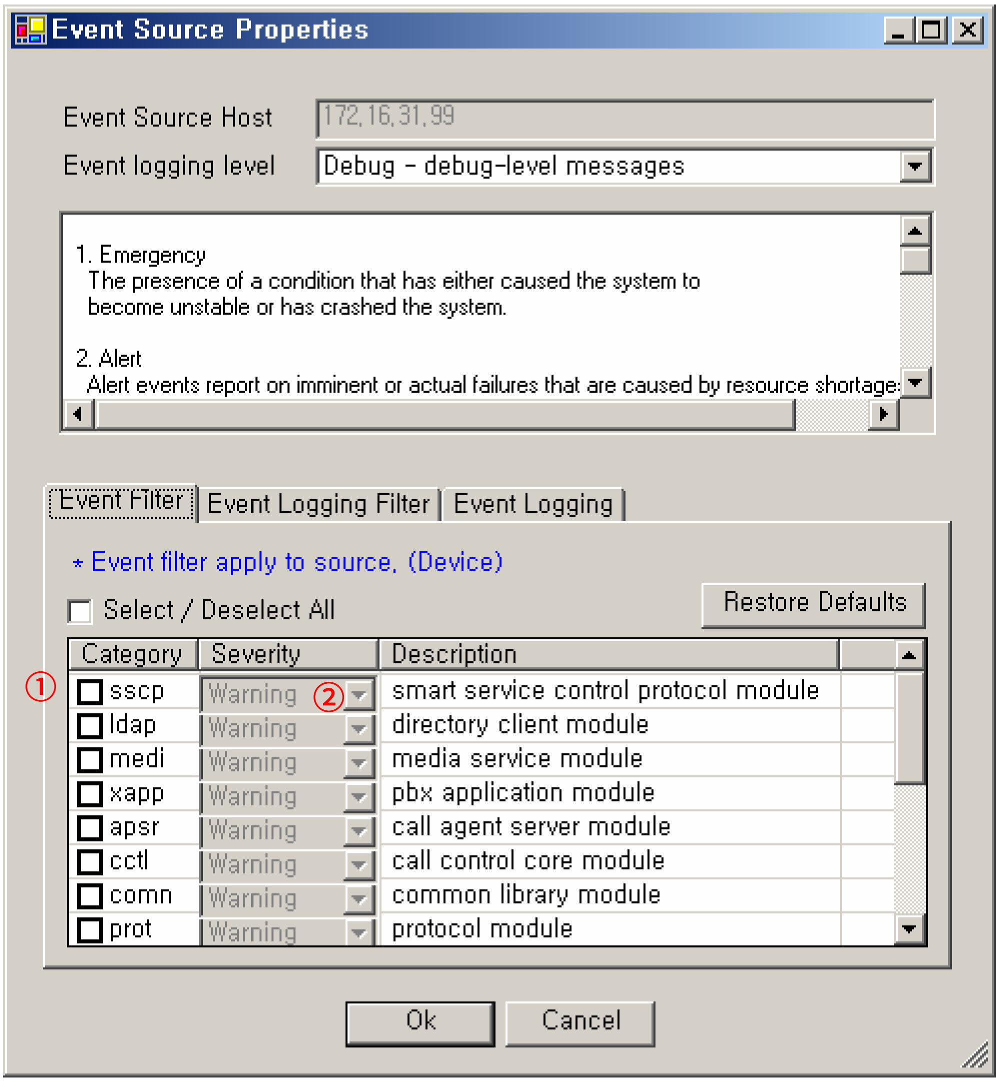


Figure 3-3 Event Source - Event Filter Properties

Table 3-2 Event Source – Description of Event Filter Properties

Field	Description
1	Select an option whether to All Select / DeSelect All for the categories listed in the section 3.
2	Select a level of the Category. There are Error, Warning, Information, Debug for the levels

Event Logging Filter setting is applied to Event Manager to be used as a function to configure and save Event Logging on category basis.

* When Event source is set to Call Manager, it is not displayed

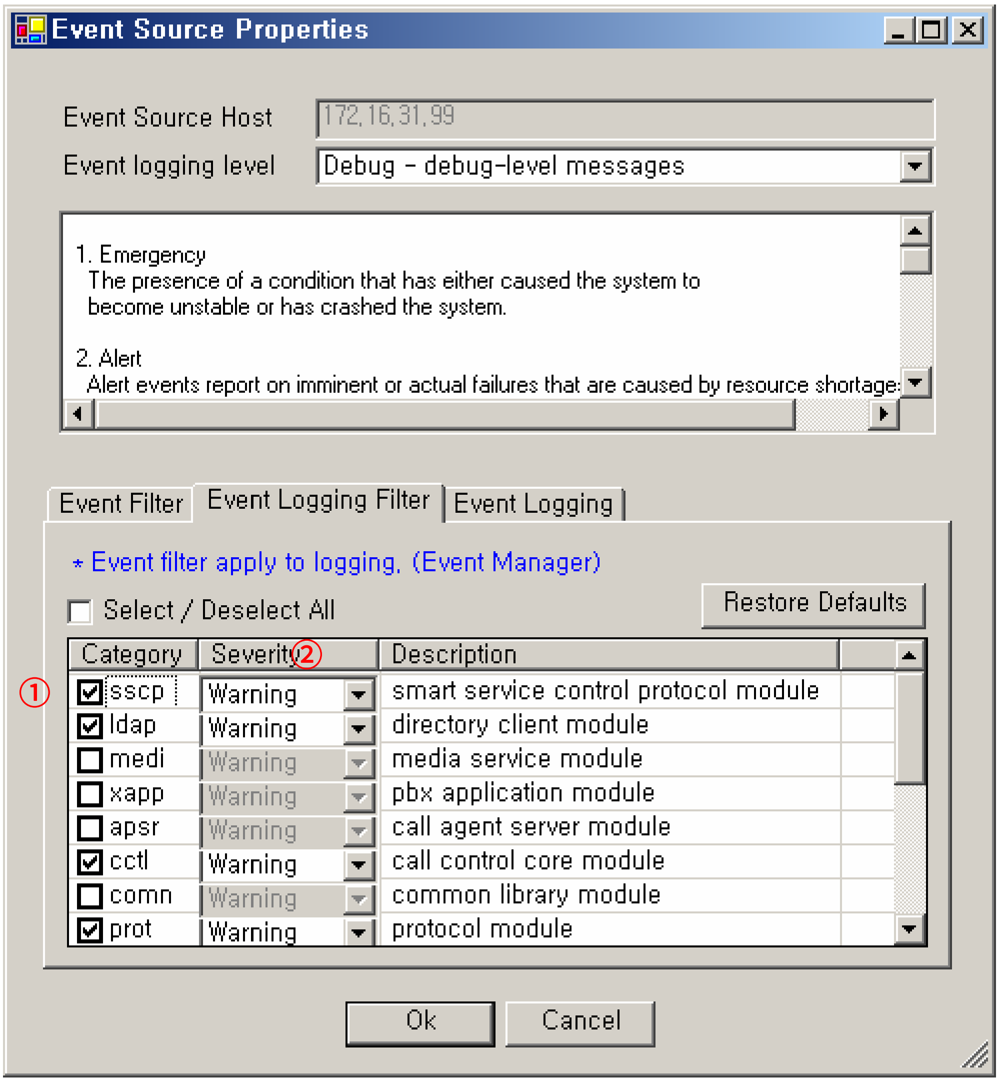


Figure 3-4 Event Source - Event Logging Filter Properties

Table 3-3 Event Source - Event Logging Filter Properties Description

Field	Description
1	A user can Select/ Deselect a detail to which that Event is to be saved for each Category.
2	Select a level of severity to save Event from the corresponding category. The options of the level are Error, Warning, Information, Debug

Event Monitoring

Event Monitoring is a function to monitor Event for the operational status of IPNext PBX on real-time basis. The details of information of for the actual fault are displayed on Event monitoring Screen.

As it is shown in the Figure below, Event & Monitoring item is selected on the left tree, then Event Monitoring in the Sub Tree is chosen. As a result, the window of Event Monitoring is created in the lower part of the screen.

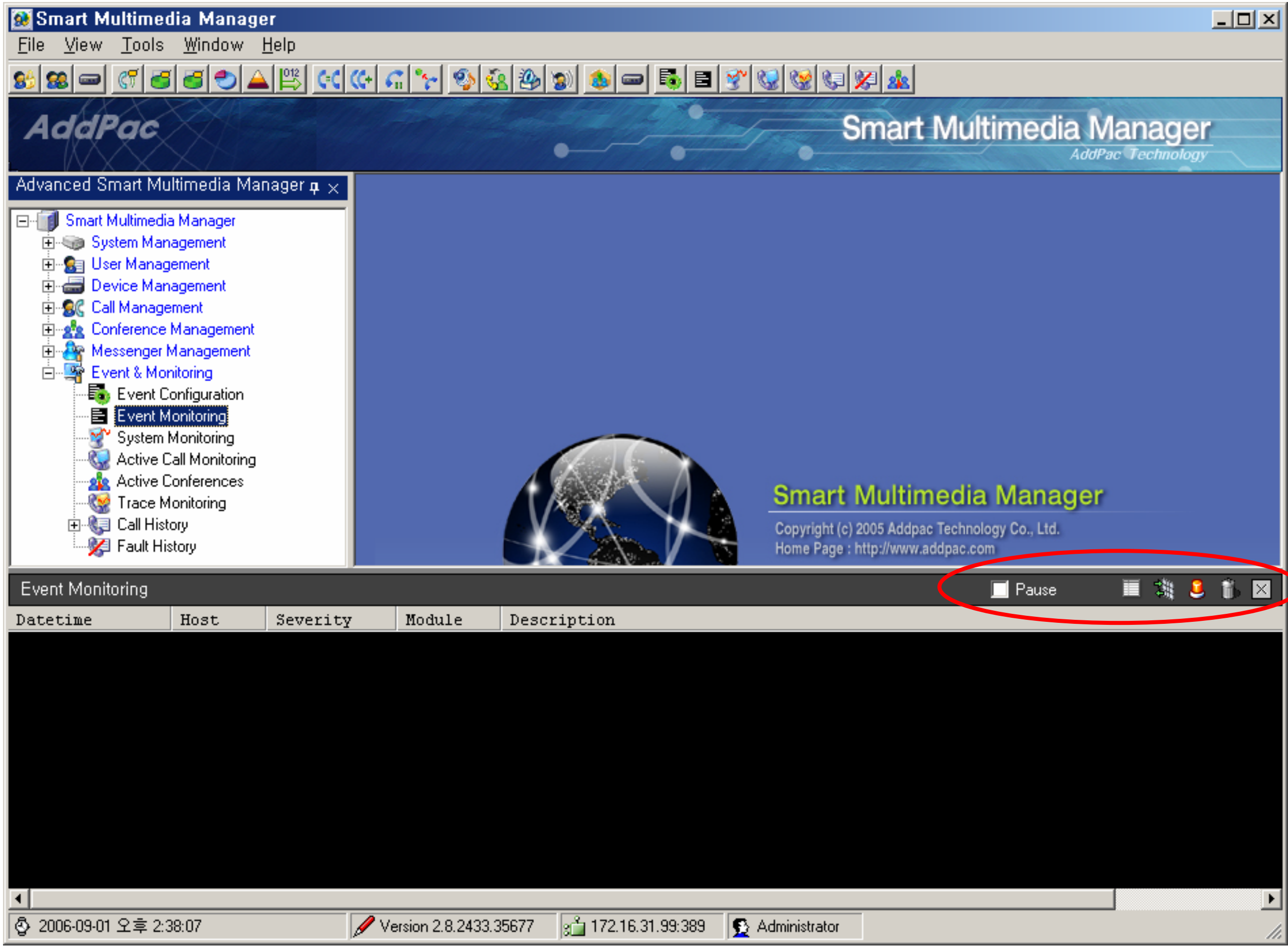


Figure 3-5 Event Monitoring Screen

The icons of setting functions of Event on the right side of Event Monitoring Screen is used for: pause, start, stop and Event filtering setups, fault monitoring and real time display

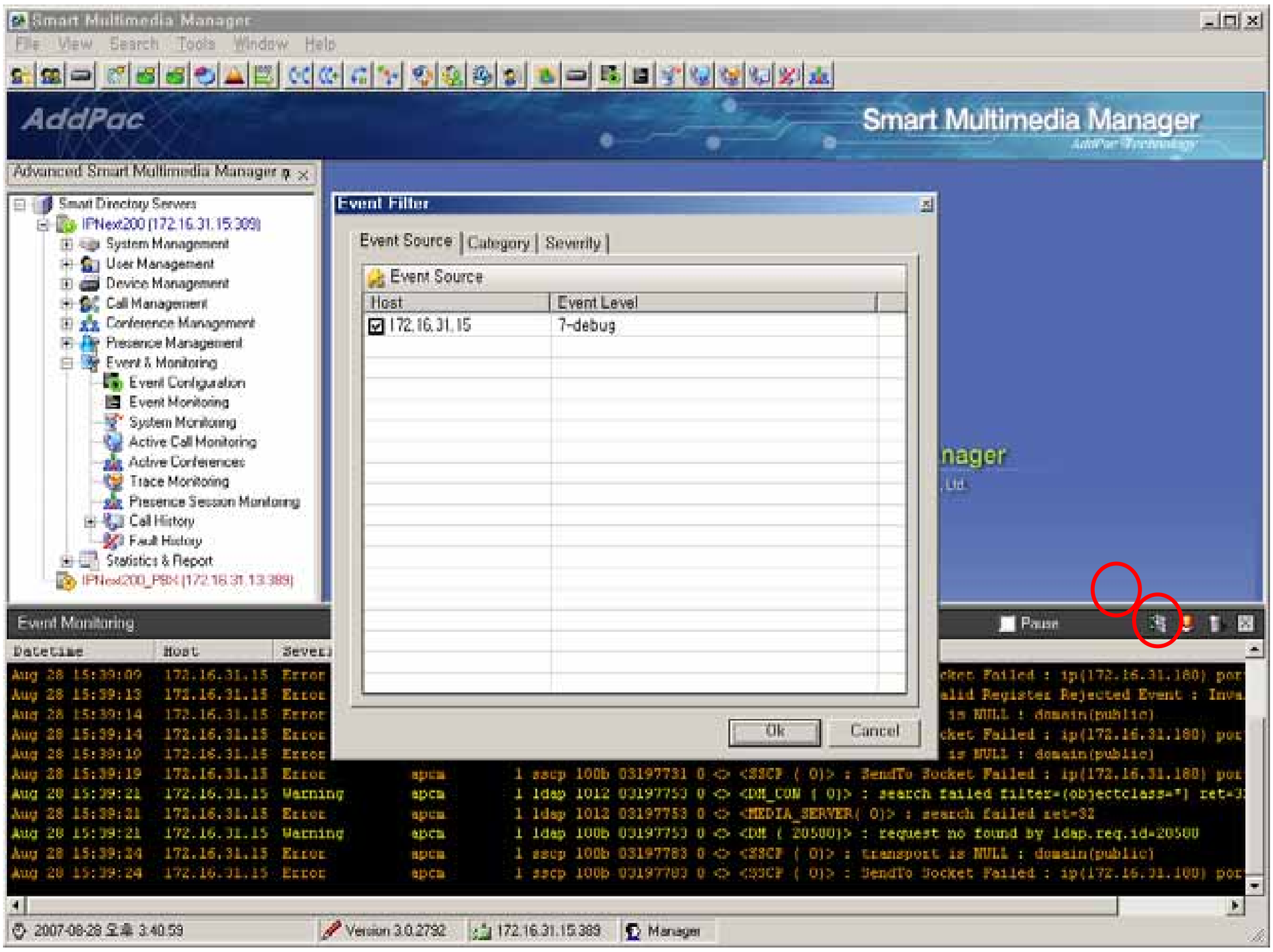


Figure 3-6 Event Filter Configuration Screen

When the icon with the red circle is selected, Event Source List comes out on the screen. Then the current Logging status and the button with the yellow highlight for Event source are used to start or stop Event.

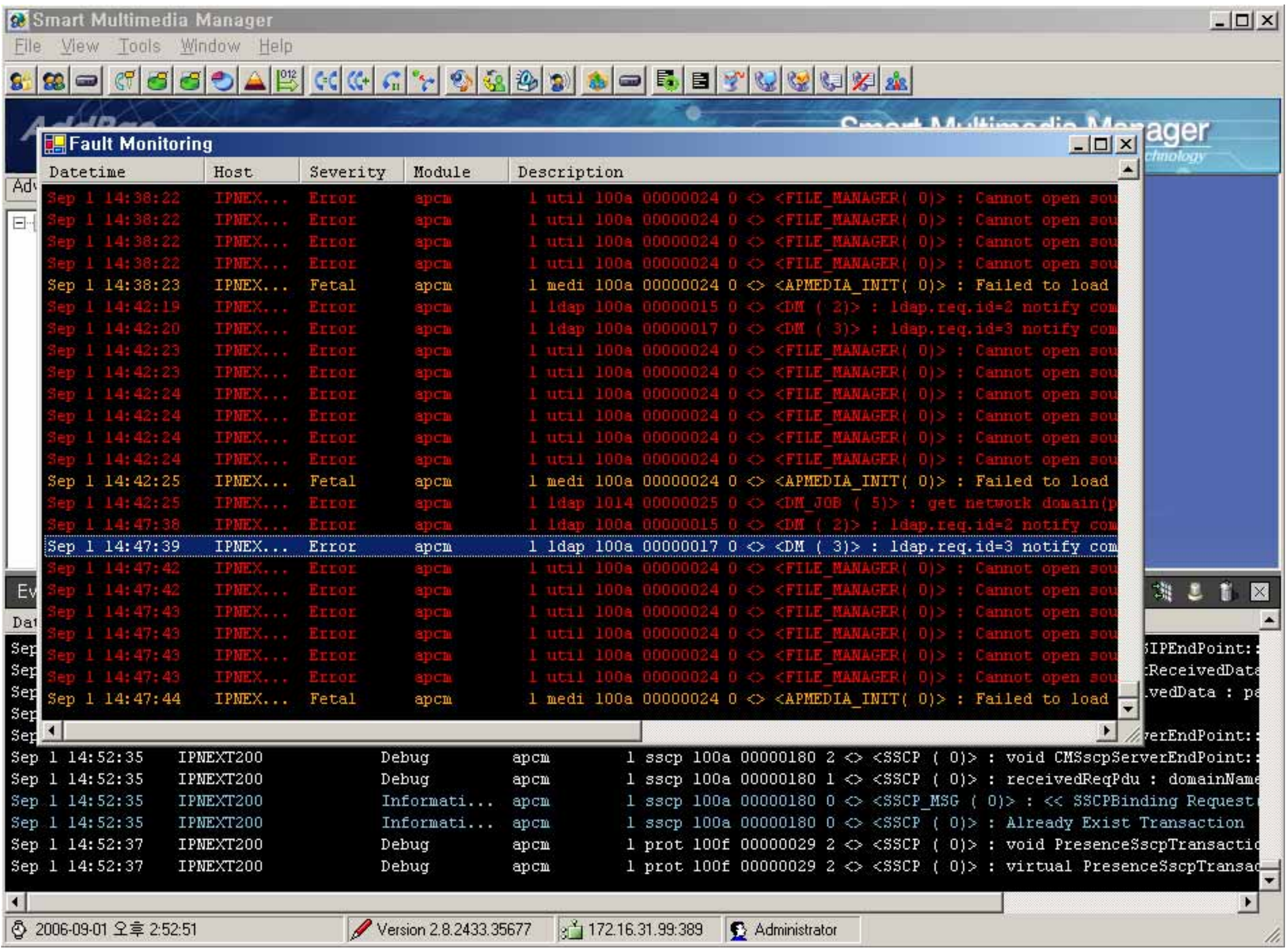


Figure 3-7 Fault Monitoring Screen

The above figure shows SMM Event Monitoring for itself. Only the filtered Event Log information is displayed on the screen by the selected Event Log for each chosen category.

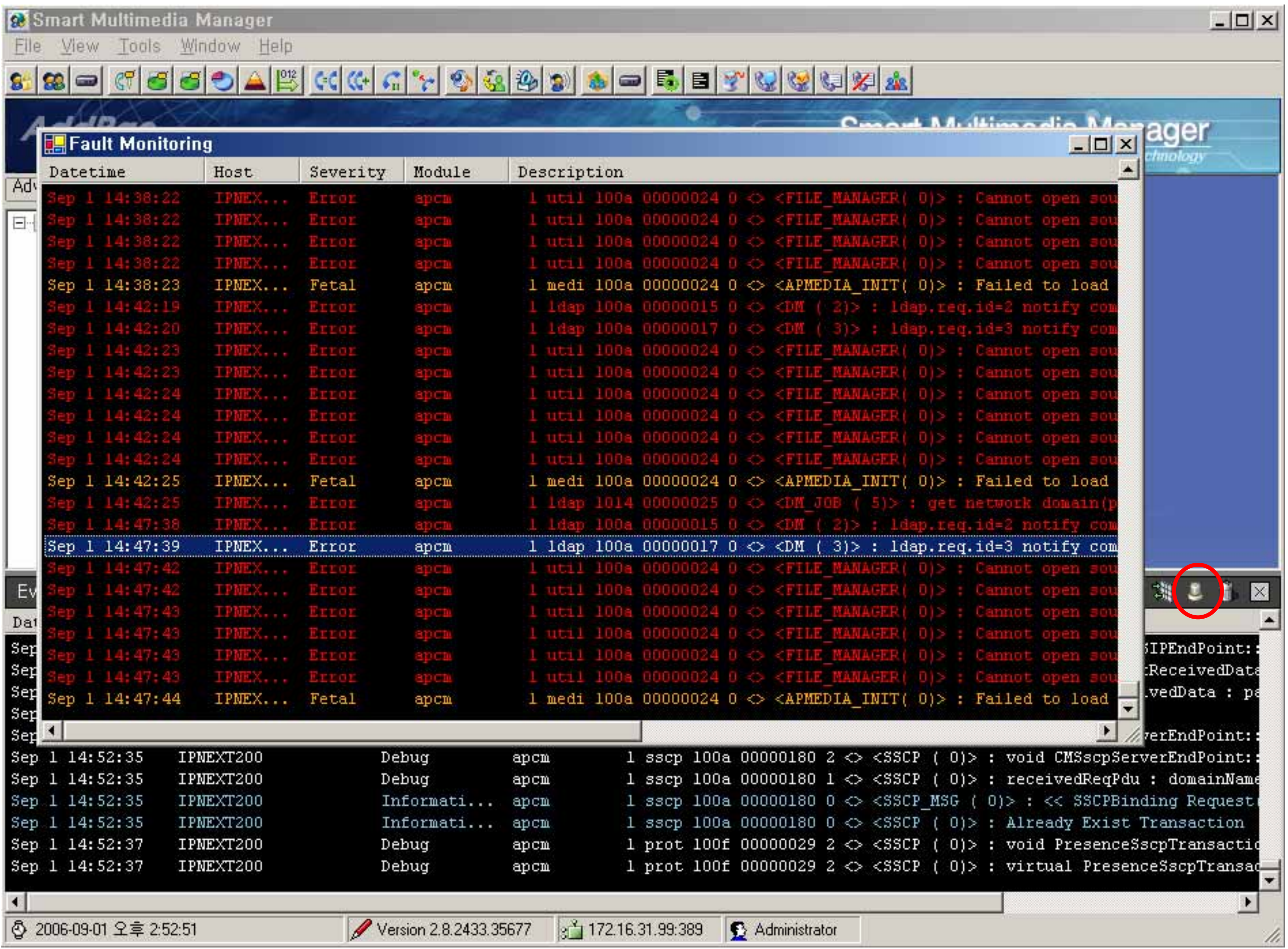


Figure 3-8 Fault Monitoring Screen

The above figure shows when a fault is found in the IPNext PBX, the fault is recognized and processed on a basis of Fault Level for Event Configuration. Then the icon with the red circle blinks. When it is selected, the details of fault are displayed.

System Monitoring

System Monitoring is a function to provide the information of CPU and Memory Usage of the system as to monitor the system resource of IPNext PBX.

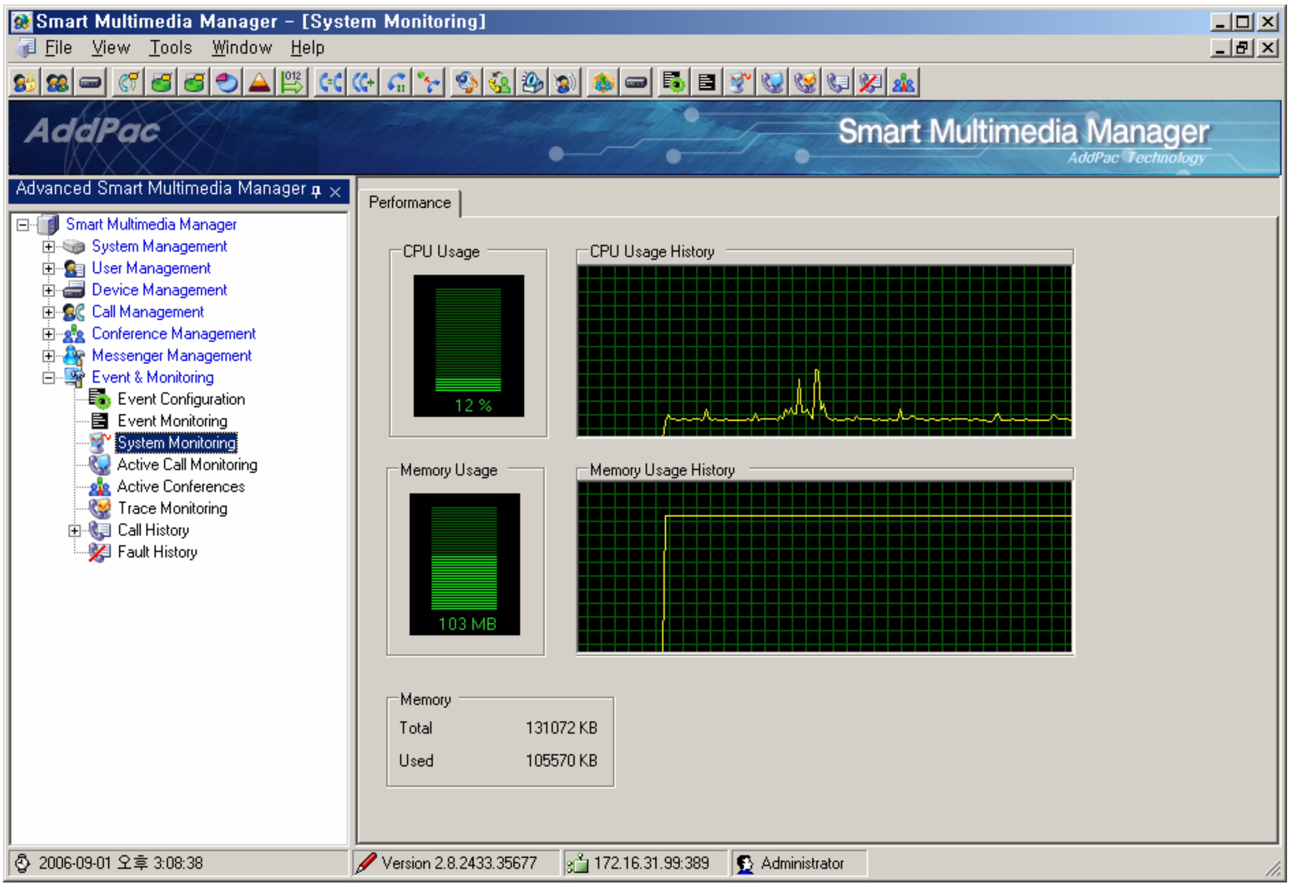


Figure 3-9 System Monitoring Screen

Active Call Monitoring

Active Call Monitoring is a function to identify the usage information of all the calls in the progress through all IPNext PBX in operation.

As it is shown in the Figure below, IP address of Call Manager is indicated and Call ID is also indicated to inform how many calls have been made previously through IPNext PBX. Also checking the point of time for connection of the corresponding call and the calling time in present progress on real-time basis is possible and it is also possible to identify who the caller and receiver are.

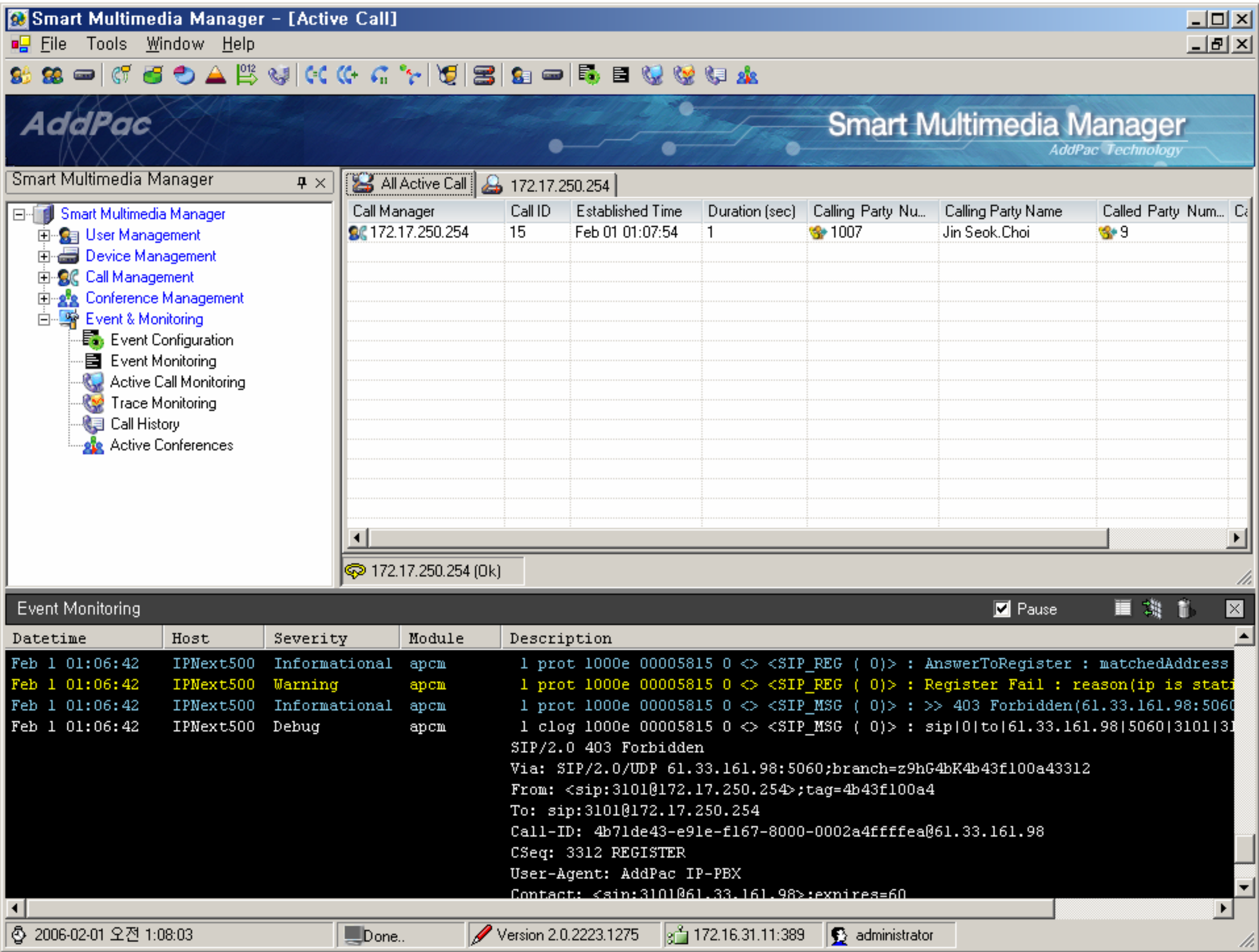


Figure 3-10 Active Call Monitoring Screen

Active Conference

Active Conferences is a function to monitor a list of Conference, on-going status and list of participants. Active Conference provides as to perform the functions of Conference settings and Layout changes, adding and deleting the participants, provisioning and others while it is monitoring.

The following screen presents how the functions of Active Conferences can be carried out (Event & Monitoring > Active Conferences)

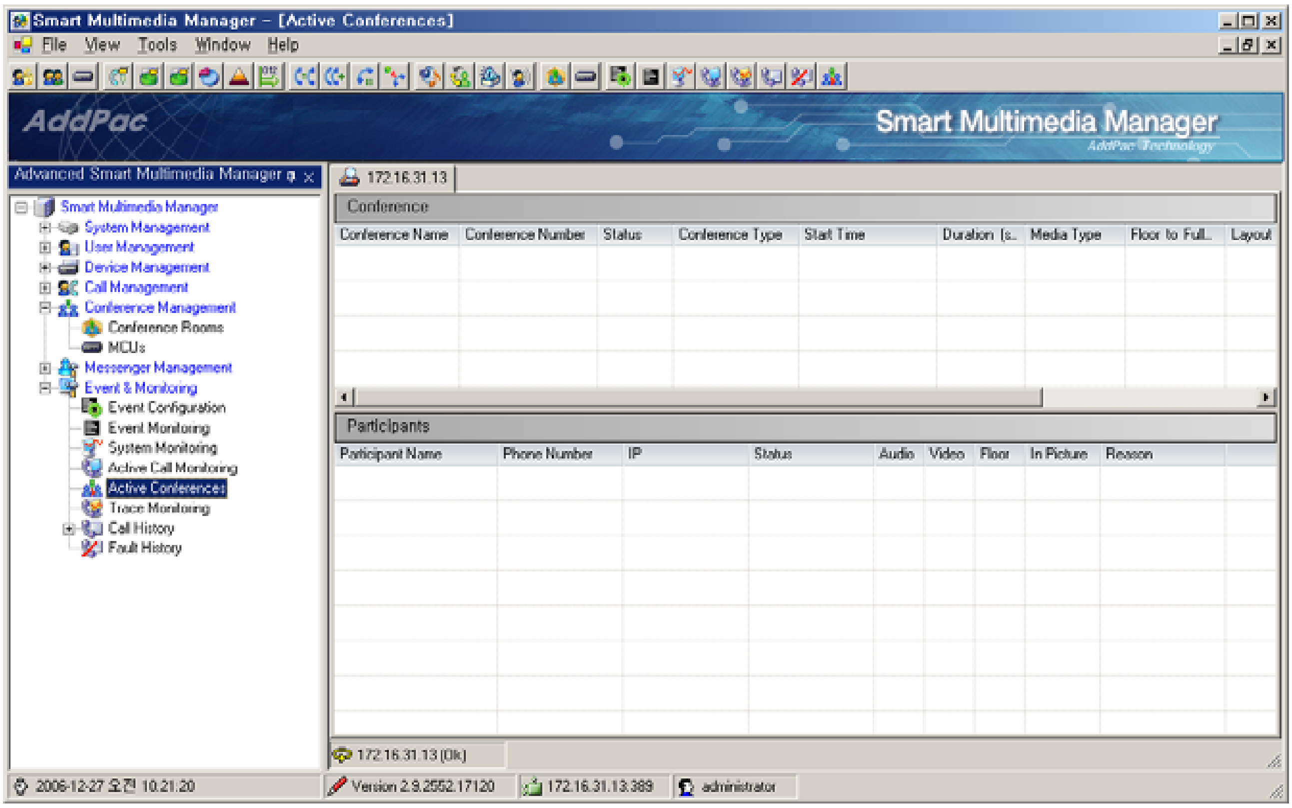


Figure 3-11 Active Conference Screen 1

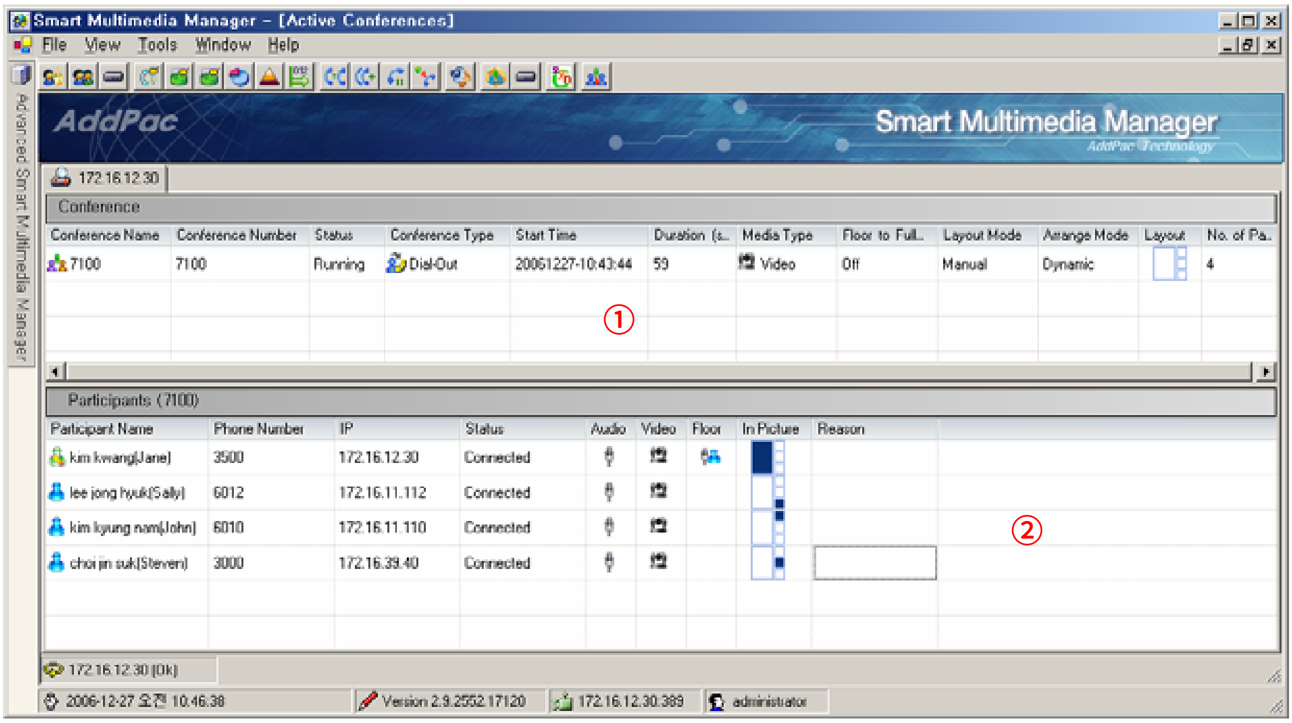


Figure 3-12 Active Conference Screen 2

Table 3-4 Active Conference Screen Description

Field	Description
1	This is the list Conference in progress, - Conference Name : a name of Conference to be displayed - Conference Number : a number of Conference to be displayed - Status : Display status - Conference Type : Types of Conference (Ad-Hoc, Dial-Out, Meet-Me..) - Start Time : starting time - Duration : the time for progress - Media Type : Audio or Audio + Video - Floor to Full-screen : The full screen for the user with the floor (on/off). - Layout Mode : Layout mode(Auto(symmetric / asymmetric), Manual) - Arrange Mode : Arrange mode(Dynamic, Fixed, Fixed-Center Reserved) - Layout : Video Layout - Field of Participant : the total number of participants
2	When the Conference is double-clicked, a list of participants for the corresponding Conference is to be displayed - Participant Name : a name of participant - Phone Number : a telephone number of the participant - IP : IP address of the partipating terminal

- Status : status of participation (Connect, Disconnect..)
- Audio : indicated for Audio or Mute
- Video : indicated for Video or Mute
- Floor : the person with the floor
- In Picture : location setting and display of Layout for the participants
- Reason

The followings are Menu Description of Conference and Participants settings in Active Conference.

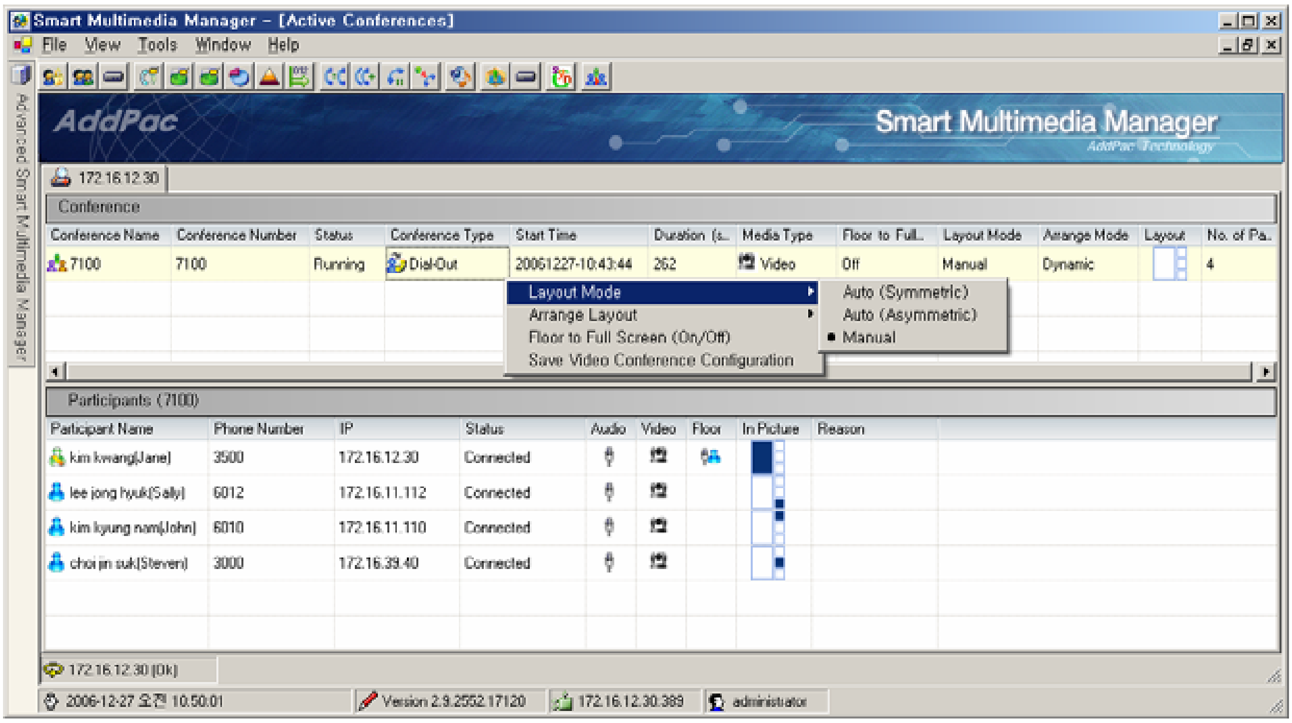


Figure 3-13 Active Conference – Conference Menu

Table 3-5 Active Conference - Conference Menu Description

Field	Description
1	- Layout Mode : Change Layout Mode of Conference (Auto(symmetric/asymmetric), Manual) - Arrange Layout : Change Arrange mode (Dynamic, Fixed, Fixed-Center Reserved) - Floor to Full Screen : An option whether to display the person with the floor in Full Screen - Save Video Conference Configuration : Save any change made to Video settings in Conference.

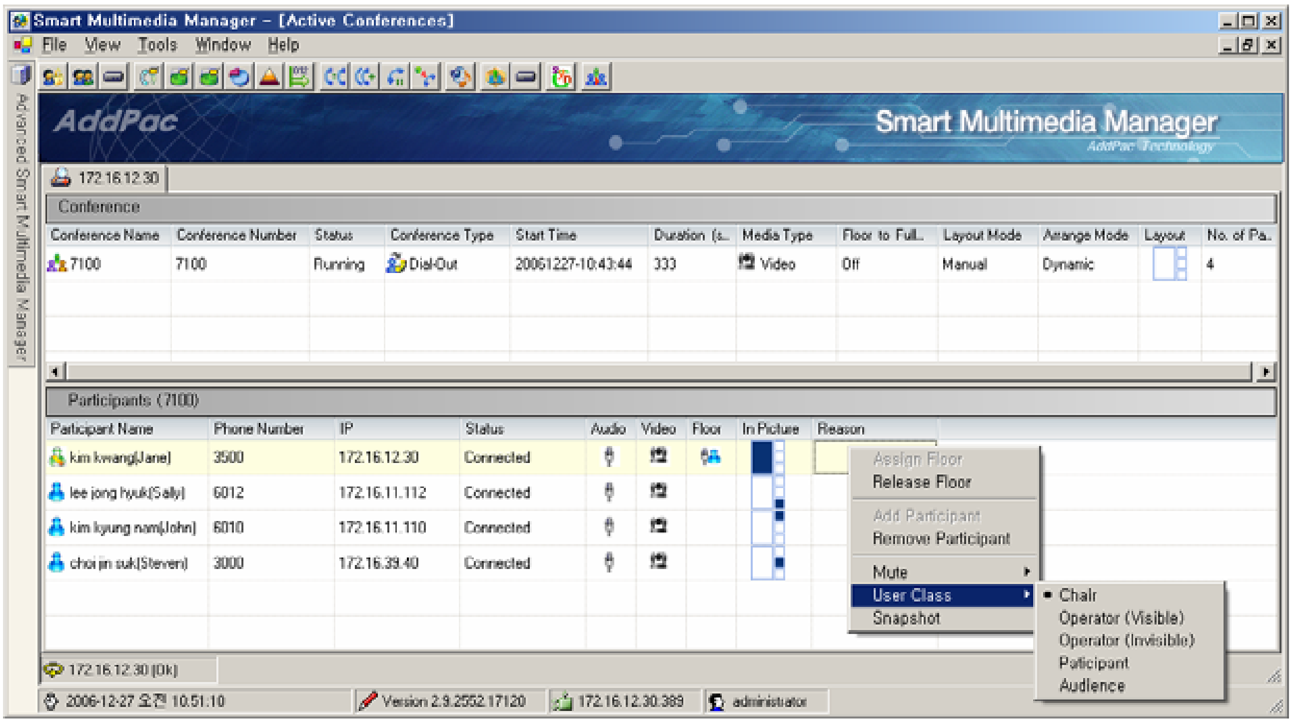


Figure 3-14 Active Conference - Participants Menu

Table 3-6 Active Conference – Participants Menu Description

Field	Description
1	- Assign Floor : give the floor to one of the participants - Release Floor : take the floor away from the participant - Add Participant :adding another new user who is not participating - Remove Participant : exit from the conference - Mute : mute/unmute for Audio, Video - User Class : Change the participant's class (Char, Operator(visible), Operator(Invisible), Participant, Audience) - Snapshot : start or stop snapshot

When Conference is set to Video, the details of progress can be monitored through SMM by using Snapshot of Screen for Conference. Snapshot Screen can be updated automatically or manually. The screen is as to follow:

Snapshot

During videoconferencing, Snapshot feature is used to take a picture of the conference to verify the process. Snapshot screen can be updated either manually or automatically. The screen describes as to follow:

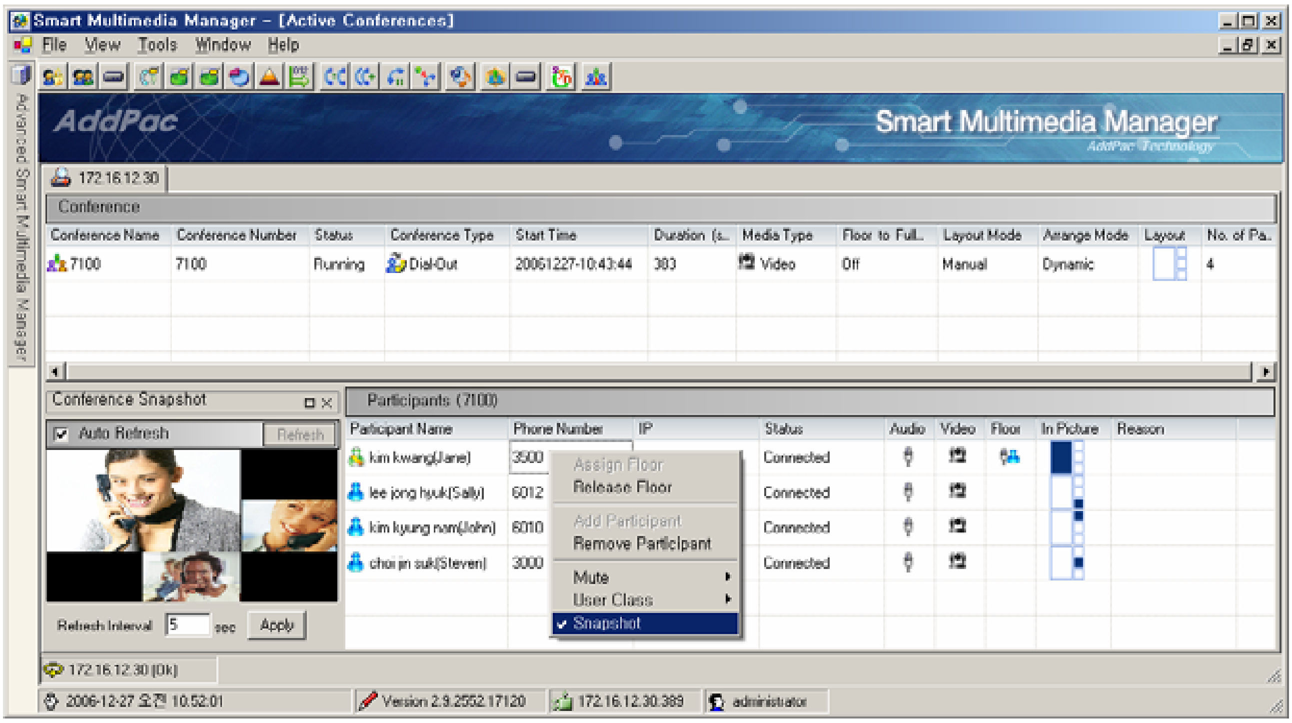


Figure 3-15 Active Conference – Snapshot Screen

Layout View

During videoconferencing, Layout View allows to set up the progress status of the participants. The screen describes as to follow:

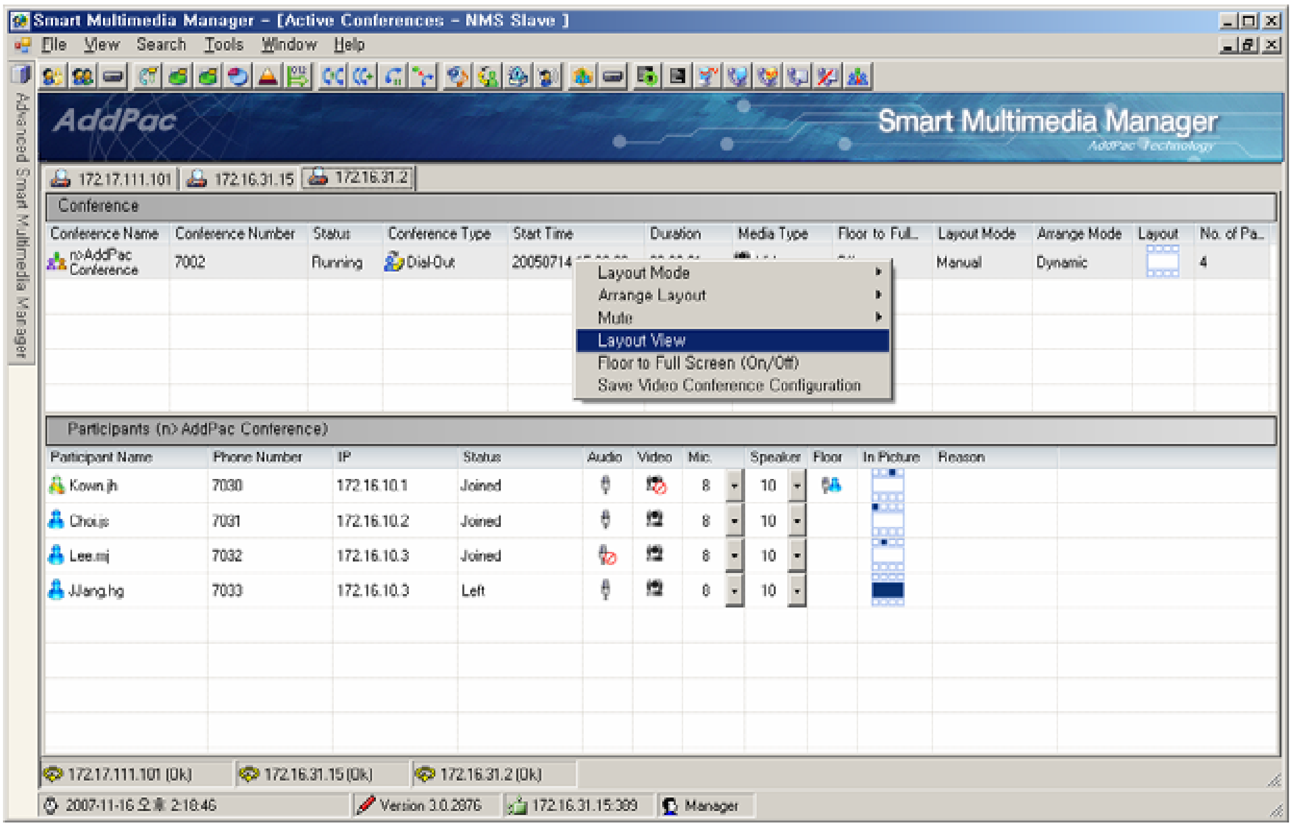


그림 3-16 Active Conference – Layout View 실행

You can change a position of layout easily as to drag and drop by mouse. You can set up the floor, add and exit participants. The screen describes as to follow:

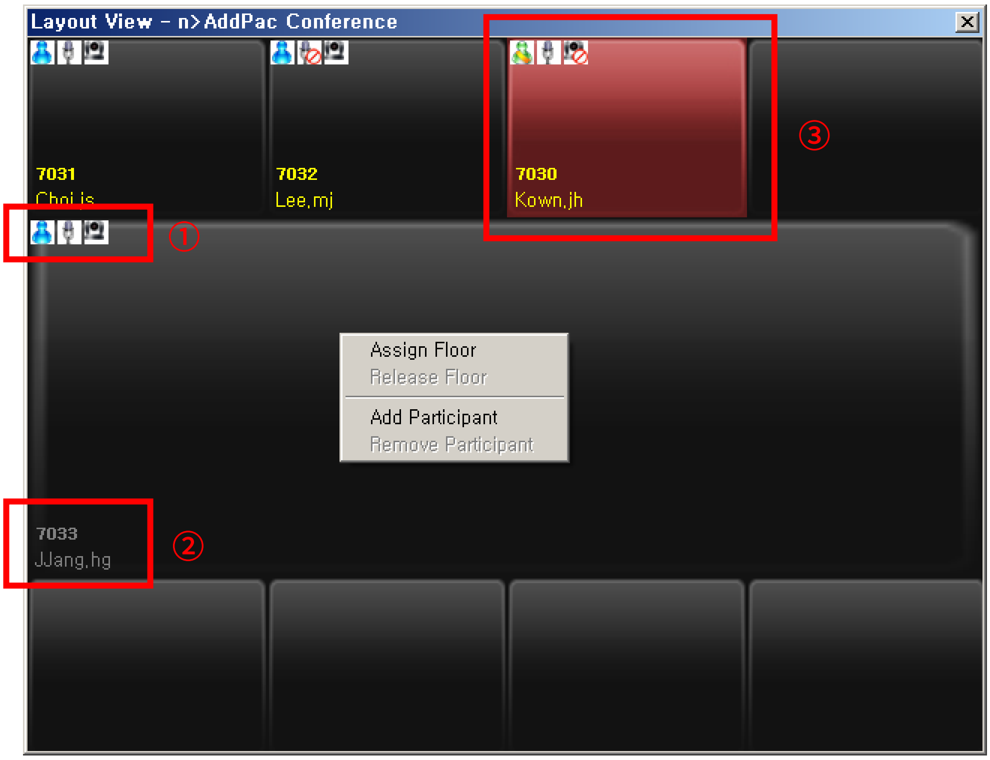


Figure 3-17 Layout View

Table 3-7 Description of Layout View

No.	Description
1	Layout indicators display Mute setup (on/off) and status for the audio/video and each User Class of each participant from the
2	Layout status displays each participant's details (phone number, nick name) and participating status by colors of the letters (yellow: participating, gray: not participating)
3	Display a status of the floor with red color of the background

Trace Monitoring

Trace monitoring allows the user to check Call Flow performed by IPNext PBX presently.

When a call is made, as it is shown the Figure below, the corresponding time can be known. Using call ID, the user can check how many calls have been made. Also identifying Protocol (SIP/H.323/SSCP) is possible. For instance, monitoring is possible for how SIP message is delivered in the following:

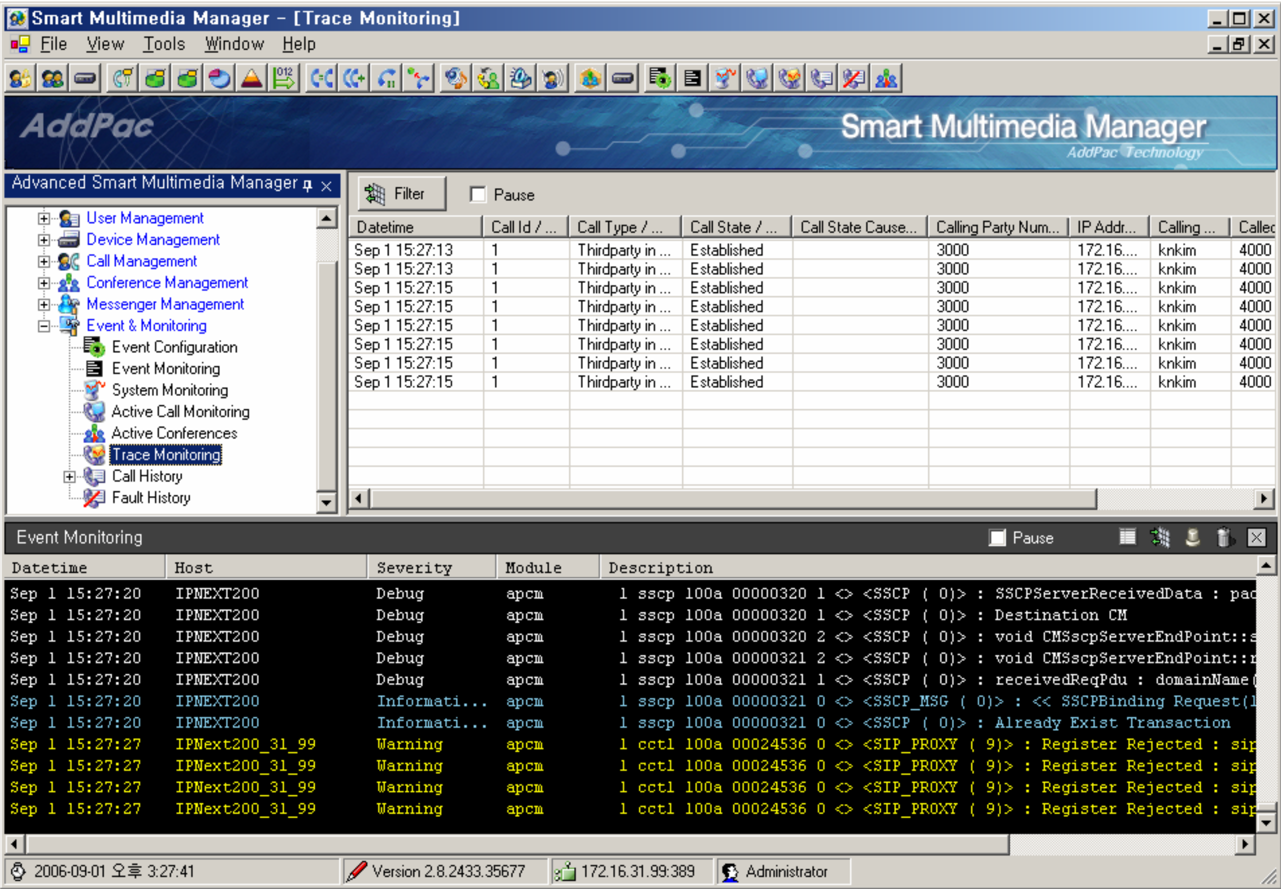


Figure 3-18 Trace Monitoring Screen 1

If the user wants to know the detailed message, the corresponding message can be double-clicked as it is shown in the following figure.

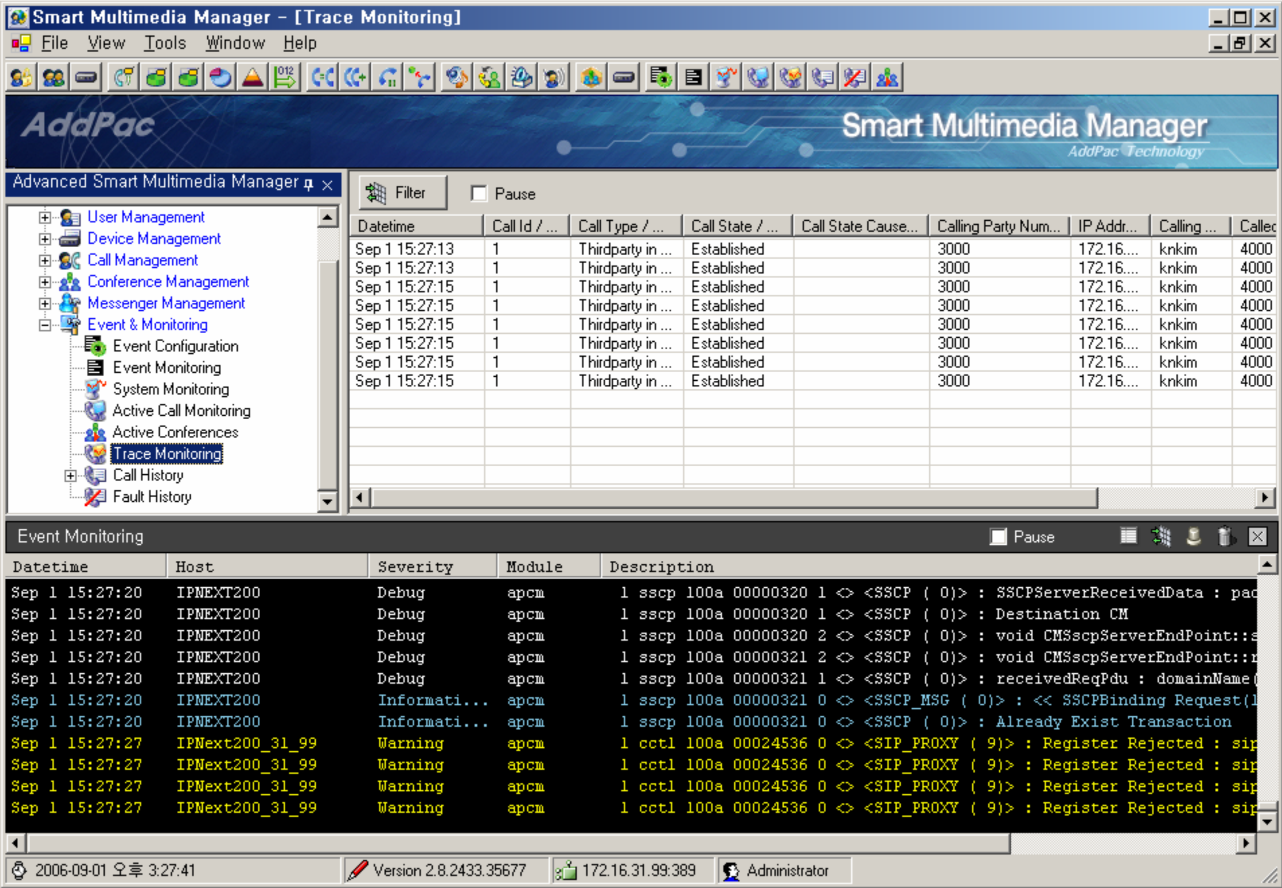


Figure 3-19 Trace Monitoring Screen 2

When too much information of trace monitoring is displayed on the screen, the user can configure the settings for the filter of the desired type (Host, Phone Number), then only the corresponding information of trace is displayed.

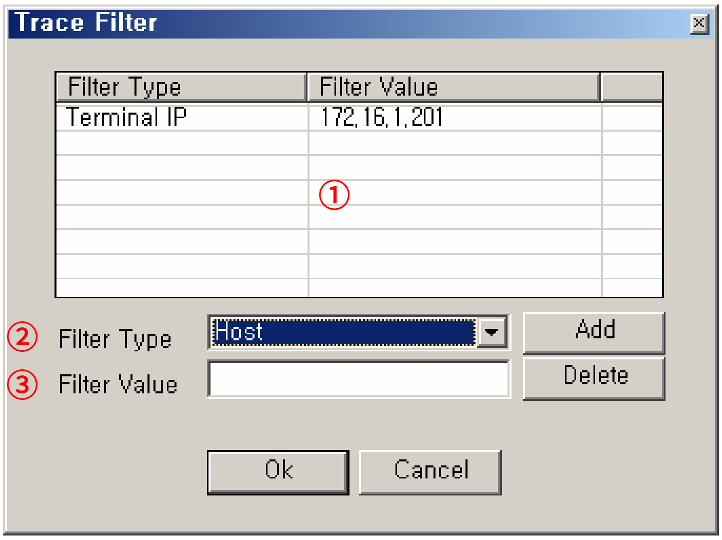


Figure 3-20 Trace Filter Screen

Table 3-8 Trace Monitoring – Trace Filter Properties

Field	Description
1	This is a list to filter trace information.
2	Select a tilter type(Host, Phone Number).
3	Choose a filter value for the filter type.

Presence Session Monitoring

Presence Session Monitoring monitors session of the terminals connected to presence server when an external Presence Server interoperates with IPNext PBX. This feature provides access information of Smart Messenger and the phone user.

resence Session is divided by Messenger and IP-Phone and can be displayed by Presence Group, in Summary format. You can check the session details by selecting a category of Presence Group. You can also update the session basing on the session details in the group (Directory, SpeedButton Profile). The screen describes as to follow:

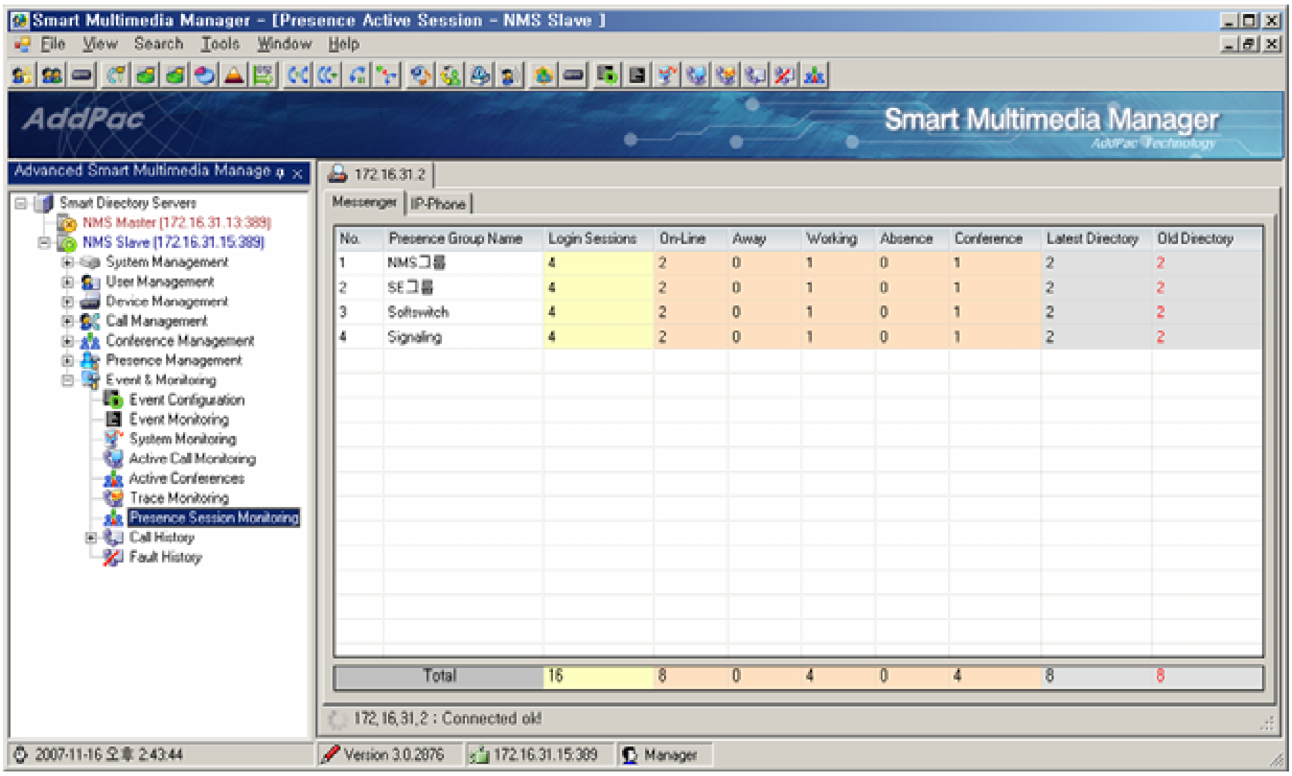


Figure 3-21 Presence Active Session Screen

The following figure displays the relevant session details when a category is selected for a specific Presence Group. The contents include session status, user information, organization, IP address, Presence Group, Directory Version.

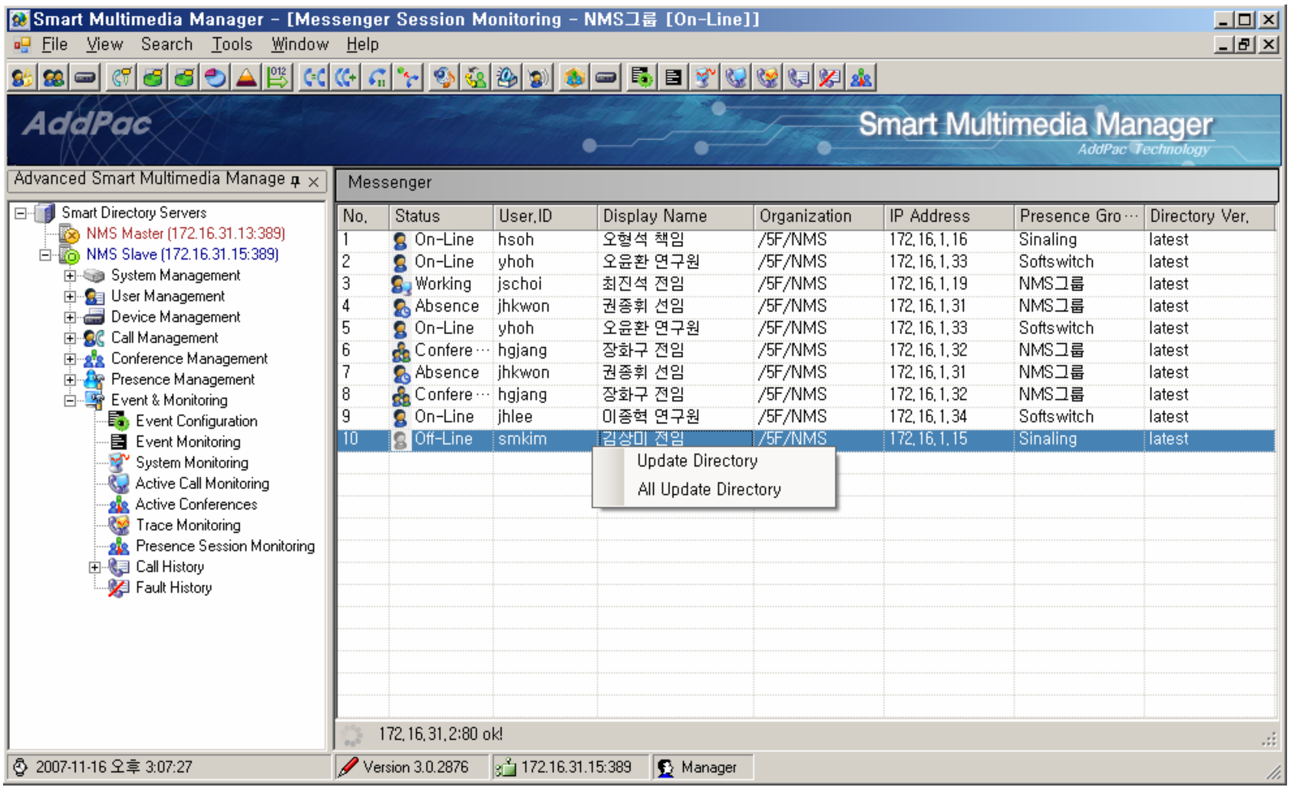


Figure 3-22 Messenger Session Monitoring Screen

When the session list has an old version, the version can be updated by Update Directory.

Call History

Call history is a function of displaying on screen as to process the call history of event log file saved in SMM or SEM(Smart Event Manager).

The administrator can designate and bring the event log file.

Through SEM, the administrator can bring event log information remotely which can match with the condition.

The following figure shows the call history (Event & Monitoring > Call History) performed on the screen.

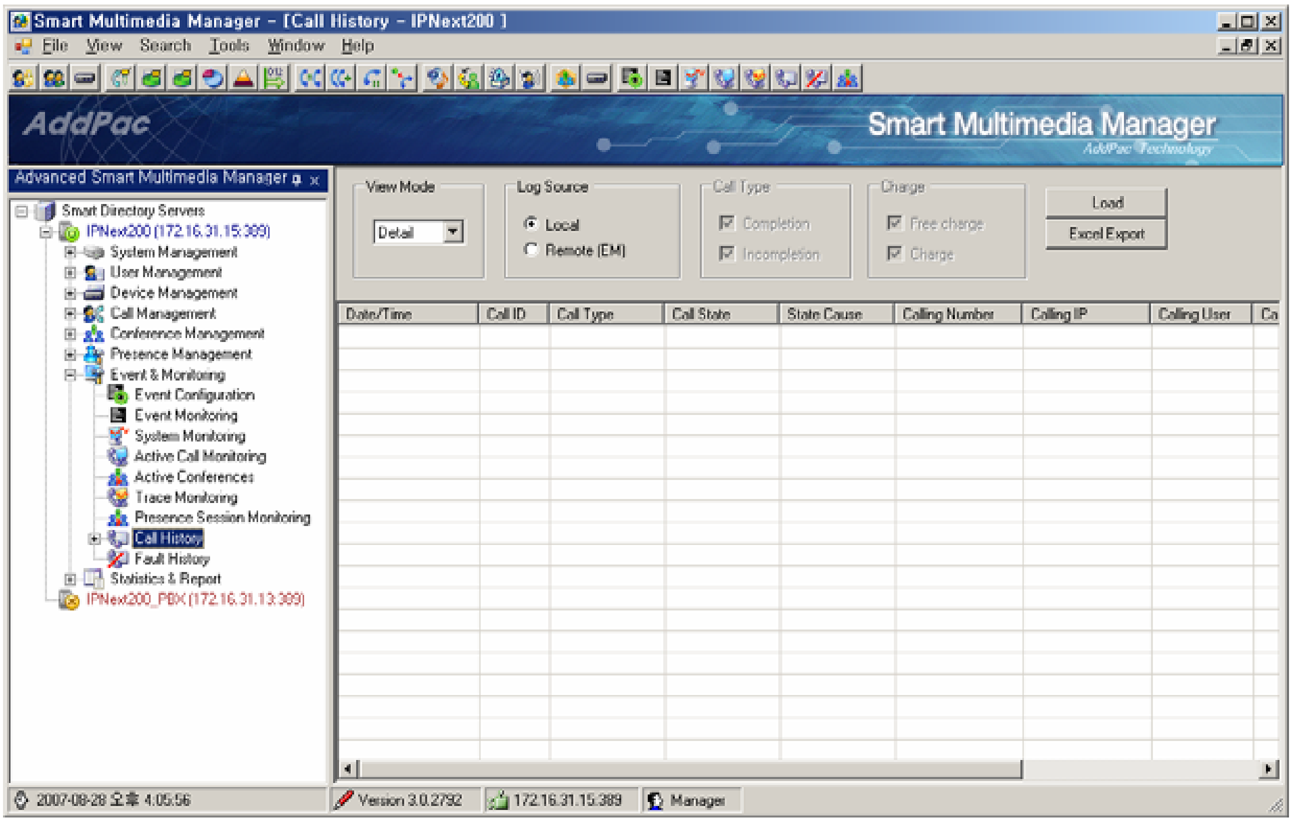


Figure 3-23 Call History Screen 1

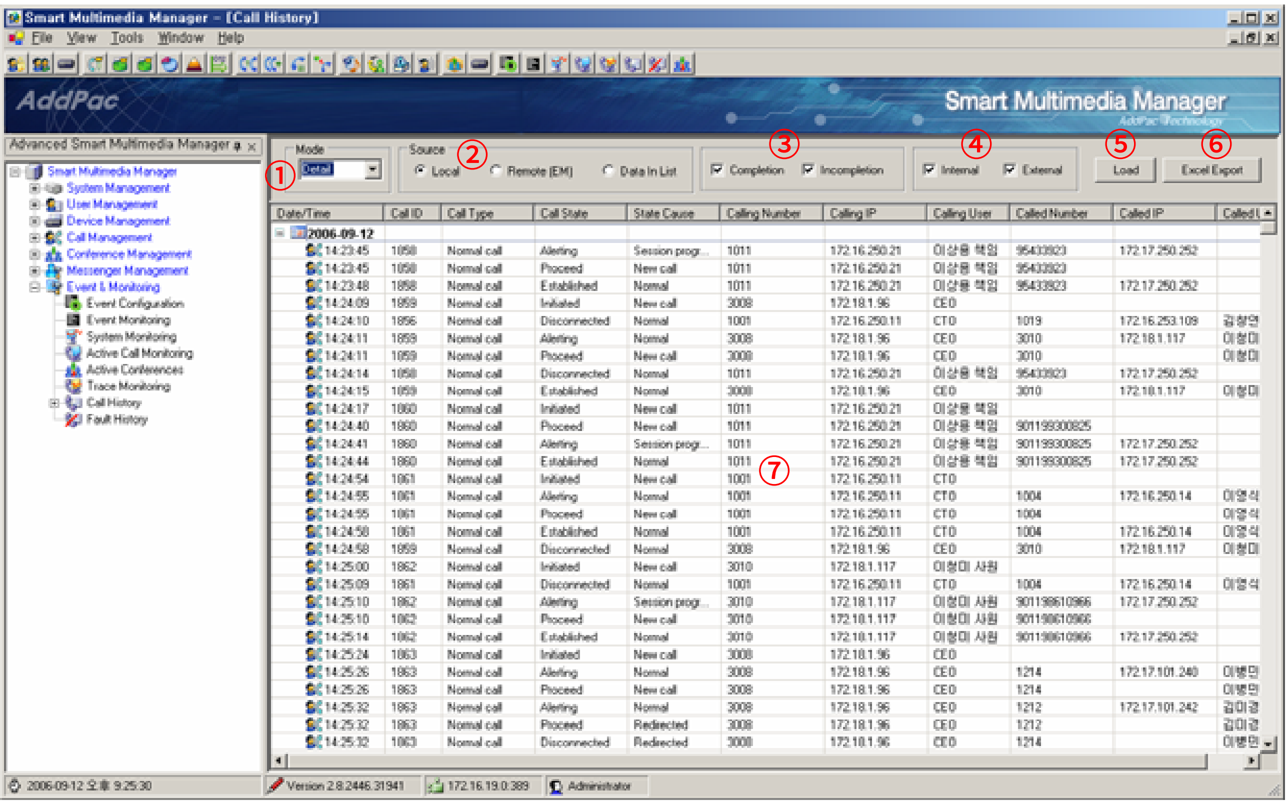


Figure 3-24 Call History Screen 2

Table 3-9 Call History Screen Description

Field	Description
1	Select a type of screen view (detail, summary)
2	Select a source to bring the call history. - Local : bring an event log file from a local PC - Remote(EM) : bring the event log by a remote access to SEM - Data In List : filter the event log that has been brought in
3	Select an option of displaying complete/incomplete call (when the mode is set to summary)
4	Select an option of displaying internal/external call (when the mode is set to summary).
5	This is a button to query the even log.
6	Save the queried file with call log information in the Microsoft Excel file
7	Call history can be displayed by a sequence of dates. When the sequence is expanded, the corresponding call history is displayed

When 'Load' button is clicked to check the call history, the conditional settings for the history filter is displayed on the screen in the following figure. When 'OK' button is clicked after the setting of the filter conditions, an event log file can be selected or the call history can be queried from SEM depending on which source (Local, Remote, Data In List) is selected.

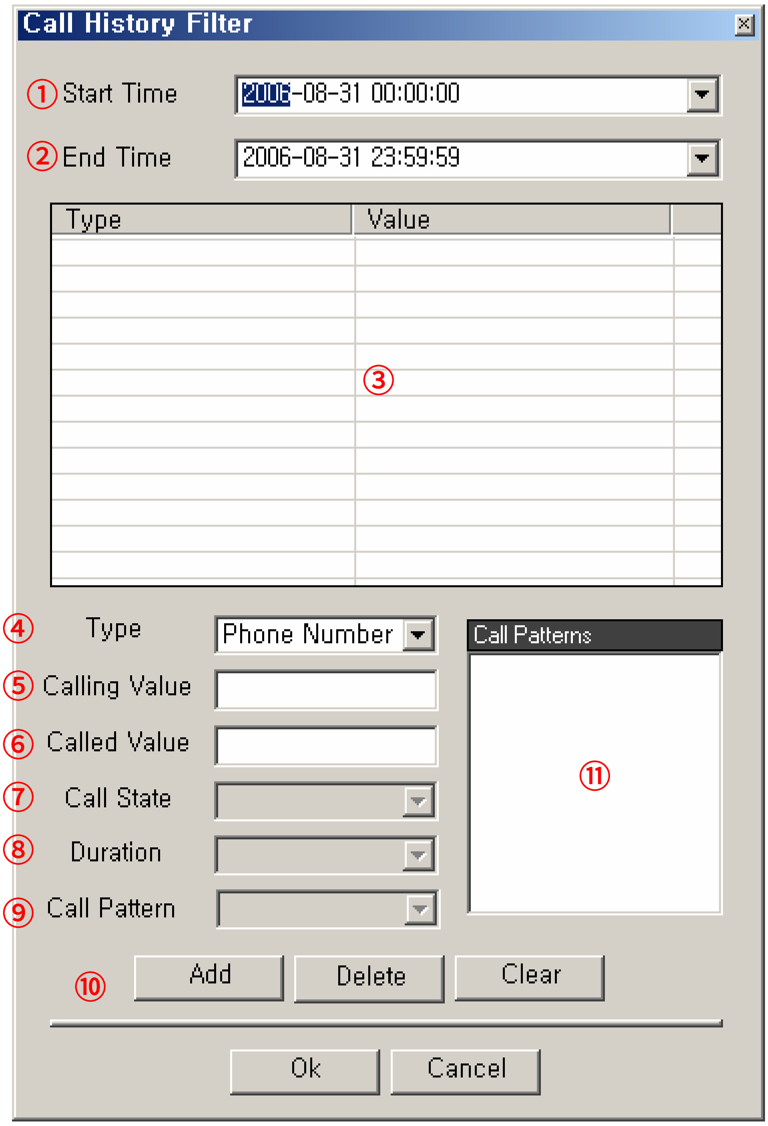


Figure 3-25 Call History Filter Properties

Table 3-10 Call History Filter Properties Description

Field	Description
1	Set up a starting time of the call history to be queried.
2	Set up an ending time of the call history to be queried.
3	This is a list to filter the call history.
4	Select the conditions to filter (Host, Phone Number, Mac address, User Name, Call Id and others)

5	Enter an information on a caller's to be queried basing on the conditions to filter
6	Enter an information on a receiver basing on the condition to filter
7	Set up the search conditions for Call status when the filter condition is Call State
8	Select when Filter condition is Duration (seconds)
9	Select when Filter condition is Call Pattern. Call Pattern can be listed when it is defined prior to search
10	Register, delete, delete all the search conditions
11	Call Pattern numbers can be listed for the selected Call Pattern and the desired Call Pattern number can selected to be searched.

Configuring Call Pattern

Call pattern is a function to search by a pattern in call history. For instance, call pattern of a long distance call is to be registered to search for a number of a long distance call. Search by a pattern is possible by registering call pattern as a search condition of call history.

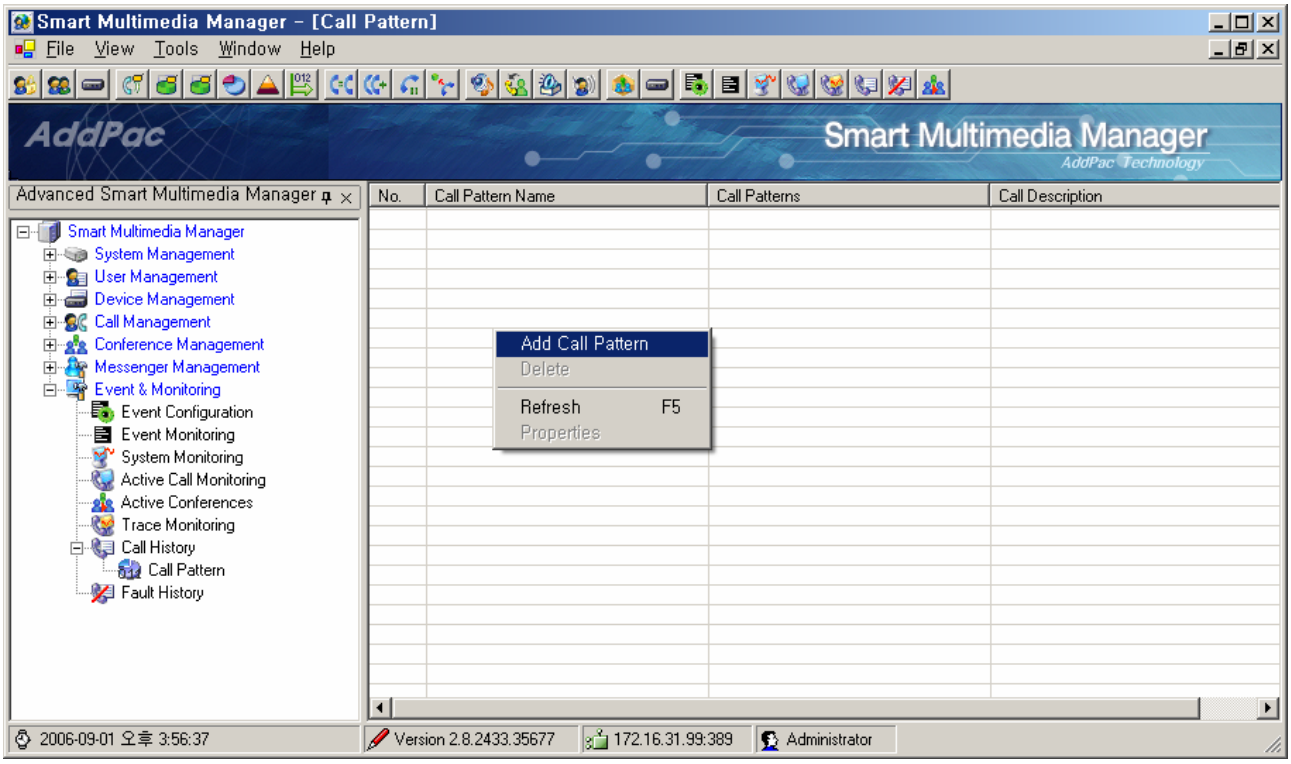


Figure 3-26 Call Pattern Configuration Screen

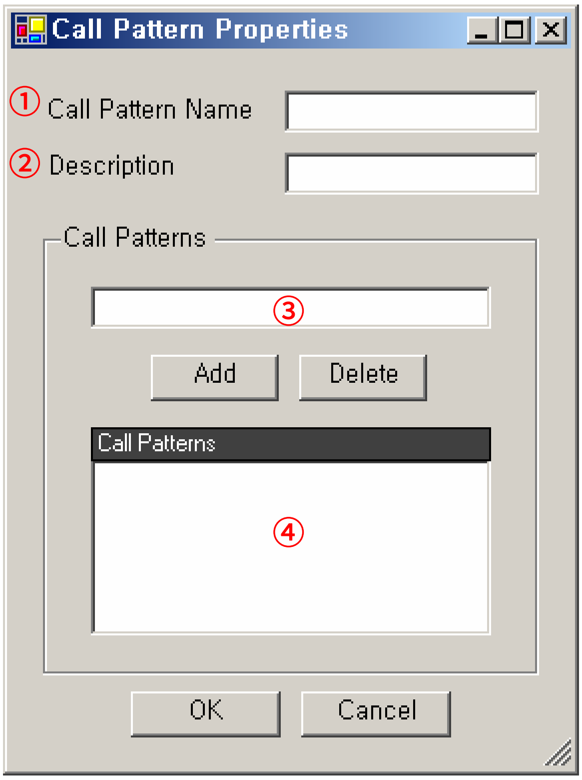


Figure 3-27 Call Pattern Properties

Table 3-11 Call Pattern Properties Description

Field	Description
1	Register a name of call pattern.
2	Register a description of the call pattern.
3	Register a number of the call pattern.
4	This is a list of the registered call patterns

Fault History

Fault history is a saved log file of SMM for the system failures or a function of the event manager to query the fault history within a determined time period.

The way to save the log in SMM can be described in the following manner as it is shown in the figure below:

- Set Source to 'Local', then click 'Load'.
- Select a file (with a date) to be queried for the fault history

The way to manage the log by the event manager can be described in the following manner as it is shown in the figure below:

- Fix a time period and request to the event manager
- Receive the corresponding fault history from the event manager
- Display on SMM screen

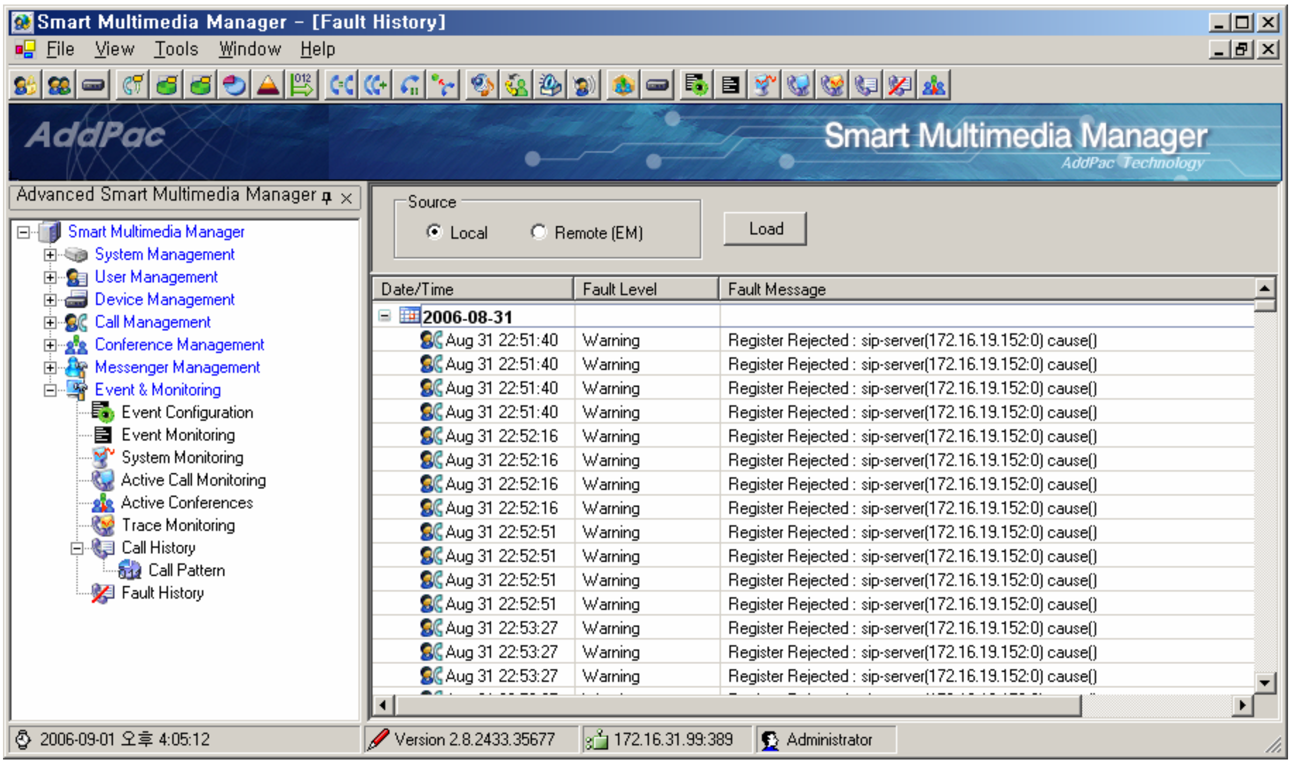


Figure 3-28 Fault History Screen

Chapter 4. Statistics & Report

SMM provides and reporting functions and the statistical data of Call Usage, Number of Calls, Incompletion Call and Rank, basing of the basic information that IP-PBX provides. Smart Event Manager (SEM) collects the events of IPNext PBX and allows this feature to organize call statistic database. Thus, the event source of SMM must be configured first in order to operate this feature.

* When the event source is provided from Call Manager, Statistics & Report Menu can not be operated.

Call Usage Statistics

A statistic data for call usage of the IP-PBX, in a specified time period, can be generated by search conditions and divided by date, month and provided in graph and table formats. The following figure shows the statistic screen:

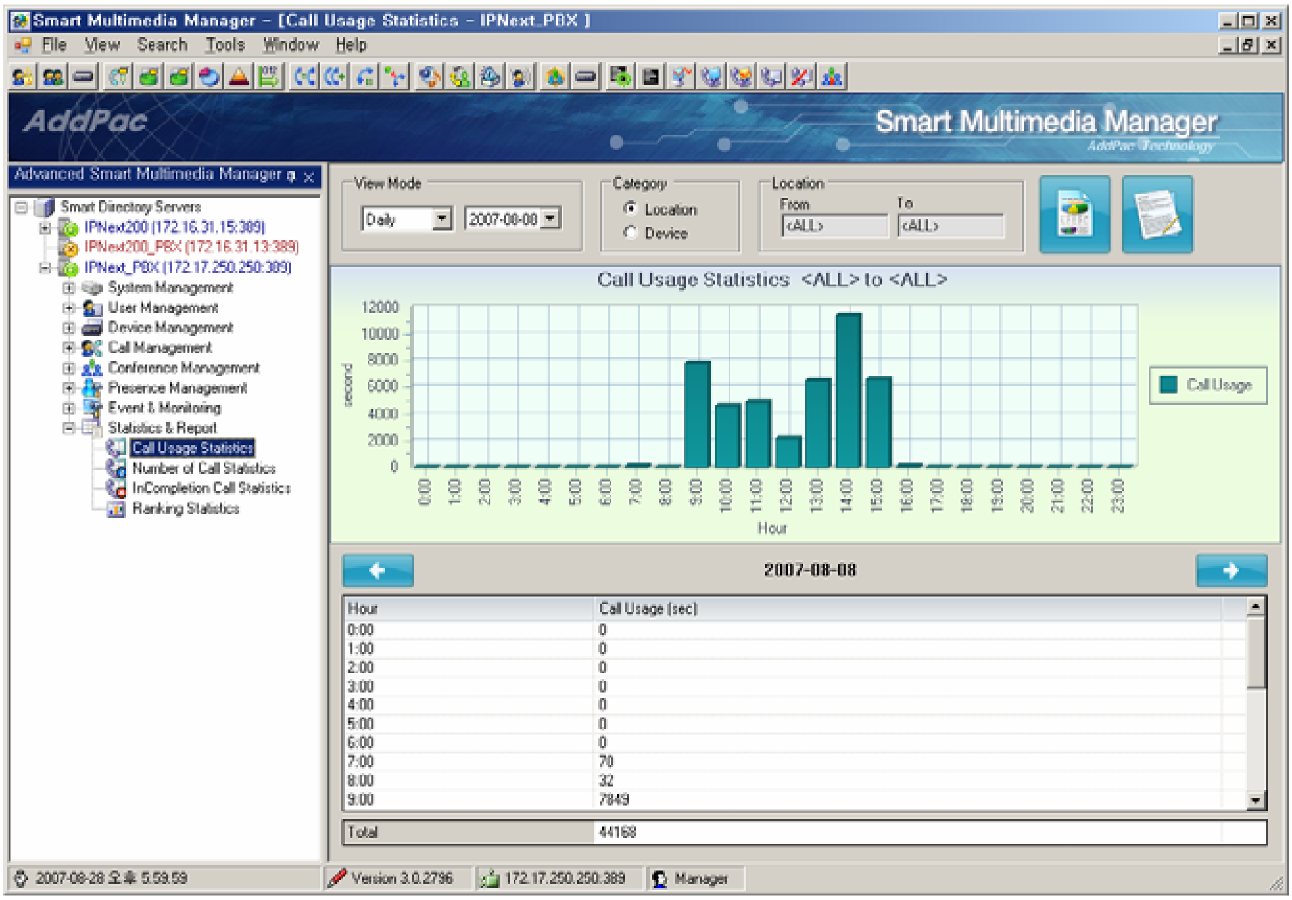


Figure 4-1 Call Usage Statistics

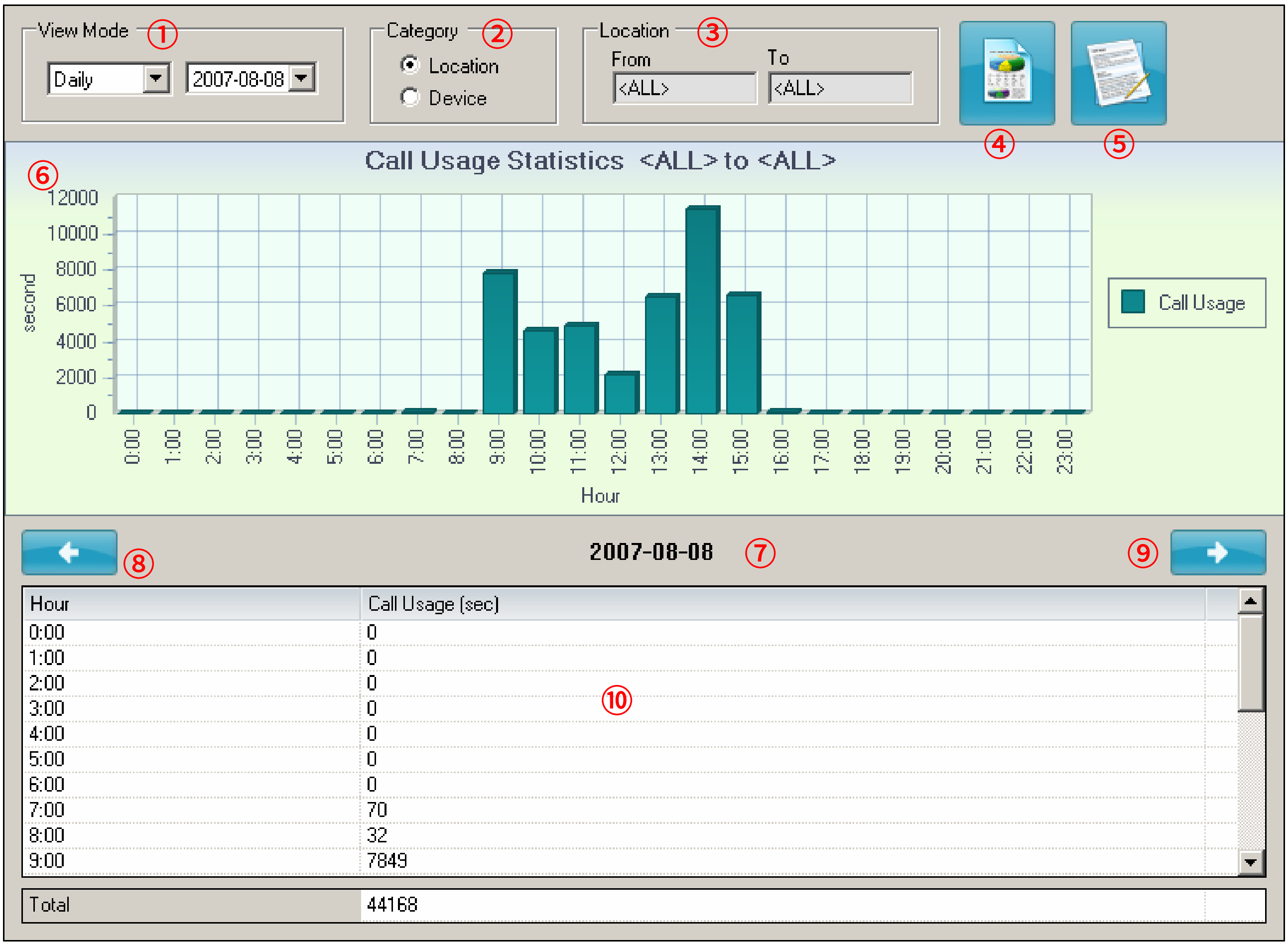
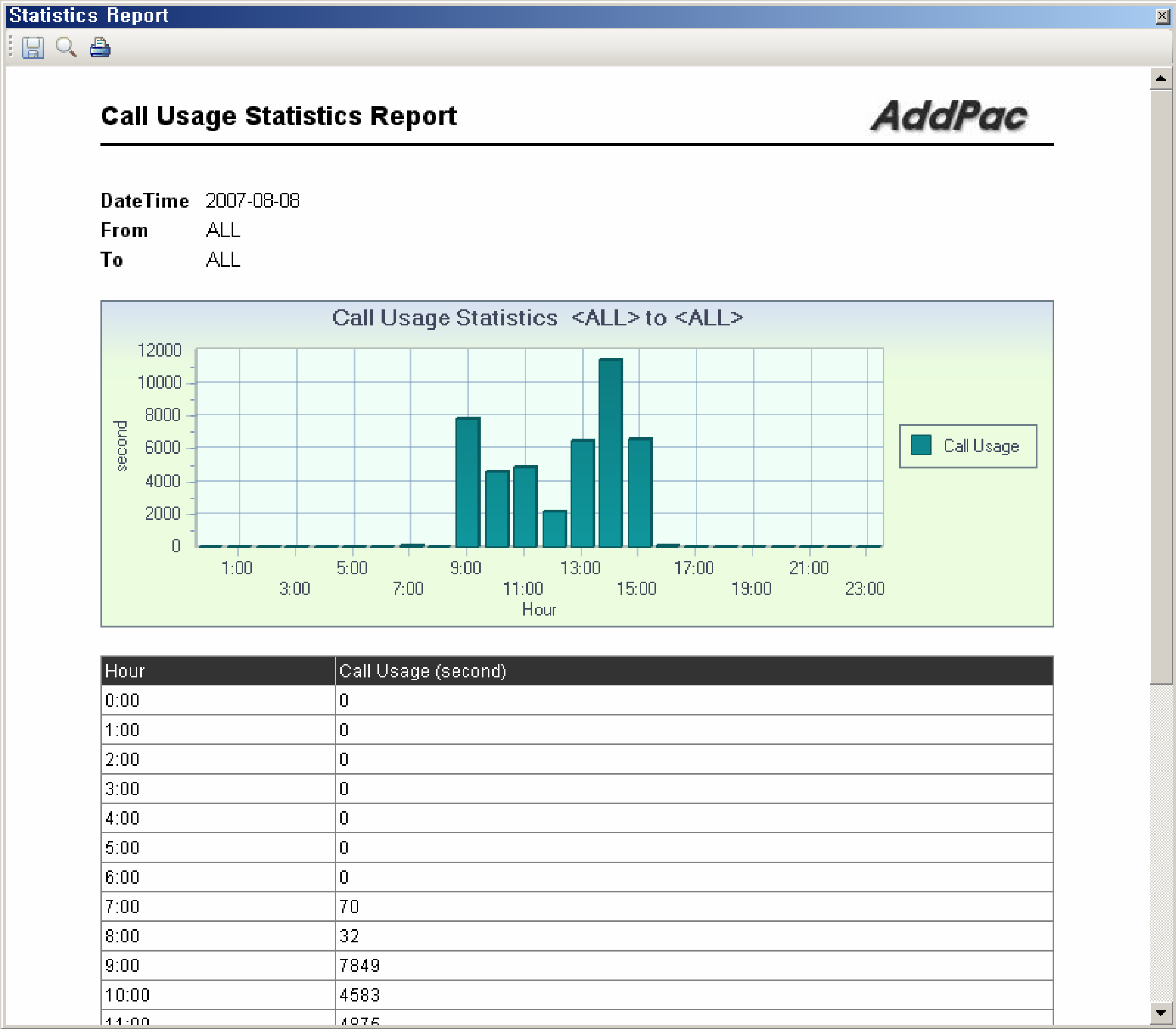


Figure 4-2 Call Usage Statistics

Table 4-1 Description of Call Usage Statistics

No.	Description
1	Select the standards for the statistic data to be displayed: - Daily : one day basis (24 hours) - Monthly : one month basis (31 days) - hourly : the statistic data in a specified day
2	Select a search standard (Location, Device)
3	Select a search condition basing on the search standard (From, To)
4	Execute the statistic data basing on the search conditions which have been entered
5	Display the statistic data which has been searched on a report
6	Display the static data by graph
7	The norm for the statistic data
8	Search the previous day or month basing on the norm for output (daily, monthly)
9	Search the next day or month basing on the norm for output (daily, monthly)
10	Display static data on the table

The statistic data can be displayed in the following by clicking the report button:



Hour	Call Usage (second)
0:00	0
1:00	0
2:00	0
3:00	0
4:00	0
5:00	0
6:00	0
7:00	70
8:00	32
9:00	7849
10:00	4583
11:00	4875

Figure 4-3 Call Usage Statistics Report

Number of Calls Based Statistics

A statistic data for number of calls based statistics of the IP-PBX, in a specified time period, can be generated by search conditions and divided by date, month and provided in graph and table formats. The following figure shows the statistic screen:

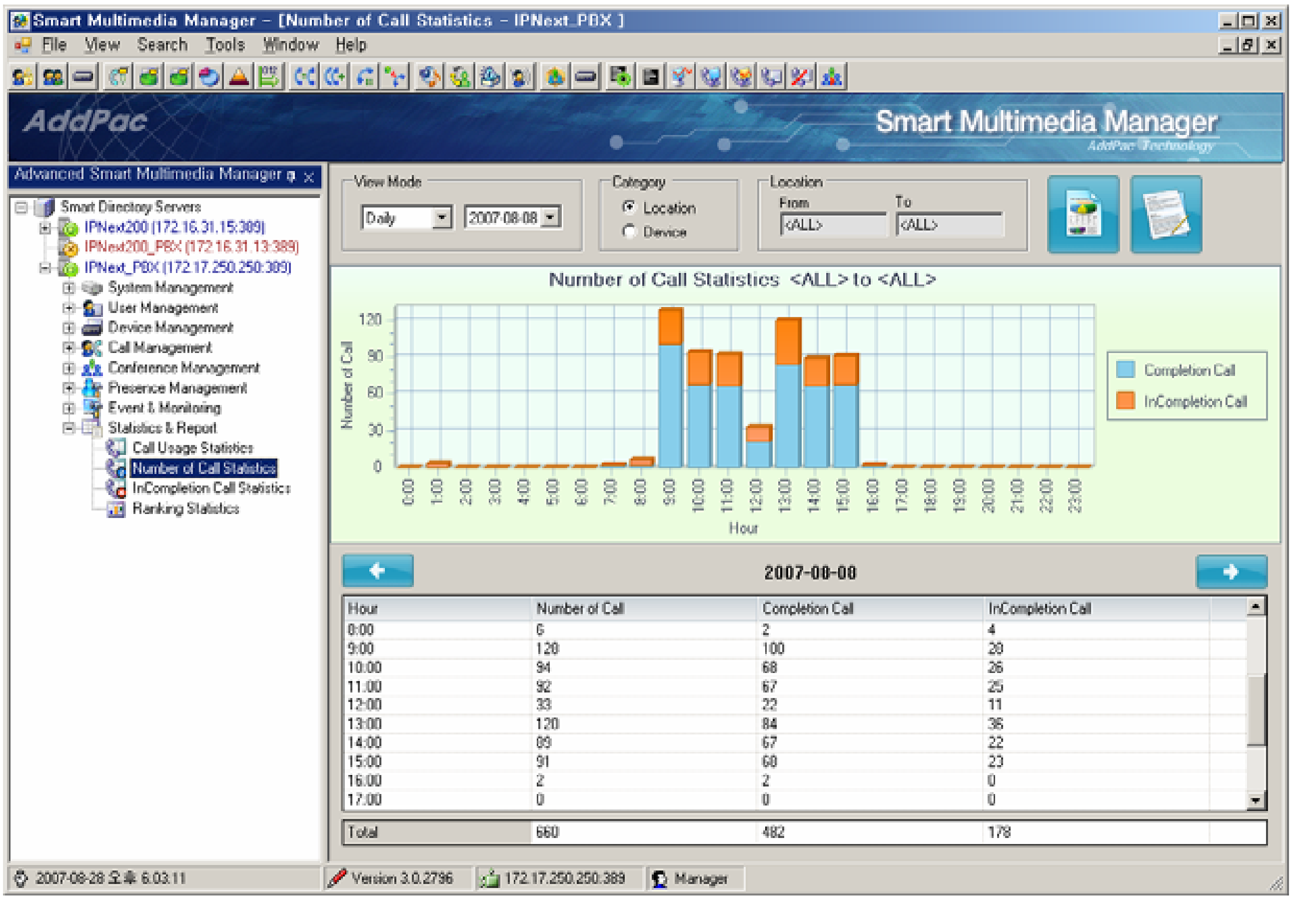


Figure 4-4 Number of Calls Based Statistics

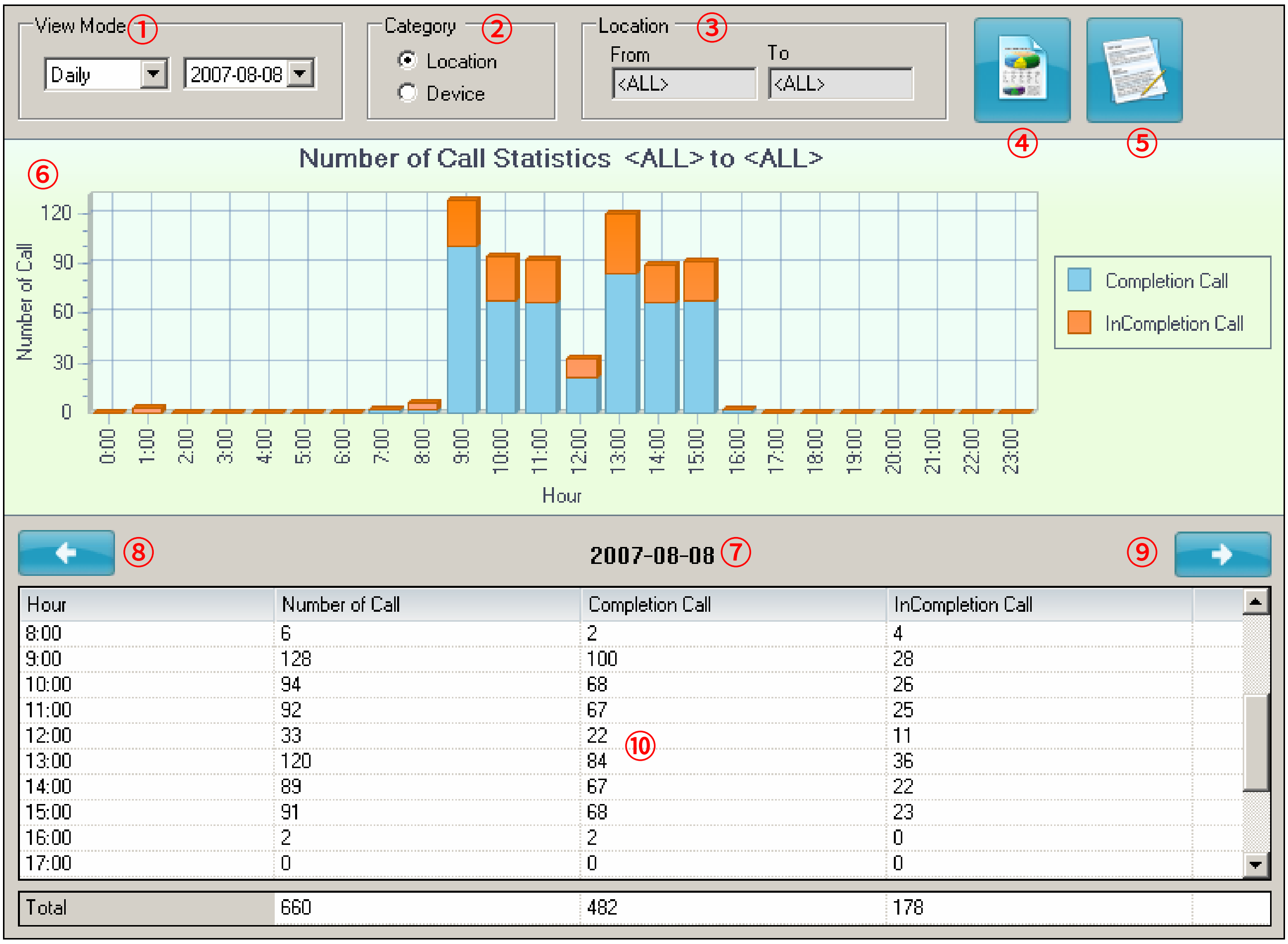
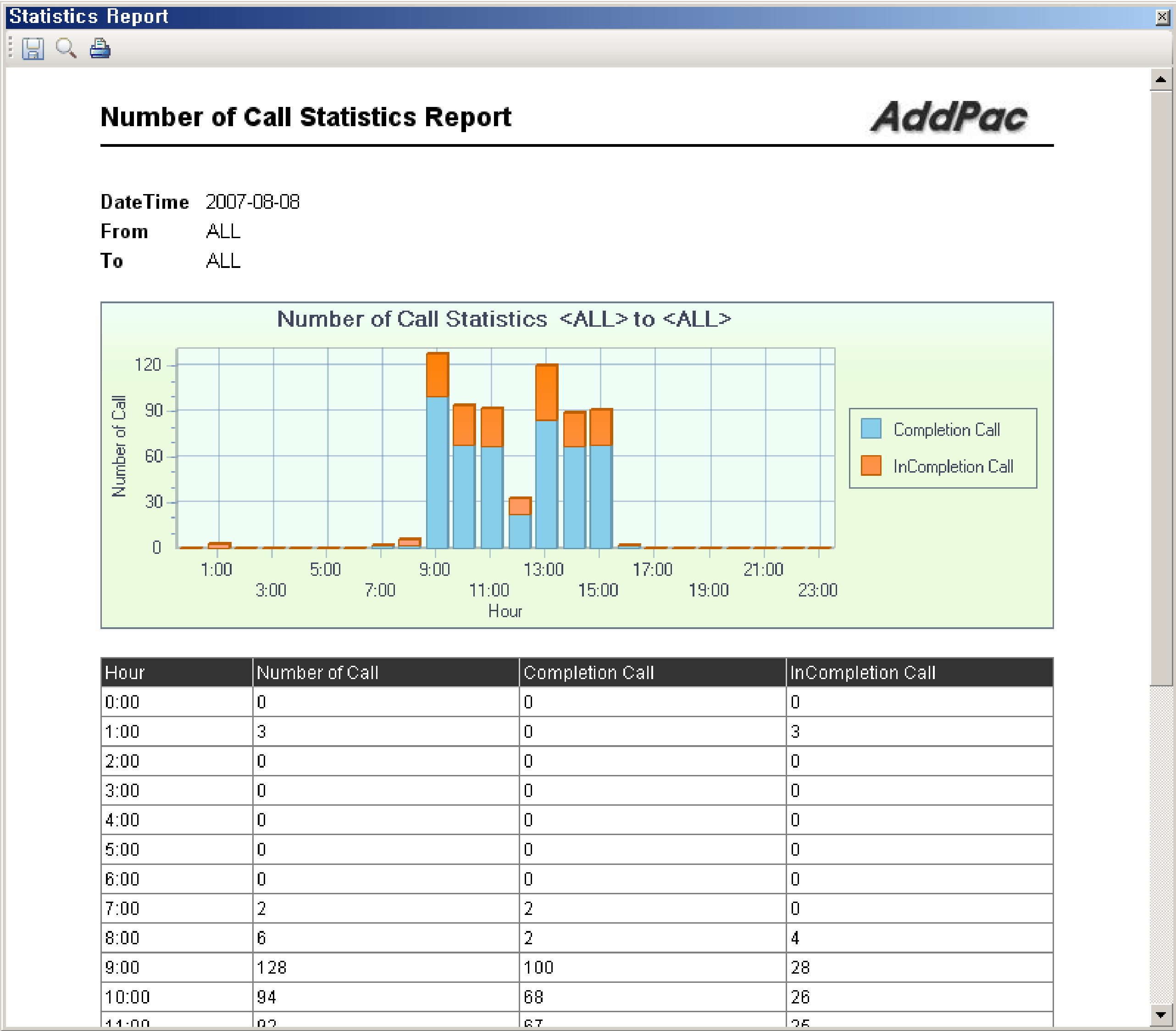


Figure 4-5 Statistics by Number of Calls

Table 4-2 Description of Statistics by Number of Call

No.	Description
1	Select the standards for the statistic data to be displayed: - Daily : one day basis (24 hours) - Monthly : one month basis (31 days) - hourly : the statistic data in a specified day
2	Select a search standard (Location, Device)
3	Select a search condition basing on the search standard (From, To)
4	Execute the statistic data basing on the search conditions which have been entered
5	Display the statistic data which has been searched on a report
6	Display the static data by graph
7	The norm for the statistic data
8	Search the previous day or month basing on the norm for output (daily, monthly)
9	Search the next day or month basing on the norm for output (daily, monthly)
10	Display static data on the table

The statistic data can be displayed in the following by clicking the report button:



Hour	Number of Call	Completion Call	InCompletion Call
0:00	0	0	0
1:00	3	0	3
2:00	0	0	0
3:00	0	0	0
4:00	0	0	0
5:00	0	0	0
6:00	0	0	0
7:00	2	2	0
8:00	6	2	4
9:00	128	100	28
10:00	94	68	26

Figure 4-6 Number of Calls Based Statistics Report

Incompletion Call Statistics

A statistic data for incompletion of the IP-PBX, in a specified time period, can be generated by search conditions and divided by date, month and provided in graph and table formats. The following figure shows the statistic screen:

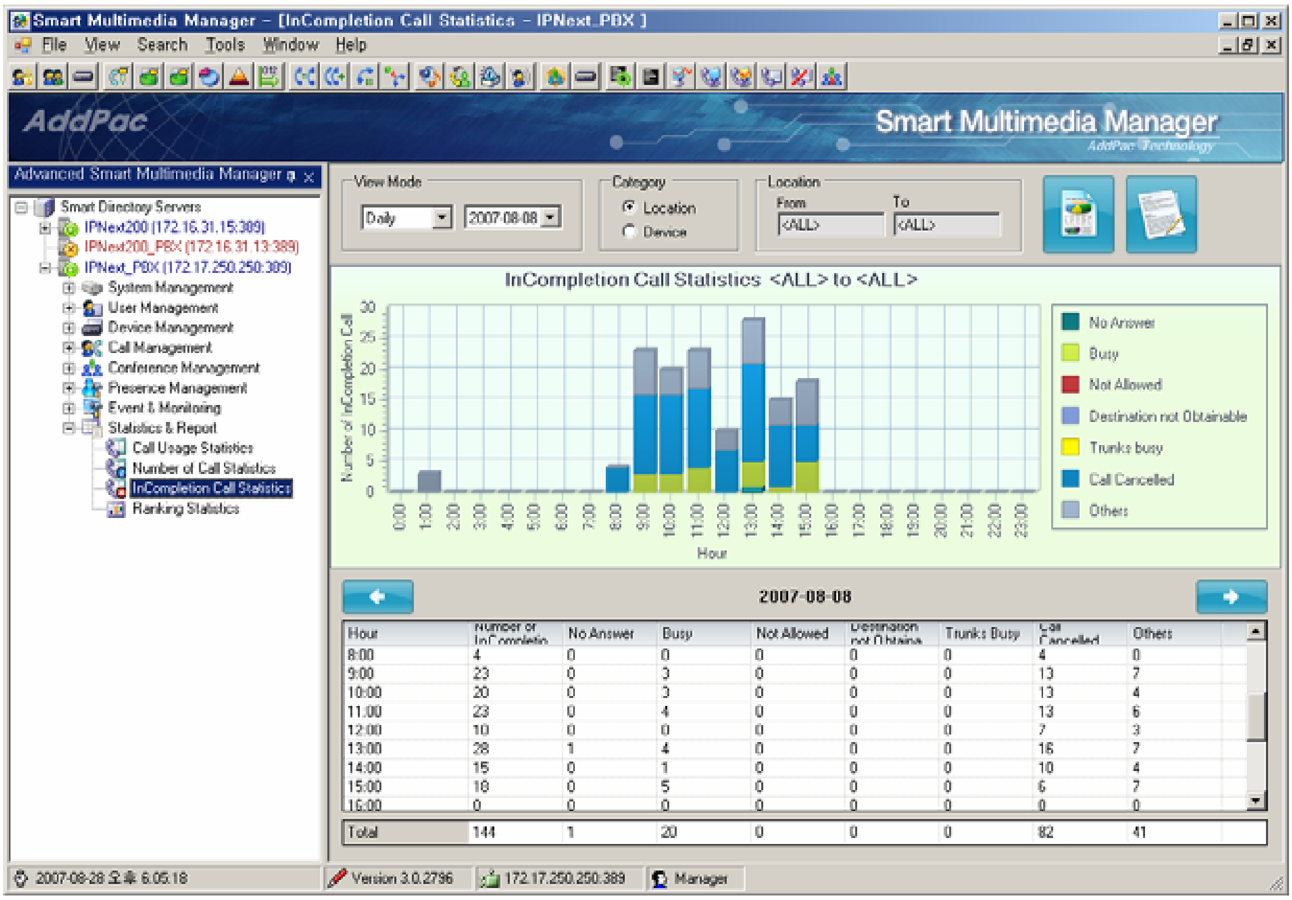


Figure 4-7 Incompletion Call Statistics

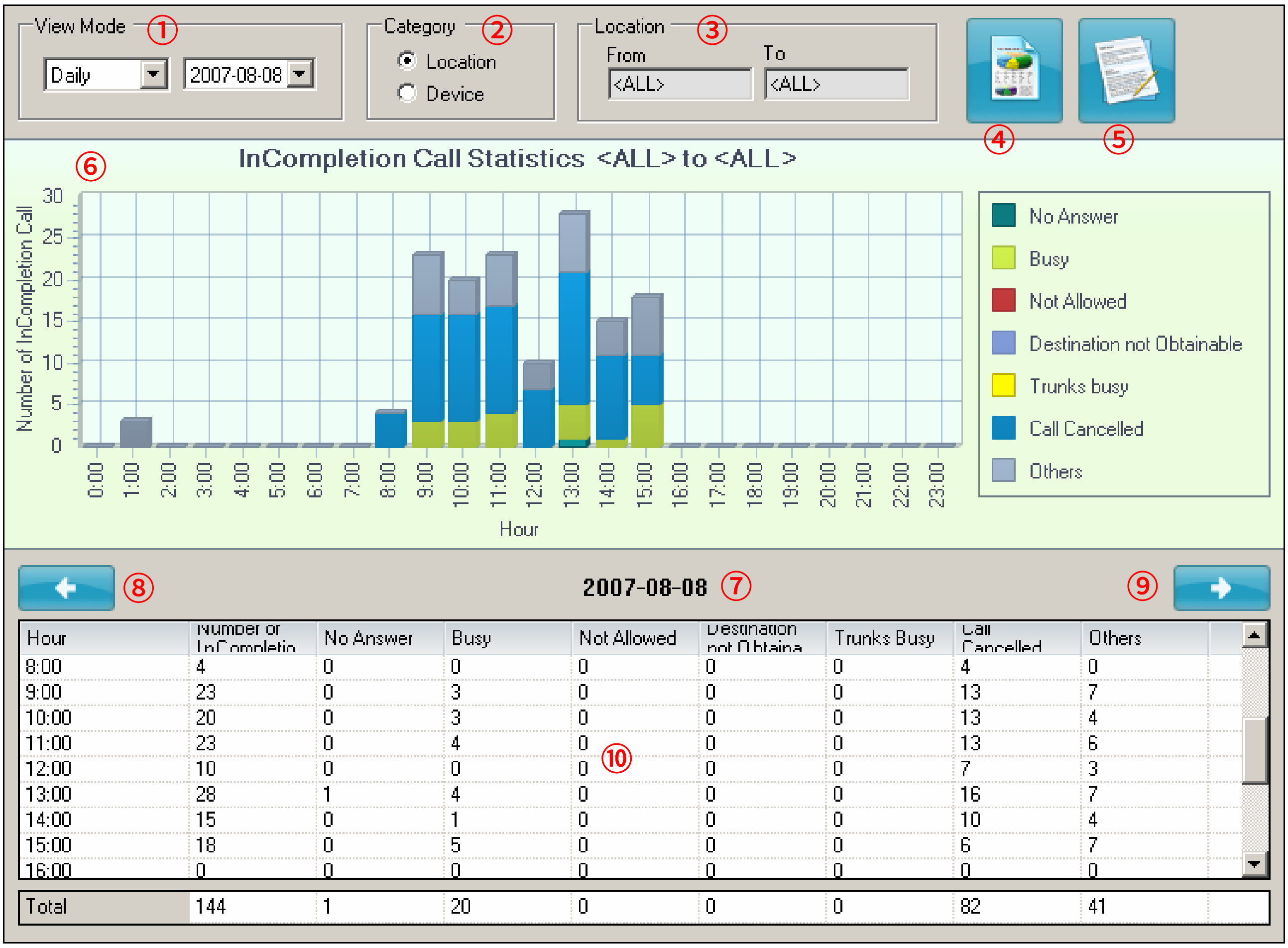


Figure 4-8 Incompletion Call Statistics Screen

Table 4-3 Description of Incompletion Call Statistics Screen

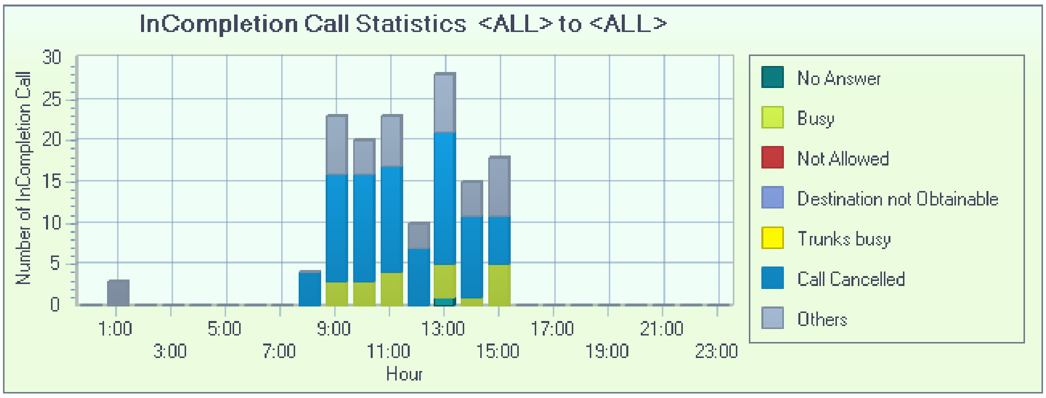
No.	Description
1	Select the standards for the statistic data to be displayed: - Daily : one day basis (24 hours) - Monthly : one month basis (31 days) - hourly : the statistic data in a specified day
2	Select a search standard (Location, Device)
3	Select a search condition basing on the search standard (From, To)
4	Execute the statistic data basing on the search conditions which have been entered
5	Display the statistic data which has been searched on a report
6	Display the static data by graph
7	The norm for the statistic data
8	Search the previous day or month basing on the norm for output (daily, monthly)
9	Search the next day or month basing on the norm for output (daily, monthly)
10	Display static data on the table

The statistic data can be displayed in the following by clicking the report button:

Statistics Report

InCompletion Call Statistics Report AddPac

DateTime 2007-08-08
From ALL
To ALL



Hour	Number of InCompletion Call	No Answer	Busy	Not Allowed	Destination not Obtainable	Trunks Busy	Call Cancelled	Others
0:00	0	0	0	0	0	0	0	0
1:00	3	0	0	0	0	0	0	3
2:00	0	0	0	0	0	0	0	0
3:00	0	0	0	0	0	0	0	0
4:00	0	0	0	0	0	0	0	0
5:00	0	0	0	0	0	0	0	0
6:00	0	0	0	0	0	0	0	0
7:00	0	0	0	0	0	0	0	0
8:00	4	0	0	0	0	0	4	0

Figure 4-9 InCompletion Call Statistics Report Screen

Ranking Statistics

A statistic data for number of call ranking of the IP-PBX, in a specified time period, can be generated by search conditions and divided by date, month and provided in graph and table formats. The following figure shows the statistic screen:

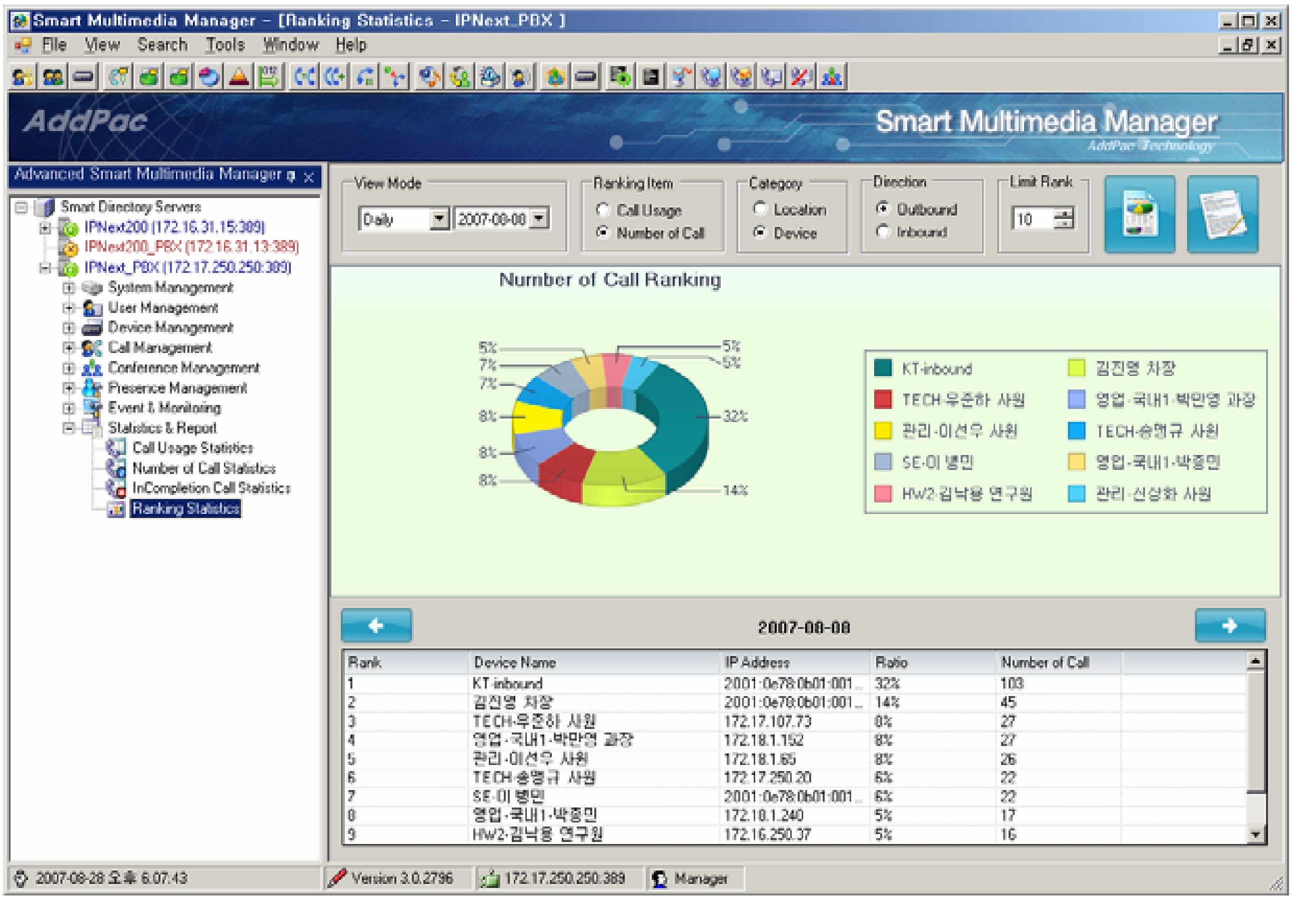


Figure 4-10 Ranking Statistics

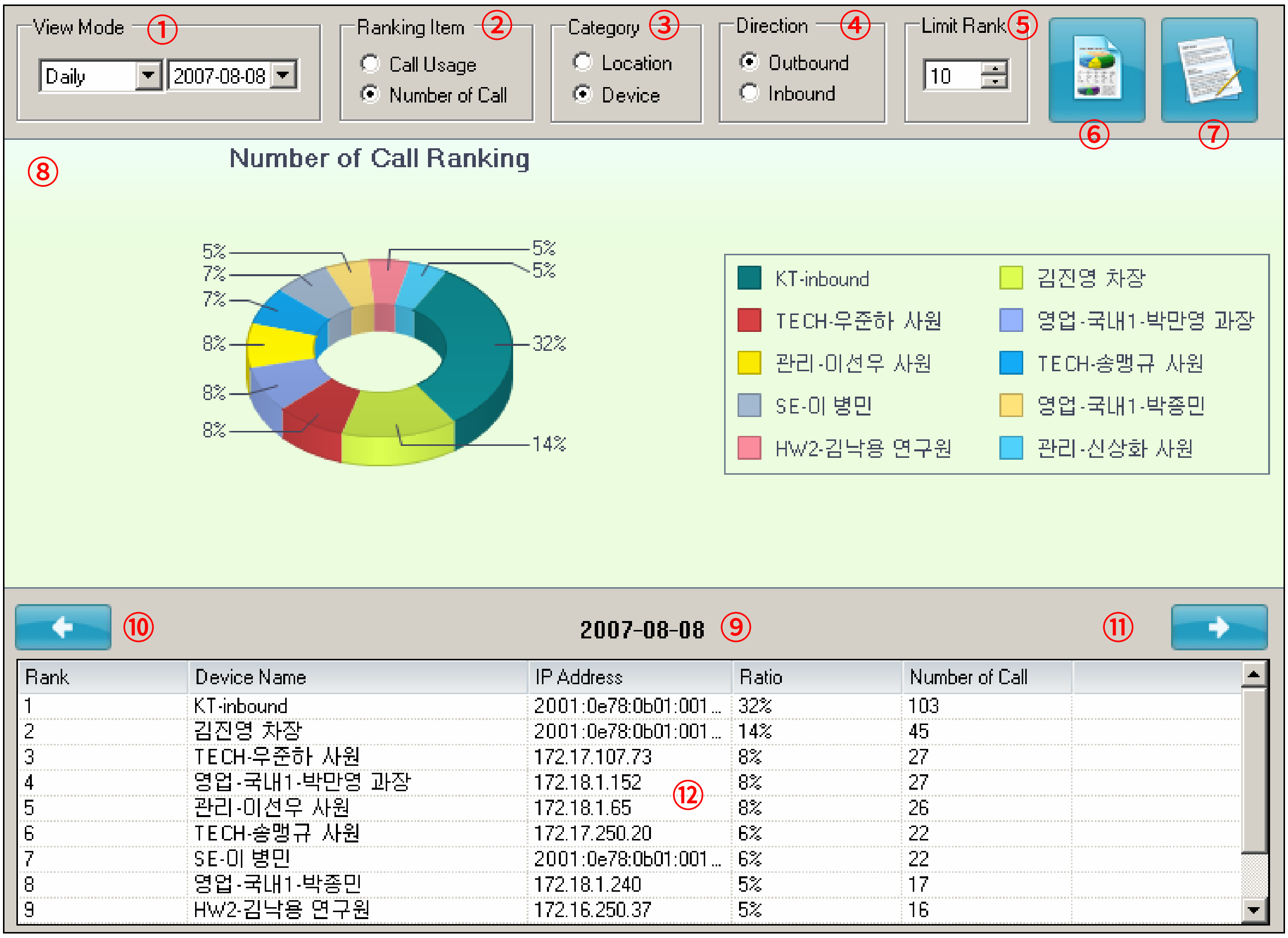


Figure 4-11 Ranking Statistics Screen

Table 4-4 Description of Ranking Statistics Screen

No.	Description
1	Select the standards for the statistic data to be displayed: - Daily : one day basis (24 hours) - Monthly : one month basis (31 days) - hourly : the statistic data in a specified day
2	Select a search standard (Location, Device)
3	Select a search condition basing on the search standard (From, To)
4	Execute the statistic data basing on the search conditions which have been entered
5	Display the statistic data which has been searched on a report
6	Display the static data by graph
7	The norm for the statistic data
8	Search the previous day or month basing on the norm for output (daily, monthly)
9	Search the next day or month basing on the norm for output (daily, monthly)
10	Display static data on the table

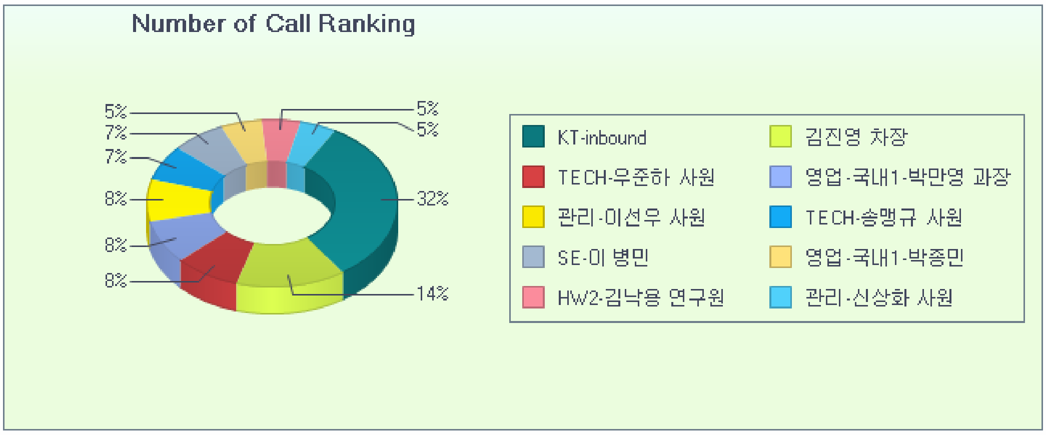
The statistic data can be displayed in the following by clicking the report button:

Statistics Report

Ranking Statistics Report

AddPac

DateTime	2007-08-08
Ranking Item	Number of Call
Direction	Outbound



Rank	Device Name	Host	Ratio	Call Usage (sec)
1	KT-inbound	2001:0e78:0b01:0017:0250:0000:0000:0252	32%	103
2	김진영 차장	2001:0e78:0b01:0017:0250:0000:0000:0010	14%	45
3	TECH-우준하 사원	172.17.107.73	8%	27
4	영업-국내1-박만영 과장	172.18.1.152	8%	27
5	관리-이선우 사원	172.18.1.65	8%	26
6	TECH-송맹규 사원	172.17.250.20	6%	22
7	SE-이 병민	2001:0e78:0b01:0017:0101:0000:0000:0240	6%	22
8	영업-국내1-박종민	172.18.1.240	5%	17
9	HW2-김낙용 연구원	172.16.250.37	5%	16
10	관리-신상화 사원	172.18.1.105	4%	15

Print Time : 2007-08-28 오후 6:08:55

Figure 4-12 Ranking Statistics Report Screen

Chapter 5. SMM Data Management

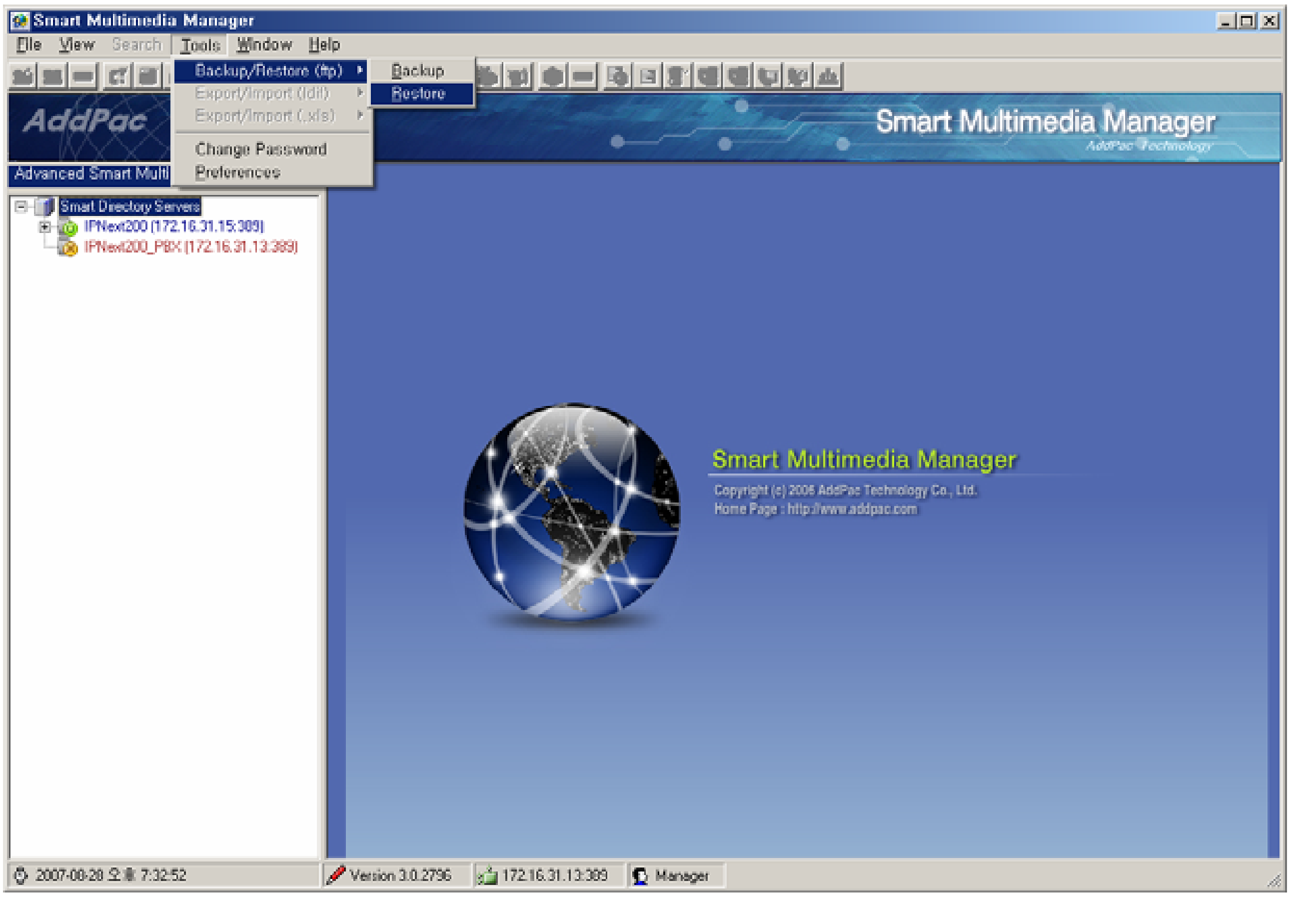
SMM provides initialization, backup and restoration of the basic data and database information for IPNext PBX. This chapter describes how to initialize IPNext PBX, backup and restore the data in operation.

Data Initialization & Recovery

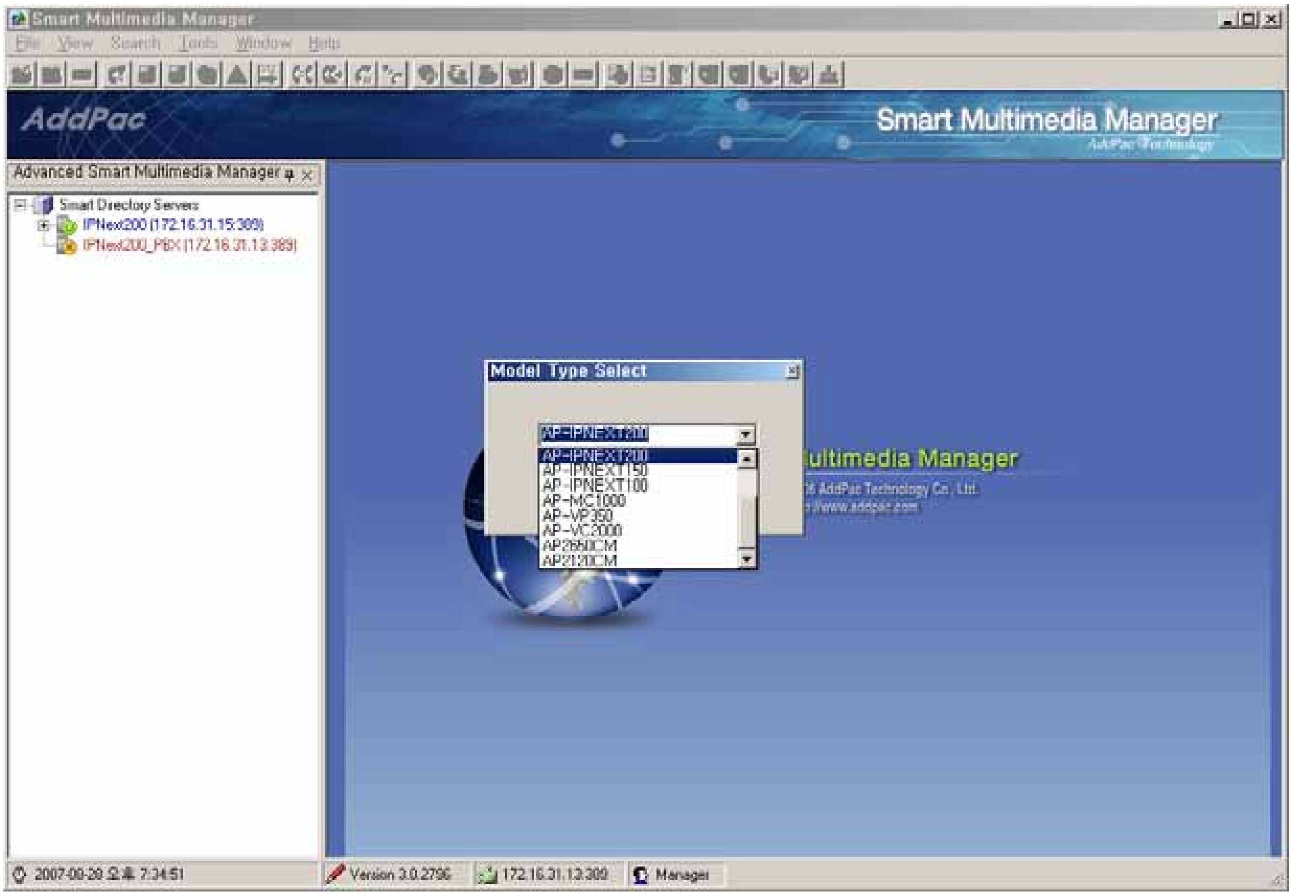
Data (ment files of Call Manager, data base schema and initial file) needs to be initialized or recovered when HDD or Flash Memory is damaged or lost by an unpredictable error while IPNext PBX is running.
The followings describe a process to initialize or recover the data:

* This process is used for exceptional cases that Smart Directory Server is damaged or needs to be upgraded with a new version. The process initializes all the data, so a special attention is required.

1) Choose 'Tools > Backup/Restore (ftp) > Restore' from the main menu.

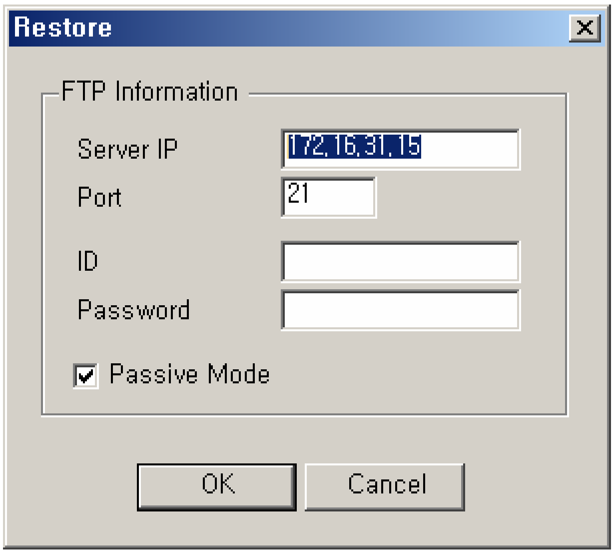


2) Select a model of IPNext PBX to be restored, then click ‘OK’ button.



* Select Directory Server on the left. You may skip this step if you already know the model of IPNext PBX

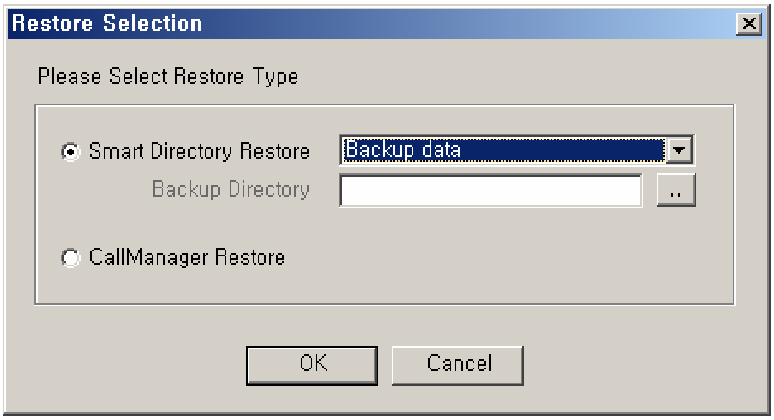
3) Enter FTP information of the IPNext PBX to be initialized.

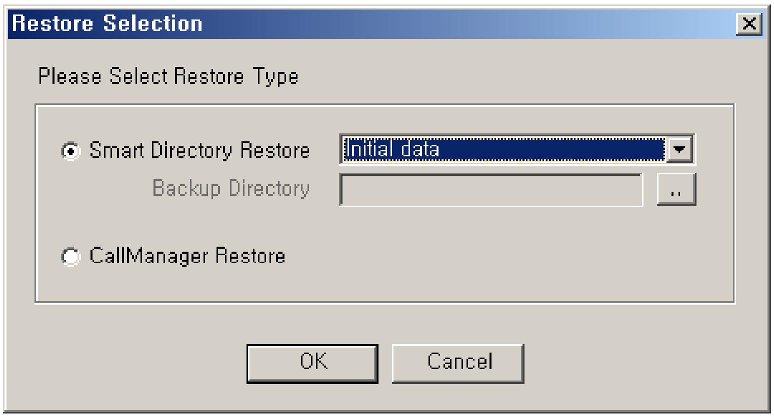


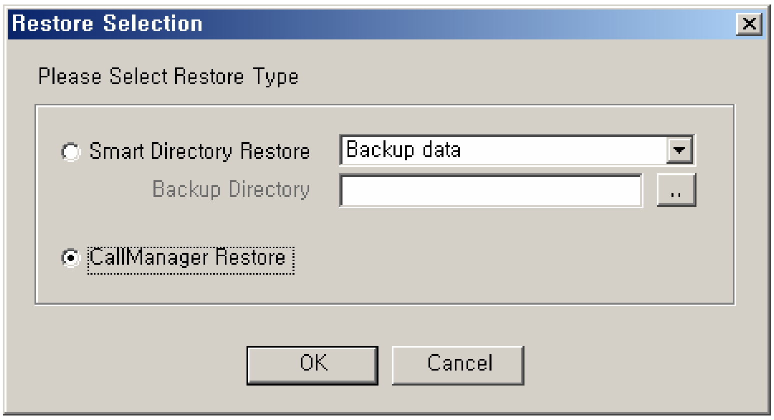
4) Select a data recovery option

* The following table describe the data recovery options

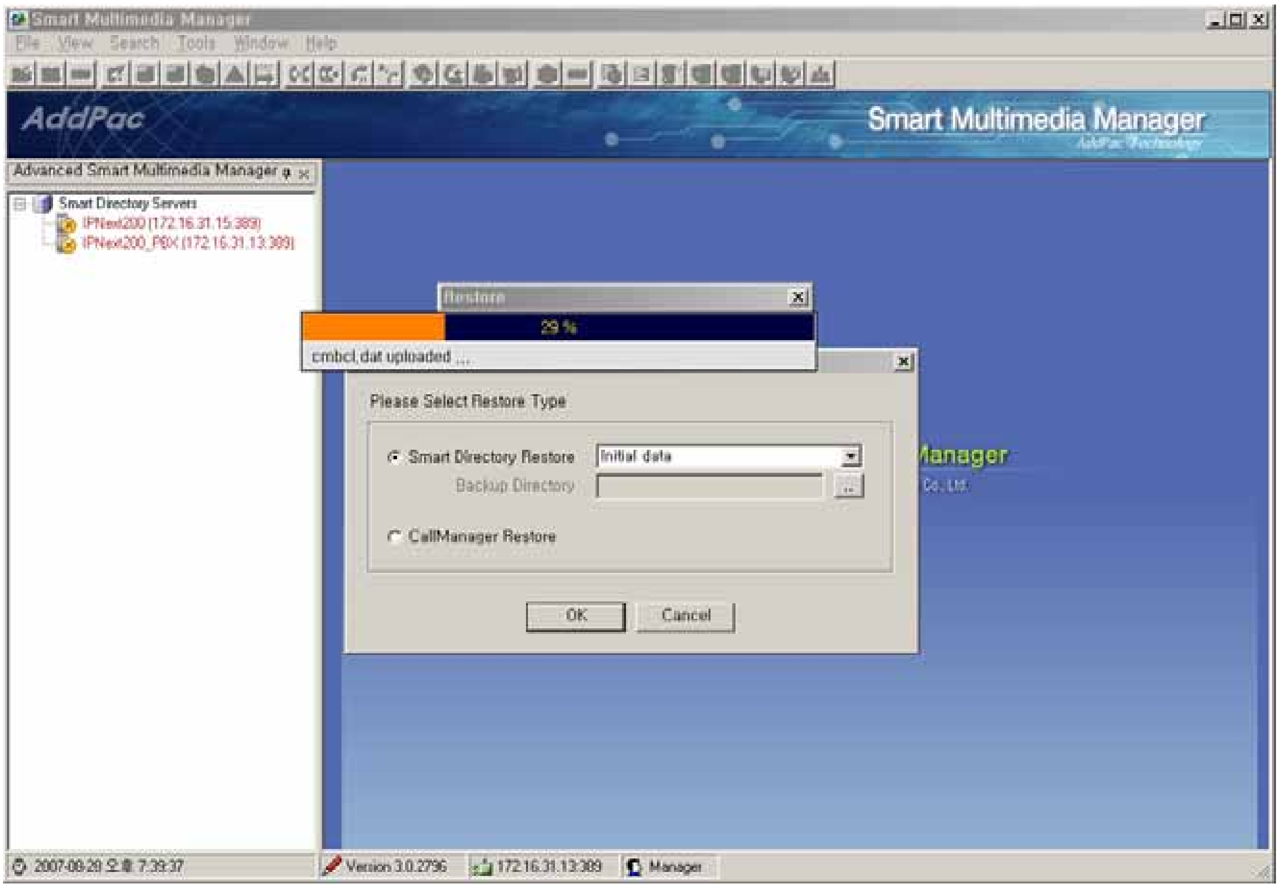
Option	Sub-Option	Description
Smart Directory Restore	Backup data	Backup the file related to the database (ldap) of IPNext PBX. (Specify a directory path for backup)
	Initial data	Initialize the basic file and schema related the database if IPNext PBX
Call Manager Restore		Restore the data related to the database, basic and ment files of IPNext PBX Restore the data (restores to the factory default mode)



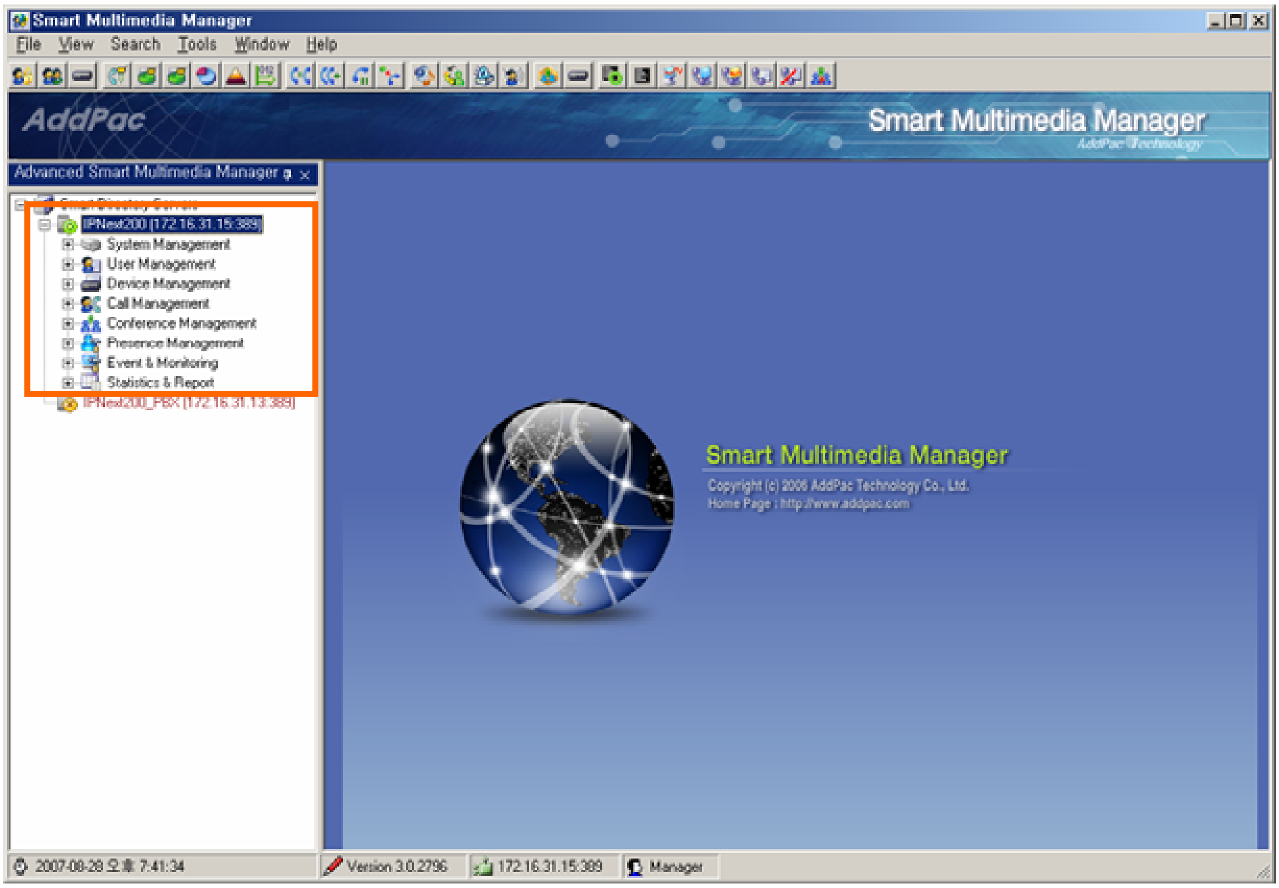




5) Process initialization of the basic data or database of IPNext PBX



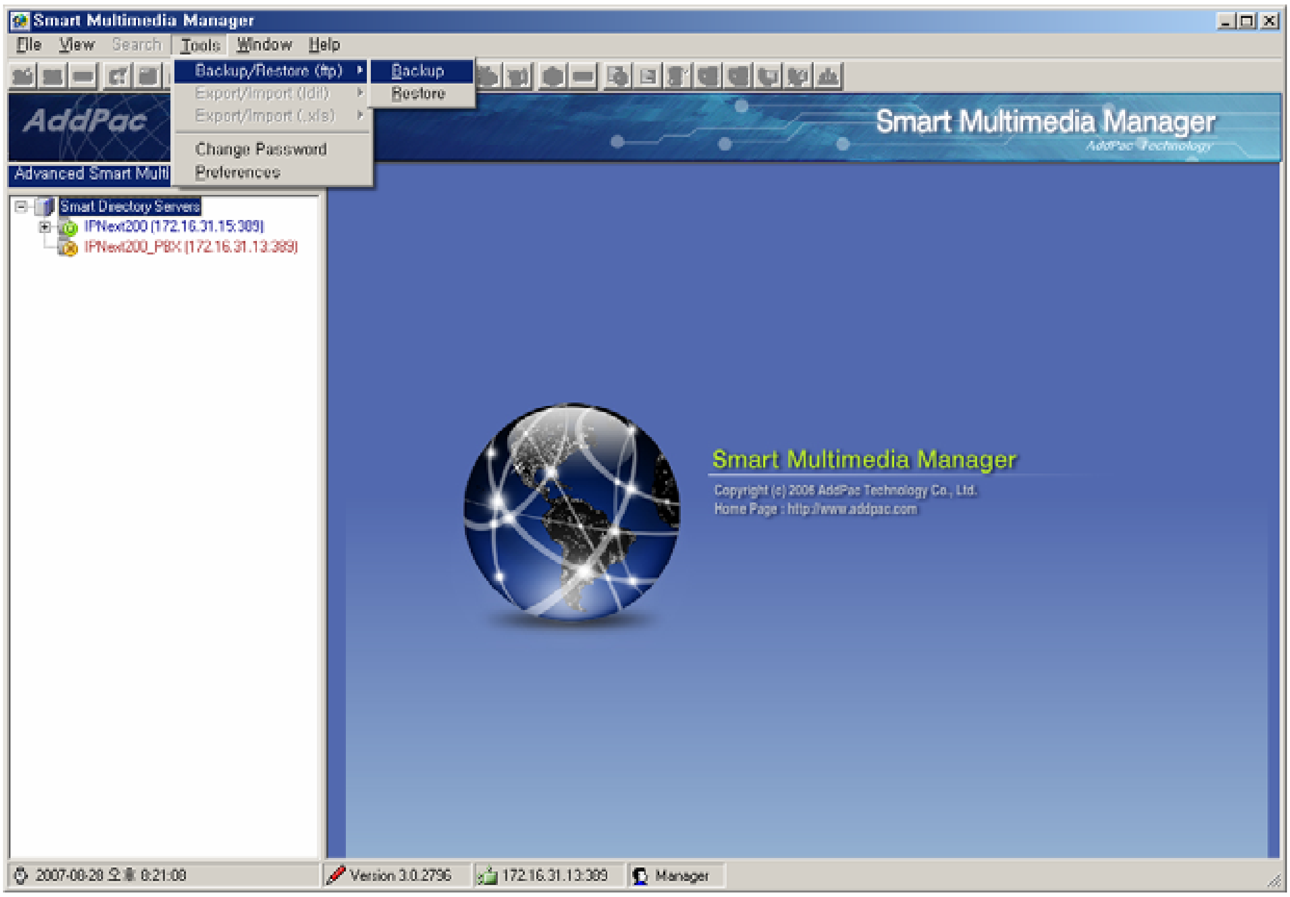
6) When the process is completed, IPNext PBX reboots automatically. After the reboot is completed and IPNext PBX is registered to SMM, Directory Server is access as to follow:



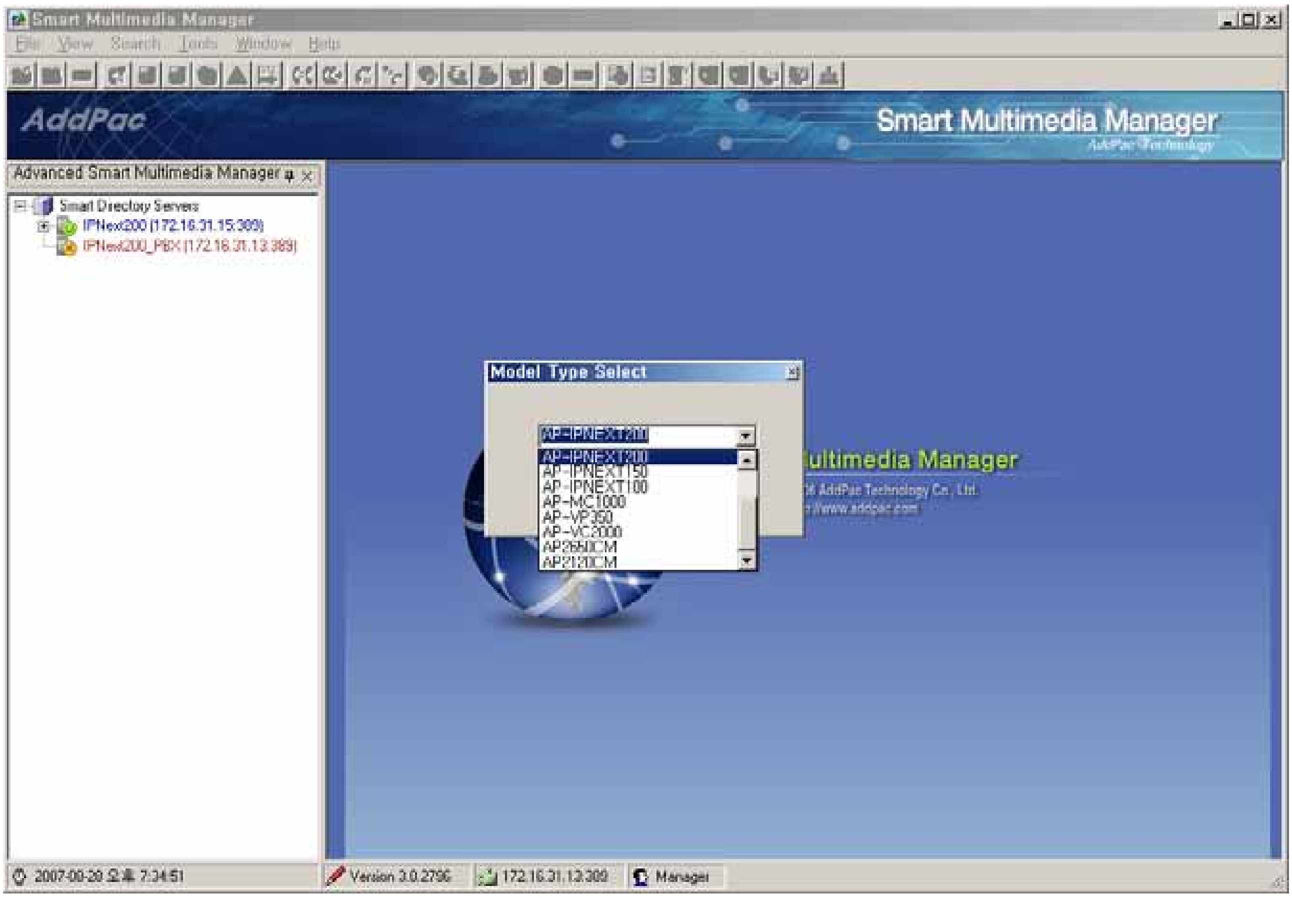
Data Backup

The data (ldap data file)needs to be backed up in case of an unexpected damage and loss of the data by an error.

1) Select 'Tools > Backup/Restore (ftp) > Backup' of the main menu

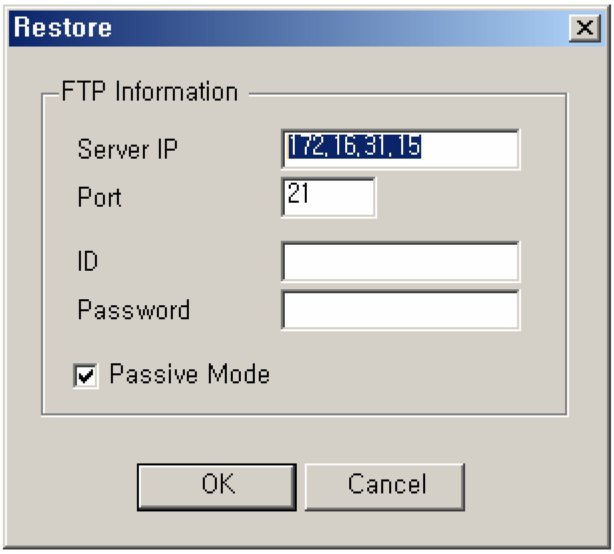


2) After selecting IPNext PBX model, then click OK button.

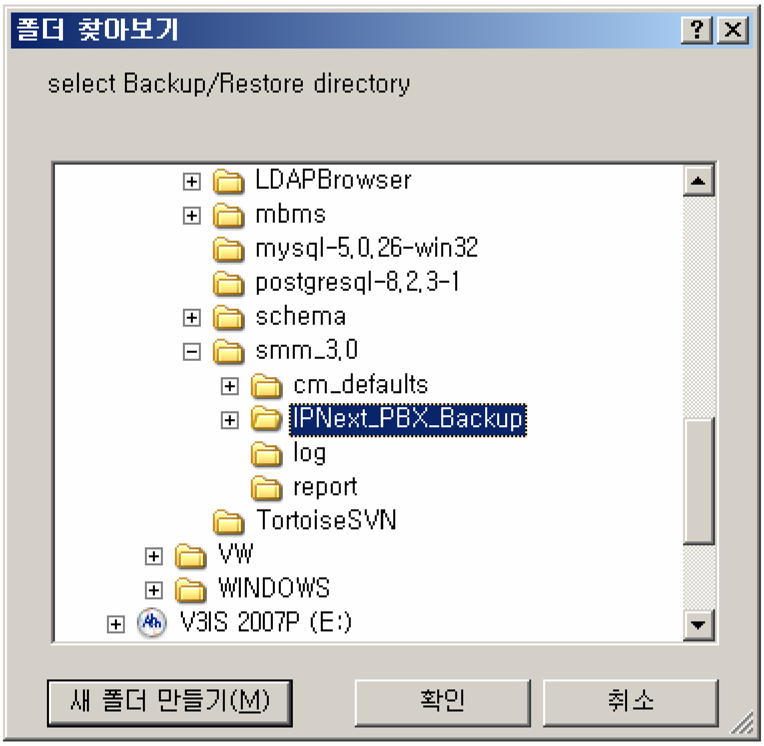


* After selecting Directory Server, you can skip the next step if you know the IPNext OBX model.

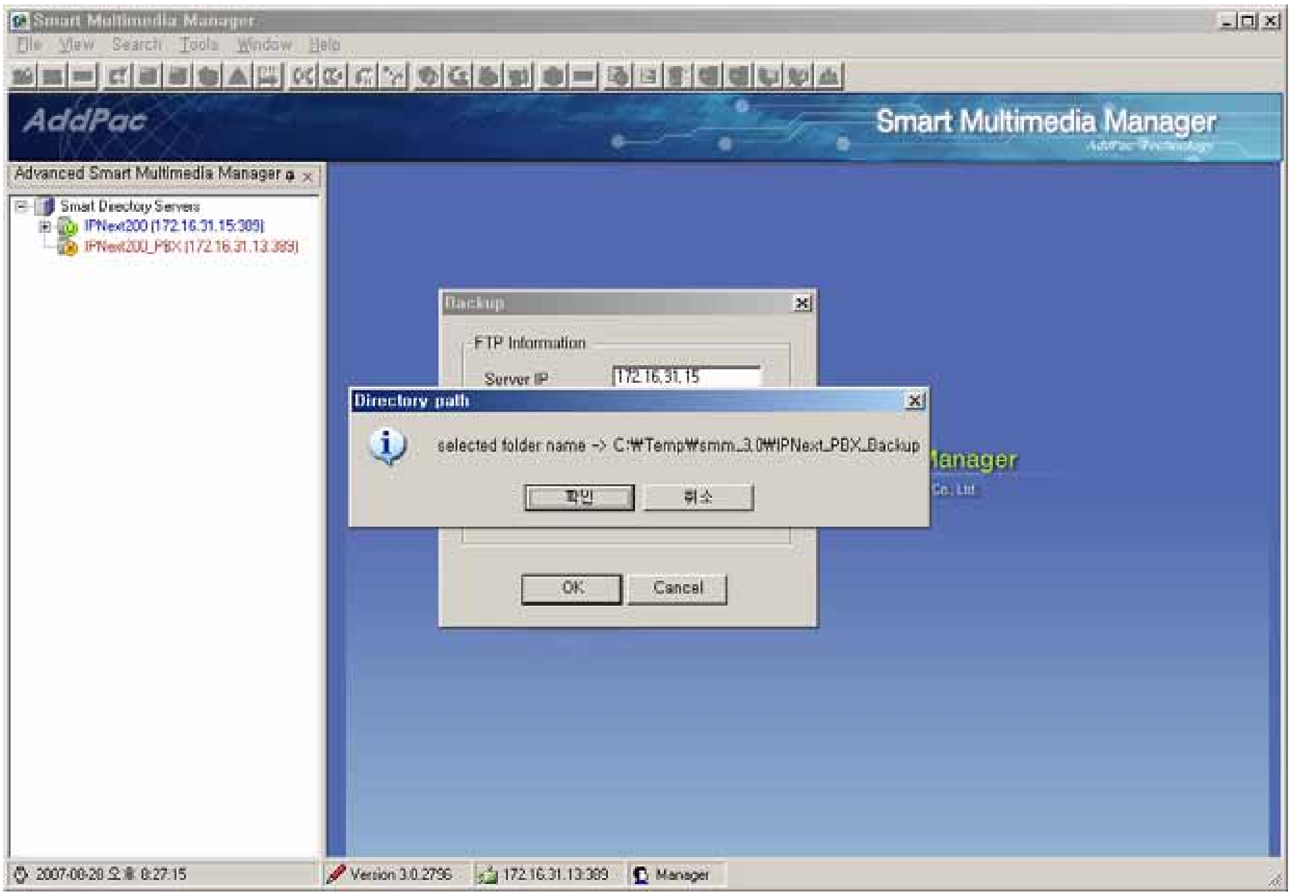
3) Enter FTP access information of IPNext PBX for backup



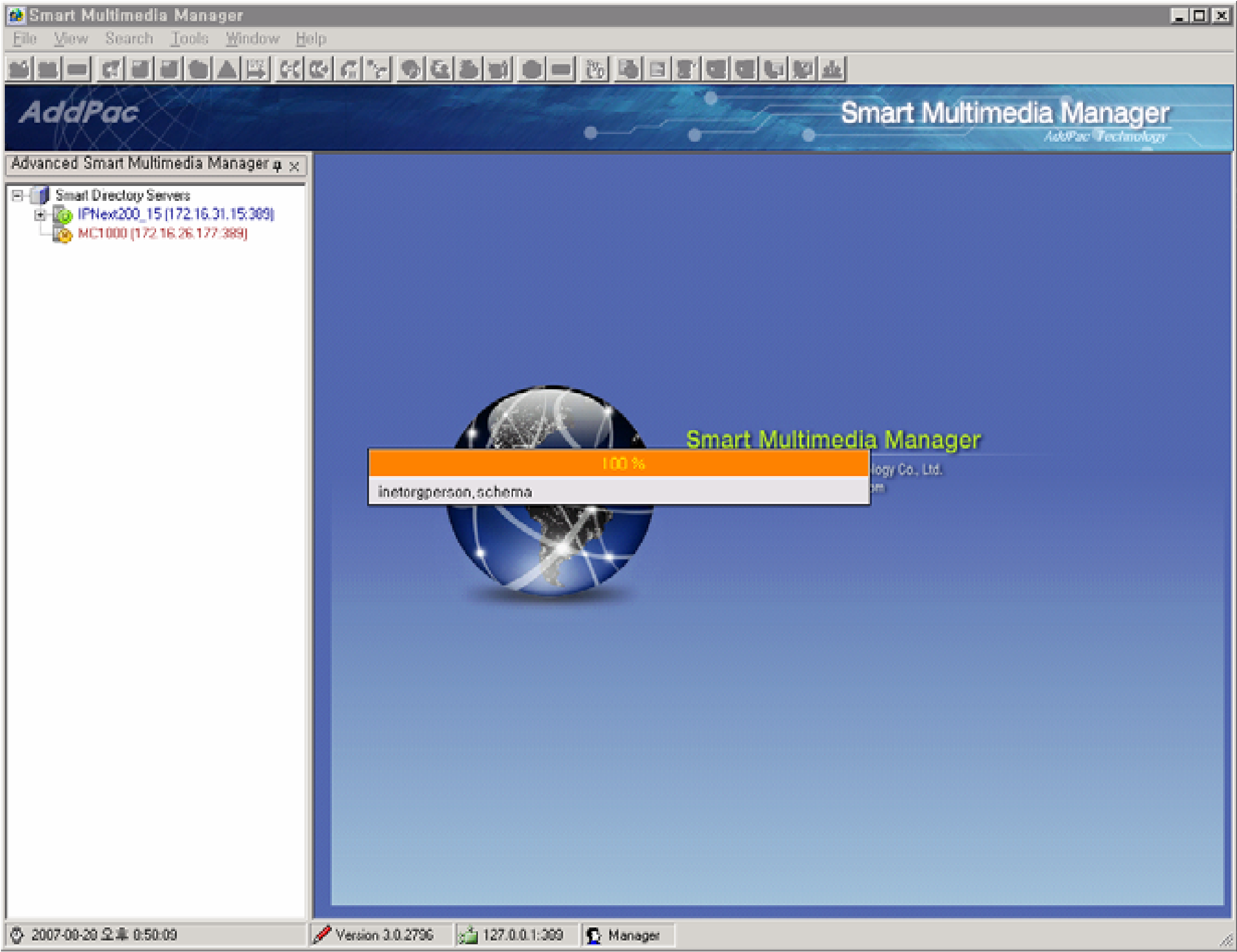
4) Specify directory file path for backup



5) Verify the selected path



6) Process backup for the database information in IPNext PBX.



LDIF Export/Import

LDIF(LDAP Data Interchange Format) is a file format to save the data. In SMM, the data can be saved and recovered as LDIF.

This function is located in the main menu of Menu Tools > Backup/Restore(ldif)

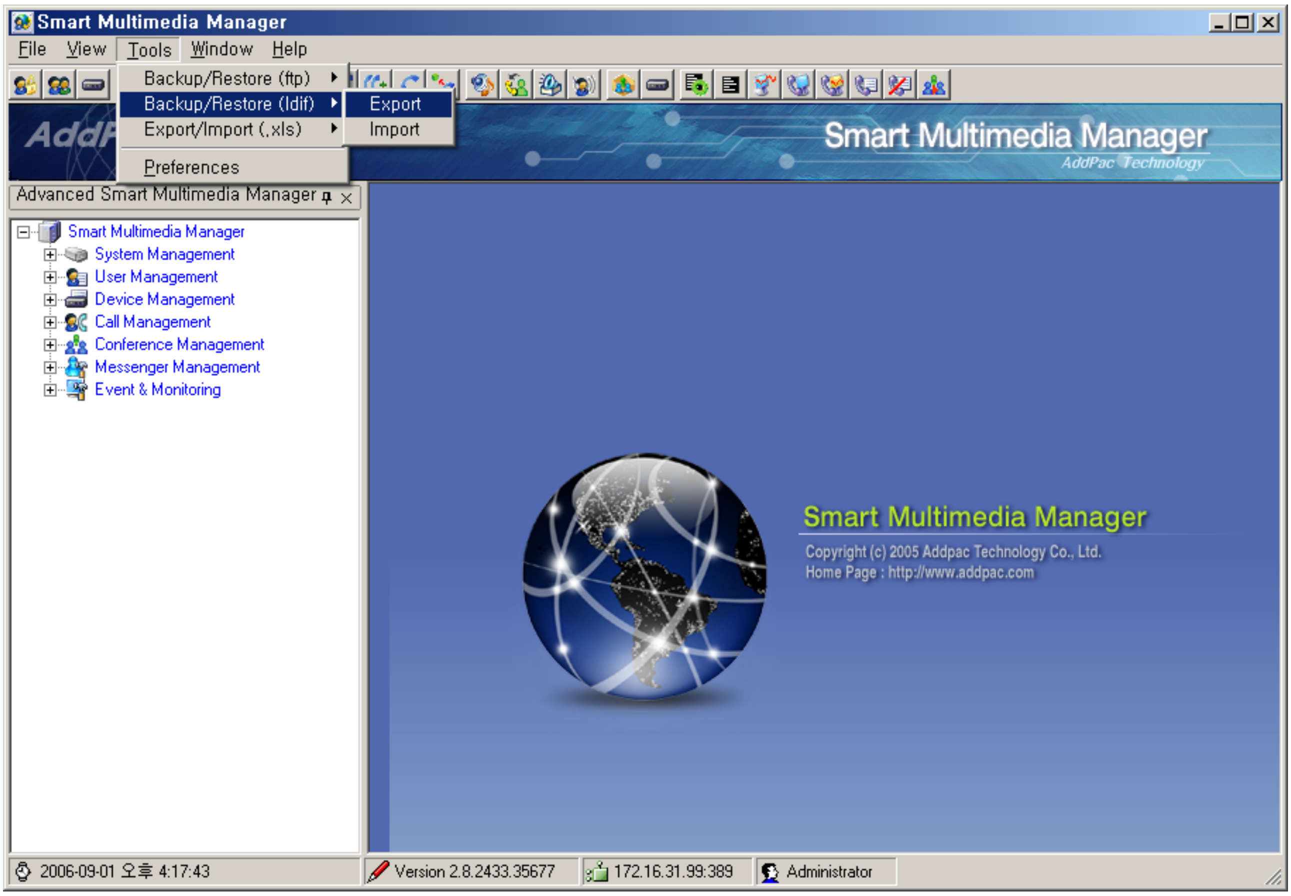


Figure 5-1 LDIF Export / Import Menu

Coose Tools > Backup/Restore(ldif) > Export then enter a file name and location to be exported as the process of LDIF Export is shown in the following figure:

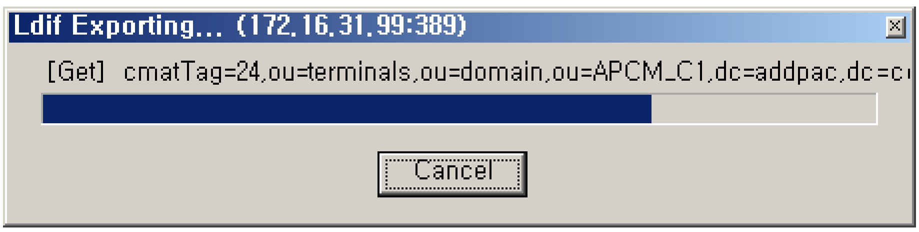


Figure 5-2 LDIF Export Process Screen

Select and perform Tools > backup/Restore(ldif) > Import to use LDIF file to recover the data.

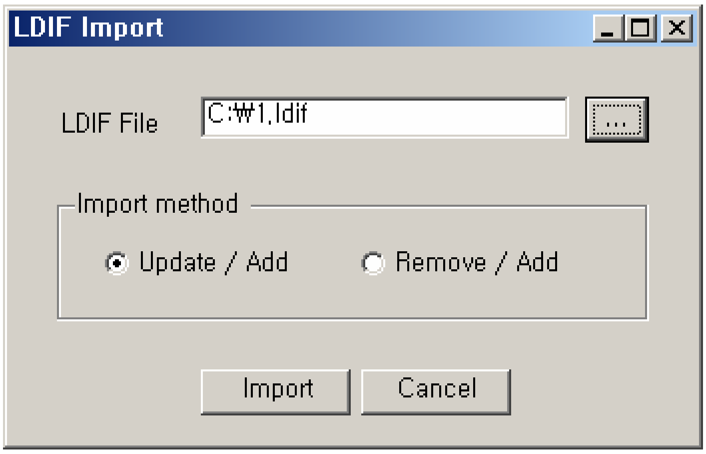


Figure 5-3 LDIF Import Process Screen 1

When to import, some precautions should be taken for the following methods:

- Updat/Add: maintain the existing information and update the parts that have been changed
- Remove/Add: delete all the existing data and register LDIF file

Figure 5-4 LDIF Import Process Screen 2

When the process of import is completed, the changes of data need to be implemented normally by rebooting the IPNext PBX.

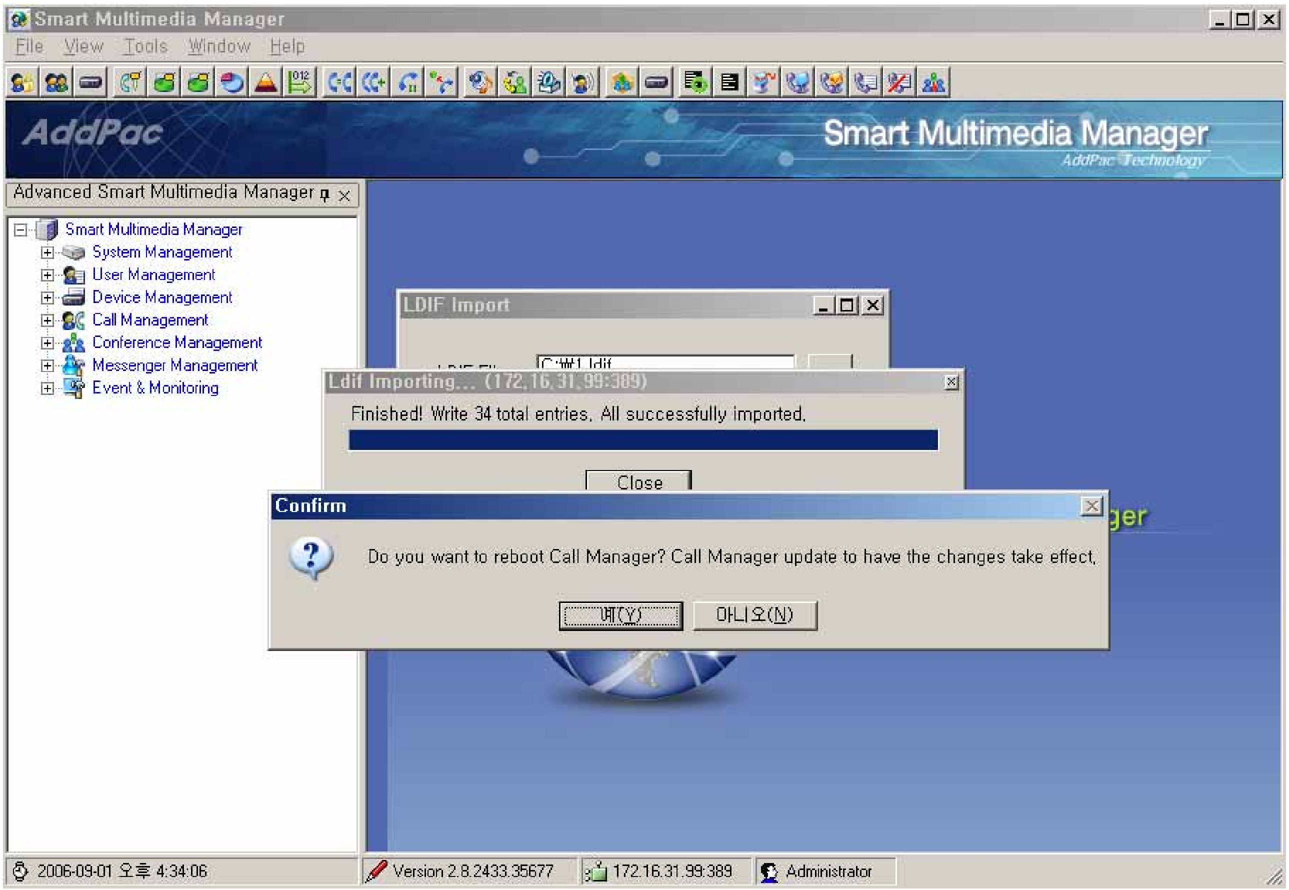


Figure 5-5 LDIF Import Completion Screen

Excel Export/Import Function

SMM provides the data management functions by preparing, registering and extracting data in Excel file format which is user-friendly. It also provides a function to upgrade the data of IPNext PBX in operation. The registered data in the server can be exported for a new schema upgrade, then it can be imported for restoring the server with upgrade of the data in the existing operation. The data is to be saved in each excel sheet, the user can add, edit and delete easily, because the excel format is user-friendly.

Export

The data in operation of the IPNext PBX generates the spreadsheet corresponding to each menu with the user friendly excel format, the work of adding, editing and deleting can be done. This data can be imported for upgrading the data in operation. After accessing to SMM, click Tool > Export/Import(.xls) > Export Menu to carry out this function. The process of Excel Import is shown in the figure below:

1) After accessing SMM, click Tool > Export/Import(.xls) > Export on the menu.

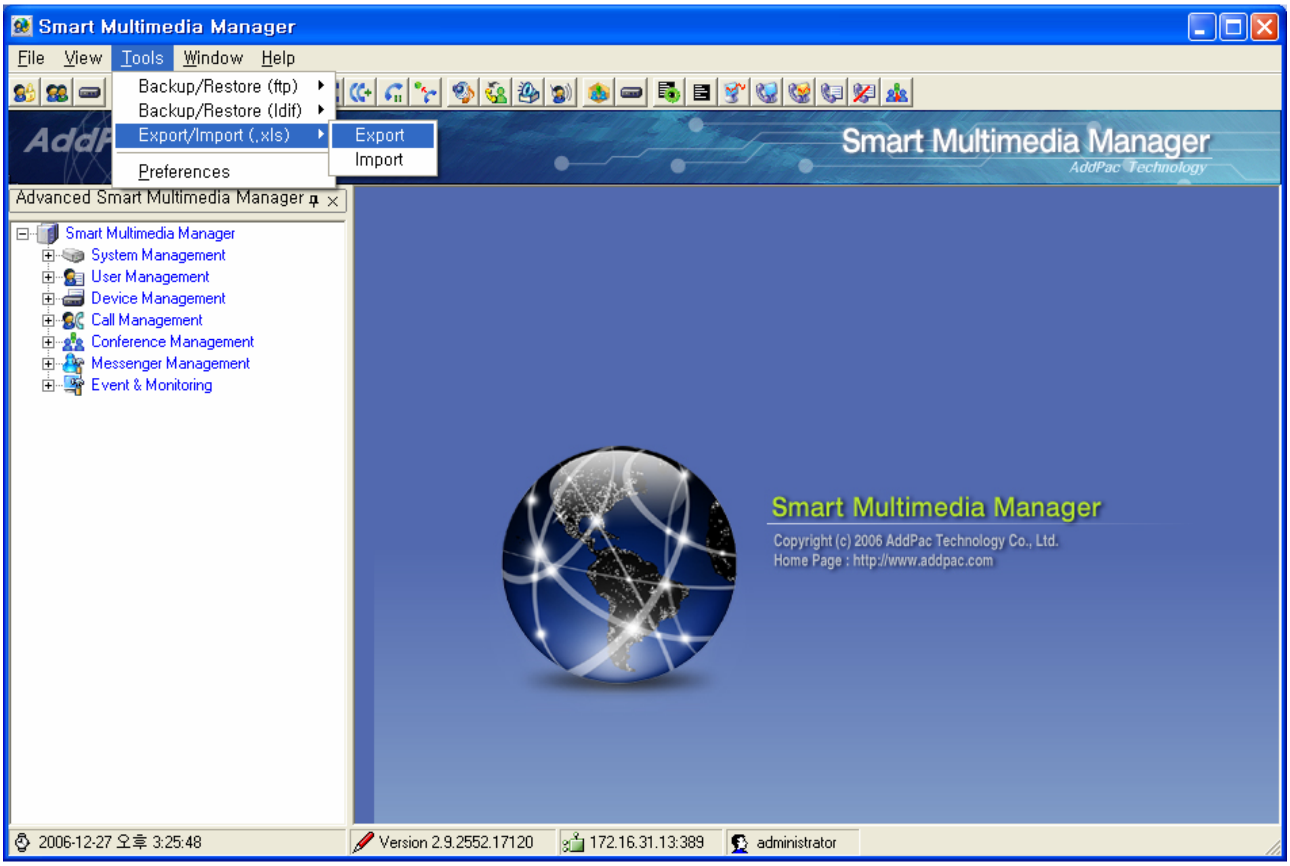


Figure 5-6 Excel Export Menu

2) Set the path to save the Excel file, create a file name.

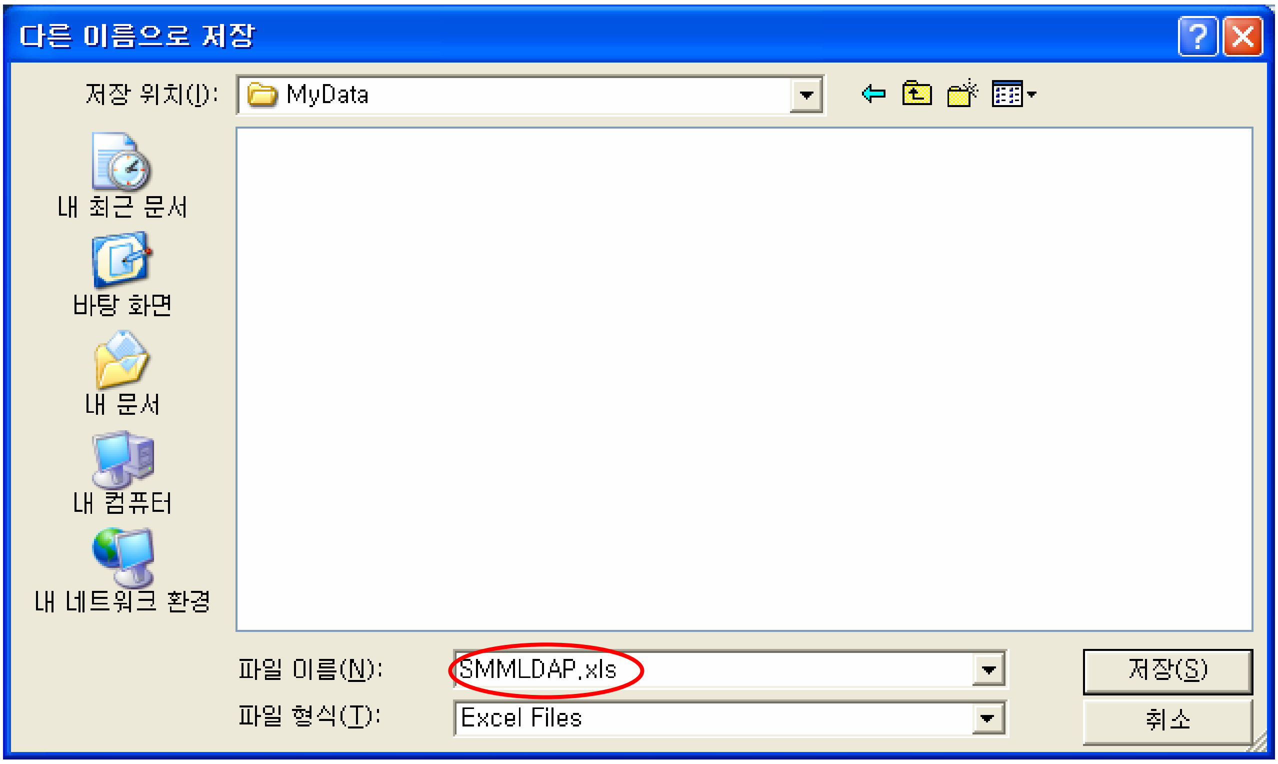


Figure 5-7 Screen for Saving Excel Export file

3) The status of export process is indicated by each category, so the present progress can be checked.

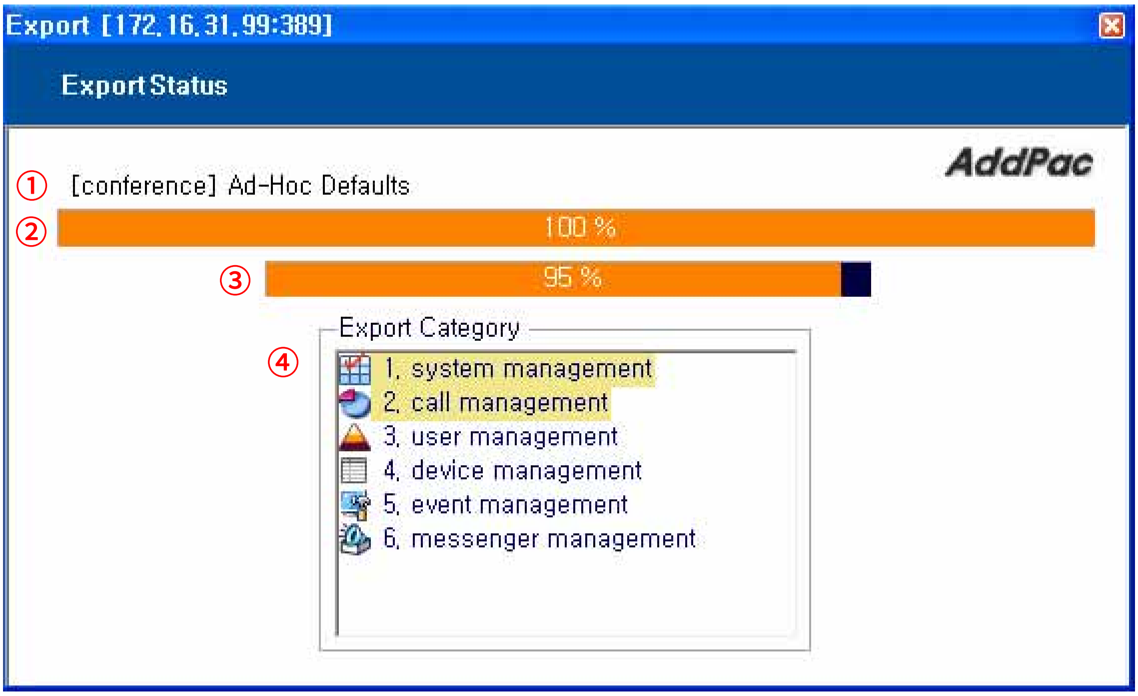


Figure 5-8 Excel Export Process Screen

Table 5-1 Description of the Procedures for Exporting an Excel File

Field	Description
1	A detailed information of the data in progress
2	Indicating a degree of data in progress basing on a single spread sheet.
3	Indicating a degree of sheet in progress basing on a single category
4	Indicating a category in progress.

4) When the process of export is finished normally, a message of ‘‘Finished, All Successfully Exported’ is displayed

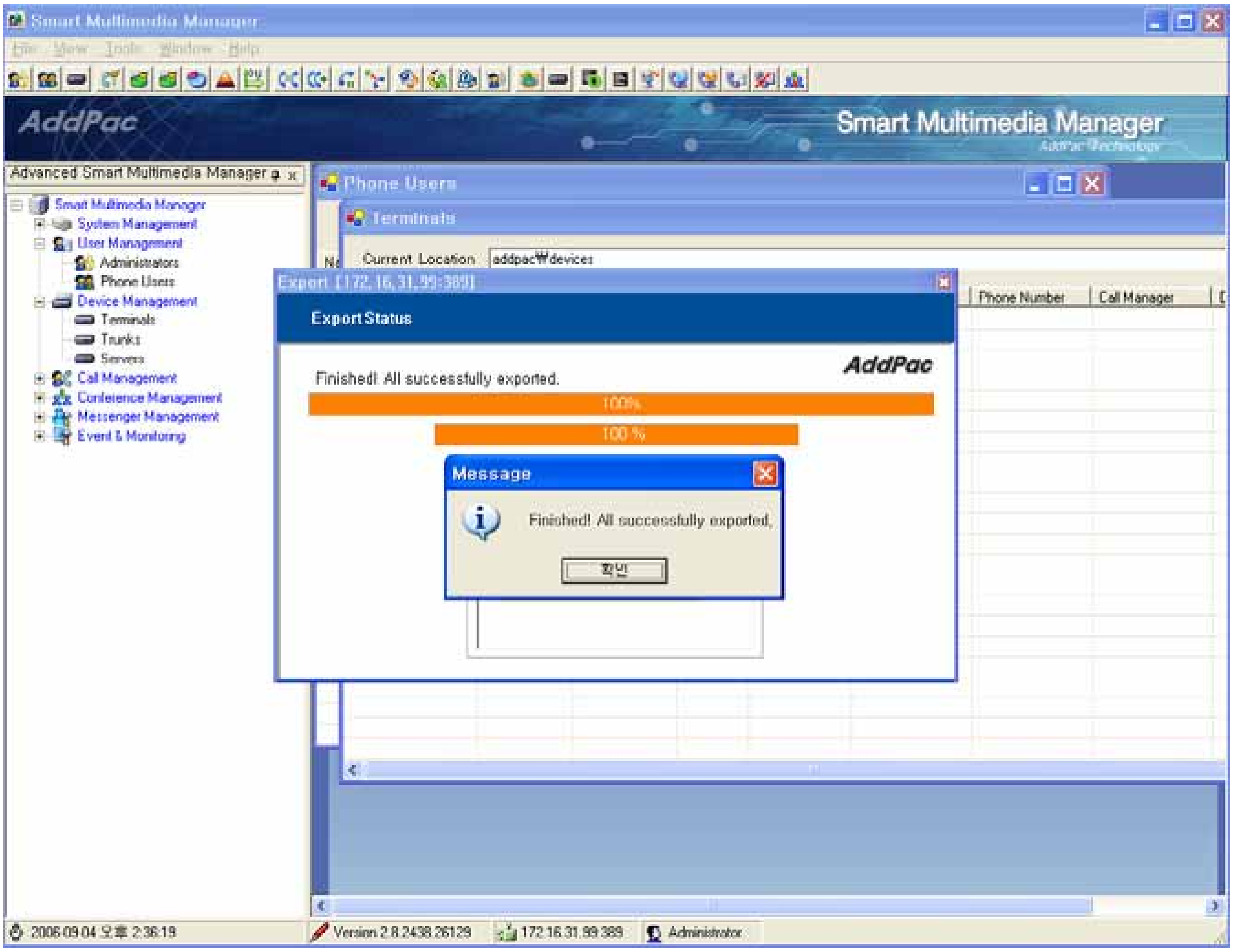


Figure 5-9 Excel Export Completed Screen

5) When ‘OK’ button is clicked, the process of export is ended. The saved excel file can be opened to check the completed data.

Import

The data that has been exported from SMM can be restored in the server with the current version or upgraded version. Also the data can be imported in addition to the existing one in operation.
When the data is added, an excel file is created and the corresponding excel file is to be imported from 'Import' in the menu.
When all the data needs to be restored, the IPNext PBX needs to be initialized first prior to importing the excel file. After initialization is completed, the job of importing the excel file is to be proceeded.
The following figure shows the process of importing the excel file after initialization.

1) Click Tool > Export/Import(.xls) > Import Menu after accessing to SMM.

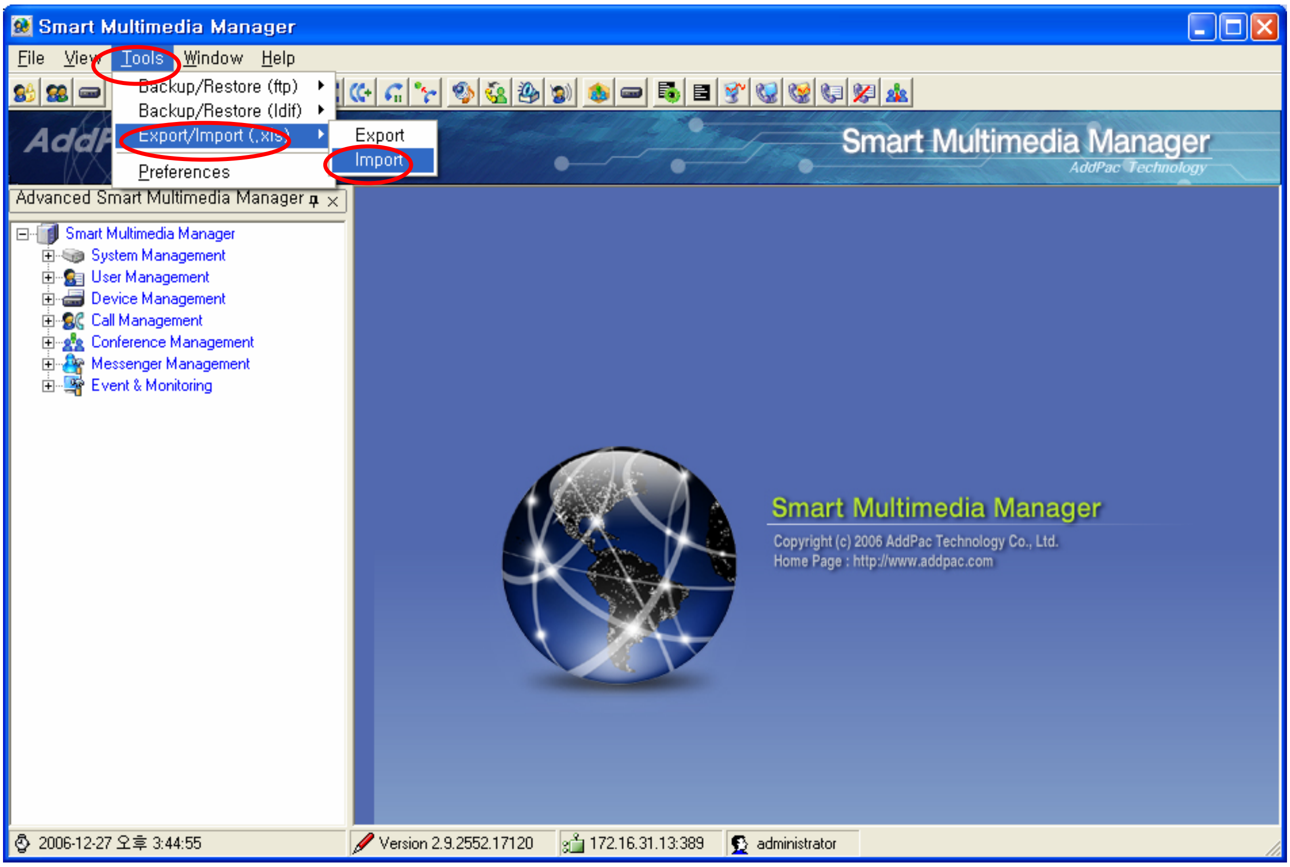


Figure 5-10 Excel Import Menu

2) Set a new path to import the excel file. Then select the file.

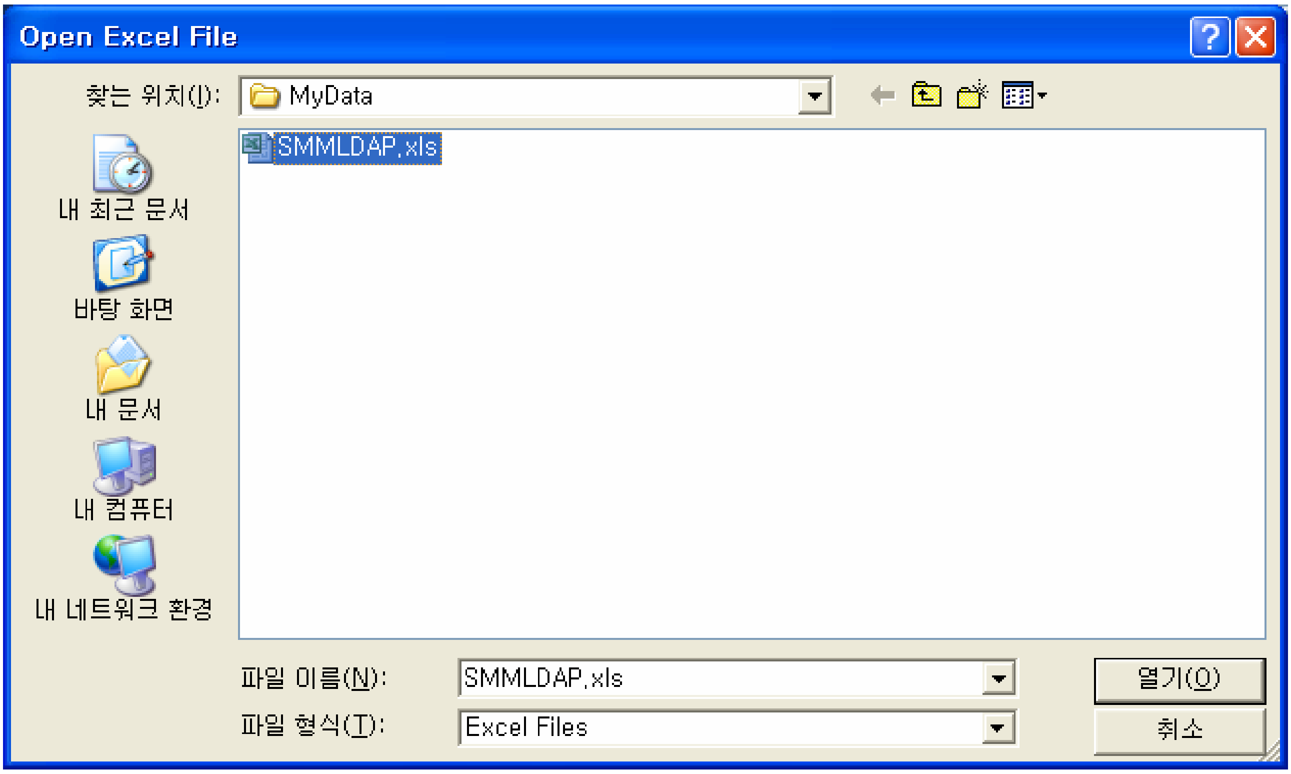


Figure 5-11 Screen for the Excel File to be Opened

3) The progress status of import is indicated for each category. The progress status up to the present can be checked at a glance.

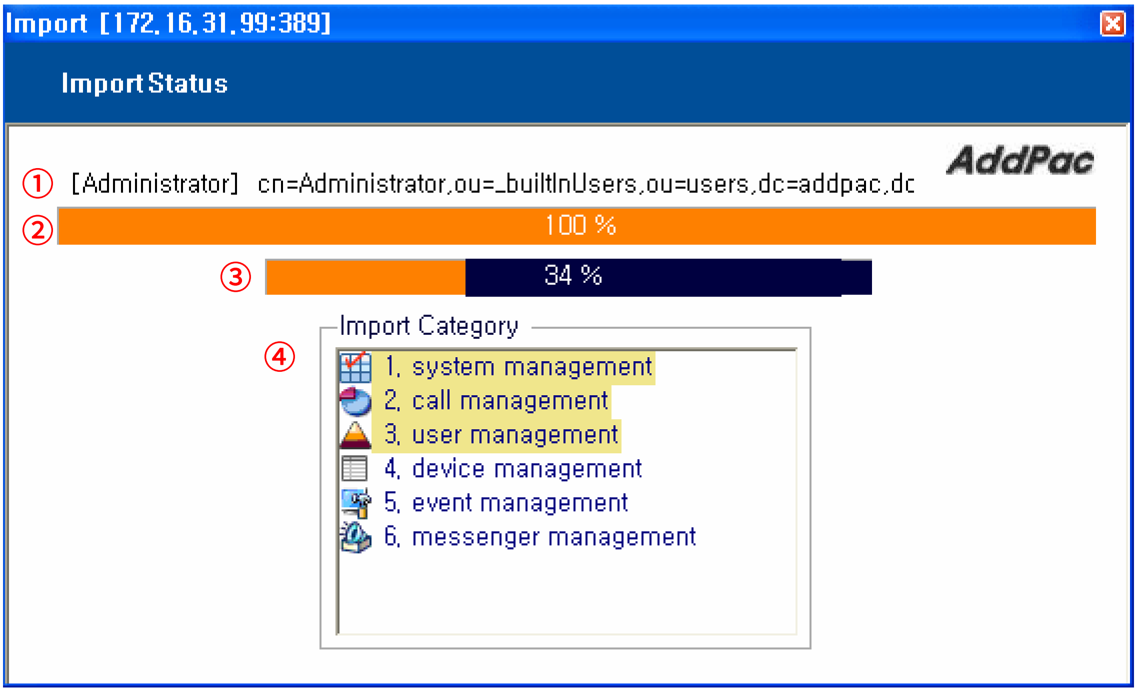


Figure 5-12 Excel Import Progress Screen

Table 5-2 Description of Excel Import Progress Screen

Field	Description
1	The detailed information of the data in progress.
2	A degree of the data in progress basing on a single spread sheet of excel is indicated.
3	A degree of the sheet in progress basing on a single category is indicated.
4	The category in progress is indicated.

4) When the importing is done, a message can be seen as ''Finished, All Successfully Imported' as it is shown in the following:

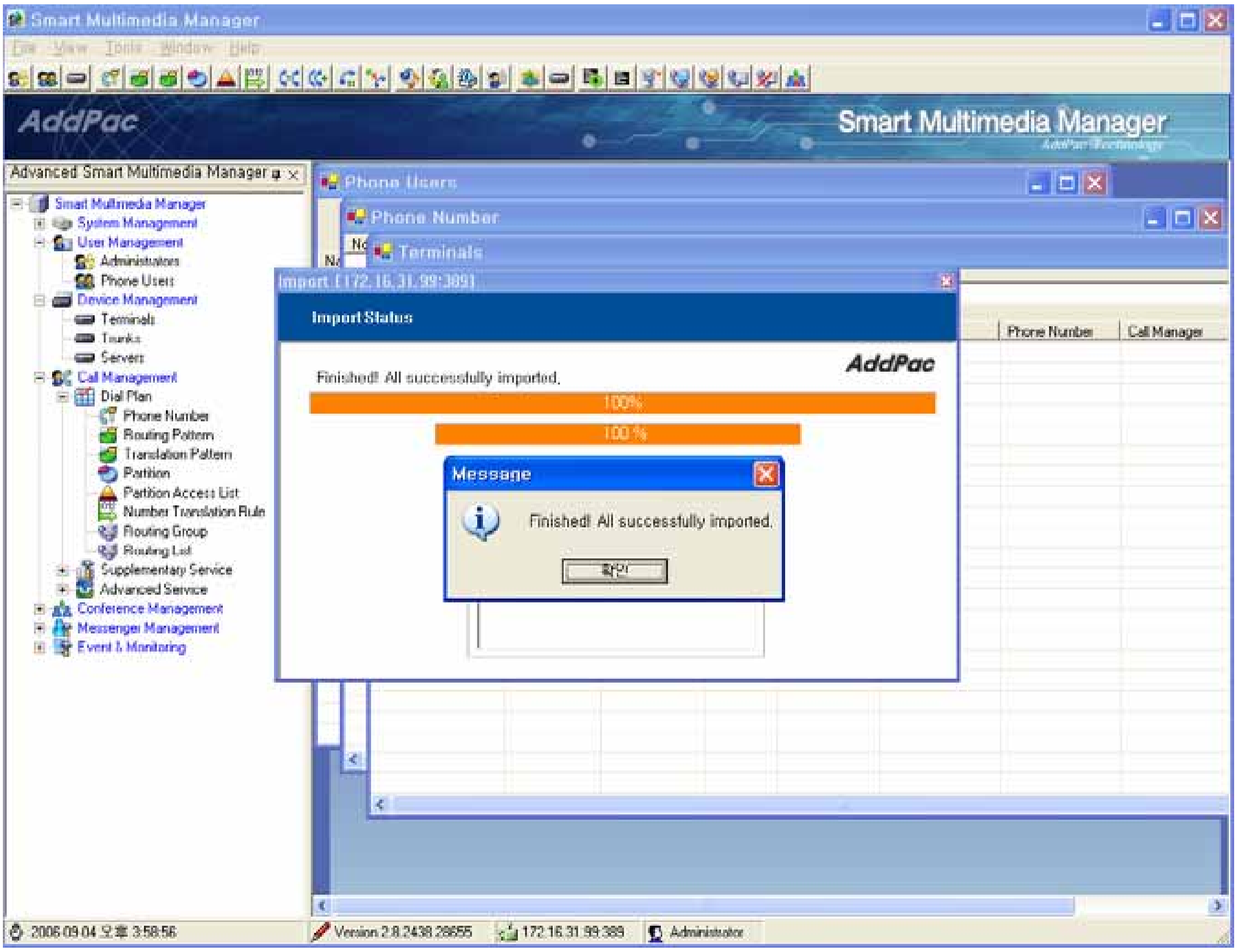


Figure 5-13 Excel Import Completed Screen

6) When 'OK' button is clicked, the job of import is ended. The saved import file can be opened to check the completed data.

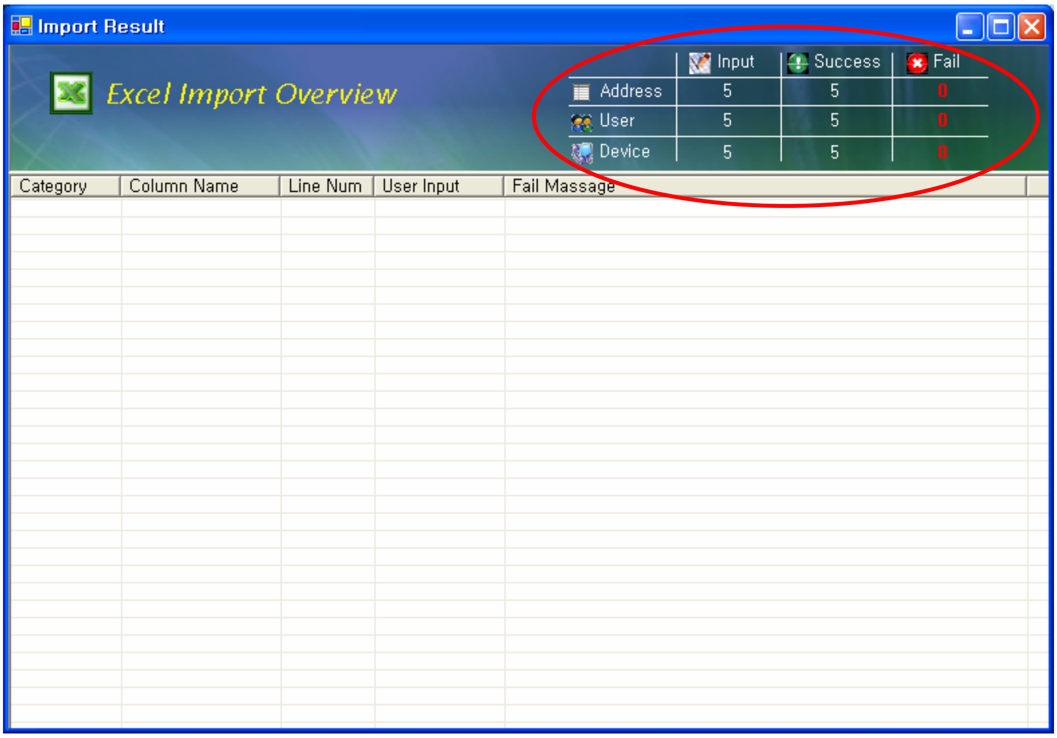


Figure 5-14 Excel Import Result Screen

5) If the user entered a wrong value of the data, a list of failure messages is displayed.

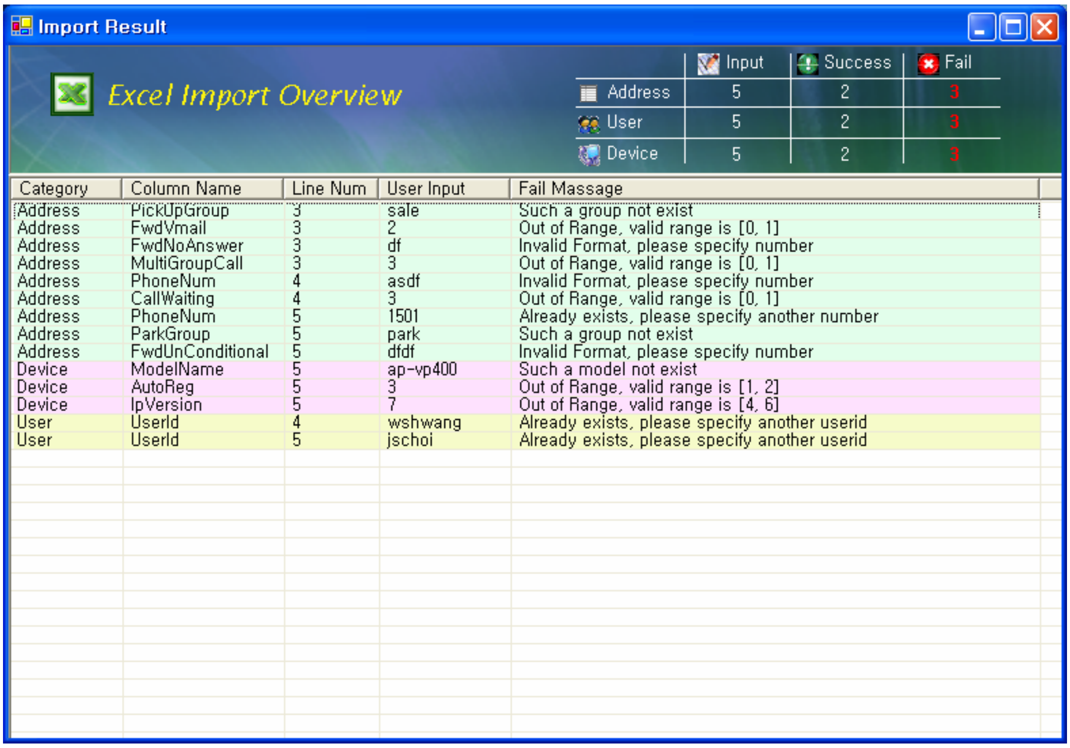


Figure 5-15 Excel Import Result Screen - 1

6) The user can double-click an item on the list to move to the corresponding cell to make a correction

on the wrong value.

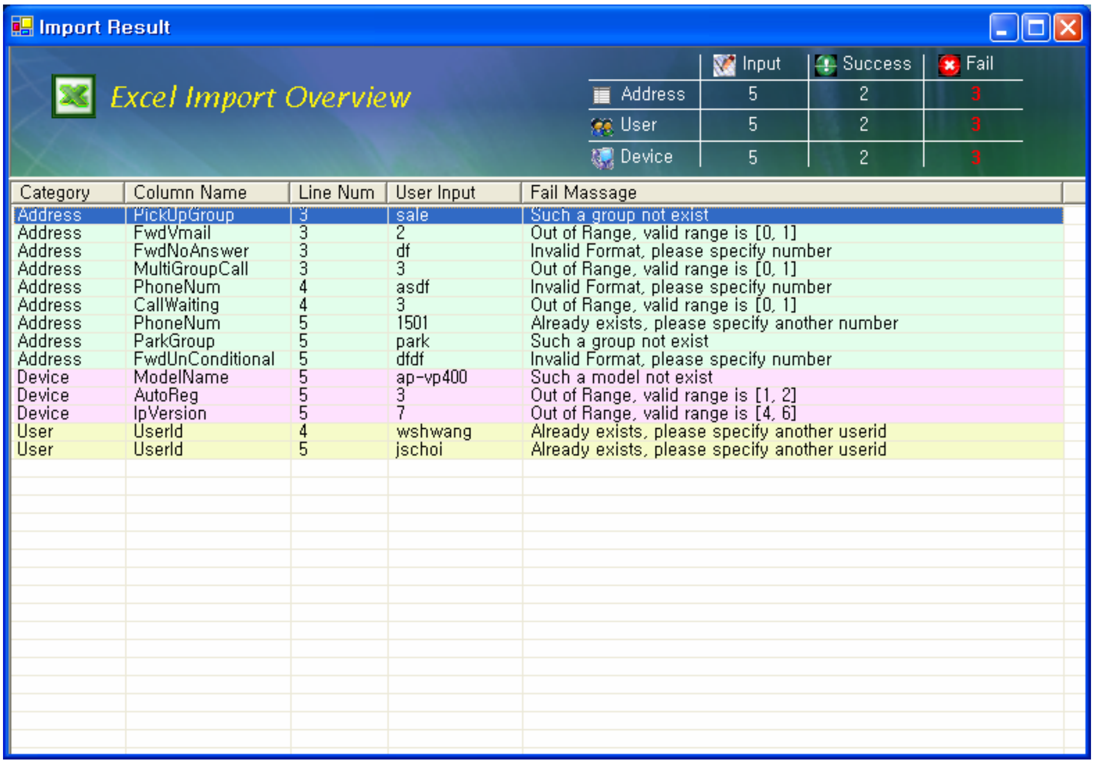


Figure 5-16 Excel Import Result Screen – 2

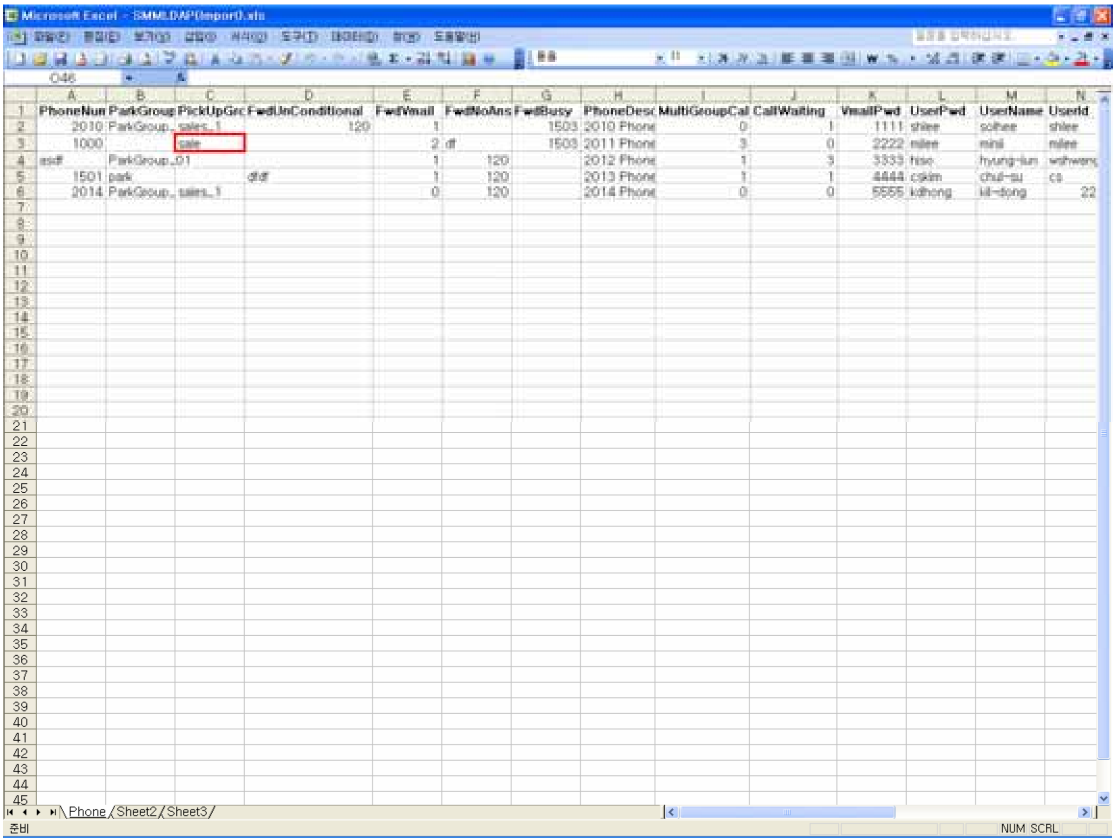


Figure 5-17 The Screen for Moving to the Corresponding Cell for the Failed Import Result

7) When the window of ‘Import Result’ is closed, another window is displayed for asking whether to

reboot. After the data is restores, the server needs to go through rebooting process to be operated with the upgraded data.

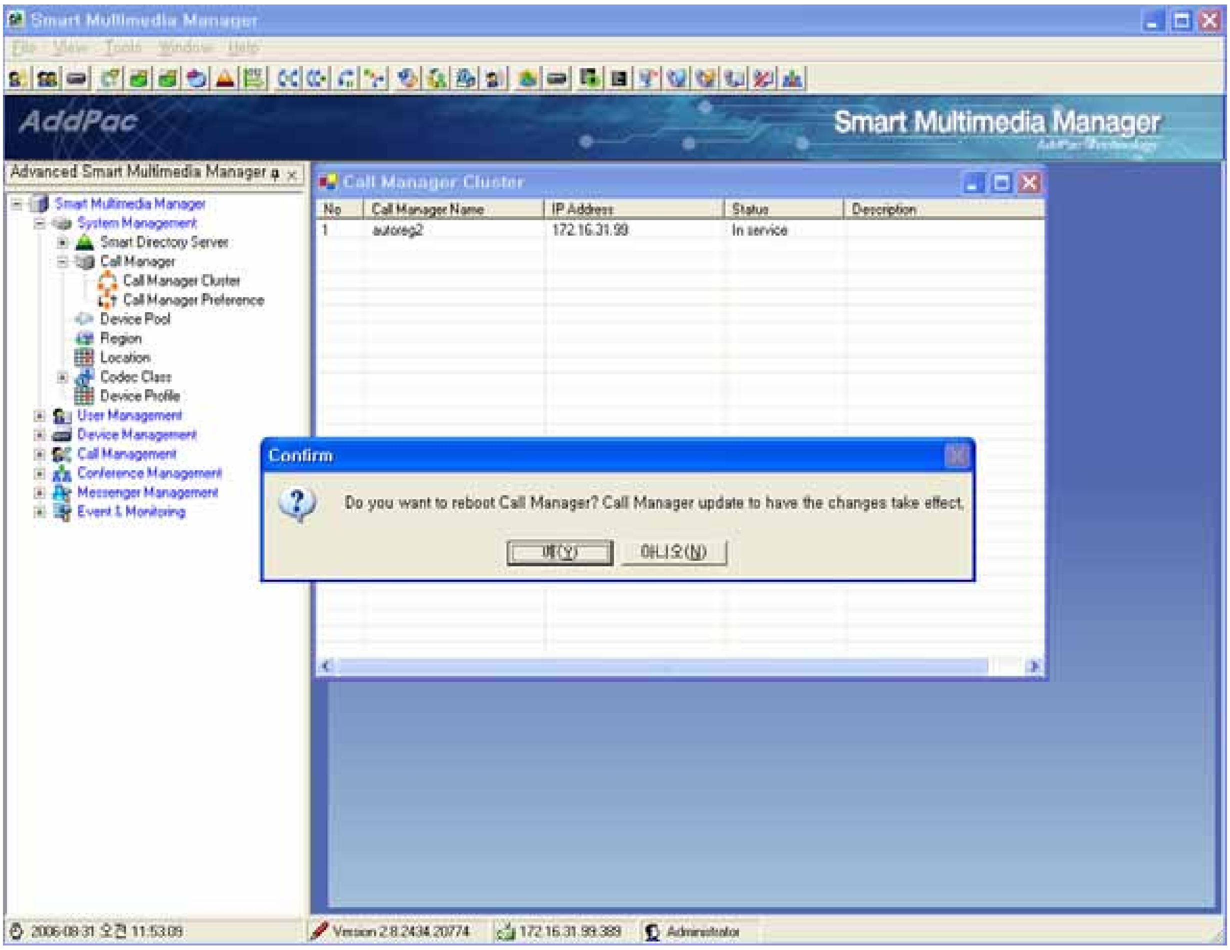


Figure 5-18 Screen for Asking whether to Reboot after Excel Import

8) When 'yes' button is clicked in the message window, the server processes the rebooting process. Then the mode of SMM login is to be performed. Afterwards the imported data can be checked through the login process.

Excel File Field and Input Format

Information of Excel Sheet for each SMM Menu

The menu for registering the data from SMM can be largely structured in 7 different categories. These 7 categories has many menu functions and each menu connected to the excel sheet. Basing on the entire menu, the one excel file with the name '.xls' can be extracted with an assigned sheet names and field names.

The following table shows the corresponding Excel sheet name for each menu of SMM:

1. System Management

Section	Menu	Function	Sheet Name
1.1	Smart Directory Server	Smart Directory Cluster	**sd cluster**
1.2		Smart Directory Preference	**sd preference**
1.3	Call Manager	Call Manager Cluster	**cm cluster**
1.4		Call Manager Terminal	**cm terminal**
1.5		Call Manager Preference	**cm preference**
1.6	Device Pool	Device Pool	**device pool**
1.7	Region	Region	**Region**
1.8	Codec Class	Codec Policy	**codec policy**
1.9		Audio Codec	**audio codec**
1.10		Video Codec	**video codec**
1.11	Location	Location	**location**
1.12	Device Profile	Device Profile	**device profile**

2. User Management

Section	Menu	Function	Sheet Name
2.1	Administrator	Administrator	**administrator**
2.2	Phone Users	Users	**phone user**
2.3		Organization	**user group**

3. Device Management

Section	Menu	Function	Sheet Name
3.1	Terminals	Terminals	terminal
3.2		Organization	terminal group
3.3	Trunks	Trunks	trunk
3.4	H.323 Gatekeeper	H.323 Gatekeeper	gate keeper
3.5	SIP Proxy Server	SIP Proxy Server	proxy server
3.6	MCU Server	MCU Server	server
3.7	Add Upgrade Server	Auto Upgrade Server	upgrade server
3.8	Broadcasting Server	Broadcasting Server	broadcast server

4. Call Management

Section	Menu	Function	Sheet Name
4.1	Dial Plan	Phone Number	phone number
4.2		Routing Pattern	route pattern
4.3		Partition	partition
4.4		Partition Access List	partition access list
4.5		Number Translation Rule	translation rule
4.6		Translation Pattern	translation pattern
4.7		Routing Group	routing group
4.8		Routing List	routing list
4.9	Supplementary Service	Hunt Group	hunt group
4.10		Pickup Group	pickup group
4.11		Park Address Pool	park address pool
4.12		Park Group	park group
4.13	Advanced Service	Music & Announcement	music announce
4.14		Auto Attendant Service	auto attendant
4.15		Voice Mail Service	voice mail
4.16		IVR Service	ivr service
4.17		Service Code	service code

5. Conference Management

Section	Menu	Function	Sheet Name
5.1	Conference Rooms	Conference	**conference**

6. Messenger Management

Section	Menu	Function	Sheet Name
6.1	Messenger Configuration	Messenger Configuration	**msg config**
6.2	User	Messenger Users	**msg user**
6.3	Organization	Messenger Organization	**msg group**

7. Event & Monitoring

Section	Menu	Function	Sheet Name
7.1	Event Configuration	Event Configuration	**event config**
7.2	Call Pattern	Call Pattern	**call pattern**

Field Description for Each Excel Sheet

The table below shows the fields organized in the excel sheet. The field names, description and input format can be verified.

1. System Management

1.1 sd cluster Sheet

Excel Header	Field Name	Description	Input Format
A	Name	A Name of Smart Directory Server	
B	Model	A Model Information of Smart Directory Server	IPNEXT - 500 IPNEXT - 200 AP - MC1000 AP - VP350MCU
C	Ip	An IP Address of Smart Directory Server	
D	ControlPort	Control Port(HTTP port) of Smart Directory Server	
E	Desc	A Description of Smart Directory Server	
F	LoginId	Access Information(ID) of Smart Directory Server	
G	LoginPwd	Access Information of (password) Smart Directory Server	
H	ServerPort	Access Information (port) of Smart Directory Server	
I	FTPId	FTP Access Information of (ID) Smart Directory Server	
J	FTPPwd	FTP Access Information(password)Smart Directory Server	
K	FTPPort	FTP Access Information (port) of Smart Directory Server	
L	FTPPassiveMode	An option whether to set FTP Passive	TRUE or FALSE

1.2 sd preference Sheet

Excel Header	Field Name	Description	Input Format
A	Name	Smart Directory Preference Name	
B	Desc	Smart Directory Preference Description	
C	Preference	Smart Directory Server List of the Registered Preference	**Refer to below 1.2 1)**

*** Reference 1.2 1) More than one Preference can be configured for Smart Directory Preference. The section between one preference to the other can be divided by using '/'.**

ex) IP-PBX1/IP-PBX2

1.3 cm cluster

Excel Header	Field Name	Description	Input Format
A	Name	Call Manager Name	
B	SIPRetryCount	SA number of attempts for SIP reconnection	1 - 10
C	SIPTimeout	Expiration time for transmitting SIP message	100 - 4000
D	SIPInviteExpire	Expiration time of SIP invite	5 - 600
E	SIPRegisterExpire	Expiration time of SIP Register	0 - 86400
F	SIPMinSessionExpire	The minimum value of expiring time of session	30 - 86400
G	SIPResponseType	A type of response to Invite	default alert progress
H	SIPSDPType	Type of SDP	with-sdp without-sdp default
I	SIPForceNAT	To support or not support the terminals under NAT setting	TRUE or FALSE
J	SSCPKeepAlive	The time interval of KeepAlive SSCP terminals	10 - 86400
K	SSCPRetryCount	A number of retransmitting SSCP message	1 – 10
L	SSCPRetryTimeout	Expiration time of retransmitting SSCP message	1000 - 1000000
M	SSCPPacketSize	SSCP Packet Size	800 - 1472
N	HTTPAuth	Whether to authenticate HTTP access to communicate with Call Manager	TRUE or FALSE
O	ControlPort	HTTP access information (port) to communicate with Call Manager	default : 80
P	UserID	HTTP access information (ID) to communicate with Call Manager	
Q	UserPassword	HTTP access information (password) to communicate with Call Manager	
R	FTPPort	FTP access information (port) to communicate with Call Manager	default : 21
S	FTPUserID	FTP access information to communicate with Call Manager (ID)	
T	FTPUserPassword	FTP access information to communicate with Call Manager (password)	
U	InterDigitTimeout	The time allowed for no input of DTMF tone	1000-100000
V	InitialDigitTimeout	The time for waiting the initial digit to be entered	1000-100000
W	InternalCall	A ringer sound for the internal call	**Renference below 1.3 1)**
X	ExternalCall	A ringer sound for the external call	
Y	InternalFwdCall	A ringer sound for the forwarded internal call.	
Z	ExternalFwdCall	A ringer sound for the forwarded external call	
AA	CallingParty	The default setting of displaying a caller	ALLOWED RESTRICT

AB	MaxRecentCall	The maximum number of saving the most recent call list for each user	
AC	CACMode	The maximum or minimum bandwidth to be reserved for a call	min or max
AD	OffnetTransfer	Allowing a call to be delivered to the outside while it is on the line with the outside	TRUE or FALSE
AE	MacAddress	MAC address of Call Manager	
AF	SaveRecentCall	Whether to save an incoming/ outgoing telephone list	TRUE or FALSE
AG	SipPort	SIP signaling port	5060
AH	DefaultLanguage	call-manager default language	(en, ko, jp ..)
AI	AgentInfo	Not in use	
AJ	UpdateTimeout	Not in use	
AK	NetworkDomain	Network domain list information	Public/Private
AL	FTPPassiveMode	Whether to set to FTP passive mode	TRUE or FALSE

*** Reference 1.3 1) For Distinctive Ring, one of the following options can be set.**

Distinctive Ring
dr1
dr2
dr3
dr4
dr5
silence
phone bell
ring
door bell
bike bell
harp
chirp
sonar

1.4 cm terminal Sheet

Excel Header	Field Name	Description	Input Format
A	CMName	Call Manager Name	
B	NetworkDomainName	Network Domain Name	
C	Ip4	IPv4 Netowrk domain address	
D	Ip6	IPv6 Network domain address	

E	Ver4Ip	IPv4 address	
F	Ver4UserInputIp	IPv4 User Input IP address	
G	Ver4SipPort	SIP port	
H	Ver4IsEnable	Whether to include Cluster of Call Manager	TRUE or FALSE
I	Ver4IsInDomain	Whether Call Manager interface is NATSetting or not	TRUE or FALSE
J	Ver4UseKeepAlive	Whether to use Keep alive for Terminal	TRUE or FALSE
K	Ver4RegT	Registration type	
L	Ver4Version	Call Manager version	
M	Ver4RegCM	Call Manager Terminal ID	
N	Ver4NetProtocolT	Protocol type	
O	Ver4TerminalState	Status of the terminal	
P	Ver6Ip	IPv6 address	
Q	Ver6UserInputIp	IPv6 address for User Input	
R	Ver6SipPort	SIP port	
S	Ver6IsEnable	Whether to include Cluster of Call Manager	TRUE or FALSE
T	Ver6IsInDomain	Whether Call Manager interface is NAT Setting or not	TRUE or FALSE
U	Ver6UseKeepAlive	Whether to use Keep alive of Terminal	TRUE or FALSE
V	Ver6RegT	Registration Type	
W	Ver6Version	Call Manager version	
X	Ver6RegCM	Call Manager Terminal ID	
Y	Ver6NetProtocolT	Protocol Type	
Z	Ver6TerminalState	State of Teminal	

1.4 cm preference

Excel Header	Field Name	Description	Input Format
A	Name	Call Manager Preference Name	
B	Desc	Call Manager Preference Description	
C	Preference	Call Manager list included in Preference	**Reference below 1.4 1)**

*** Reference1.4 1) More than 1 Preference of Call Manager can be configured. The section between one preference to the other can be divided by using '/'.**

ex) IP-PBX1/IP-PBX2

1.5 cm terminal

Excel Header	Field Name	Description	Input Format
A	CMName	Call Manager Name	
B	NetworkDomain	Call Manager의 Network Domain	
C	Ip4	Call Manager Terminal version 4 IP Address	
D	Ip6	Call Manager Terminal version 6 IP Address	
E	Ver4Ip	Call Manager Terminal version 4 IP Address	
F	Ver4UserIp	Call Manager Terminal version 4 사용자 입력 IP Address	
G	Ver4SipPort	Call Manager Terminal version 4 SIP 포트	**default : 5060**
H	Ver4IsEnable	Call Manager Terminal version 4 Service 여부	**TRUE/FALSE**
I	Ver4IsInDomain	Call Manager Terminal version 4 Domain 설정 여부	**TRUE/FALSE**
J	Ver4UseKeepAlive	Call Manager Terminal version 4 KeepAlive 여부	**TRUE/FALSE**
K	Ver4RegT	Call Manager Terminal version 4 Registration 타입	**1 - Static Registreation** **2 – Dynamic Registration**
L	Ver4Version	Call Manager Terminal version 4 APOS 버전	
M	Ver4RegCM	(현재 사용하지 않음)	
N	Ver4NetProtocolT	Call Manager Terminal version 4 Network Protocol 타입	**4 - Version 4** **6 - Version 6**
O	Ver4TerminalState	Call Manager Terminal version 4 Terminal 상태	**1 - In Service** **2 - Out of Service**
P	Ver6Ip	Call Manager Terminal version 6 IP Address	
Q	Ver6UserIp	Call Manager Terminal version 6 사용자 입력 IP Address	
R	Ver6SipPort	Call Manager Terminal version 6 SIP 포트	**default : 5060**
S	Ver6IsEnable	Call Manager Terminal version 6 Service 여부	**TRUE/FALSE**
T	Ver6IsInDomain	Call Manager Terminal version 6 Domain 설정 여부	**TRUE/FALSE**
U	Ver6UseKeepAlive	Call Manager Terminal version 6 KeepAlive 여부	**TRUE/FALSE**
V	Ver6RegT	Call Manager Terminal version 6 Registration 타입	**1 - Static Registration** **2 – Dynamic Registration**
W	Ver6Version	Call Manager Terminal version 6 APOS 버전	
X	Ver6RegCM	(현재 사용하지 않음)	

Y	Ver6NetProtocolT	Call Manager Terminal version 6 Network Protocol 타입	**4 - Version 4** **6 - Version 6**
Z	Ver6TerminalState	Call Manager Terminal version 6 Terminal 상태	**1 - In Service** **2 - Out of Service**

1.6 device pool

Excel Header	Field Name	Description	Input Format
A	Name	A name of Device Pool	
B	Desc	A description Device Pool	
C	sm_preference	Smart Directory Preference to be applied to Device	
D	cm_preference	Call Manager Preference to be applied to Device	
E	region	Region to be applied to Device	

1.7 region

Excel Header	Field Name	Description	Input Format
A	Name	A name of Region	
B	Desc	A description of Region	

1.8 location

Excel Header	Field Name	Description	Input Format
A	Name	a name of Location	
B	Desc	A description of Location	
C	Bandwidth	Set the limits to bandwidths of the devices in the location (**Total**)	**Set to Unlimited -1** **Setting Bandwidth- Above 0**
D	Audio_Bandwidth	Set the limits to bandwidths of the devices in the location (**Audio bandwidth**)	**Above 0**
E	Video_Bandwidth	Set the limits to bandwidths of the devices in the location (**Video bandwidth**)	**Above 0**
F	Location_Group	Set up the Location Group included in the Location	

1.9 location group

Excel Header	Field Name	Description	Input Format
A	Name	a name of Location	
B	Desc	A description of Location	
C	Bandwidth	Set the limits to bandwidths of the devices in the location (**Total**)	**Set to Unlimited -1** **Setting Bandwidth- Above 0**
D	Audio_Bandwidth	Set the limits to bandwidths of the devices in the location (**Audio bandwidth**)	**Above 0**
E	Video_Bandwidth	Set the limits to bandwidths of the devices in the location (**Video bandwidth**)	**Above 0**

1.10 codec policy

Excel Header	Field Name	Description	Input Format
A	region name	A name of registered Region	
B	audio codec	A type of Audio Codec set to Region	G711A G723 G726 G729 G711U
C	video codec	A type of Video Codec set to Region	MPEG4 H263 H264
D	picture size	Picture Size set to Region	0 - QVGA 1 - HVGA 2 – VGA
E	frame rate	A Frame Rate set to Region	9 - 30
F	target rate	A Target Rate set to Region	[64 - 2048] 6의 배수
G	audio class	A name of Audio Class set to Region	
H	video class	A name of Video Class set to Region	

1.11 audio codec

Excel Header	Field Name	Description	Input Format
A	Name	A name of Audio Codec Class	
B	Desc	Audio Codec Class Description	
C	Preference	Audio Codec List assigned toCodec Class	**Reference below 1.8 1)**

*** Reference 1.8 1) More than 1 Preference of Call Manager can be configured. The section between one preference to the other can be divided by using '/'.**

ex) G711A/G729

1.12 video codec

Excel Header	Field Name	Description	Input Format
A	Name	A name of Video Codec Class	
B	Desc	A description of Video Codec Class	
C	Video Info	Video Codec list assigned to Codec Class	**Reference below 1.9 1)**

A Video Codec can be configured to more than 1 including the details below. The section between one video codec to another to the other can be divided by using '/' the details of video codec can be divided by '&'

ex) video.codec=1&video.picture.size=1&video.frame.rate=30&video.target.rate=12

/video.codec=2&video.picture.size=2&video.frame.rate=30&video.target.rate=12

Detail	Setting Value
video.codec	1 - MPEG4 2 - H263 3 - H264
video.picture.size	0 - QVGA 1 - HVGA 2 – VGA
video.target.rate	[64 - 2048] 6의 Multiples
video.frame.rate	9 - 30

1.13 device profile

Excel Header	Field Name	Description	Input Format
A	Name	A Device Profile name	
B	Desc	A description of the Device Profile	
C	FaxProtocol	Set to VoIP Fax Protocol	0 - bypass 1 - inband-t38 2 - t38
D	FaxRate	Set to VoIP Fax Rate	disable 2400 4800 7200 9600 12000 14400
E	DTMFRelayType	Set to VoIP DTMF Relay	0 - rtp 4 - rtp-2833 5 - dual-mode 6 - info 7 - cisco-info
F	SidEnable	When mute status is recognized, select whether transmit packet	TRUE or FALSE
G	VadEnable	When mute status is recognized, select not to transmit voice packet	TRUE or FALSE
H	ComfortNoise	Set to make Comport Noise for mute	TRUE or FALSE
I	EchoCancel	Set to cancel Echo cancellation	TRUE or FALSE
J	G.711PacketInterval	To set a number of frame per packet for each codec	1 - 5
K	G.726PacketInterval	To set a number of frame per packet for each codec	1 - 8
L	G.723 PacketInterval	To set a number of frame per packet for each codec	1 - 8
M	G.729 PacketInterval	To set a number of frame per packet for each codec	1 - 8
N	RetryCount	A number of retransmitting SIP message	3 - 10
O	RetryTimeout	Expiration time for retransmitting SIP message	100 - 4000
P	InviteTimeout	Expiration time for sip invite	5 - 600
Q	RegisterTimeout	A setting value of Register Expire time of SIP terminals	10 - 86400
R	RegisterRetryTimeout	Expiration time of retransmitting SIP Register	10 - 86400
S	SSCPRetryCount	A number of retransmitting SSCP message	1 - 10

T	SSCPRetryTimeout	Expiration time of retransmitting SSCP message	
U	SSCPRegisterRetryTimeout	A number of times that SSCP register message to retransmit after it fails in a number of times as many as retry count	1000 - 1000000

2. User Management

2.1 administrator

Excel Header	Field Name	Description	Input Format
A	UserId	An administrator's account ID	
B	UserPwd	An administrator's password	
C	Desc	An administrator's description	

2.2 phone user

Excel Header	Field Name	Description	Input Format
A	UserName	A user's name	
B	UserId	A user's ID	
C	UserPwd	A user's password	
D	VmailPwd	A user's password for Voice Mail	4 digit input is mandatory
E	PhoneNum	A phone number assigned to a subscriber	**Reference below 2.2 1)**
F	UserDesc	Description A description of a subscriber	
G	DisplayName	A nick name of a subscriber to be displayed on Smart Messenger	
H	HomePhone	A subscriber's home number	
I	MobilePhone	A subscriber's telephone number	
J	Email	A subscriber's e-mail address	
K	GroupPath	A subscriber's group path	
L	DisplayOrder	An order of a subscriber's group or sub-group list to be displayed on Smart Messenger	
M	AddressPartition	An information of Partition list for the phone number assigned to a subscriber (to be registered 1:1 with the phone number)	**Reference below 2.2.2)**

*** Reference 2.2.1) More than one phone number can be assigned to a user. One preference to another can be divided by using '/'.**

ex) 1000/1001/1002

***Reference 2.2.2) AddressPartition is the partition information of phone number and it corresponds to the phone number 1:1. One partition to another can be divided by using '/'.**

ex) partition_01/ /partition_02 (if the number 1001 is not set to the partition, it is treated as a blank)

2.3 user group

Excel Header	Field Name	Description	Input Format
A	GroupName	Registration of a group name	
B	GroupDesc	Register a description of the group	
C	Path	the path of the group that a subscriber belongs to	
D	BusinessCategory	Whether to set up Presence Notify group from Smart Messenger	TRUE or FALSE
E	DisplayOrder	An order to be displayed on Smart Messenger for the group in a lower rank or the group to which the subscriber belongs to	

3. Device Management Field Information

3.1 terminal

Excel Header	Field Name	Description	Input Format
A	DeviceName	A terminal name	
B	DeviceDesc	A description of the terminal	
C	ModelName	a terminal model	**Reference below 3.1 1)**
D	SignalingProtocal	A type of VoIP Signaling Protocol to be used from the terminal	0 - SIP 1 - H.323
E	AutoReg	In case of Static Register, FALSE is to be selected and IP Version, IP address, Signaling Port, Network Domain of Call Manager is to be entered. In case of Dynamic Register, TRUE is to be selected.	TRUE or FALSE
F	MacAddr	SIP signaling port of the corresponding device is to be entered for Static Register. In case of IP-Phone, the terminal can be registered automatically with MAC address.	
G	IpVersion	IP Version is to be entered in case Auto Registration is FALSE	4 - version4 6 - version6
H	IpAddr	When Auto Registration is FALSE, the IP address is to be entered.	
I	SignalingPort	When Auto Registration is FALSE, Signaling Port is to be entered	
J	PhoneNum	A phone number to assigned to the corresponding port	**Reference below 3.1 2)**
K	ModelType	A model type of the terminal	IPPhone or Gateway
L	GroupPath	A group path of the corresponding terminal	
M	DevicePool	Select a Device Pool	
N	Location	Select a Location	
O	NetworkDomain	When Auto Registration is FALSE, Network Domain is to be entered	
P	RTPProxyRequired	A force setup to use RTP Proxy when a call is on line	TRUE or FALSE
Q	TerminalAuth	Whether to authenticate when the corresponding terminal is registered to IPNext PBX by SIP Register	TRUE or FALSE
R	UserInfo	An IP and password to be used for authentication when the terminal is registered to IP-PBX with SIP (SIP Registration)	**Reference below 3.1 3)**

S	DeviceProfile	Select a Device Profile	
T	PartitionAccessList	Select Partition Access List	
U	DefaultPhoneNum	One phone number to be transmitted basically, when the corresponding terminal makes an outbound call	**Reference below 3.1 2)**
V	InternalCall	A ringer sound between the internal calls	**Reference below 3.1 4)**
W	ExternalCall	A ringer sound for the incoming external call	**Reference below 3.1 4)**
X	InternalForwardedCall	A ringer sound for the forwarded internal call	**Reference below 3.1 4)**
Y	ExternalForwardedCall	A ringer sound for the forwarded external call	**Reference below 3.1 4)**
Z	CallManager	Call Manager information of the corresponding terminal (cm terminal ID)	
AA	AddressPartition	A telephone number to be assigned to the terminal for the information of Partition List (to be registered with PhoneNum 1:1)	**Reference below 3.1.5)**
AB	DefaultAddressPartition	A partition list information of the phone number with the default setting assign to the terminal	**Reference below 3.1.6)**

- **Reference 3.1 1) The model name can be chosen one of the following values**

IPPhone	Gateway
AP - VP200	AP - VG1000
AP - VP300	AP2620
AP - VP350	AP2520
AP - IP300	AP2120
AP - IP200	AP1601
GeneralIPPhone	AP1200
	AP1100F
	AP1100
	AP1005
	AP1002
	AP1000
	AP300
	AP200
	AP200E
	AP200C
	AP190
	AP2650

*** Reference 3.1 2) The terminal can have more than one port, each port can be assigned with many phone numbers. One phone number to another can be divided by using /'. One port to another can be divided by using ','.**

ex) 1000/1001, 1002/1003

*** Reference 3.1 3) One port can select one Useinfo. ID, Password can be divided by '/' each port can be divided by ','.**

ex) root/router, admin/router

*** Reference 3.1 4) For Distinctive Ring, one of the setting values in the followings can be selected**

Distinctive Ring
dr1
dr2
dr3
dr4
dr5
Silence
phone bell
Ring
door bell
bike bell
harp
chirp
sonar

*** Reference 3.1 5) Address Partition is the partition information of phone numbers and it corresponds to the phone number 1:1. One partition to another can be divided by using '/'.**

For the following phone numbers: "1000/1001, 1002/1003"

ex) partition_01/partition_02, partition_03/ (if the number 1001 is not set to the partition, it is treated as a blank)

*** Reference 3.1 6) Default Address Partition represent a partition information of the default phone number for each port. When there are many ports, each port can be divided by ',' as it is shown in the following:**

ex) partition_01,partition_02

3.2 terminal group

Excel Header	Field Name	Description	Input Format
A	GroupName	A name of terminal organization	
B	GroupDesc	A description of the organization	
C	Path	A path of the organization	

3.3 trunk Sheet

Excel Header	Field Name	Description	Input Format
A	Name	A name of Trunk Gateway	
B	Desc	A description of Trunk Gateway	
C	SignalingProtocol	Select VoIP protocol	0 - SIP 1 - H.323
D	IpVersion	ISelect IP version	4 - version4 6 - version6
E	IpAddress	Register IP address	
F	SignalingPort	The port corresponding to VoIP protocol	
G	CalledNumber	The incoming number to which Translation Rule is to be applied to the number translation for the Inbound call.	
H	CallingNumber	The outgoing number to which Translation Rule is to be applied to the number translation for the inbound call.	
I	DTMFRelay	A type of DTMF-Relay to be transmitted on the trunk **Rtp-2833** : DTMF to be transmitted with accordance with RFC-2833 standard **Inband** : Inband DTMF tone is to be transmitted through RTP **Out-of-band** : DTMF is to be transmitted by using SIP INFO, H.245	0 - Rtp-2833 1 - Inband 2 - Out-of-band
J	CallPriority	The call priority order is to be set for the Inbound call.	0 - 7
K	MusicOnHold	This is a setup to provided MOH for the hold request of a terminal	TRUE or FALSE
L	NortelHold	This a setup to provide MOH as to be connected with Nortel softswitch	TRUE or FALSE
M	RtpProxyRequired	This is a force setup to use RTP Proxy while the call on line	TRUE or FALSE
N	ExternalDevice	Configure an option whether the trunk is to be connected with the outside	TRUE or FALSE
O	MRBT	Select whether to provide MRBT(Media Ringback tone) for the Inbound call	TRUE or FALSE
P	DevicePool	Select Device Pool	
Q	Location	Select Location	
R	NetworkDomain	Select Network Domain of Call Manager	
S	PartitionAccessList	Select Partition Access List for provisioning the Inbound call	
T	CallingParty	Select and option to display the outgoing number for the Outbound	0 - Default

	Presentation	call **Default** : Follow Default setup of Call Manager (Call Manager Cluster > Options > Calling Party Presentation) **Allowed**: Display my telephone number to the other **Restricted** : Not to display my phone to the other party	1 - Allowd 2 - Restricted
U	CallerIDDN	This is a force setup of the outgoing number (ex: general directory number)	
V	MonitoringFault	Configure an option whether the trunk is to be connected with the outside	TRUE or FALSE

3.4 gate keeper

Excel Header	Field Name	Description	Input Format
A	Name	A name of gatekeeper	
B	Desc	A description of the gatekeeper	
C	CallStartMode	Call Setup mode of H.323	0 - FastStart 1 - SlowStart 2 - PreferredSlow
D	H.245Tunneling	Tunneling mode to set up H.245	TRUE or FALSE
E	CallPriority	This is the call priority order for the Inbound call.	0 - 7
F	H.323ID	Register H.323 ID	
G	H.323Password	Register H.323 password	
H	KeepGKonPRJ	This is a setup to keep registering Gatekeeper even the response is rejected	TRUE or FALSE
I	RtpProxyRequired	This is a force setup to use RTP proxy when the call is on line	TRUE or FALSE
J	ExternalDevice	Set up an option whether the trunk is to be connected with the outside	TRUE or FALSE
K	Register	Set up an option whether a Phone Number is to be registered to Gatekeeper	TRUE or FALSE
L	CalledNumber	The incoming number to which Translation Rule is to be applied to the number translation for the Inbound call.	
M	CallingNumber	The outgoing number to which Translation Rule is to be applied to the number translation for the inbound call.	
N	PhoneNumber	The phone number to be registered to the Gatekeeper	**Reference below 3.4 1)**
O	TechPrefix	A prefix of a telephone number to be registered to Gatekeeper	**Reference below 3.4 1)**
P	DevicePool	Select a device pool	
Q	Location	Select a location	
R	NetworkDomain	Select a network domain	
S	H.323GKList	The priority order can be given by registering IP address of Gatekeeper to the list	**Reference below 3.4 2)**
T	PartitionAccessList	Select Partition Access lList for provisioning the Inbound call	

U	CallingParty Presentation	Select and option to display the outgoing number for the Outbound call **Default** : Follow Default setup of Call Manager (Call Manager Cluster > Options > Calling Party Presentation) **Allowed**: Display my telephone number to the other **Restricted** : Not to display my phone to the other party	0 - Default 1 - Allowd 2 - Restricted
V	CallerIDDN	A force setup of the outgoing number (ex: general directory number)	
W	MonitoringFault	Whether to treat a failed registration as a system failure of Call Manager	TRUE or FALSE

*** Reference 3.4 1) Gatekeeper can be assigned with more than one phone number and prefix. Each number can be divided by using '/'.**

ex) 1000/1001/1002

*** Reference 3.4 2) Gatekeepr can be assigned with more than one H.323GKList. Each Serverip, Serverport, Serverid can be divided by using ',' and the list can be divided by using '/'.**

ex) 1.1.1.1, 8020, root/1.1.1.2, 8020, admin

3.5 proxy server

Excel Header	Field Name	Description	Input Format
A	Name	A name of SIP Proxy Server	
B	Desc	A description of SIP Proxy Server	
C	DevicePool	Select a device pool	
D	Location	Select a location	
E	SIPUserName	Register a SIP User Name	
F	SIPPassword	Register of a SIP password	
G	LocalDomain	Register of a local domain	
H	NetworkDomain	Select a network domain	
I	CalledNumber	An incoming number to be applied with Translation Rule for the number translation of an inbound call	
J	CallingNumber	An outgoing number to be applied with Translation Rule for the number translation of an inbound call	
K	CallPriority	A setup of a call priority order for an inbound call	0 - 7
L	SIPProxyServerList	IP address and port list of SIP Proxy server can be configured with a priority order	**Reference below 3.5 1)**
M	RTPProxyRequired	A force setup of using RTP Proxy while the call is on line	TRUE or FALSE
N	UseLocalHostName	Use a local host name instead of the domain name of the proxy	TRUE or FALSE

		server to be registered To : 7000@local hostname From : 7000@local hostname	
O	UseUserName	Register a user name instead of e164 to be registered to the proxy server To : jschoi@172.16.32.40 From : jschoi@172.16.32.40 Contact : jschoi@172.16.32.70	TRUE or FALSE
P	Register	Set up an option whether to register a phone number to the proxy server	TRUE or FALSE
Q	UseMusicOnHold	A setting to provided MOH for a hold request from a terminal	TRUE or FALSE
R	NortelHold	A setting to provide MOH from connection with Nortel softswitch.	TRUE or FALSE
S	ExternalDevice	Set up an option whether the trunk is to be connected to out side	TRUE or FALSE
T	AddressInfo	A list of phone numbers to be registered to SIP Proxy Server **UserName** : a user name for phone number authentication **Password** : a password for phone number authentication	**Reference below 3.5 2)**
U	PartitionAccessList	Select a partition access list for provisioning an inbound call	
V	CallingParty Presentation	Select and option to display the outgoing number for the Outbound call **Default** : Follow Default setup of Call Manager (Call Manager Cluster > Options > Calling Party Presentation) **Allowed**: Display my telephone number to the other **Restricted** : Not to display my phone to the other party	0 - Default 1 - Allowd 2 - Restricted
W	CallerIDDN	A force setup of the outgoing number (ex: general directory number)	
X	MonitoringFault	Whether to treat a failed registration as a system failure of Call Manager	TRUE or FALSE
Y	UseP-AssertedIdentityHeader	An option for using P-Asserted-Identity Header	TRUE or FALSE
Z	RegisterExpireTimeout	A setting value of expiration time for registration is used when SIP Proxy Server is registered	
AA	DTMFRelay	Select a type of DTMF-Relay to be transmitted by the Trunk	0 - Rtp-2833 1 - Inband 2 - Out-of-band

*** Reference 3.5 1) More than one SIPProxyServerList can be configured to ProxyServer. Each Serverip, Serverport, Serverid can be divided by using ‘,’ and each list can be divided by using ‘/’.**

ex) 1.1.1.1, 8020, root/1.1.1.2, 8020, admin

*** Reference 3.5 2) More than one phone number can be assigned to Proxyserver. Each phone number can be configured with RegisterUserId, RegisterUserPassword. Each PhoneNumber, RegisterUserId, RegisterUserPassword is divided by using ‘,’ and each list is divided by using ‘/’.**

ex) 1000,root,router/1001,admin,router

****

3.6 server

Excel Header	Field Name	Description	Input Format
A	Name	A Server name	
B	Desc	A Description of the Server	
C	DevicePool	Select a Device Pool	
D	ModelName	Select a Server model	IPNEXT - 500 IPNEXT - 200 AP - MC1000 AP - VP350MCU
E	Location	Select a Location	
F	ServiceIpAddress	Register an IP address of the Server	
G	ServiceControlPort	Register a control port to communicate the IPNext PBX with the Sever	default : 5080
H	ControlPort	HTTP access port which can control a MCU Server	default : 80
I	SessionCapacity	a number of session for the MCU Server	default : 2
J	PartyCapacity	A number of participants in each session of the MCU Server	Default : 32
K	MacAddress	MCU Server의 Mac 주소 A Mac address of the MCU Server	
L	NetworkDomain	Select a Network Domain to be used for connection with MCU Server	
M	Authentication	An option whether to use HTTP account authentication to control the MCU Server.	TRUE or FALSE
N	UserId	HTTP account access information to control the MCU Server (ID)	default : root
O	UserPassword	HTTP account access information to control the MCU Server (password)	default : router

P	EnableService	An option whether to enable or disable service of the MCU server	TRUE or FALSE

3.7 upgrade server

Excel Header	Field Name	Description	Input Format
A	Name	A Server name	
B	Desc	A Description of the Server	
C	DevicePool	Select a Device Pool	
D	Location	Select a Location	
E	ServerURL	Configure URL for an Upgrade Server ex) http://[2001:230:20c:6011::2]/upgrade/	
F	UpgradeServerType	Select a type of the Upgrade Server	0 - Addpac Web Server 1 - General Web Server
G	FTPPort	Set FTP access information of the Upgrade Server (FTP access port).	default : 21
H	FTPUserID	Set FTP access information of the Upgrade Server (ID)	
I	FTPUserPassword	Set FTP access information of the Upgrade Serve (password)	
J	HTTPAuthentication	Select an option whether to use HTTP authentication when the Upgrade Server is accessed or not	TRUE or FALSE
K	ServerEnable	Select an option whether to enable the Upgrade Server or not	TRUE or FALSE
L	UpgradeInfo	Register a model to be upgraded by the Upgrade Server for uploading the firmware. The current program version and most recent time of uploading are displayed on the list of each model	**Reference below 3.7 1)**
M	FTPPassiveMode	An option whether to set the Upgrade Server to FTP Passive Mode or not	TRUE or FALSE

*** Reference 3.7 1)** The Upgrade Server manages a history of uploaded APOS image or program. The history consists of more than one list. Each detail can be divided by using ‘,’ and history can be divided by using ‘&’.

ex) model=AP-VP300,version=8.36.004,datetime=2006-08-20 18:01:13,url=http://61.33.161.11/upgrade/AP-VP300/_g2_v8_36_004.bin,upgrade_mode=1&model=AP- VP300/vp300_g2_v8_36_004,datetime=2006-08-20 17:11:08,url=http://61.33.161.11/upgrade/AP-IP200/ip200_g2_v8_36_004.bin,upgrade_mode=0

3.8 broadcast server sheet

Excel Header	Field Name	Description	Input Format
A	Name	A Server name	
B	Desc	A description of the Server	
C	ModelName	A model name of the Server	
D	DevicePool	An information of Device Pool	
E	Location	An information of Location	
F	IpAddress	An IP address of the Server	
G	ControlPort	A control port to be used for interoperability of Broadcasting Server with Call Manager	
H	NetworkDomain	A network domain to be used for interoperability of Broadcasting Server with Call Manager	Netdomain.name=public

3.9 presence server 시트

Excel Header	Field Name	Description	Input Format
A	Name	Register a name of Presence Server	
B	Desc	Register a description of Presence	
C	ModelName	Select a server model	**AP-PS2000**
D	DevicePool	Select a Device Pool	
E	Location	Select a Location	
F	IpAddress	Register an address of Presence Server	
G	ControlPort	Register a control port for communication between the server and IPNext PBX	**default : 5080**
H	NetworkDomain	Select a network domain	
I	UserInputIP	Register User Input address to Presence Server	
J	HTTPPort	Set up HTTP access information of Presence Server (port number)	**default : 80**
K	HTTPID	Set up HTTP access information of Presence Server (user ID for authentication)	
L	HTTPPassword	Set up HTTP access information of Presence Server (password for authentication)	
M	HTTPAuth	Set up HTTP access information of Presence Server (an	**TRUE/FALSE**

	option to be authenticated)

4. Call Management

4.1 phone number

Excel Header	Field Name	Description	Input Format
A	PhoneNum	Register a phone number	
B	PickUpGroup	Configure a Pickup Group to which the corresponding phone number to be used	
C	ParkGroup	Configure a Park Group to which the corresponding phone number to be used	
D	FwdUnConditional	Unconditional Call Forwarding	**Reference below 4.1 1)**
E	FwdBusy	Busy Call Forwarding	**Reference below 4.1 1)**
F	FwdNoAnswer	No Answer Call Forwarding	**Reference below 4.1 1)**
G	CallWaiting	Select an option whether to use Call Waiting or not Call Waiting is a function to send a receiver a signal when another call arrives while the receiver is already on the phone. The receiver can place Hold on the call which is already on the line and gets through the new call by pressing Hook Flash	TRUE or FALSE
H	MultiGroupCall	Select an option whether to use Multi Group Call Parking or not. Provisioning the corresponding phone number for whether to support Call Park feature to a different Park Group.	TRUE or FALSE
I	PhoneDesc	A description of the phone number	
J	CallingParty Presentation	Select and option to display the outgoing number for the Outbound call **Default** : Follow Default setup of Call Manager (Call Manager Cluster > Options > Calling Party Presentation) **Allowed**: Display my telephone number to the other **Restricted** : Not to display my phone to the other party	0 - Default 1 - Allowd 2 - Restricted
K	CallerIDDN	A force setup of the outgoing number (ex: general directory number)	
L	CallPriority	Assign the call priority order to a phone number	0 - 7
M	PartitionAccessList	Configure Partition Access List to the corresponding phone number	
N	Partition	Configure a Partition to be used for the corresponding phone number	
O	Emergency	When Emergency is selected, the call priority order is set to the highest	TRUE or FALSE

*** Reference 4.1 1) Each Unconditional, Busy, No Answer option of a phone number to be set and**

each detail of Forwarding Number, Partition Access List, Voice Mail is divided by using '/'.

When the Phone Number is set to No Answer, more details of setting options of No Answer Timeout and the timeout for No Answer are to be added

ex) TRUE/1000/Partition_01/TRUE/TRUE/10

Details Section	Options	Forwarding Number	Partition Access List	Voice Mail	No Answer Timeout Setting Options	No Answer Timeout
Unconditional	TRUE	1000	Partition_01	TRUE		
Busy	FALSE	1000	Partition_01	FALSE		
No Answer	TRUE	1000	Partition_01	TRUE	TRUE	10

4.2 digit map 시트

Excel Header	Field Name	Description	Input Format
A	DigitPattern	Enter a pattern that represents many phone numbers	**<[0-9#*]|[].TF>**
B	Desc	Enter a description	
C	Partition	Pattern 전화번호가 해당하는 Partition을 선택 Select a partition for the pattern of the phone number	

4.3 route pattern

Excel Header	Field Name	Description	Input Format
A	Pattern	Configure a Pattern to be routed for a particular number	
B	Desc	A Description of Routing Pattern	
C	Partition	Enter a Partition to assign a call limit on each class basis.	
D	CalledNumberRule	Translation Rule to be applied to the incoming for Number Translation of the Inbound call	
E	CallingNumberRule	Translation Rule to be applied to the outgoing for Number Translation of the Inbound call	
F	RoutingMode	Select a Routing mode of the Route List **Preference** : This is a mode to select a device in the priority order. When a Setup Message is transmitted the device with the highest priority first and no response is received in a certain time/ count, the setup messages is redirected to the device with the next highest priority.	0 - Preference 1 - Sequential

		Sequential : selecting one at a time regardless of the priority order	
G	Emergency	Select an option whether to set the highest priority order to the outbound call or not	TRUE or FALSE
H	OutSideDialTone	Select an option whether to provide a virtual Dialtone	TRUE or FALSE
I	BlockThisPattern	Select an option whether to block Routing on the Pattern.	TRUE or FALSE
J	Trunk/RoutingList	Select a Trunk device or Routing List with priority to be applied to the Routing Pattern	

4.4 translation pattern

Excel Header	Field Name	Description	Input Format
A	TranslationPattern	Register a Translation Pattern	
B	Desc	A Description of Translation Pattern	
C	Partition	Enter a Partition to assign a call limit on each class basis.	
D	PartitionAccessList	Configure Partition Access List which is to be used by the corresponding Pattern	
E	CalledNumber	An incoming number to be applied with Translation Rule for the number translation of an inbound call	
F	CallingNumber	The outgoing number to which Translation Rule is to be applied to the number translation for the inbound call.	
G	OutsideDialTone	Select an option whether to provide a virtual Dialtone	TRUE or FALSE
H	Emergency	When Emergency is selected, the call priority order is set to the highest	TRUE or FALSE
I	BlockThisPattern	Select an option whether to block Routing on the Pattern.	TRUE or FALSE

4.5 partition

Excel Header	Field Name	Description	Input Format
A	Name	A Partition name	
B	Desc	A Description of Partition	

4.6 partition access list

Excel Header	Field Name	Description	Input Format
A	Name	A name of Partition Access List	
B	Desc	A Description of Partition Access List	

C	PartitionList	The Partition list included in Partition Access List	**Reference below 4.4 1)**

*****Reference 4.4 1) Partition Access List can be assigned with more than one PartitionList. Each Partition can be divided by using '/'.

ex) first_class/second_class

4.7 translation rule

Excel Header	Field Name	Description	Input Format
A	Name	A name of Number Translation Rule	
B	Desc	A Description of Number Translation Rule	
C	TranslationRule	A registered list with Input Matched Pattern (An input number of the pattern to be used for Number Translation) and Substituted Pattern (A Pattern which is subject to Number Translation by an input number of the Pattern) for the Rule.	**Reference below 4.5 1)**

*** Reference 4.5 1) Translation Rule can be configured with more than one. Each Input Matched Pattern, Substituted Pattern can be divided by using ','. Each Rule can be divided by using '/'.**

ex) 0700,01/0800,02

Input Matched Pattern	Substituted Pattern
0700	01
0800	02

4.8 routing group

Excel Header	Field Name	Description	Input Format
A	Name	A name of Routing Group	
B	Desc	A Description of Routing Group	
C	TrunkDeviceList	A list of trunk devices which are not included the present Group	**Reference below 4.7 1)**

* **Reference 4.7 1) Routing Gorup can be configured with more than one TrunkDevice and Each Device can be devided by using '/'.**

ex) trunk_01/trunk_02

4.9 routing list

Excel Header	Field Name	Description	Input Format
A	Name	A name of Routing List	

B	Desc	A Description of Routing List	
C	RoutingGroupList	The Routing Group which are registered to the present Routing List	**Reference below 4.8 1)**

*** Reference 4.8 Routing list can be configured with more than one RoutingGroup. Each RoutingGroup, CallingNumber, CalledNumber is divided by using ‘,’ and each Rule is divided by using ‘/’.**

ex) a,1000,1001/b,1002,1003

4.10 aar group

Excel Header	Field Name	Description	Input Format
A	GroupName	A name of AAR Group	
B	Desc	A description of AAR	
C	Prefix	The digits added in the front of the actual dialing number for rerouting by AAR	
D	KeyNumber	Enter a key number for replacing a path and routing	
E	ConnectTimer	Set the timer for counting the actual time and AAR two stage dialing after 1st stage dialing. 2nd stage dialing is started after connect timer value is started	**0-10000**
F	TwoStageDialingMode	Connect two stage dialing for AAR	**TRUE/FALSE**

4.11 hunt group

Excel Header	Field Name	Description	Input Format
A	Number	A Hunt Group number	
B	Desc	A Description of the Hunt Group	
C	Mode	Set up a Hunt Mode **Preference** : Call in the priority order which has been set up **Simultaneous** : Call all at once regardless of the priority order **Random** : Call randomly regardless of the priority order	PREFERENCE SIMULTANEOUS RANDOM
D	Group Chain	Assign another Hunt Group to be connected with the Hunt Group	
E	Phone Number	A Phone Number list included in the Hunt Group	**Reference below 4.9 1)**
F	Partition	Select a Partition	

G	AddressPartition	The corresponding Partition List for the Phone Number	**Reference below 4.9.2)**

*** Reference 4.9.1) Hunt Group can be assigned with more than one Phone Number and each Phone Number can be divided by using '/'.**

ex) 1000/1001/1002

*** Reference 4.9.2) More than one Partition List for each Phone Number can be registered to Hunt Group. Each Partition can be divided by using '/'.**

ex) partition_01/partition_02 (If 1001 of the partition is not set up, each partition can be divided by a blank)

4.12 pickup group

Excel Header	Field Name	Description	Input Format
A	Name	A name of Pickup Group	
B	Number	Register a Pickup Number	**Reference below 4.10 1)**
C	Desc	A Description of the Pickup Group	
D	Partition	Select a Partition	

*** 4.10 1) Pickup Group can be assigned with more than one Phone Number. Each Phone Number can be divided by using '/'.**

ex) 1000/1001/1002

4.13 park address pool

Excel Header	Field Name	Description	Input Format
A	Number	Register a Park Number. Either one phone number can be assigned or many. By using '[ ]' , many phone numbers can be assigned. For instance, when 800[09] is used, 10 Park Numbers can be assigned from 8000~8009.	
B	Desc	A Description Park Address Pool	
C	Partition	Select a Partition	

4.13 park group

Excel Header	Field Name	Description	Input Format
A	Name	A Park Group name	

B	Desc	A Description of the Park Group	

4.15 music announce

Excel Header	Field Name	Description	Input Format
A	ServiceEnable	Select an option whether to enable or disable Music & Announcement service	TRUE or FALSE
B	Codec	Select a type of codec matches with VoIPSetting	pcm - G.711 ulaw alaw - G.711 alaw g729 - G.729
C	DialTonePolicy	Set up Dial-tone policy	LOCAL REMOTE LOCAL_REMOTE
D	FramePerPacket	Set a number of frame per packet	1 – 5 for G.711 setting 1 – 8 for G.729 setting

4.16 auto attendant sheet

Excel Header	Field Name	Description	Input Format
A	ServiceEnable	Select an option to enable or disable Auto Attendant service	TRUE or FALSE
B	AttendantNumber	Set up the phone numbers for Auto Attendant service	
C	OperatorNumber	Set up an extension in Auto Attendant service for the incoming call to be directed to	
D	Codec	Select a type of audio codec to be used to Auto Attendant service	pcm - G.711 ulaw alaw - G.711 alaw g729 - G.729
E	FramePerPacket	Set a number of frame per packet	1 – 5 for G.711 setting 1 – 8 for G.729 setting
F	Partition	Select a Partition	

4.17 voice mail

Excel Header	Field Name	Description	Input Format
A	ServiceEnable	Select an option to enable or disable Voice Mail service	TRUE or FALSE

B	VoiceMailNumber	Set a phone number for Voice Mail service	
C	Codec	Select a type of audio codec to be used to Voice Mail Service	pcm - G.711 ulaw alaw - G.711 alaw g729 - G.729
D	FramePerPacket	Set a number frame per packet	1 – 5 for G.711 setting 1 – 8 for G.729 setting
E	Partition	Select a Partition	

4.18 ivr service Sheet

Excel Header	Field Name	Description	Input Format
A	Name	A scenario name	
B	Desc	A description of IVR Service	
C	Number	ISet a number of IVR Service	
D	Codec	Set a type of Audio Codec	pcm - G.711 ulaw alaw - G.711 alaw g729 - G.729
E	ScenarioFolder	Set a location of IVR Service to be saved	
F	EnableService	Select an option whether to enable or disable IVR Service	TRUE or FALSE
G	FramePerPacket	Set a number of frames per packet	1 – 5 for G.711 setting 1 – 8 for G.729 setting
H	Partition	Select a Partition	

4.19 service code

Excel Header	Field Name	Description	Input Format
A	Park	A code setup of a call to Park	<*#><0-9>
B	Pickup	A code setup of a call to Pickup	## or **
C	DirectPickup	A code setup of a call, which doe not come from the same Group, to Pickup	<*#><0-9>
D	CWActivation	A code setup of Call Waiting	<*#><0-9><0-9>
E	CWDeactivation	A code setup to deactivate Call Waiting	
F	CFwdAllRegister	A code setup to register a number of unconditional forwarding	
G	CFwdBusyRegister	A code setup to register Call Forwarding Busy	

H	CFwdNoAnswerRegister	A code setup to register a number of Call Forwarding for No Answer
I	CFwdCancel	A code setup to cancel Call Forwarding
J	CFwdAllActivation	A code setup to Unconditional Call Forwarding
K	CFwdAllDeactivation	A code setup to deactivate Unconditional Call Forwarding
L	CFwdBusyActivation	A code setup for Call Fowarding Busy
M	CFwdBusyDeactivation	A setup to release the code for Call Forwarding Busy
N	CFwdNoAnswerActivation	A code setup of Call Forwarding for No Answer
O	CFwdNoAnswerDeactivation	A setup to release the code of Call Forwarding for No Answer
P	CCBSRegistration	CCBS[8] Setup
Q	CCBSCancel	Cancel CCBS
R	AbsenceActivation	A code setup for Absence
S	DoNotDisturbActivation	A code setup to reject all the incoming calls
T	CallRejectDeactivation	A setup to release the code of rejecting all the incoming calls

[8] **CCBS**(Termination of Calls to Busy Subscriber) - If A makes a call to B and B is busy on the line, CCBS allows A to connect B after B is finished with the call without calling back

5. Conference Management Field Information\

5.1 media class

Excel Header	Field Name	Description	Input Format
A	MediaClass	A name of Media Class	
B	Desc	A description of Media Class	
C	PartyInfo	Set up Audio Party, Video Party	**Refer to 5.1 1)**

*** Reference 5.1 1) Audio party and video party can be set in the media class. Use '&' between each attribute**

ex) audio.codec=5&video.codec=2&video.target.rate=64&video.frame.rate=30&video.picture.size=1

Category	Settings
audio.codec	0 - G711A 1 - G723 2 - G726 3 - G729 4 - G711U 5 - G722 6 - G722.1 7 - G728
video.codec	1 - MPEG4 2 - H263 3 - H264
video.target.rate	[64 - 2048] 6의 배수
video.frame.rate	9 - 30
video.picture.size	video codec = MPEG4 0 – 320*240 1 – 640*240 2 – 640*480 video codec = H.263 0 - QCIF 1 - CIF 2 - 4CF video codec = H.264 0 – 320*240 1 – 640*240 2 – 640*480

5.2 conference sheet

Excel Header	Field Name	Description	Input Format
A	Name	A Conference name	
B	Desc	A Description of the Conference	
C	Number	Conference number	
D	Type	Select a type of Conference	AD-HOC DIAL-OUT MEET-ME AD-HOC DIAL-OUT
E	Partition	Select a type Partition	
F	MediaType	Select a type of media for Ad-Hoc Default	AUDIO AUDIO-VIDEO
G	AudioCodec	Select a type of Audio Codec	G.711A G.723 G.726 G.729 G.711U
H	CloseOnChairOut	This is an option to select whether the conference call is to be terminated when Chair Man, who invites the Conference Call, terminates a call when he/she ends. When the option is selected, Conference Call ends when Chair Man terminates his/her call.	TRUE or FALSE
I	Participants	A list of the phone user's telephone numbers registered to a conference room **Start Auth** - An authority to start a conference for Dial-out conference **Type** - Internal or External	**Reference below 5.1 1)**
J	VideoCodec	Select a type of Video Codec	MPEG4 H.263 H.264
K	VideoTargetRate	Set a Video Target Rate	[64 - 2048] 6 multiples
L	VideoFrameRate	Set a Video Frame Rate	9 - 30
M	VideoPictureSize	Set a Video Picture Size	0 – 320x240 1 – 640x240 2 – 640x480
N	VideoEndTargetRate	Set a Video End Target Rate	[64 – 4096] 6의 배수
O	VideoEndFrameRate	Set a Video End Frame Rate	9 - 30

P	VideoEndPictureSize	Set a Video End Picture Size	0 – 320x240 1 – 640x240 2 – 640x480
Q	VideoDynCtrl	Select an option whether to provide Video Dynamic Control or not	TRUE or FALSE
R	VideoType	Select a Video Type	1 – Continuous Presence 2 - Video Switching
S	VideoConfLayout	Select a type of Video Conference LayOut	1 - Layout1 2 - Layout2 3 - Layout3 4 - Layout4
T	SecretRoom	Select an option whether to open a conference room to the public or not	TRUE or FALSE
U	SecretPasswd	Set a password for the closed room	
V	MaxParty	Set the maximum number of participants	1 - 16
W	AddressPartition	Partition List for each participant's phone number	**Reference below 5.1 2)**
X	LayoutMode	Conference Layout Mode Setup	1 - Auto (Symmetric) 2 - Auto (Asymmetric) 3 - MANUAL
Y	LayoutType	Layout Shape Setup	S4, S9, S16, S25, S36, S64, A2_A, A2_B, A2_C, A3_A, A3_B, A3_C, A4_A, A4_B, A5_A, A5_B, A6_A, A6_B, A7_A, A8_A, A8_B, A9_A, A9_B, A12_A, A16_A, A20_A, A28_A, A48_A
Z	ArrangeMode	Arrange mode setup	1 - Dynamic 2 - Fixed 3 - Fixed (Center Reserved)
AA	FloorFullScreenSymetric	Select an option whether to provide the full screen of the participant with the floor in the Symmetric shape of Layout	TRUE or FALSE
AB	FloorFullScreenAsymmetric	Select an option whether to provide the full screen of the participant with the floor in the Asymmetric shape of Layout	TRUE or FALSE
AC	MicOffMute	Whether to turn on the microphone for the participants who are not	TRUE or FALSE

		diplayed	
AD	BorderEnable	An option for color section of the Border line.	TRUE or FALSE
AE	FloorEnable	An option for color section of the Floor line	TRUE or FALSE
AF	BackgroundEnable	An option for color section of the Background line	TRUE or FALSE
AG	BorderColor.Y	Border line Color Y	[0-255]
AH	BorderColor.CR	Border line Color Cr	
AI	BorderColor.CB	Border line Color Cb	
AJ	FloorColor.Y	Floor line Color Y	
AK	FloorColor.CR	Floor line Color Cr	
AL	FloorColor.CB	Floor line Color Cb	
AM	BackgroundColor.Y	Background line Color Y	
AN	BackgroundColor.CR	Background line Color Cr	
AO	BackgroundColor.CB	Background line Color Cb	
AP	DisplayNameEnable	An option whether to display a participant's name	TRUE or FALSE
AQ	DisplayNameLayout	A location of the participant's name to be displayed	0 – Top Left 1 – Top Center 2 – Top Right 4 – Bottom Left 5 – Bottom Center 6 – Bottom Right
AR	FloorAssignMode	A setup to request for the floor	1 – After release 2 – On request
AS	VirtualAudienceInfo	An information on a list of Virtual Audience	**Reference below 5.1 3)**

*** Reference 5.1.1) Many Participants can be registered to Conference. Each participant can be divided by '/' as it is shown in the following input format.**

- **Format**

Type(Internal or External), PhoneNumber, UserClass(1~5), MediaType(1~3), Position, DisplayName

- **ex) Internal,1000,1,3,-1,User1/Internal,1001,4,3,-1,User2/External,2000,4,3,-1,User3**

***** Reference 5.1.2) Register a Partition for each telephone number. Each Partition can be divided by '/'.

- **ex) partition_01/partition_02/**

*** Reference 5.1.3)** Many Virtual Audiences can be registered. Each Virtual Audience can be divided by '/' as it is shown in the following input format

- **Format**

Broadcasting Server Name, Media Type(1~2), Audio Port, Video Port

- ex) BroadcastingServer_01,1,8000,/VideoServer,2,8001,8002

6. Messenger Management Field Information

6.1 msg config

Excel Header	Field Name	Description	Input Format
A	MaxStorageCount	The maximum number of the messenger notes to be saved	default : 200
B	MaxStorageDays	The maximum time to save the messenger note	default : 30

6.2 msg user

Excel Header	Field Name	Description	Input Format
A	Name	A name of the outside registered user who is allowed share the messenger	
B	Desc	A description of outside user registered to the messenger	
C	Office Phone	A company telephone number of outside registered user who is allowed share the messenger	
D	Mobile	A cell phone number of outside registered user who is allowed share the messenger	
E	E-Mail	An e-mail address of the outside registered user who is allowed share the messenger	
F	Path	A group of outside registered user who is allowed share the messenger	
G	DisplayOrder	An order for displaying the outside registered user who is allowed share the messenger	

6.3 msg group

Excel Header	Field Name	Description	Input Format
A	Name	A name of the outside group who is registered and allowed to share messenger	
B	Desc	A description of the outside group who is registered and allowed to share messenger	
C	Path	A location of the outside group who is registered and allowed to share messenger	
D	DisplayOrder	An order for displaying the outside group who is registered and allowed to share the messenger	

6.4 ps preference

Excel Header	Field Name	Description	Input Format
A	Name	A name for Presence Profile	
B	Desc	A description of Presence Profile	
C	Preference	Presence Server List which has been set in the Profile	**Refer to 6.4 1)**

*** Reference 6.4 1) presence server preference can be set in ps preference. Use '/' to devide each detail of presence server.**

ex) PS2000_1/PS2000_2

6.5 speed button

Excel Header	Field Name	Description	Input Format
A	Name	A name of Speed Button Profile	
B	Desc	A description of Speed Button Profile	
C	SpeedButton	Enter Display name, number type, phone number of each Speed Button (1~25)	**Refer to 6.5 1)** **Home phone.type=0** **Office phone.type=1** **Mobile phone.type=2** **Extension phone.type=3** **Trunk phone.type=4**

*** Reference 6.5 1) 25 speed buttons can be set Speed button. Each speed button can be divided by '/', use '&' for dividing each attribute.**

ex)btn.num=1&display.name=testuser&phone.type=1&phone.num=1010/btn.num=2&display.name =testuser&phone.type=2&phone.num=1008

6.6 ps group

Excel Header	Field Name	Description	Input Format
A	Name	A name of Presence Group	
B	Desc	A description of Presence Group	
C	PSServerPreference	Preference information of Presence Server	
D	DirectoryBoundary	Configure the user's organization for provisioning	**Refer to 6.6 1)**

		each use.	
E	MonitoringBoundary	사용자간의 접속 상태를 모니터 하기 위한 접근 권한을 부여하기 위하여 사용자 조직을 설정. Directory Boundary에서 선택한 조직의 하위조직만을 설정할 수 있음	**참조 6.6 1)**
F	SpeedButton	Presence Group에 포함된 Speed Button Profile 리스트	**참조 6.6 2)**

*** Reference 6.6 1) DirectoryBoundary, MonitoringBoundary can be set in ps group Entry is in DN Path format.**

ex) ou=users,dc=addpac,dc=com or ou=1층, ou=users,dc=addpac,dc=com

*** Reference 6.6 2) More than one speed button can be configured in ps group. Use '/' to divide each detail.**

ex) 1st floor speed button /2nd speed button/3rd floor speed button

7. Event & Monitoring Field Information

7.1 event config

Excel Header	Field Name	Description	Input Format
A	DeviceIp	A IP address in where the event manager is located	
B	IsLogging	Choose an option whether to save a log	TRUE or FALSE
C	CallLogTimeSpan	A time length (minutes) for saving a call log	1 - 1440
D	CallLogFileSize	A file size (MB) for saving the call log	1 - 1000
E	FaultLogTimeSpan	A time length (minutes) to save a fault log	1 - 1440
F	FaultLogFileSize	A file size(MB) for saving a fault log	1 - 1000
G	DebugTimeSpan	A time length for saving a debug log	1 - 1440
H	DebugLogFileSize	A file size for saving a debug log	1 - 1000
I	sscp	Choose a level from the corresponding category	**Reference below 7.1 1)**
J	ldap	Choose a level from the corresponding category	
K	medi	Choose a level from the corresponding category	
L	xapp	Choose a level from the corresponding category	
M	apsr	Choose a level from the corresponding category	
N	cctl	Choose a level from the corresponding category	
O	comn	Choose a level from the corresponding category	
P	prot	Choose a level from the corresponding category	
Q	sdpm	Choose a level from the corresponding category	
R	srvc	Choose a level from the corresponding category	
S	test	Choose a level from the corresponding category	
T	tefr	Choose a level from the corresponding category	
U	user	Choose a level from the corresponding category	
V	util	Choose a level from the corresponding category	
W	xmlp	Choose a level from the corresponding category	
X	xivr	Choose a level from the corresponding category	
Y	clog	Choose a level from the corresponding category	
Z	pres	Choose a level from the corresponding category	
AA	xcdr	Choose a level from the corresponding category	

*** Reference 7.1 1)** Event configuration allows a label setting of each category for event monitoring

Label	Description
128	Emergency

131	Error
132	Warning
134	Informational
135	Debug
1351	Fine
1352	Finest

7.2 call pattern

Excel Header	Field Name	Description	Input Format
A	Name	A name of a call pattern	
B	Desc	A description of a call pattern	
C	Pattern	Choose a pattern	**Reference below 7.2 1)**

*** Reference 7.2 1) Call pattern can be configured with more than one pattern and each pattern can be divided by using '/'.**

ex) 011/016/00700/001/002

Chapter 6. Appendix

IPNext PBX Clustering (Redundancy)

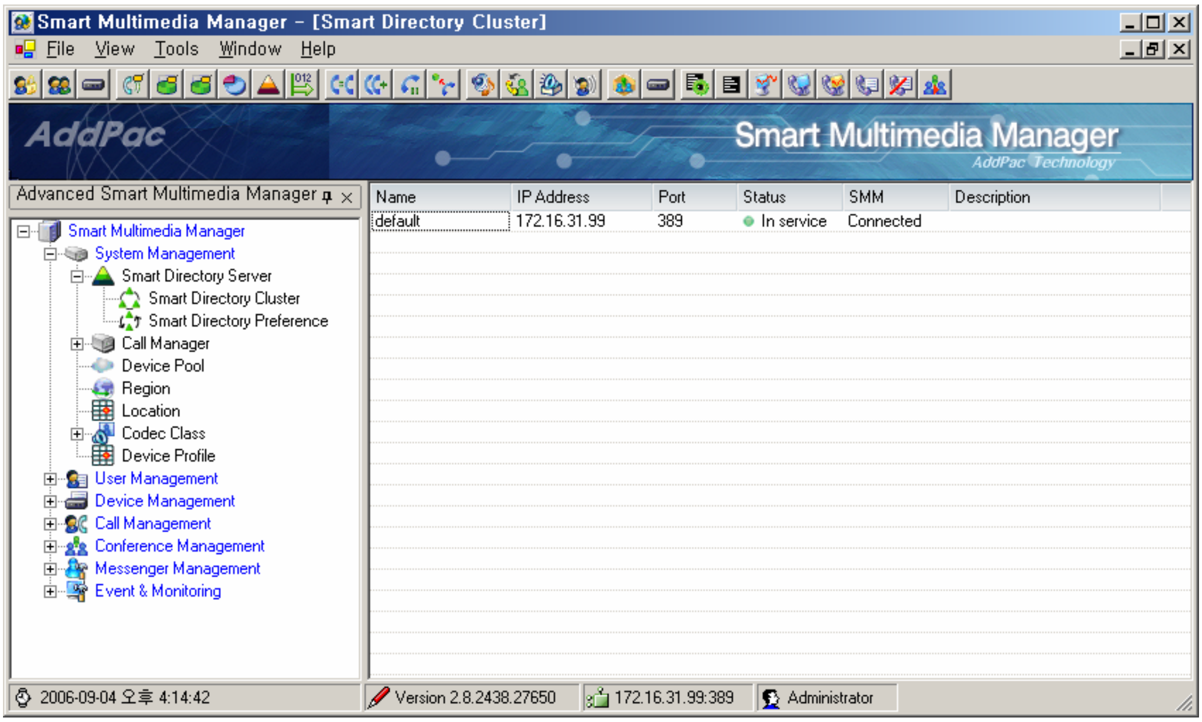
IPNext PBX is to be configured and operated in redundancy for its data and system stability. In this chapter, the ways to configure and manage redundancy of IPNext PBX is described in details.

Adding IPNext PBX

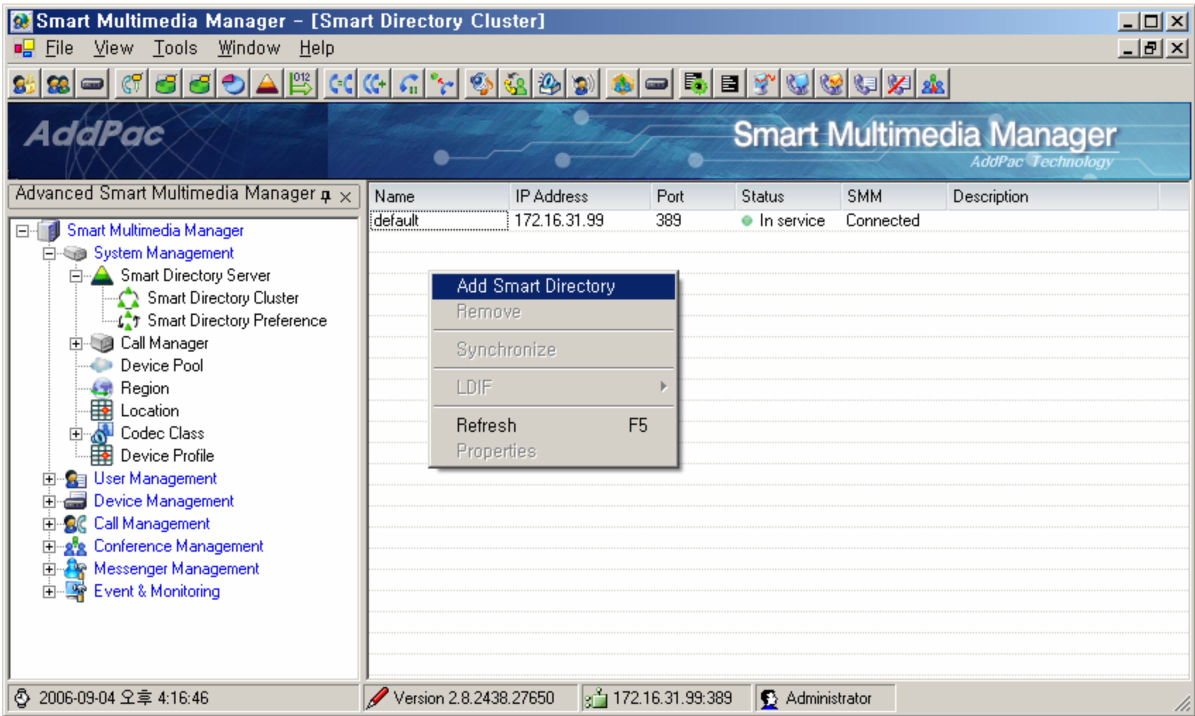
For further detailed description of registering another IPNext PBX to the standalone type of IP PBX in the beginning for redundancy configuration, you may refer to the page 34 of Smart directory sever in SMM Operation Guide Manual.

The following are the description of the procedures to configure the IPNext PBX as to be added and registered in redundancy:

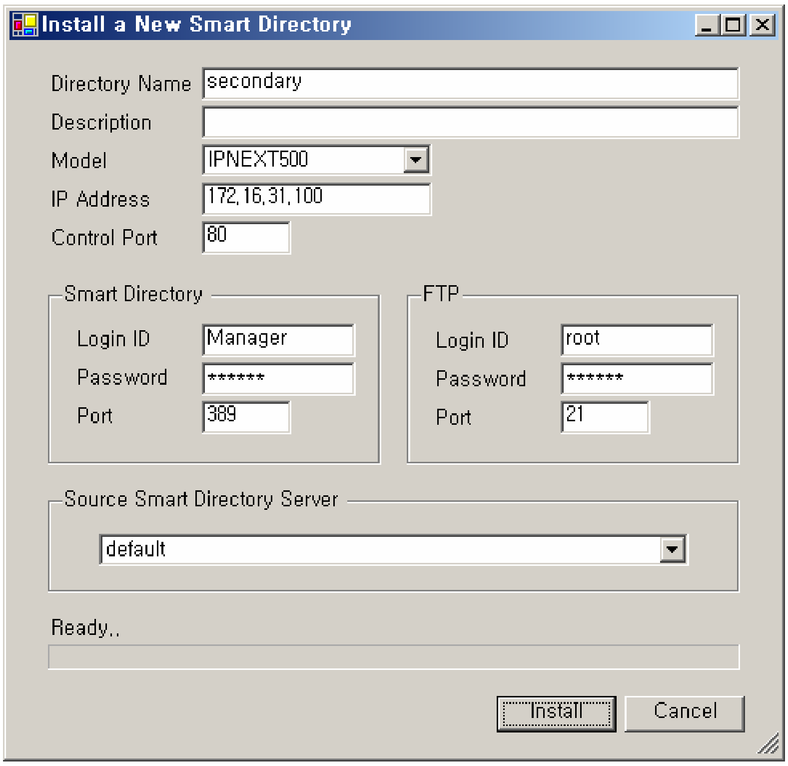
1) Check the basic configuration of a new IPNext PBX to be added and set up through its console
- IP address registration and domain information registration
- Check whether to enable FTP, HTTP, LDAP services
- Check the registration of Call Manager Interface
- HDD mount settings and others
2) Access to the IPNext PBX in operation by performing SMM.
3) Select a menu of Smart Directory Cluster(System Management > Smart Directory Server > Smart Directory Cluster) in the advanced mode



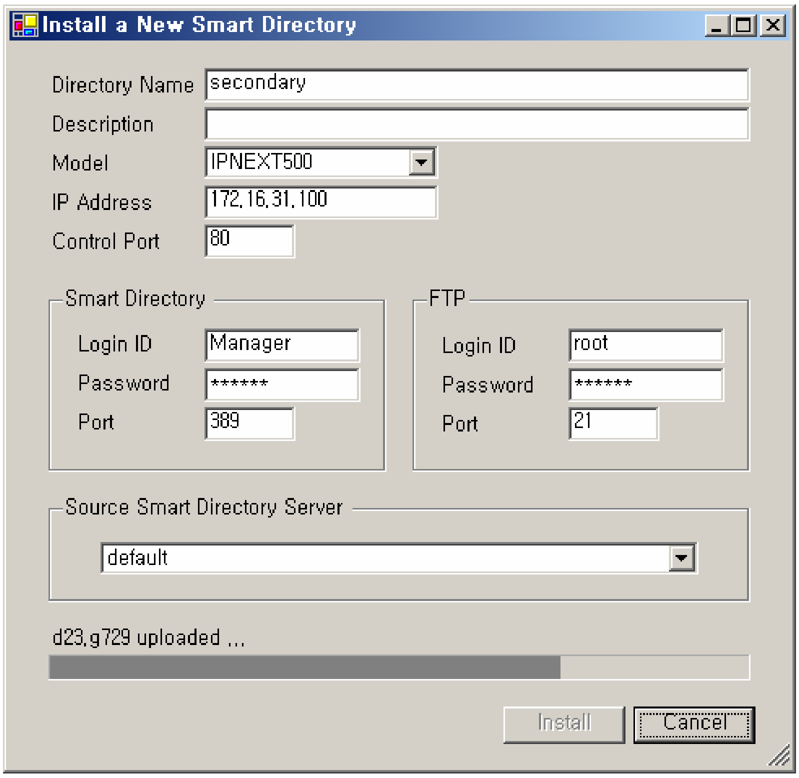
4) Perform Add Smart Directory Server from a pop-up menu of Smart Directory ClusterScreen.



5) after entering the Properties of the new IPNext PBX to be registered, click 'Install' button

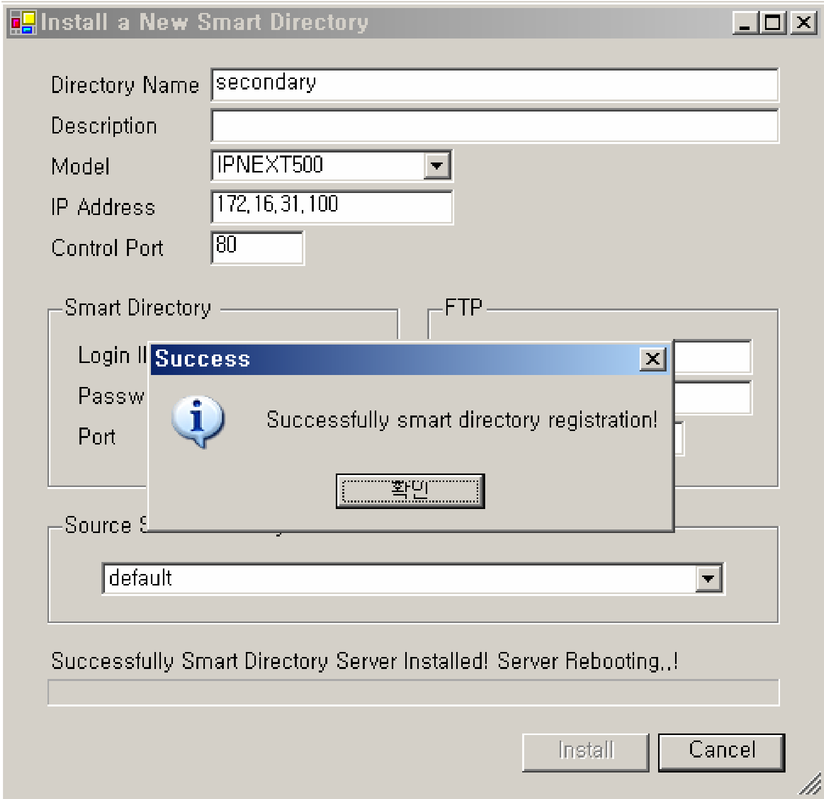


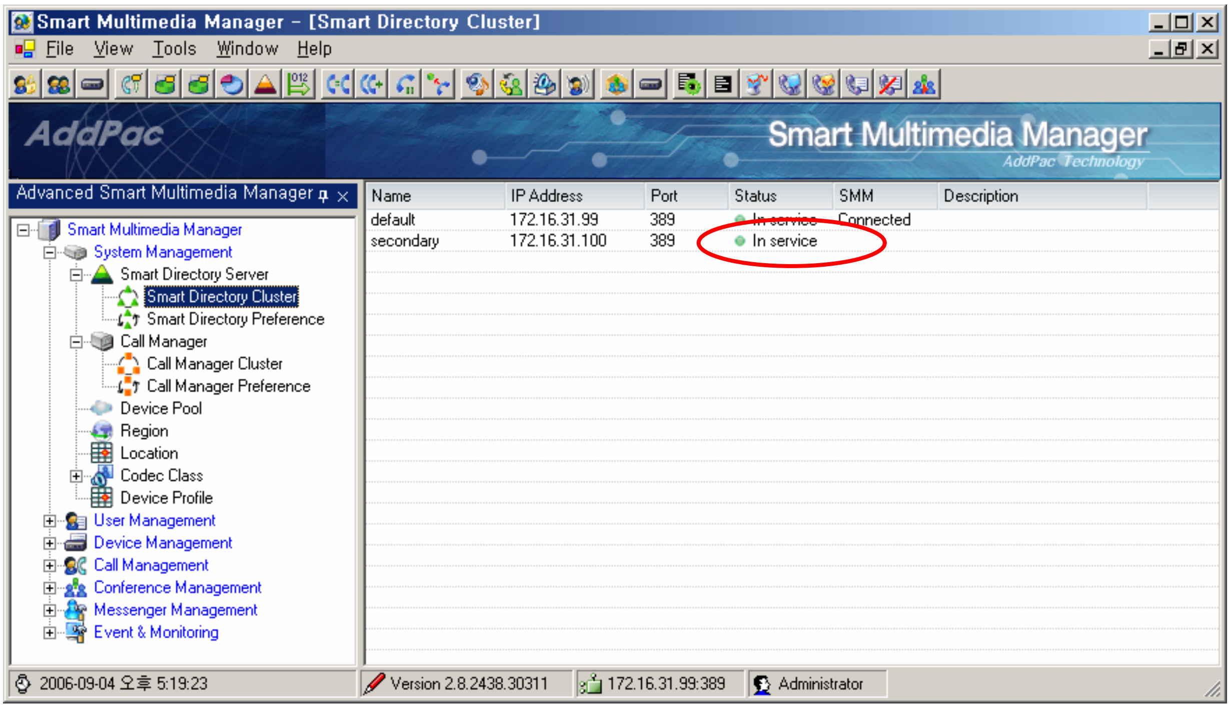
6) When 'Install' button is clicked, the data installation and redundancy configuration is processed (it takes about 3~5 minutes).



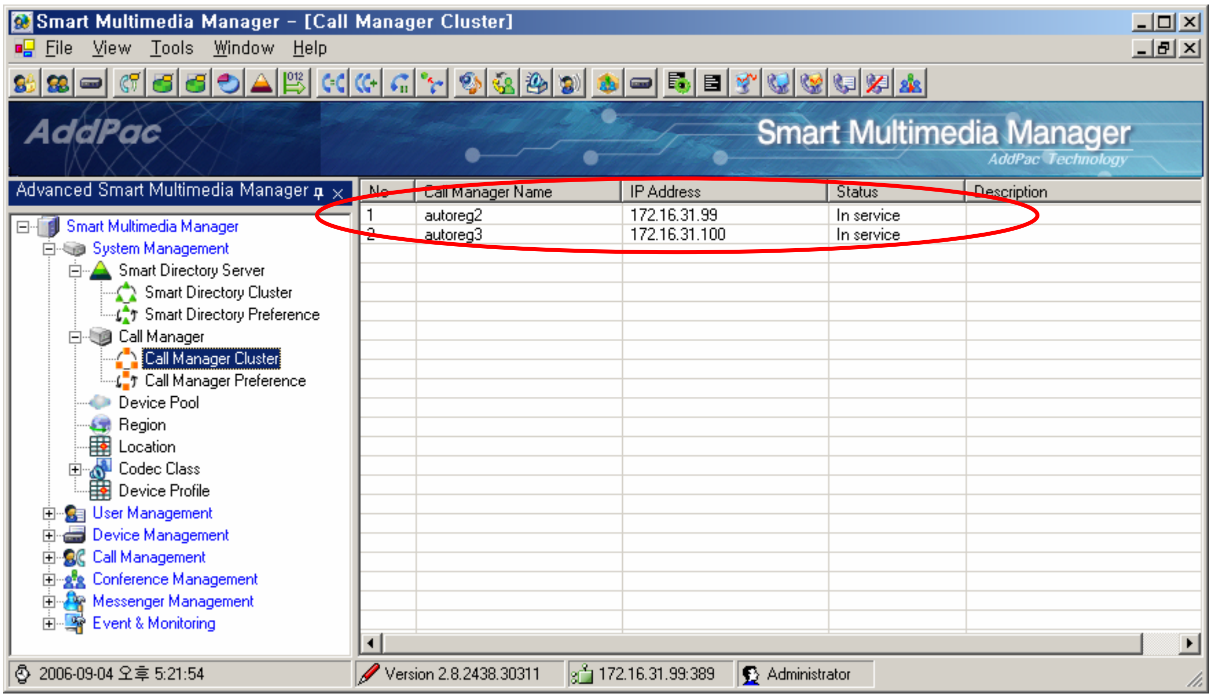
7) When the setting is completed, rebooting of the new IPNext PBX is processed. When it is

comp-leted, whether the IPNext PBX is in service from the cluster screen of Smart Directory.



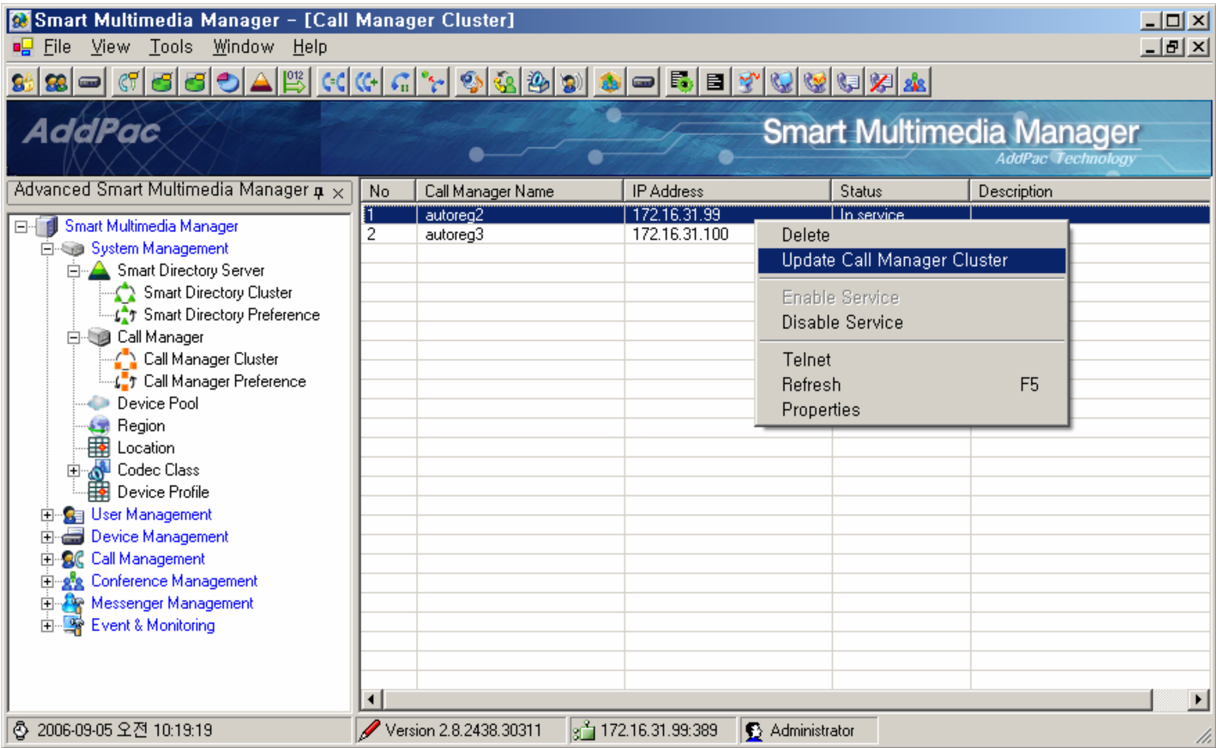


8) Now check whether the new IPNext PBX is registered Call Manager Cluster(System Management > Call Manager > Call Manager Cluster).



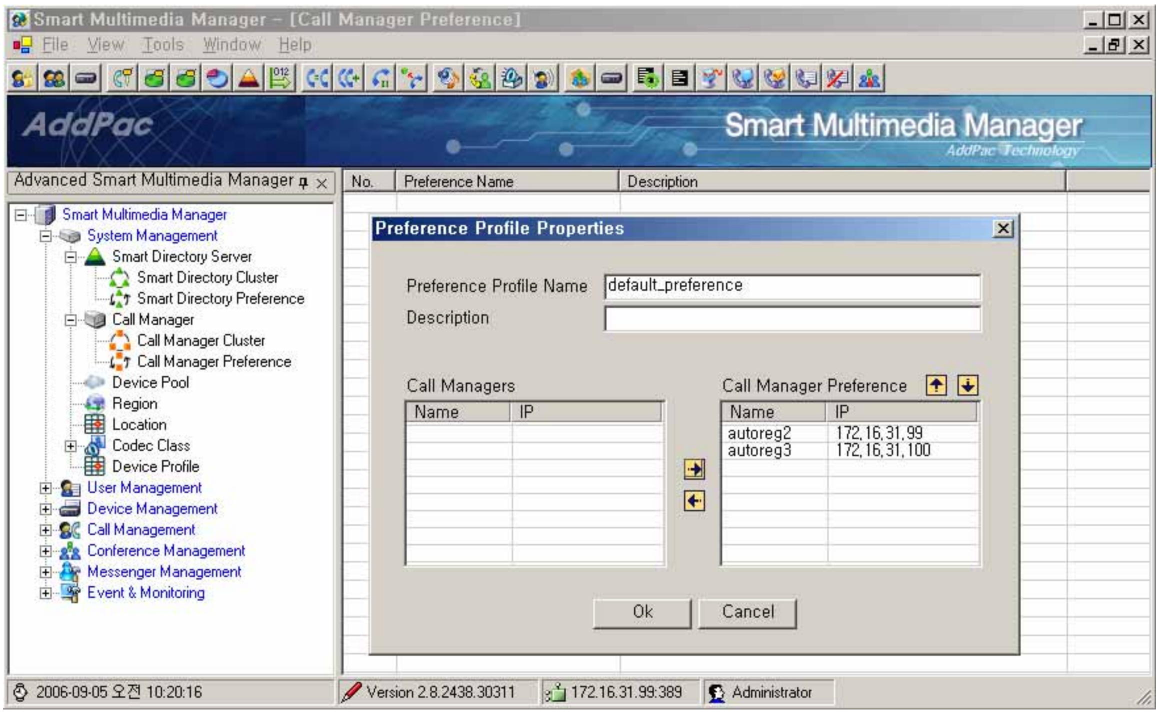
9) To apply the existing IPNext PBX in operation, the following procedures can be conducted:

- Select the existing IPNext PBX from Call Manager ClusterScreen
- Use the pop-up menu to perform Update Call Manager Cluster (supported from v2.8)
  - If there is no such function, please reboot the existing IPNext PBX

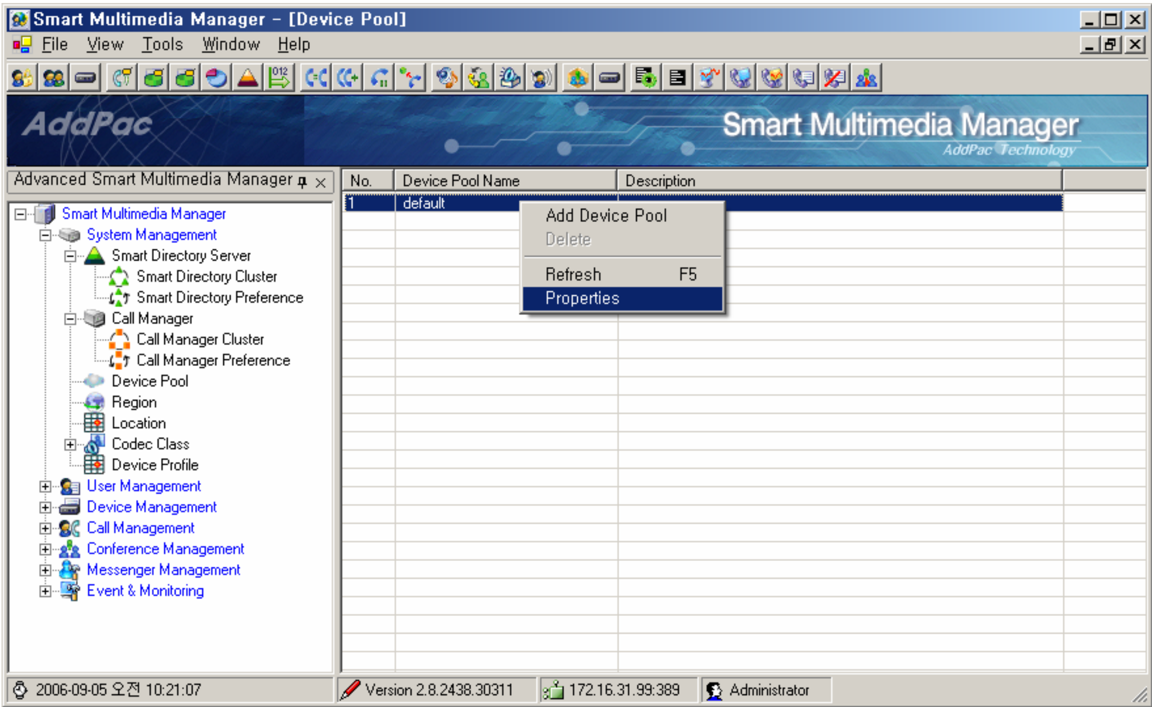


10) Configure Call Manager Preference from the Device Pool which is to be applied to the terminal by make a new registration of Call Manager Preference

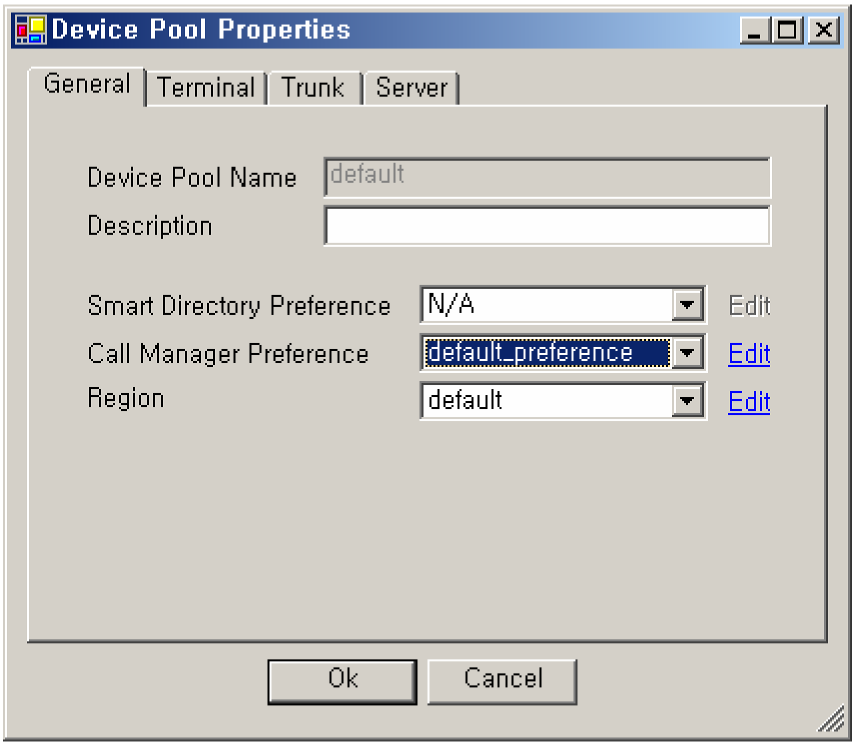
- A new Call Manager Preference registration



- Perform Properties of default Device Pool



- Set up Call Manager Preference in the Device Pool Properties



11) At this time, check whether the Call Manager Preference is applied to the terminal (if it is not applied, each terminal can be reset through SMM to apply the redundancy configuration).

- A way to check : Through OSD(sscp setup) of the terminal or console(show running-config > sscp dynamic CM list)Screen, the information of Call Manager can be checked.

12) Check the call is in normal process through the new registration of IPNext PBX.

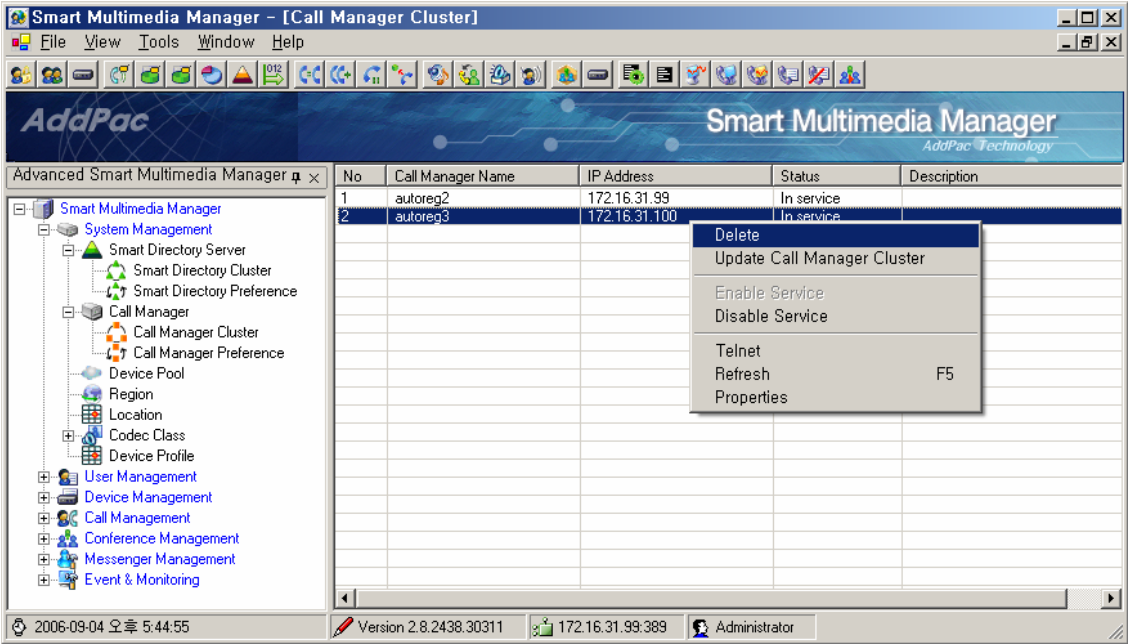
13) The data backup (Excel or LDIF) of the current configuration can be performed.

14) Now the configuration of the new IP Next PBX is completed.

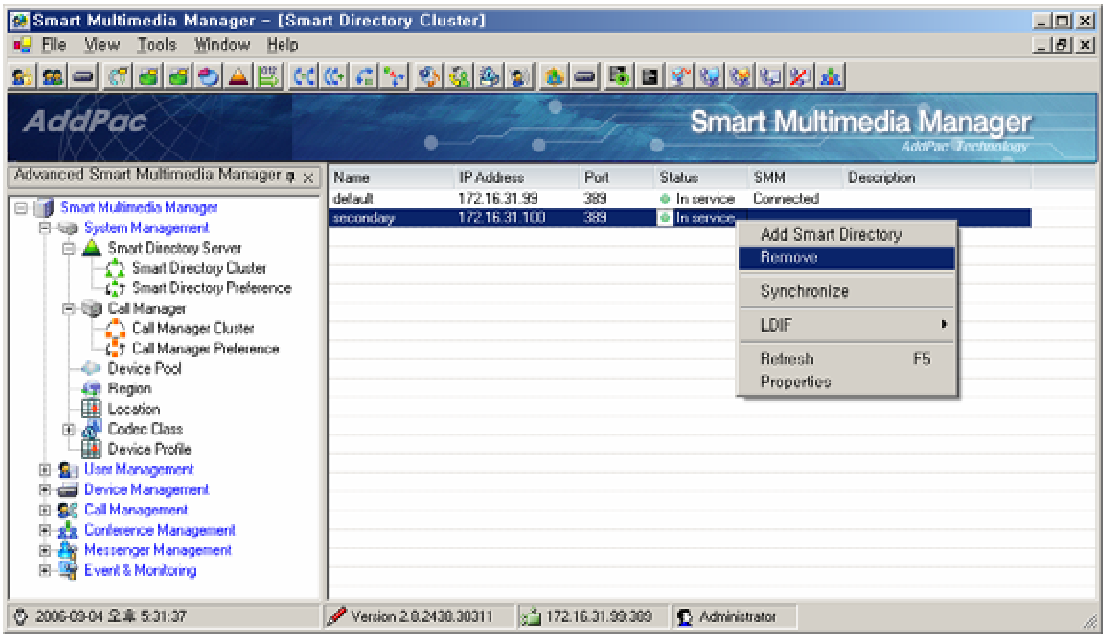
Deleting IPNext PBX

This section is to describe a procedure of deleting a specific IPNext PBX in the redundant operation for an unavoidable circumstance

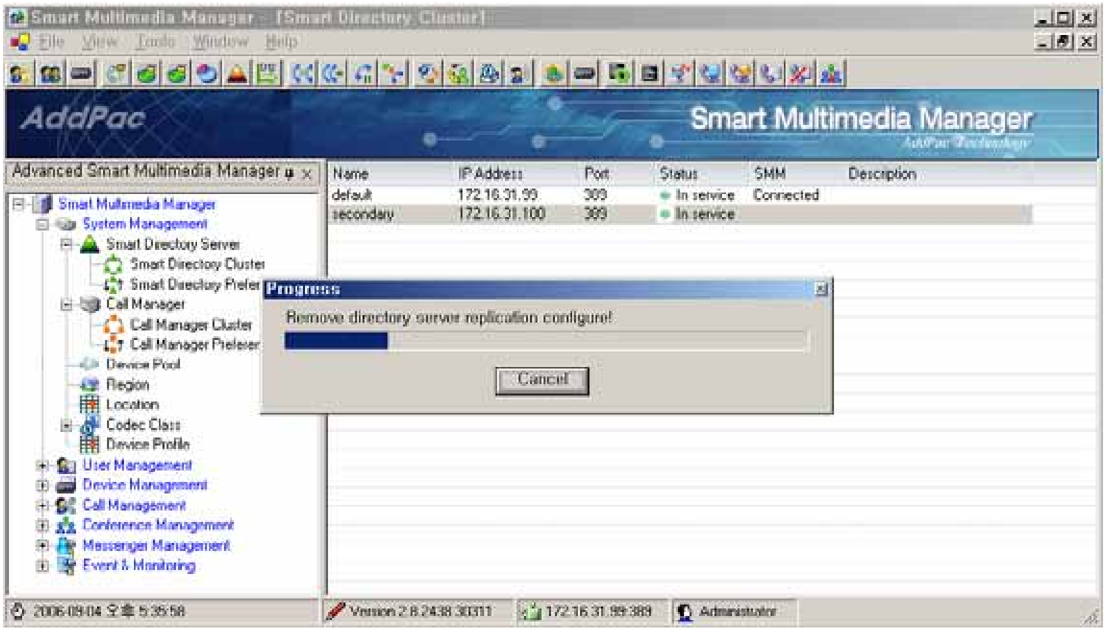
1) Select the Call Manager to be deleted by performing Call Manager Cluster(System Management > Call Manager > Call Manager Cluster) then select the Call Manager and perform 'Delete' of the pop menu to delete.



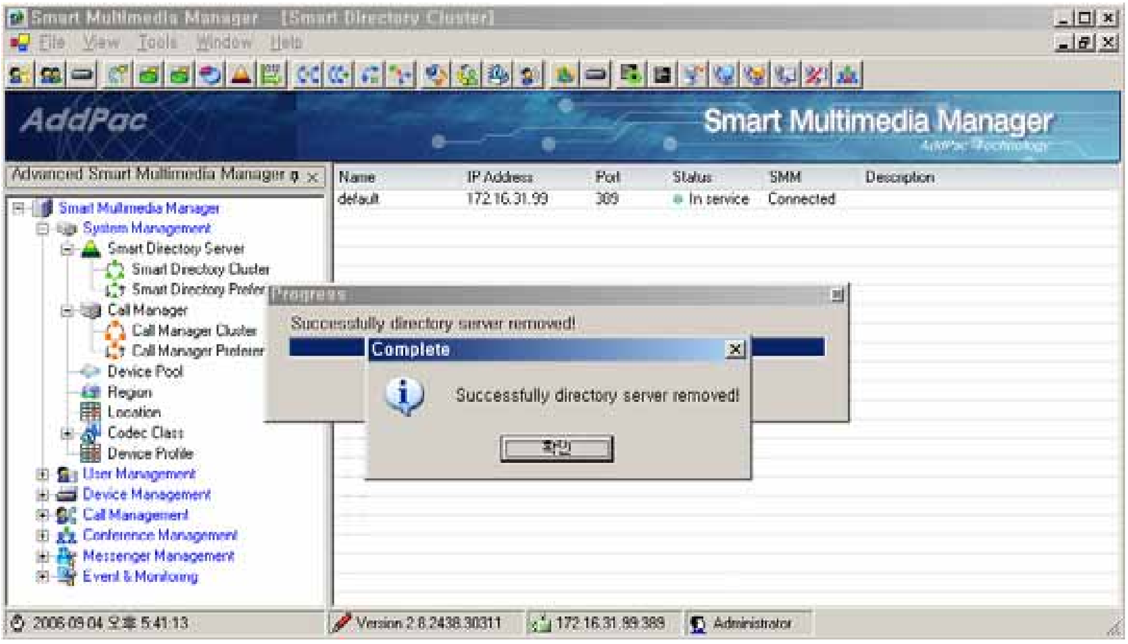
2) Select the Directory Server to be deleted from the screen of Smart Directory Cluster(System Management > Smart Directory Server > Smart Directory Cluster) then perform 'Remove' of the pop-up menu to delete the Directory Server.



3) Deleting the Directory Server is being processed.



4) When the Directory Server is deleted, the IPNext PBX operates as a standalone type and it is also deleted from Smart Directory Cluster.



5) The redundant IPNext PBX is deleted.

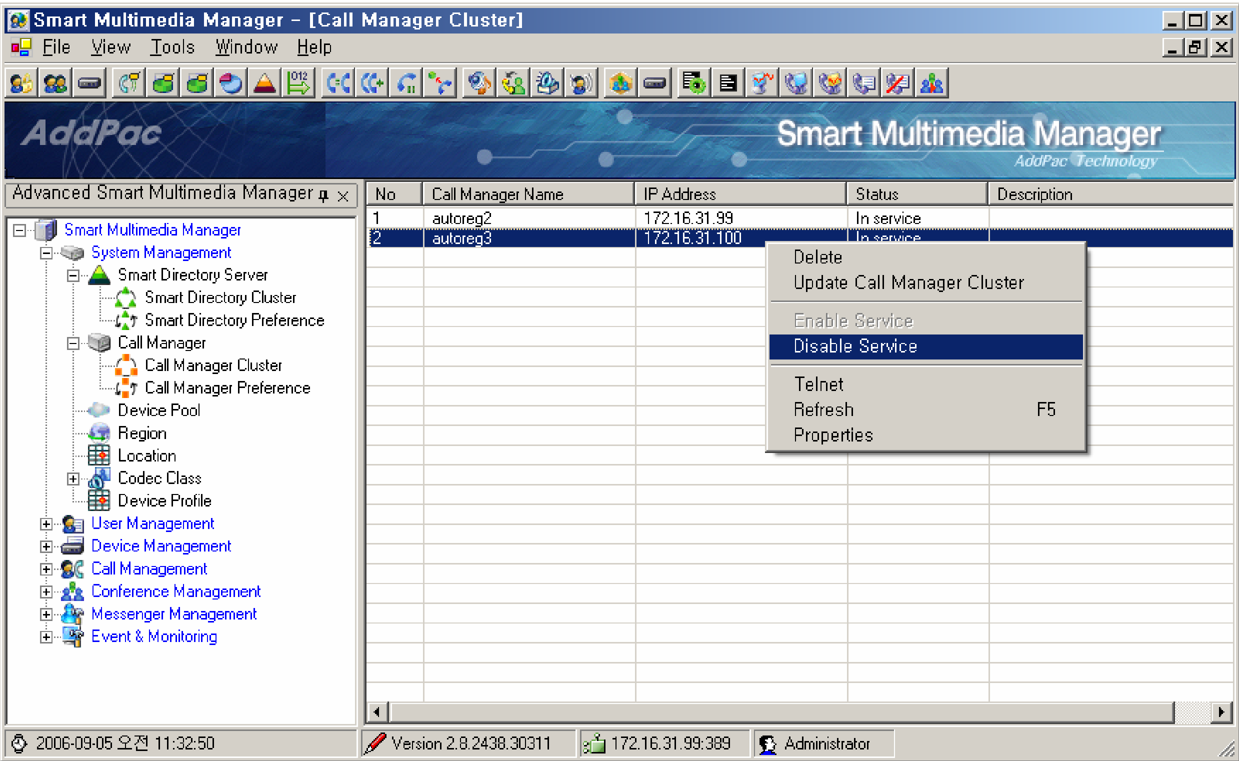
Failure Recovery of the IPNext PBX with Redundancy

A procedure of the failure recovery is to be performed for the IPNext PBX system with redundancy in case it is unable to provide service due to a failure. In this case, the recovery work is to be conducted when the failure is perceived that IPNext PBX can provide a normal service by initialization.

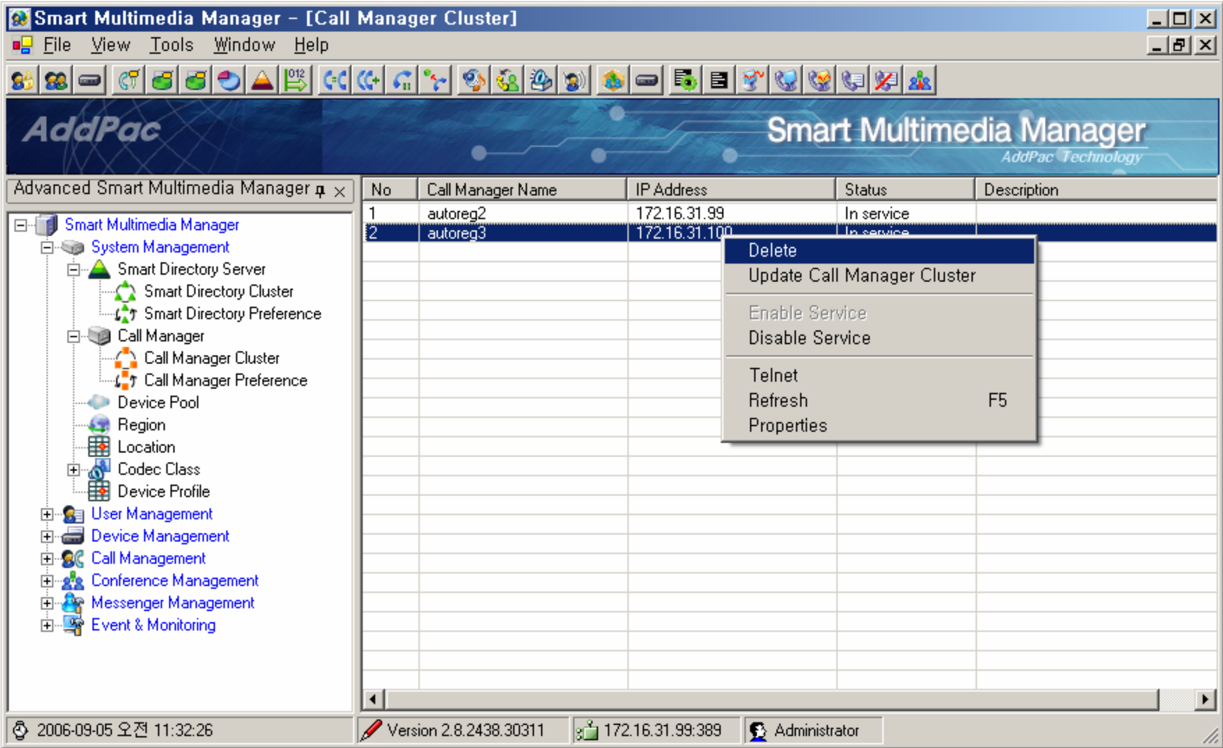
The recovery process can be described in the following redundant configuration (with assumption that 2 IPNext PBX is configured with CM A, CM B) as the service can not be used due to a failure of CM B

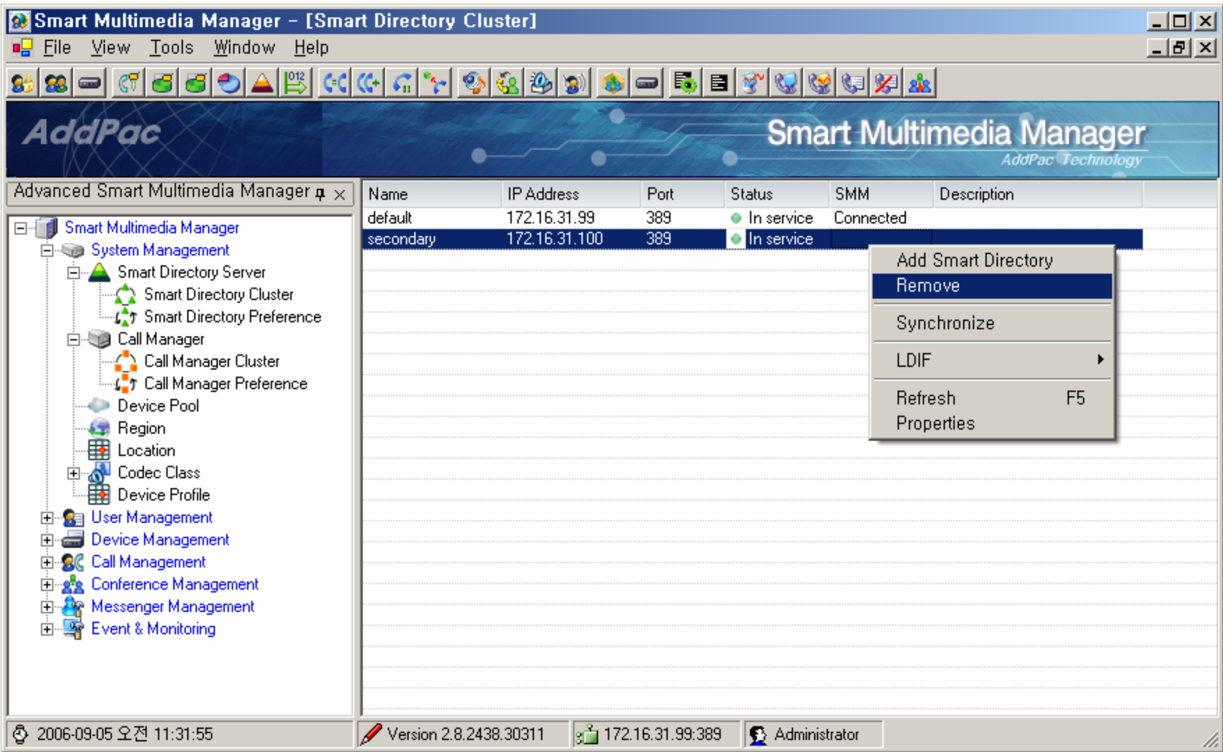
1) Access to CM A by using SMM

2) Disable CM B by selecting 'Disable Service' from Call Manager Cluster(System Management > Call Manager > Call Manager Cluster). In here, the terminals which were receiving the service from CM B can take the service from CM A.



3) Delete the corresponding Call Manager CM B from Call Manager Cluster with the redundant configuration then delete CM B with the failure from Smart Directory Cluster (CM B is to be set to a standalone)





4) The initial procedure for CM B is to be performed. This procedure is same as Adding the IPNext PBX after on the 오류! 참조 원본을 찾을 수 없습니다. of the page **254**. Following this procedure, CM B can be configured with redundancy.

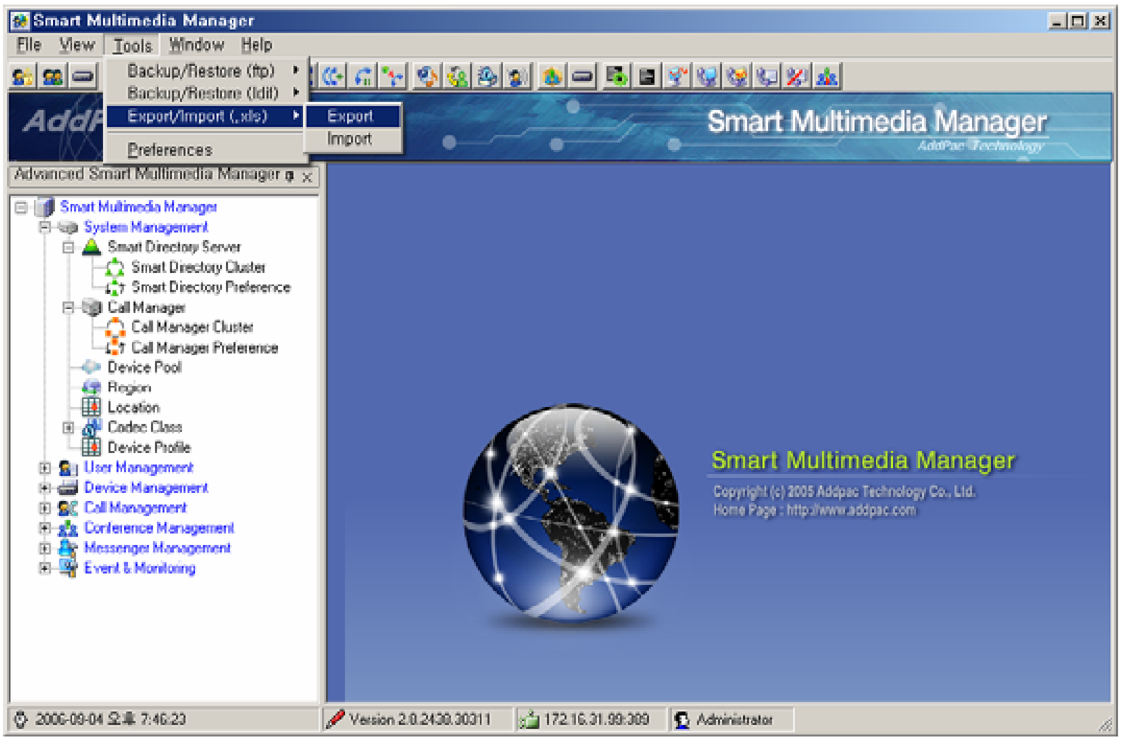
Upgrading the IPNext PBX with the Redundant Configuration

Upgrading the IPNext PBX can be described in a way of operating as a standalone and another way of operating in the redundant configuration.
If upgrading the IPNext PBX does not require any change in the schema of Smart Directory Server and is satisfied with a minor patch by changing APOS image, all it has to do is to reboot after the upload of APOS image by FTP, in a simple way.

Upgrading When the IP-Next PBX is a Standalone Type

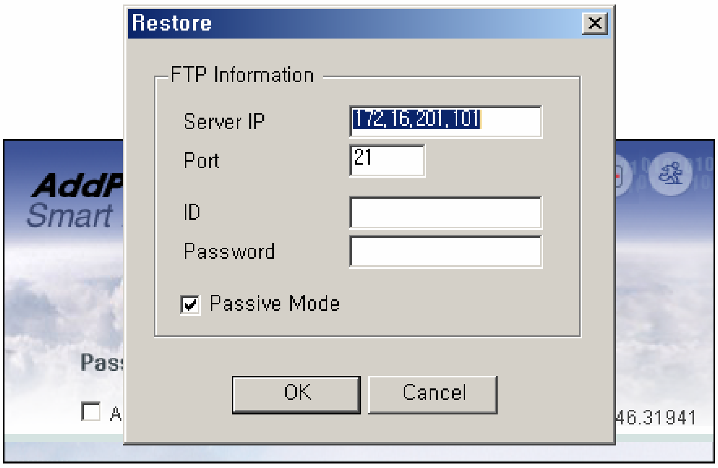
This section is to describe the procedures to upgrade the IPNext PBX system operating as a standalone type.

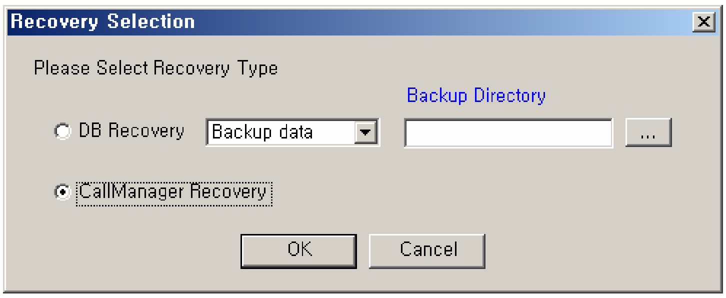
1) By using the function Excel Export of the existing SMM software, the database of the IPNext PBX is to be backed up.



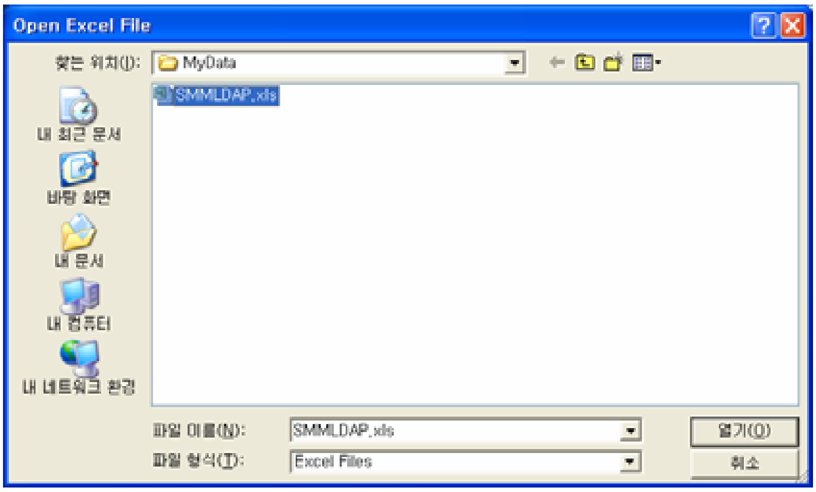
2) Prepare and install the programs necessary for the upgrade.
- By using FTP, the most recent APOS image can be uploaded.
- Install the most recent SMM software matches with the image.

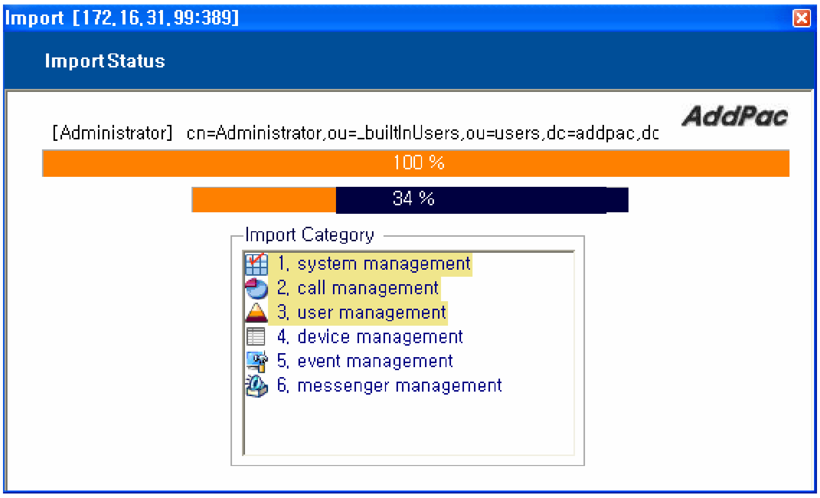
3) Initialize Call Manager Recovery of the IPNext PBX by recovering the most recent SMM program



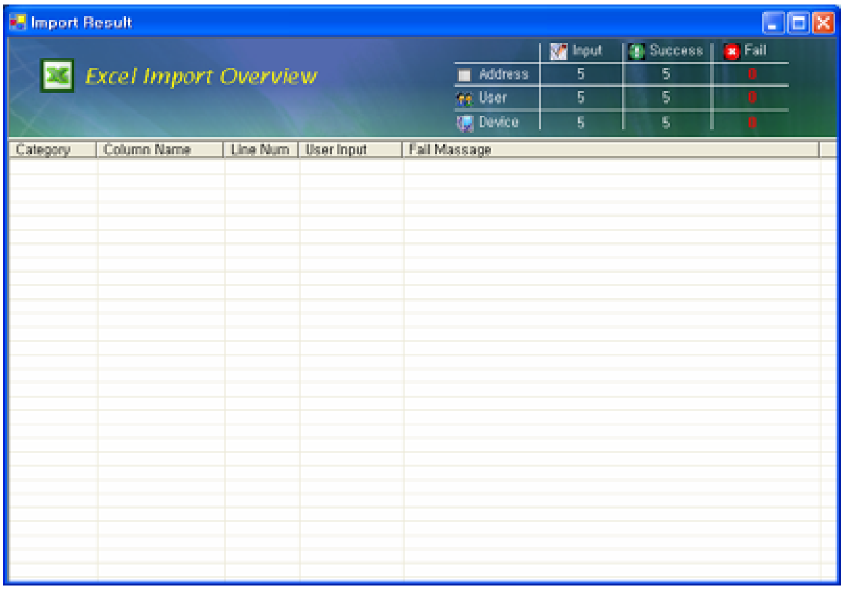


4) When initialization is completed, the IPNext PBX reboots automatically.
5) When rebooting is completed, log in to the IPNext PBX by accessing to the most recent SMM.
6) Upgrade the data by selecting the Excel data which is previously saved by using Excel Import.

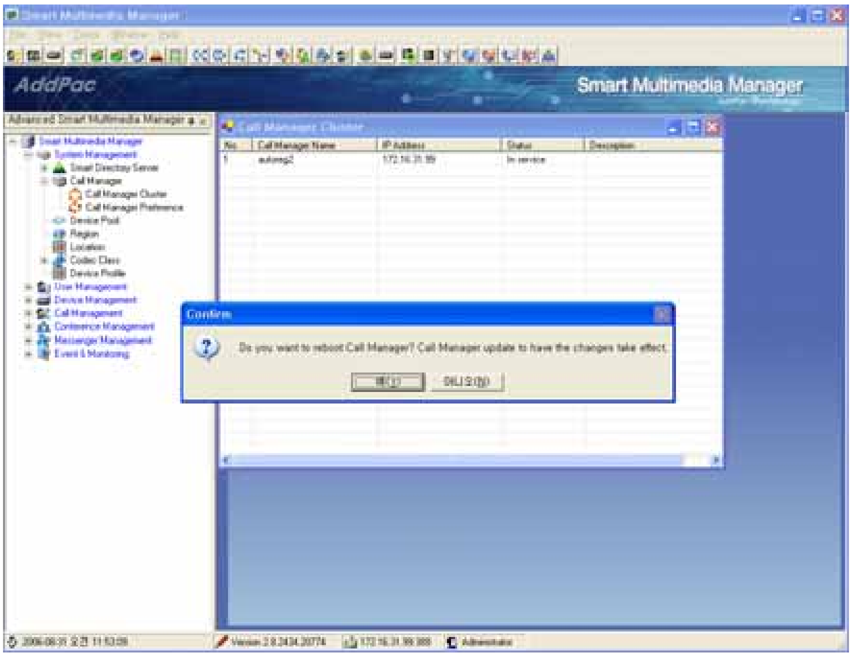




7) When Import of the Excel data is completed, the window for the result is displayed.



8) When the window with the results is finished. the IPNext PBX needs to be rebooted to apply the data



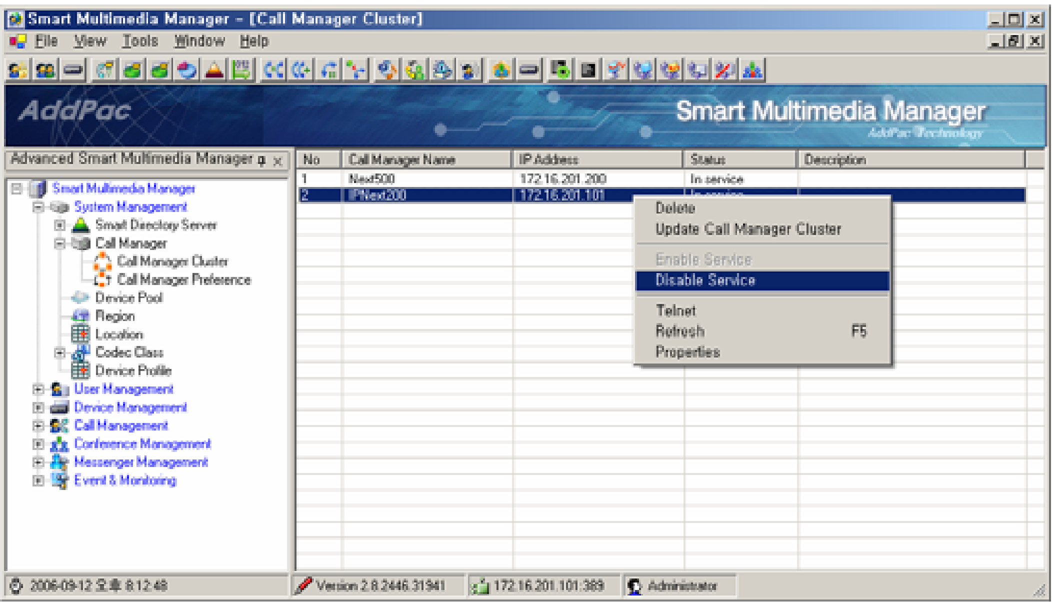
9) The upgrade is completed when the IPNext PBX completes rebooting

Upgrading the IPNext PBX Configured with Redundancy

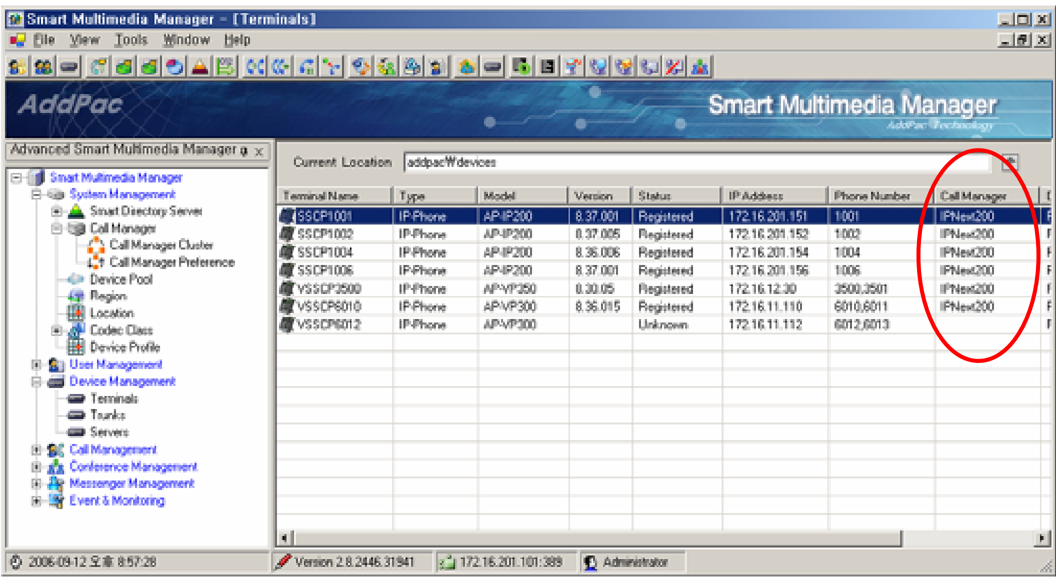
This section is to describe the procedure of upgrading the redundant IPNext PBX system in operation. With assumption that 2 systems (CM A, CM B) of IPNext PBX are in operation, the following procedures can be taken:

- Upgrade CM B first and then it can be configured to provide the service
- Upgrade CM A

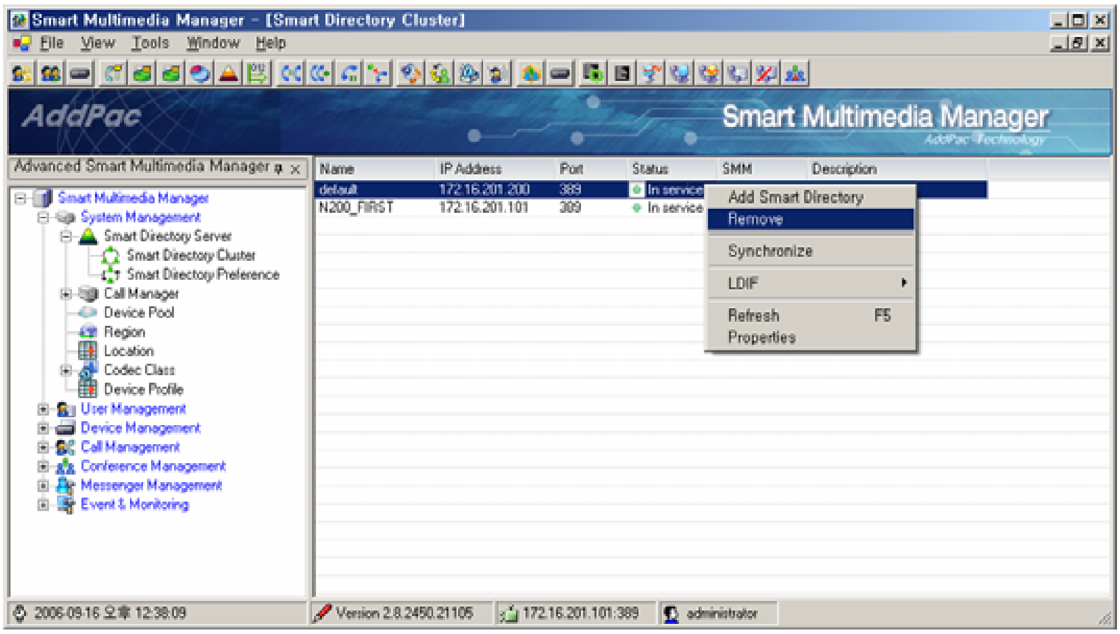
1) After accessing to CM B through the existing SMM, select CM B (System Management > Call Manager > Call Manager Cluster) and perform ‘Disable Service’.



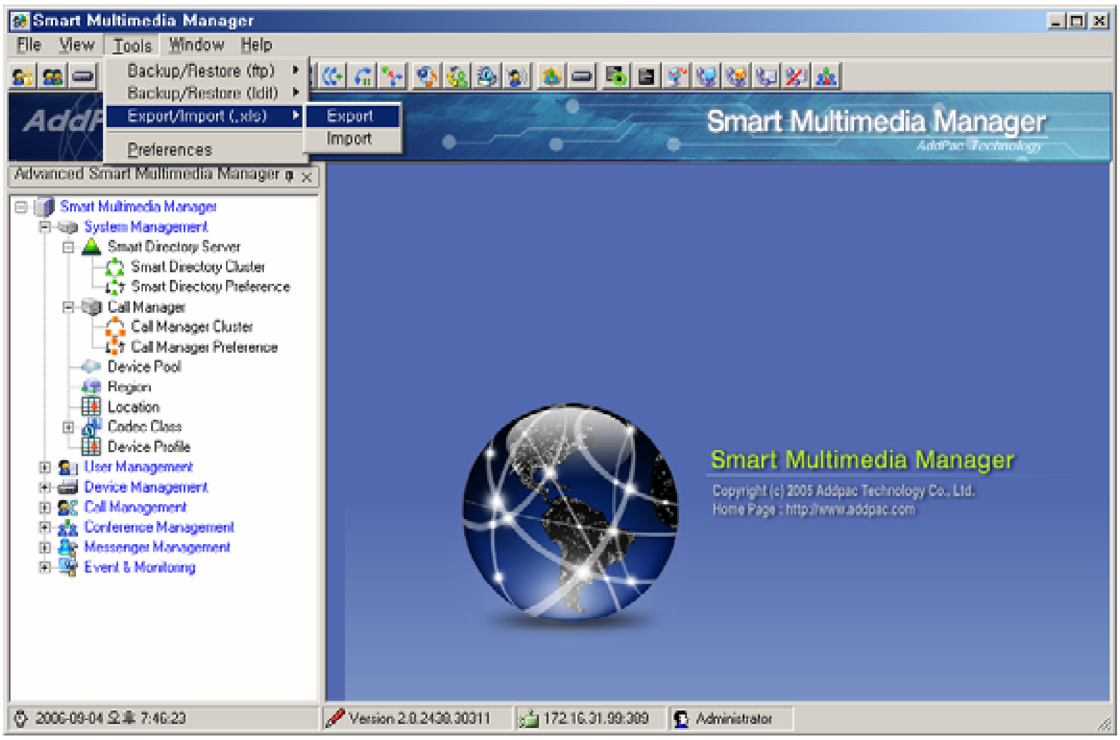
2) The terminals, which were registered to CM B, can be registered to CM A to be operated (Registration of the terminals can be checked by performing Device Management > Terminals)



3) Delete CM A from Smart Directory Cluster(System Management > Smart Directory Server > Smart Directory Cluster) through the existing SMM.



4) Export the data information of the IPNext PBX in an Excel file, by using a function of Excel Export (Tools > Export/Import (xsl) > Export)

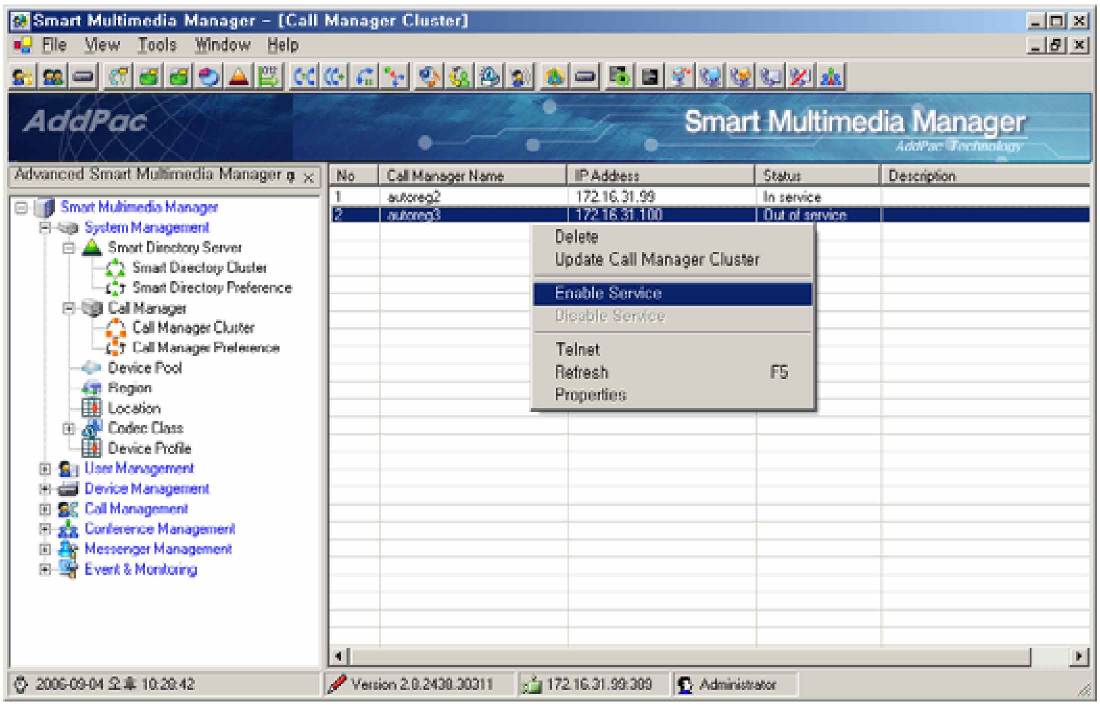


5) Upload a new APOS image of the IPNext PBX for CM B to be upgraded by using FTP

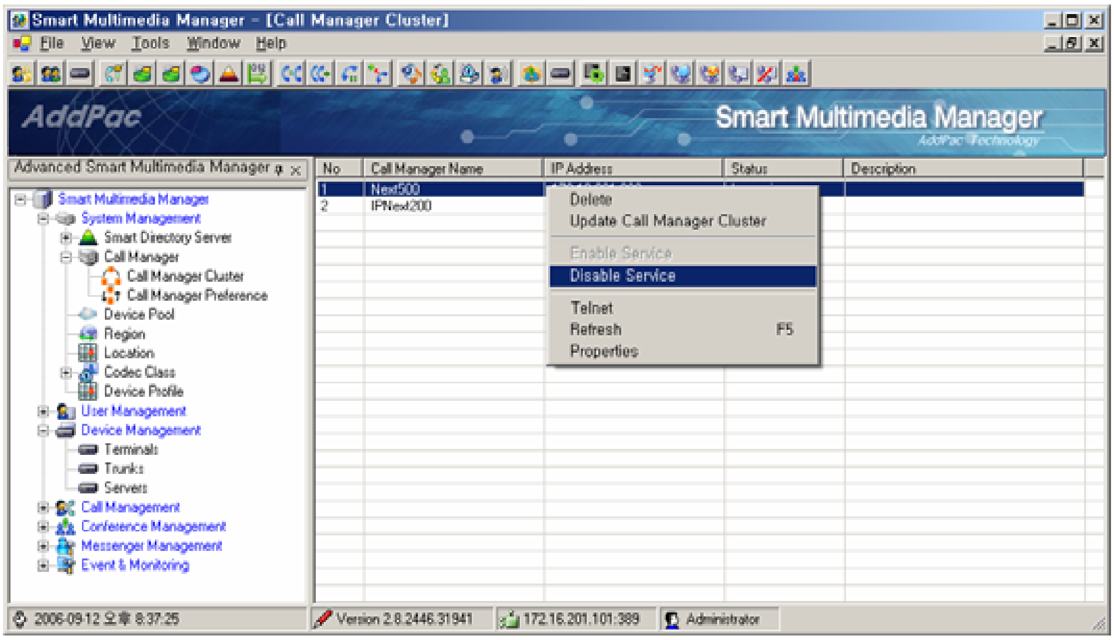
6) Access to CM B by the new SMM, then initialize the Call Manager. If this is the process for a standalone type of the IPNext PBX, the same procedure described as on the **오류! 참조 원본을 찾을 수 없습니다.** of the page**오류! 책갈피가 정의되어 있지 않습니다.** to be taken.

7) When rebooting of CM B is completed then access to the new SMM. By using the function of Excel Import, upgrade (import) the Excel data, which has been saved previously, to CM B. In case upgrading a standalone type of the IPNext PBX, this procedure is same as the one described on 오류! 참조 원본을 찾을 수 없습니다. of the page 오류! 책갈피가 정의되어 있지 않습니다..

8) By performing 'Enable Service' (System Management > Call Manager > Call Manager Cluster > Enable Service) to CM B, through the new SMM, the status can be changed to be able to provide service. In case 'Enable Service' is not activated, perform 'Disable Service' then carry out 'Enable Service' again.



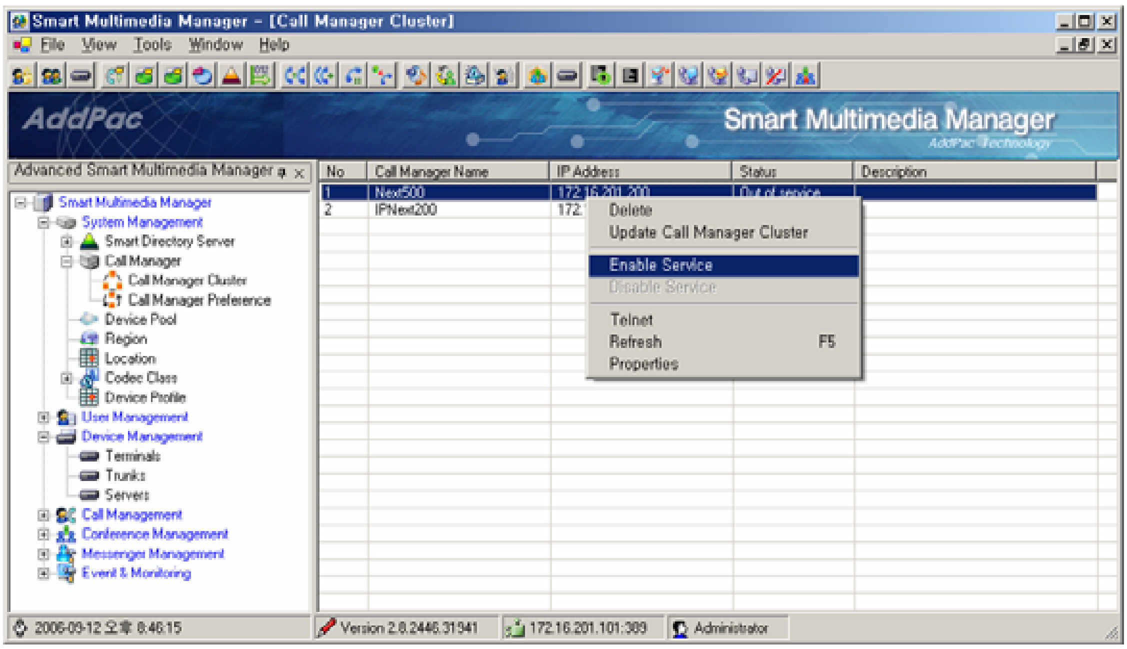
9) Stop the service of CM A by performing 'Disable Service' from Call Manager Cluster(System Management > Call Manager > Call Manager Cluster) through the new SMM (By performing Device Management > Terminals, registration of terminals to CM B can be checked.



10) Upgrade CM A with a new version of APOS by using FTP.

11) The process of redundancy can be carried out by registering CM A Directory Cluster(System Management > Smart Directory Server > Smart Directory Cluster) from the new SMM.
Please refer to the procedures from 오류! 참조 원본을 찾을 수 없습니다. to 오류! 참조 원본을 찾을 수 없습니다. on the page 314, for this function.

12) When CM A completes rebooting, perform '**Enable Service'(System Management > Call Manager > Call Manager Cluster > Enable Service)** for CM A.



13) Now the configuration of redundancy is completed. The normal operation of the redundant function can be checked by the following ways (optional):

- Ethernet Link Up/Down
- Call Manager Shutdown through SMM
- IPNext PBX Power On/Off

14) Back up (Excel, LDIF) the current configuration through the new SMM (optional).

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2543 SIP: Session Initiation Protocol. M. Handley, H. Schulzrinne, E. Schooler, J. Rosenberg. March 1999. (Format: TXT=338861 bytes) (Obsoleted by RFC3261, RFC3262, RFC3263, RFC3264, RFC3265)

2976 The SIP INFO Method. S. Donovan. October 2000. (Format: TXT=17736 bytes) (Status: PROPOSED STANDARD)

3261 SIP: Session Initiation Protocol. J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, R. Sparks, M. Handley, E. Schooler. June 2002. (Format: TXT=647976 bytes) (Obsoletes RFC2543) (Updated by RFC3265) (Status: PROPOSED STANDARD)

3264 An Offer/Answer Model with Session Description Protocol (SDP). J. Rosenberg, H. Schulzrinne. June 2002. (Format: TXT=60854 bytes) (Obsoletes RFC2543) (Status: PROPOSED STANDARD)

3265 Session Initiation Protocol (SIP)-Specific Event Notification. A. B. Roach. June 2002. (Format: TXT=89005 bytes) (Obsoletes RFC2543) (Updates RFC3261) (Status: PROPOSED STANDARD)

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0793 Transmission Control Protocol. J. Postel. Sep-01-1981. (Format: TXT=172710 bytes) (Updated by RFC3168) (Also STD0007) (Status: STANDARD)

0868 Time Protocol. J. Postel, K. Harrenstien. May-01-1983. (Format: TXT=3140 bytes) (Also STD0026) (Status: STANDARD)

1058 Routing Information Protocol. C.L. Hedrick. Jun-01-1988. (Format: TXT=93285 bytes) (Updated by RFC1388, RFC1723) (Status: HISTORIC)

1157 Simple Network Management Protocol (SNMP). J.D. Case, M. Fedor, M.L. Schoffstall, J. Davin. May-01-1990. (Format: TXT=74894 bytes) (Obsoletes RFC1098) (Also STD0015) (Status: HISTORIC)

1213 Management Information Base for Network Management of TCP/IP-based internets:MIB-II.

K. McCloghrie, M.T. Rose. Mar-01-1991.

(Format: TXT=146080 bytes) (Obsoletes RFC1158) (Updated by RFC2011, RFC2012, RFC2013) (Also STD0017) (Status: STANDARD)

1305 Network Time Protocol (Version 3) Specification, Implementation.

D. Mills. March 1992. (Format: TXT=307085, PDF=442493 bytes) (Obsoletes RFC0958, RFC1059, RFC1119) (Status: DRAFT STANDARD)

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3046 DHCP Relay Agent Information Option. M. Patrick. January 2001. (Format: TXT=30633 bytes) (Status: PROPOSED STANDARD)

Acronyms and Glossary

DSP	Digital Signal Processor. Dedicated microprocessor for digital signal process.
DTMF	Dual Tone Multi-Frequency. Using two types of voice-band tones for dialing.
E&M	An acronym for recEive and transmit or ear and mouth. E&M interface uses a RJ-48 telephone cable to connect remote calls from an IP network to PBX trunk lines (tie lines) for local distribution. It is a signaling technique for two-wire and four-wire telephone and trunk interfaces.
E1	The basic building block for European multi-megabit data rates, with a bandwidth of 2.048Mbps.
Encryption	the manipulation of a packet's data in order to prevent any but the intended recipient from reading that data.
Ethernet	Broadband LAN standard initiated by Xerox Corporation and co-developed by Intel and DEC. Utilizing CSMA/CD and the various cables of 10Mbps are used. It is similar to IEEE 802.3. Refer to: 10Base-2, 10Base5, 10Base-F, 10Base-T, 10Broad-36, Fast Ethernet, IEEE 802.3.
FAX	Short for "FACSimile." In essence, a fax machine sends an electronic "facsimile" or copy of the document. An optical scanner in the machine scans the document and the resulting bit stream is then sent to the receiving machine via telephone line. The transmission and the reproduction at a distance of still pictures printed matter and similar documented material
Frame	data that is transmitted between network points as a unit complete with addressing and necessary protocol control information. A frame is usually transmitted serial bit by bit and contains a header field and a trailer field that "frame" the data. (Some control frames contain no data.)
Frame-Relay	Switching type Data Link Layer Protocol. Using HDLC capsule, process multi-number of virtual circuits between devices.
FTP	an acronym for File Transfer Protocol, a very common method of transferring one or more files from one computer to another. Defined at RFC 959.
FXO	Foreign Exchange Office. An FXO interface connects to the Public Switched Telephone Network (PSTN) central office and is the interface offered on a standard telephone.
FXS	Foreign Exchange Station. An FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone.
G.711	Describes the 64-kbps PCM voice coding technique. In G.711, encoded voice is already in the correct format for digital voice delivery in the PSTN or through

	PBXs.
G.723.1	Describes a compression technique that can be used for compressing speech or audio signal components at a very low bit rate as part of the H.324 family of standards. This CODEC has two bit rates associated with it: 5.3 and 6.3 kbps. The higher bit rate is based on ML-MLQ technology and provides a somewhat higher quality of sound. The lower bit rate is based on CELP and provides system designers with additional flexibility.
G.726	Describes ADPCM coding at 40, 32, 24 and 16 kbps. ADPCM encoded voice can be interchanged between packet voice, PSTN, and PBX networks if the PBX networks are configured to support ADPCM. Described in the ITU-T standard in its G-series recommendations.
G.728	Describes a 16 kbps low-delay variation of CELP voice compression. CELP voice coding must be translated into a public telephony format for delivery to or through the PSTN. Described in the ITU-T standard in its G-series recommendations..
Gatekeeper	The component of an H.323 conferencing system that performs call address resolution, admission control, and subnet bandwidth management. H.323 entity on a LAN that provides address translation and control access to the LAN for H.323 terminals and gateways. The gatekeeper can provide other services to the H.323 terminals and gateways, such as bandwidth management and locating gateways. A gatekeeper maintains a registry of devices in the multimedia network. The devices register with the gatekeeper at startup and request admission to a call from the gatekeeper.
H.225	An International Telecommunication Union (ITU-T) standard for H.225.0 session control and packetization. It defines various protocols of RAS, Q.931, RTP and etc.
H.245	An International Telecommunication Union (ITU-T) standard for H.245 end-point control.
H.323	An International Telecommunication Union (ITU-T) standard that describes packet-based video, audio, and data conferencing.
HBD3	Line code type of E1 line.
HDLC	An acronym for High-Level Data Link Control. A transmission protocol for the Data Link Layer. In HDLC, data is organized into a unit (called a frame) and sent across a network to a destination that verifies its successful arrival. Variations of HDLC are also used for the public networks that use the X.25 communications protocol and for frame relay, a protocol used in both and wide area network, public and private.
Hookflash	Short on-hook period usually generated by a telephone-like device during a

	call to indicate that the telephone is attempting to perform a dial-tone recall from a PBX. Hookflash is often used to perform call transfer.
HTTP	An acronym for Hypertext Transfer Protocol. A file transfer protocol used by web browser or web server for transmitting text or graphic files.
IPSec	Internet Protocol Security protocol, a framework for a set of protocols for security at the network or packet processing layer of network communication. Earlier security approaches have inserted security at the Application layer of the communications model. IPsec is said to be especially useful for implementing virtual private networks and for remote user access through dial-up connection to private networks. A big advantage of IPsec is that security arrangements can be handled without requiring changes to individual user computers. Cisco has been a leader in proposing IPsec as a standard (or combination of standards and technologies) and has included support for it in its network routers.
IPv6	IPv6 (Internet Protocol Version 6) is the latest level of the Internet Protocol (IP) and is now included as part of IP support in many products including the major computer operating systems. IPv6 has also been called "IPng" (IP Next Generation). Formally, IPv6 is a set of specifications from the Internet Engineering Task Force (IETF). IPv6 was designed as an evolutionary set of improvements to the current IP Version 4. Network hosts and intermediate nodes with either IPv4 or IPv6 can handle packets formatted for either level of the Internet Protocol. Users and service providers can update to IPv6 independently without having to coordinate with each other.
ISP	An ISP (Internet service provider) is a company that provides individuals and other companies access to the Internet and other related services such as Web site building and virtual hosting. An ISP has the equipment and the telecommunication line access required to have a point-of-presence on the Internet for the geographic area served. The larger ISPs have their own high-speed leased lines so that they are less dependent on the telecommunication providers and can provide better service to their customers. Among the largest national and regional ISPs are AT&T WorldNet, IBM Global Network, MCI, Netcom, UUNet, and PSINet.
ITU-T	The ITU-T (for Telecommunication Standardization Sector of the International Telecommunications Union) is the primary international body for fostering cooperative standards for telecommunications equipment and systems. It was formerly known as the CCITT. It is located in Geneva, Switzerland
IVR	Interactive Voice Response (IVR) is a software application that accepts a

	combination of voice telephone input and touch-tone keypad selection and provides appropriate responses in the form of voice, fax, callback, e-mail and perhaps other media. IVR is usually part of a larger application that includes database access. Common IVR applications include: Bank and stock account balances and transfers.
LAN	A local area network is a group of computers and associated devices that share a common communications line and typically share the resources of a single processor or server within a small geographic area (for example, within an office building). LAN standard defines cable connection and signal processing on Physical Layer and Data Link Layer.
Link	Network communication channels consisting of sending and receiving devices, circuits, transmission path. Usually refer to WAN connection. Referred as Line, or transmission link.
Loopback test	A loopback test is a test in which a signal in sent from a communications device and returned (looped back) to it as a way to determine whether the device is working right or as a way to pin down a failing node in a network.
MAC Address	Standardized data link layer address that is required for every port or device that connects to a LAN. Other devices in the network use these addresses to locate specific ports in the network and to create and update routing tables and data structures. MAC addresses are 6 bytes long and are controlled by the IEEE. Also known as a hardware address, MAC-layer address, and physical address. Compare with network address.
MAN	A data network designed for a town or city. MANs are considered larger than LANs but smaller than WANs. Compare with: LAN, WAN.
MGCP	MGCP, also known as H.248 and Megaco, is a standard protocol for handling the signaling and session management needed during a multimedia conference. The protocol defines a means of communication between a media gateway, which converts data from the format required for a circuit-switched network to that required for a packet-switched network and the media gateway controller. MGCP can be used to set up, maintain, and terminate calls between multiple endpoints. Megaco and H.248 refer to an enhanced version of MGCP
NAT	NAT (Network Address Translation) is the translation of an Internet Protocol address (IP address) used within one network to a different IP address known within another network. One network is designated the inside network and the other is the outside.
NTP	Network Time Protocol (NTP) is a protocol that is used to synchronize computer clock times in a network of computers. In common with similar protocols, NTP uses Coordinated Universal Time (UTC) to synchronize

	computer clock times to a millisecond, and sometimes to a fraction of a millisecond.
PABX	Private Automatic Branch Exchange. A telephone switch for use inside a corporation. It connects offices (internal extensions) with each other and provides access (typically by dialing an access number such as 9) to the public telephone network PABX is the preferred term in Europe, PBX is used in the USA.
Packet	Packets contain a source and destination address as well as the actual message. Packets also known as Datagrams.
PBX	A PBX (private branch exchange) is a telephone system within an enterprise that switches calls between enterprise users on local lines while allowing all users to share a certain number of external phone lines.
PING	Packet INternet Groper, a packet (small message) sent to test the validity / availability of an IP address on a network
Point to Point Connection	Basic connection type. In ATM, point to point connection is half duplex connection between two ATM end systems or full duplex connection.
Pont to Multipoint Connection	Basic connection type. In ATM, point to multipoint connection is half duplex connection among one sending end system (root node) and multiple receiving end system. Compare with: point-to-point connection.
POTS	Plain Old Telephone Service. Compare with: PSTN.
PPP	The most popular method for transporting IP packets over a serial link between the user and the ISP. Developed in 1994 by the IETF and superseding the SLIP protocol, PPP establishes the session between the user's computer and the ISP using its own Link Control Protocol (LCP). PPP supports PAP, CHAP and other authentication protocols as well as compression and encryption.
Protocol Stack	Any set of communication protocols, such as TCP/IP, that consists of two or more layers of software and hardware. It's called a stack because each layer builds on the functionality in the layer below
PSTN	Public Switched Telephone Network – term for the entire, world-wide telephone network. Sometimes refers to as POTS.
PVC	Permanent Virtual Circuit or permanent virtual connection. A continuously available communications path that connects two fixed end points.
Q.931 Signaling	ITU-T specification for network layer of ISDN. Q.931 uses out-of-band signaling on the D-channel to control calls.
QoS	This refers to the assumption that data transmission rates, error rates, and other characteristics can be measured, improved, and to some degree,

	guaranteed in advance. Basically, QoS describes a collective measure of the level of service a provider delivers to its customers or subscribers.
RAM	Random-Access Memory, a non-retentive memory, whose contents get lost after a switch-off or reset. Application programs run in the random access memory and data is stored and processed.
RAS	Registration Admission Status protocol. The communication protocol used to convey registration, admission and status messages between H.323 endpoints and the gatekeeper.
RISC	Reduced Instruction Set Computing
Router	On the Internet, a router is a device or, in some cases, software in a computer, that determines the next network point to which a packet should be forwarded toward its destination. The router is connected to at least two networks and decides which way to send each information packet based on its current understanding of the state of the networks it is connected to. A router is located at any gateway (where one network meets another), including each Internet point-of-presence. A router is often included as part of a network switch. Compare with: gateway. Refer to: relay.
RS-232	Most common Physical Layer interface. Known as EIA/TIA-232.
RTCP	Real-time Control Protocol (RTCP) is a companion protocol of RTP that is used to maintain quality of service. Refer to: RTP(Real-Time Transport Protocol).
RTP	1. Routing Table Protocol, VINES routing protocol based on RIP. Distributes network topology, and aids VINES servers in finding neighboring clients, servers, and routers. Uses delay as a routing metric. Refer to: SRTP. 2. Rapid Transport Protocol. Provides pacing and error recovery for APPN data as it crosses the APPN network. With RTP, error recovery and flow control are done end-to-end rather than at every node. RTP prevents congestion rather than reacts to it. 3. Real-Time Transport Protocol. Commonly used with IP networks. RTP is designed to provide end-to-end network transport functions for applications transmitting real-time data, such as audio, video, or simulation data, over multicast or unicast network services. RTP provides such services as payload type identification, sequence numbering, time-stamping, and delivery monitoring to real-time applications.
SIP	The Session Initiation Protocol (SIP) is an Internet Engineering Task Force (IETF) standard protocol for initiating an interactive user session that involves multimedia elements such as video, voice, chat, gaming, and virtual reality. Like HTTP or SMTP, SIP works in the Application layer of the Open Systems

	Interconnection (OSI) communications model. The Application layer is the level responsible for ensuring that communication is possible. SIP can establish multimedia sessions or Internet telephony calls, and modify, or terminate them. The protocol can also invite participants to unicast or multicast sessions that do not necessarily involve the initiator. Because the SIP supports name mapping and redirection services, it makes it possible for users to initiate and receive communications and services from any location, and for networks to identify the users whatever they are. SIP is a request-response protocol, dealing with requests from clients and responses from servers. Participants are identified by SIP URLs. Requests can be sent through any transport protocol, such as UDP, SCTP, or TCP. SIP determines the end system to be used for the session, the communication media and media parameters, and the called party's desire to engage in the communication. Once these are assured, SIP establishes call parameters at either end of the communication, and handles call transfer and termination. The Session Initiation Protocol is specified in IETF Request for Comments [RFC] 2543.
SmartViewer	The real-time monitoring, statistical data search and management GUI based software developed by AddPac Technology for AP-GK1000, AP-GK2000, AP-GK3000 models.
SNMP	Simple Network Management Protocol. Network management protocol used almost exclusively in TCP/IP networks. SNMP provides a means to monitor and control network devices, and to manage configurations, statistics collection, performance, and security. Refer to: SGMP, SNMP2.
T1	A TDM physical transmission standard consisting of two twisted wire pairs and related equipment capable of carrying a 1.544 Mbps DS-1 signal. Term often used interchangeably with DS-1. Refer to: AMI, B8ZS, DS-1.
TCP/IP	Transmission Control Protocol/Internet Protocol, The protocol suit developed by DoD (USA) in 1970s for the worldwide inter-network development. TCP & IP is the most well known protocols of the suite. Refer to: IP, TCAP.
Telco	Telephone Company, referring to the company offering telephone service to customers. Typically, it refers to an individual company such as Bell operating company offering local telephone service, however, sometimes local telephony service providers are included.
Telnet	Standard Terminal Emulation program covered by TCP/IP protocol stack. Used for remote terminal connection. Via Telnet, users can log-in to the system and operate the resources as working on the local system. Defined on RFC 854.
VCI	the address or label of a VC; a value stored in a field in the ATM cell header

	that identifies an individual virtual channel to which the cell belongs. VCI values may be different for each data link hop of an ATM virtual connection.
VDSL	New DSL technology that accepts bandwidths of up to 27 Mbps over relatively short distances. VDSL, in the process of being standardized, allows symmetric or asymmetric throughputs that are much higher than other xDSL standards (up to 27 Mbps when downloading and 3 Mbps when uploading under asymmetric or 14 Mbps in symmetric), as well as the simultaneous transport of ISDN (Numeris) services but with much shorter ranges that do not exceed 900 m to 1 km. In practice, this technique may require the deployment of optical remotes and the setting up of active equipment in the local loop. Compare with: ADSL, HDSL, SDSL.
VoATM	Voice Over ATM. Voice over ATM enables an ATM switch to carry voice traffic (for example, telephone calls and faxes) over an ATM network. When sending voice traffic over ATM, the voice traffic is encapsulated using AAL1/AAL2 ATM packets.
VoFR	Voice Over Frame Relay. Voice over Frame Relay enables a router to carry voice traffic (for example, telephone calls and faxes) over a Frame Relay network. When sending voice traffic over Frame Relay, the voice traffic is segmented and encapsulated for transit across the Frame Relay network using FRF.12 encapsulation.
VoHDLC	Voice Over HDLC. Voice over HDLC enables a router to carry live voice traffic (for example, telephone calls and faxes) back-to-back to a second router over a serial line.
VoIP	VoIP (Voice delivered using the Internet Protocol) is a term used in IP telephony for a set of facilities for managing the delivery of voice information using the Internet Protocol (IP). In general, this means sending voice information in digital form in discrete packets rather than in the traditional circuit-committed protocols of the public switched telephone network (PSTN). A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service.
VPN	Virtual Private Network, VPN allows IP traffic to travel securely over a public TCP/IP network by encrypting all traffic from one network to another. A VPN uses "tunneling" to encrypt all information at the IP level.
WAN	A network that covers a large geographical area. Typical WAN technologies include point-to-point, X.25 and frame relay. Compare with: LAN, MAN.