AP-IPNext SMT Manual

AddPac

AddPac Technology OSTS

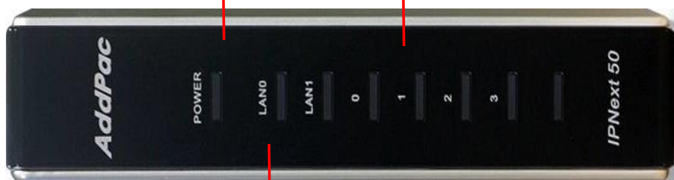
Contents

IPNext Exterior (Front View)

Front View

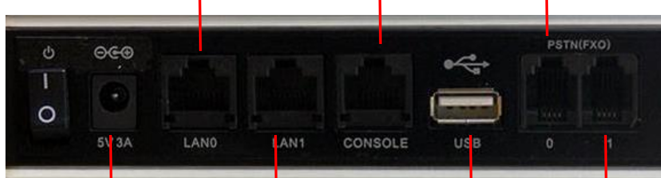
1) POWER LED
2) LAN 0,1 LED
3) FXO LED

1) POWER LED, 3) FXO LED



2) LAN0,1 LED

1) FastEthernet(Wan), 2) Console. 3) FXO 0



1) Power, 2) FastEthernet(Lan) 3) USB 4) FXO 1

Rear Side

Parts of Rear Side

1) Power Switch

Major Features of IPNext

Major Features of IP-Next

Features	**Contents**
Telephony and Service & Features	• Voice Mail • Call Parking • Call Pickup • Call Forwarding • Auto Attendant • Calling Number and Name Identification • Call Transfer – Blind, Consult by Softkey • Call Waiting Indication Call Swapping by Softkey • Call Hold by Softkey • Conference Control (internal/external MCU)

Major Features of IPNext

Features	**Contents**
Advanced Features with AddPac IP phone, Video Phone, etc	• with AddPac IP phone, Video Phone, etc • Plug and Play with Auto Discovery Function (SSCP) • Softkey Map Download and Control (SSCP) • Time and Date Setting
IP-PBX Signaling Protocols	• SIP Application Server, Proxy, Registrar and Location Server(RFC3261) • Multiple ITSP Trunk with SIP & H.323 Account Support - IP UA Client Role for Registering to ITSP SIP Server - H.323 Gatekeeper Client Role for Registering to ITSP H.323 Gatekeeper Server

Major Features of IPNext

Features	**Contents**
IVR(Interactive Voice Response) & Auto Attendant	• Default Auto Attendant Support • IVR Function - Provides with GUI-based Smart IVR Scenario Editor - Upload/Download Scenario by Smart IVR Scenario Editor - Supports Multiple Concurrent Scenarios - Support Recordable IVR Prompts
Voice Mail	• Support Voice Mail with IVR • Access from Remote Site via Trunk Support • Voice Mail Notification Support
Conference	• Ad-Hoc Conference • Dial-Out Conference • Meet-me Conference • Multiple External MCU Support (Video, Audio, etc) : AP-MC1000

Major Features of IPNext

Features	**Contents**
Music & Announcement	• Music on Hold • Replaceable Announcements • Dialing Music/Tone Service
IP-PBX User & Device Management	• Auto Discovery of IP Phones & IP Video Phones • Monitoring Status of Phones
IP-PBX Miscellaneous Function	• Auto Config & Upgrade • Personal Directory (Smart Messenger)
Basic Routing	• IPv4/IPv6 Dual Stack • Management Routing - Telnet, FTP, TFTP, SNMP, Syslog support - Packet filtering (Access-list) - Static Routing

Major Features of IPNext

Features	**Contents**
Network Management	• Standard SNMP Agent (MIB v2) Support • Traffic Queuing • Remote Management using Console, Telnet.
Security Functions	• Standard & Extended IP Access List • Enable/Disable for Specific Protocols • Auto-disconnect for Telnet/Console Sessions

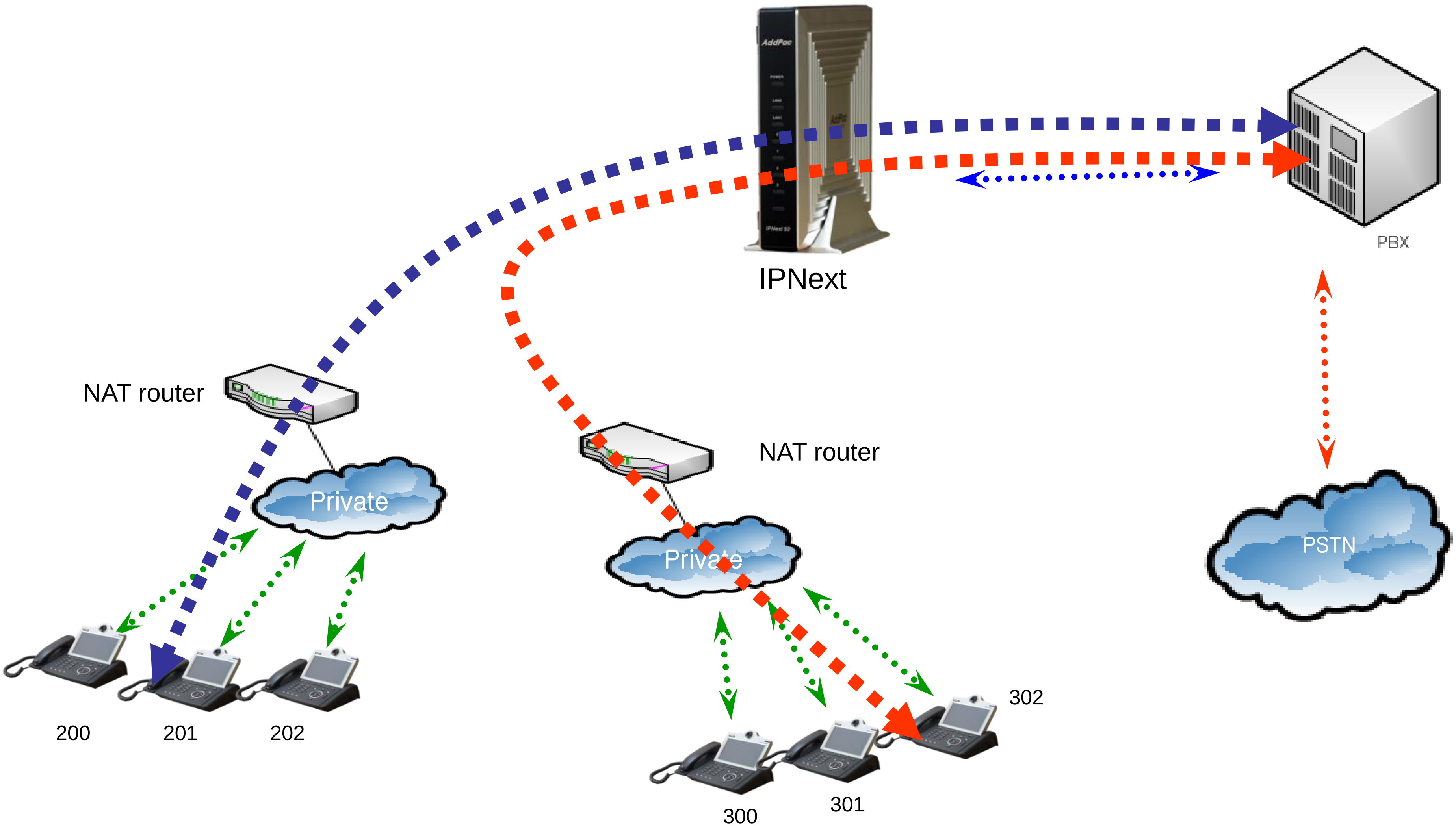
Major Features of IPNext

Features	**Contents**
Operation & Management	• System Performance Analysis for Process, CPU, Connection I/F • Configuration Backup & Restore for APOS Managements • System Booting and Auto-rebooting with Watchdog Feature • System Managements with Data Logging • IP Traffic Statistics with Accounting • Debugging command
Other Scalability Features	• DHCP Server/Client • Network Address Translation (NAT) Function • Port Address Translation (PAT) Function • Cisco Style Command Line Interface(CLI) • Network time Protocol(NTP) Support

Hardware Features of IPNext

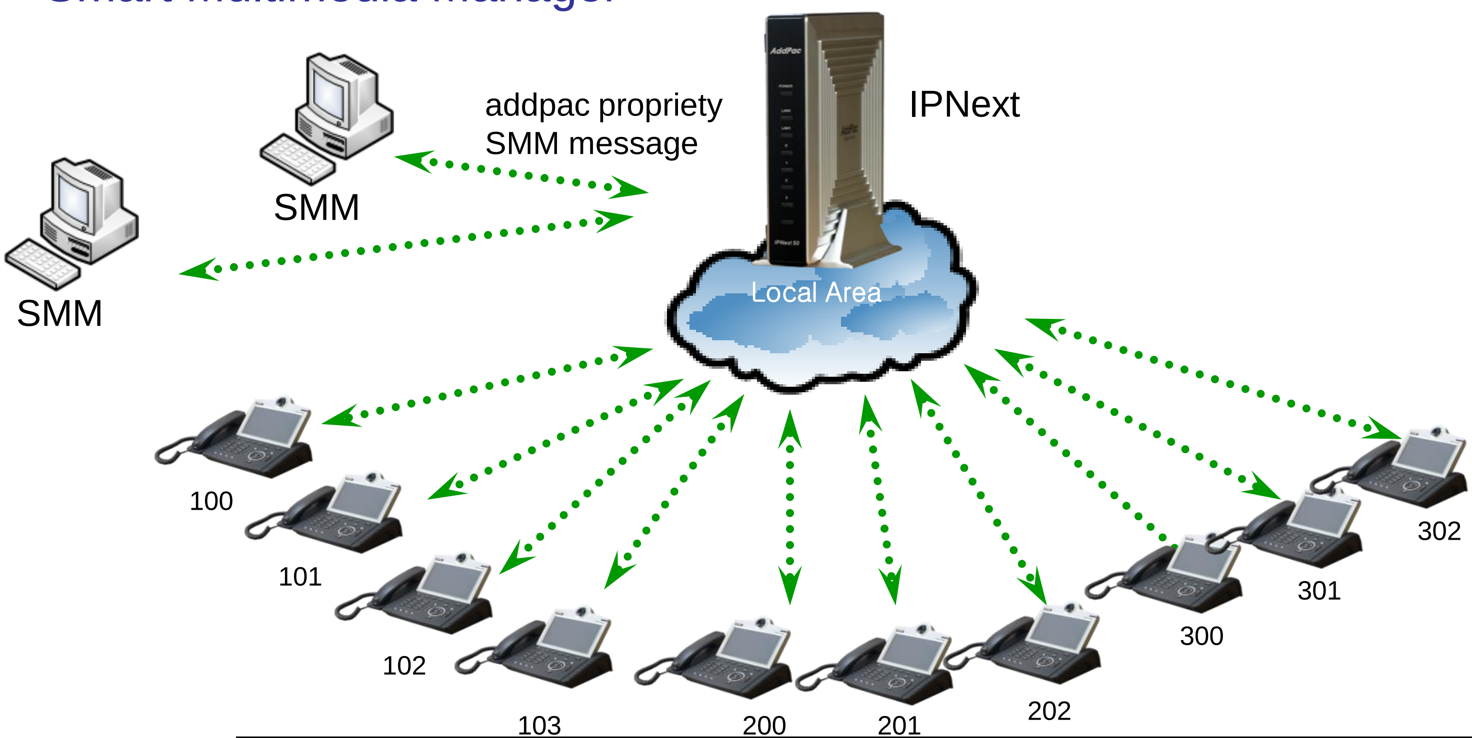
Details	**Contents**	
CPU	• High Performance RISC Integrated Host Processor	
Memory	Main Memory	• 128Mbyte SDRAM
	Flash Memory	• 512Mbyte
	SDRAM Memory	• 512Kbyte Flash Memory
LAN Interface	LAN0 Port	• One(1) Ethernet 10/100Mbps Fast Ethernet
	LAN1 Port	•One(1) Ethernet 10/100Mbps Fast Ethernet
	Console Port	•One(1) RS-232C Interface for CLI
PSTN Interface	FXO Port	•None (Model A)
		•2-Port FXO Voice Interface (2 * RJ11)(Model B)
Power & Operation Environments	• Power Requirement: Power VAC 110~220 VAC, 50/60Hz, 5V 3A • Operating Temperature: 0˚C ~ 50˚C (32˚ ~ 122˚F) • Storage Temperature: -40˚C ~ +85˚C (-40˚ ~ +185˚F) • Relative Humidity: 5% ~ 95% (Non-condensing)	
Dimensions	• 38(H) * 182(W) * 182(D)	
Weight	• 0.46 Kg	

Function of Signaling / Media Proxy (RTP Proxy)



Smart Multimedia Manager

Smart Multimedia Manager



Major Functions

- IP-Next Registration all setups related to user.
- IP-Next all setups related to service
- IP-Next status review
- IP-Next message trace

Smart Multimedia Manager

Smart Multimedia Manager

Category	Sub-Category	Description of Functions
Administration Mode	Advanced mode	Display Mode for setting up all the functions of IP-PBX
	Easy mode	Mode for Setting up only Basic Functions of IP PBX
Event monitoring	Real Time Monitoring for the Call being Processed	Monitoring Function for Enabled Calls at the present time
	Real Time message Trace	Real Time Message Tracing Function for Calls Being Processed at the Present Time
	Message logging	Storing and restoring Functions for Calls being Processed (filtering function)
Redundancy	Active-Active	The structure allowing Two IP-PBX's to provision service simultaneously
	Active-Standby	The structure allowing one of two IP PBX's become master and the other one become disbled
Terminal Registration	SSCP	The terminal using AddPac Specific Terminal Control Protocol. AP-IP-XXX, AP-VP-XXX series come under this sub- category
	Generic SIP terminal	Supporting linking protocol of SSCP and general SIP terminal or SIP gateway with Multiple Ports
	SIP Trunk	Trunk Gateway or Service Provider's SIP server for interoperability with PSTN network outside
IP	V6	IP Version 4
	V4	IP Version 6

Smart Multimedia Manager

Category	Sub-Category	Functional Description
Codec Class	codec list by group	Each codec class can be set to each group
Device Profile		In case of SSCP terminal, Reference can be made for SSCP terminal after common options are grouped and a profile is created
Account	Administrator Account/password register/delete	admin account which can access to SMM
	Subscriber account/password register/ delete	access account for Smart Messenger or voice mail
Device Management	Phone User	assigning telephone numbers to subscribers
	user General	Detailed information on telephone subscriber (one's real name, subscriber id, password and others)
	Terminal General	Setup and binding of profile/partition/VoIP Protocol for telephone subscribers
Trunk Gateway	General	Setup and binding of location/device profile/partition/VoIP Protocol for Trunk gateway
	routing pattern	Routing pattern input(Digit map)
	Call Control	Call priority / authority / MRBT/CID…
SIP Proxy Server	General	location/device profile/partition/RTP Proxy/Network domain/SIP ID/password/DTMF relay method and other setups
	Register (option)	Register Phone number

Smart Multimedia Manager

Category	Subcategory	Functional Description
MCU	Local MCU	Registering Local MCU
	Remote MCU	Registering Remote MCU
Auto Upgrade	Auto Upgrade Server	Assigning auto upgrade server outside or IP PBX
	Option	Auto upgrade group/URL/server position/ access information (id/password) and others.
Call Policy	pick-up	Setup for enabling direct pickup in the same group only for pick-up group
	partition	Used for setting ranks for outgoing calls of each group/ subscriber
	Partition access list	Used for setting ranks of outgoing calls of each group/ subscriber
	Number Translation Rule	Registering patter list of change in number
	Routing group	Routing Tag for Trunk interface
	Routing List	Routing pattern list to be applied to Routing Group

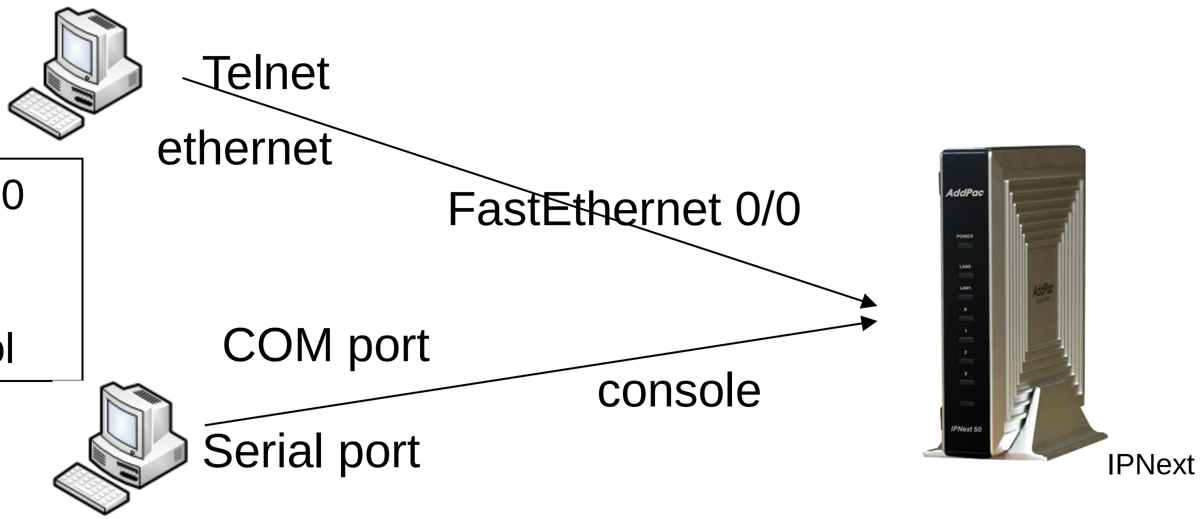
Initial Setting

1. Basic configuration(CLI)

2. LdapClient set up(CLI)

Console and Telnet Login

Baud rate 9600
No parity
1 stop bit
No flow control



Default login account/password : root/router

Command of change Root account password

```
IP-PBX> enable
IP-PBX# config t
IP-PBX(config)# no username root
IP-PBX(config)# username root password addpac administrator
```

% Caution: After no username is set and rebooted, the root account disappears and remote access becomes impossible. The only way to access is automatic login, the authority at this time is administrator

% Must save after setup.

- **Booting messages in IPNext(via console)**

System Bootstrap, Version 1.2
Decompressing the image:
############################[OK]
?System Boot Loader, Version 5.1.6
Copyright (c) by AddPac Technology Co., Ltd. Since 1999.

[DUAL-BOOT] Start application (0xbd000000)...

System Bootstrap, Version 1.2
Decompressing the image:
###
.
AddPac IP-PBX Series (IPNext_G2)
32BIT RISC Processor With 125MHz Clock
128 Mbytes System Memory.
512 Kbytes System Boot Flash Memory
32 Mbytes System Flash Memory

1 RS232 Serial Console Interface
…
Press RETURN to get started.

Basic configuration

- **Connect IPNext**

Welcome, APOS(tm) Kernel Version 8.xx.xxx.
Copyright (c) 1999-2008 AddPac Technology Co., Ltd.

Login: root **(Login ID)**
Password:****** **(Default login Password : router)**
IP-PBX>

- **Set hostname**

IP-PBX> enable
IP-PBX# configure terminal
IP-PBX(config)# **hostname <Name>** **(set Hostname)**
IP-PBX(config)#

- **Set Server Clock**

IP-PBX> enable
IP-PBX# configure terminal
IP-PBX(config)# **clock time <Year><Month><Day><Hour><Minute><Second>** **(set clock)**
IP-PBX(config)# **clock time 2010 03 15 15 28 45**
IP-PBX(config)#

Basic configuration

- **Set Interface IP**

IP-PBX> enable
IP-PBX# configure terminal
IP-PBX(config)# **interface FastEthernet 0/0**
IP-PBX(config)# **ip address <IP Address> <Subnetmask> (Set ip address)**

Example)
IP-PBX(config)# **ip address 172.17.109.200 255.255.0.0**

- **Set Gateway**

IP-PBX> enable
IP-PBX# configure terminal
IP-PBX(config)# **ip route <Destination Network IP> <Wildcard Mask><Gateway IP> (set Gateway 's ip)**
IP-PBX(config)#

Example)
IP-PBX(config)# **ip route 0.0.0.0 0.0.0.0 172.17.1.1**

LdapClient set up(CLI)

- **Necessary configuration for running Smart Multimedia Manager(WEB)**

IP-PBX#
IP-PBX# configure terminal
IP-PBX(config)#ip tcp keep-alive count 5
IP-PBX(config)# ip tcp keep-alive idle 30
IP-PBX(config)# ip tcp keep-alive interval 5

IP-PBX(config)# no http authentication
IP-PBX(config)# http document-root /hd

IP-PBX(config)# ldapclient
IP-PBX(config-ldapclient)# name IPNext
IP-PBX(config-ldapclient)# host 127.0.0.1 389
IP-PBX(config-ldapclient)# ldap enable
IP-PBX(config-ldapclient)# end
IP-PBX#

This name will be display in SmartClient
Set host IP(local loopback) + 389 (port)

IP-PBX(config)# network-domain interface ip FastEthernet 0/0 domain public
IP-PBX(config)# network-domain interface ip FastEthernet 0/1 domain private

* FastEthernet 0/0 set Public IP with “Public "domain name
FastEthernet 0/1 set Private IP with “Private” domain name for terminal which have Public IP and Private.

User's Registration

1) Creating Subscriber's Extension Numbers

2) Creating Subscriber's Extension Users

3) Opening of Subscriber's Terminals

4) Registering Subscriber's Terminal

5) Trunk Setup(including Service Provider's signaling server)

6) Number Routing table

The Procedure for Registering Subscribers

Creating New Subscriber Numbers (Phone Number)

Registering to Subscriber Terminal (Terminal)

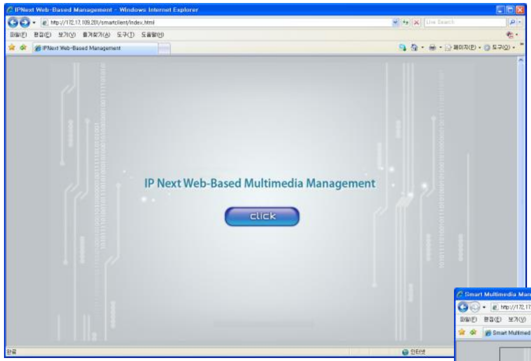
Assigning Subscriber Numbers to Subscriber Terminal

Creating Trunk (inbound/outbound SIP Server or TG)

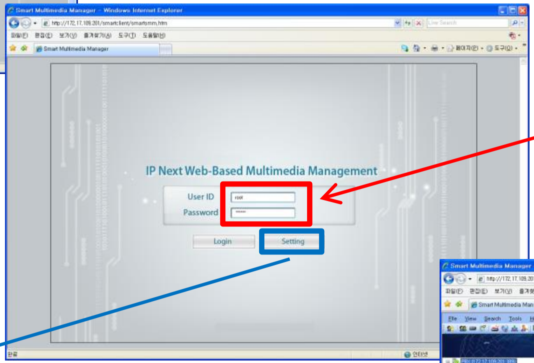
Setting up for Number Routing Map

Confirming Registration and Testing Calls

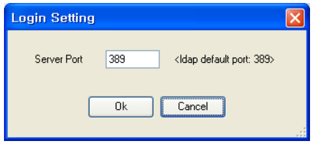
SMM Login

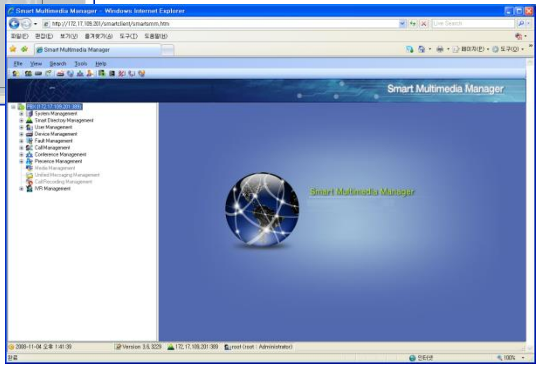


The Main Page
http://172.17.109.201 via explorer
“Click” Icon Click



ID : root
Password : router





It can change server port using Setting button

If you changed port, need to change IPNext also (via CLI)

SMM Main Page

Smart Multimedia Manager Operation

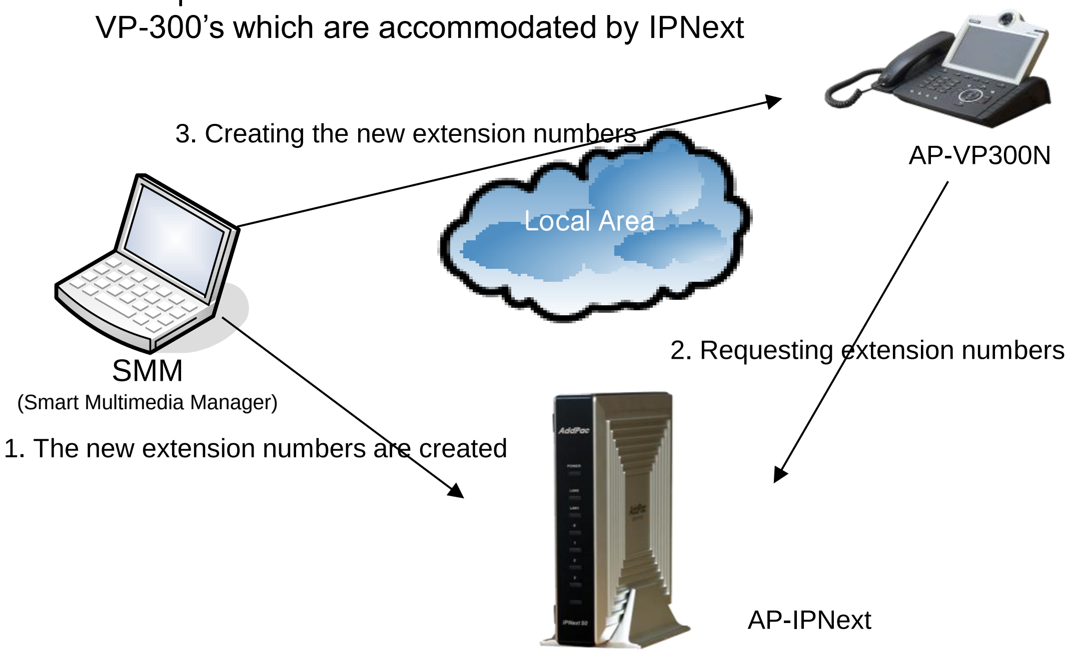
Creating New Subscriber Numbers (Overview of Creating New Extension Number Bands)

IPNext Series can create a highly sophisticated and flexible numeric system which are suitable to any client's organizational structure and operational policy.

Let's create the extension numbers for an imaginary client's company organizational structure which can be described below:

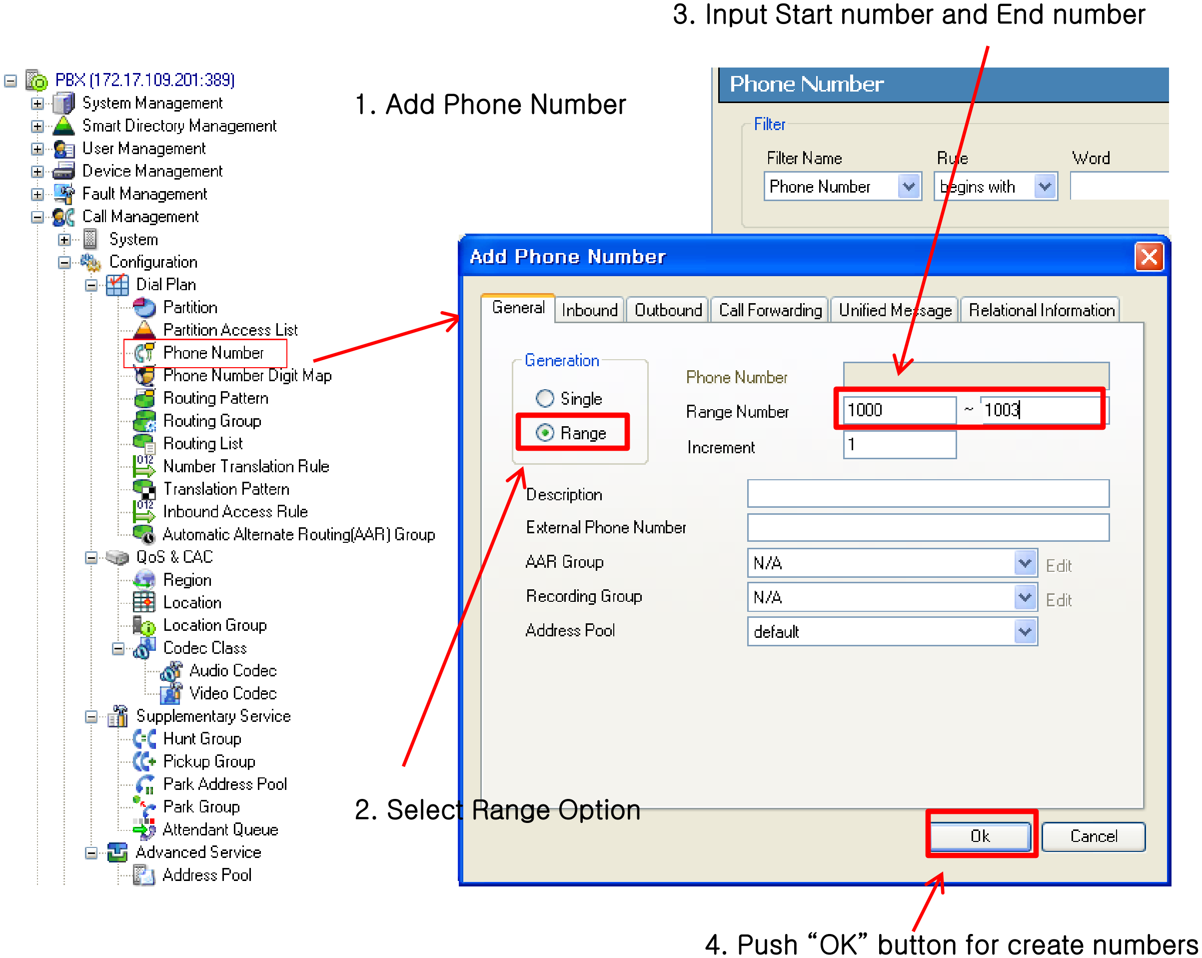
1. Each division will have 3 different extension numbers from X001 to X103
2. Each division of employees use VP-300's which are accommodated by IPNext
3. The operator allocates the numbers created for VP-300's which are accommodated by IPNext

- Business Administration (extension 1000~1003)
  - R & D (2000~2003)
  - Sales (3000~3003)
  - Production(5000~5003)
  - Technical Support(6000~6003)



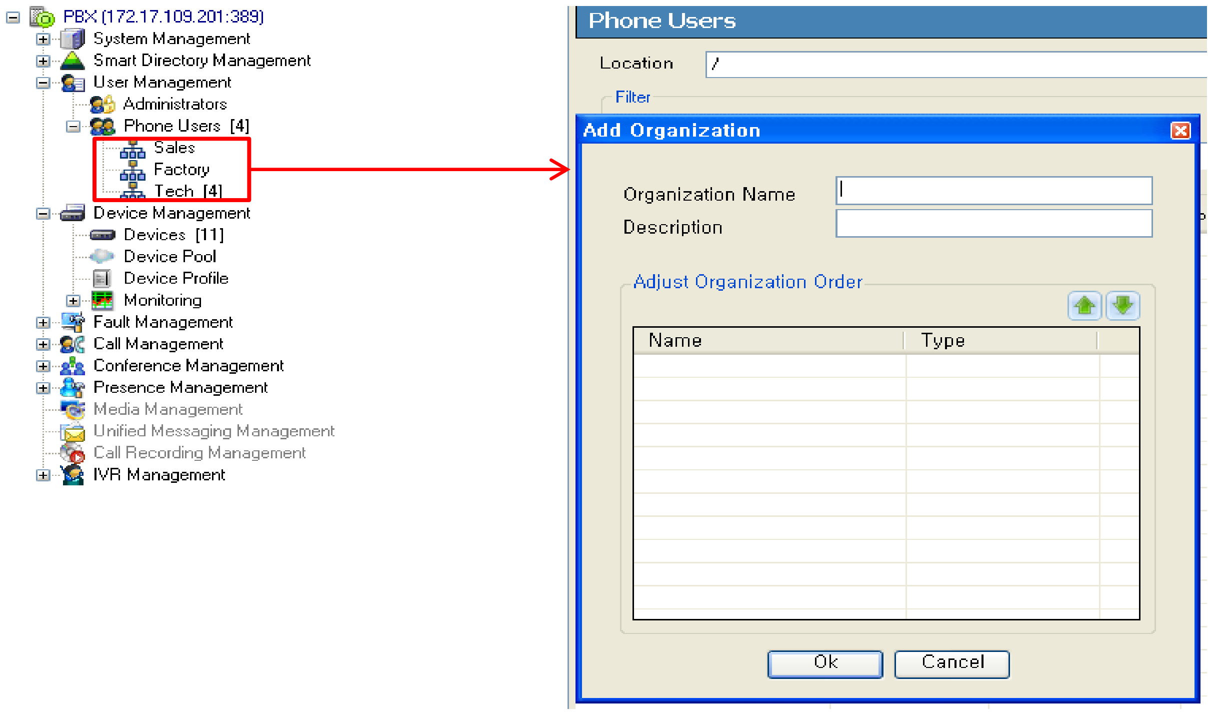
SMM – Call Manager(Phone Number)

- Select Add Phone number menu



SMM – User Management(Phone Users)

- User Group Create
  - Create user Group for management.

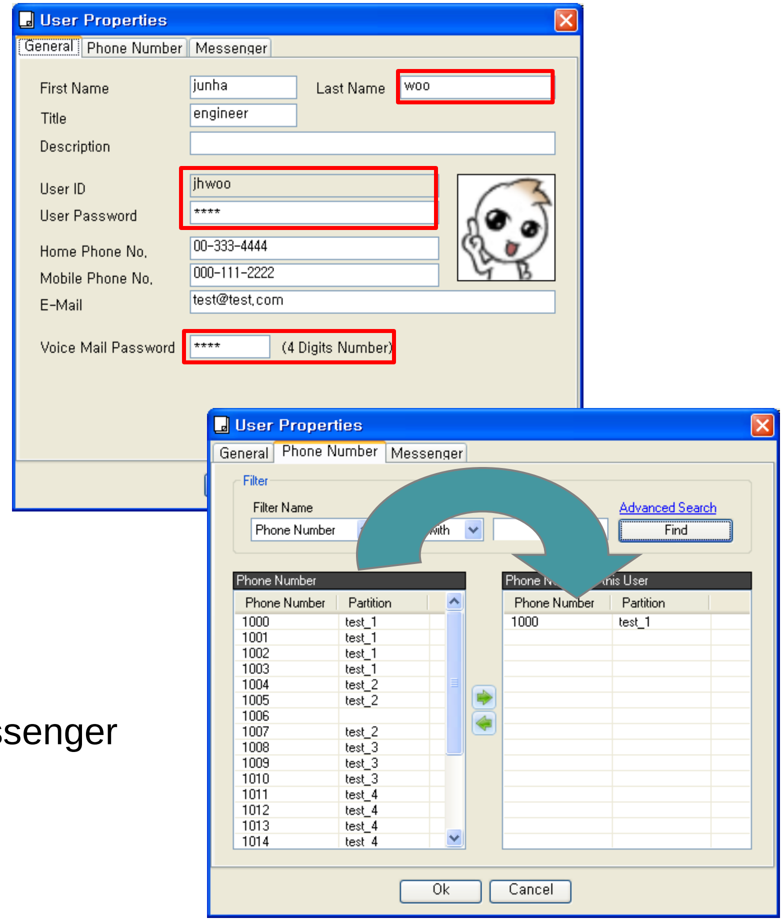


SMM – User Management(Phone Users)

1. Create User information
- Mandatory field
  - Last Name
  - User ID
  - User Password
  - Voice Mail Password

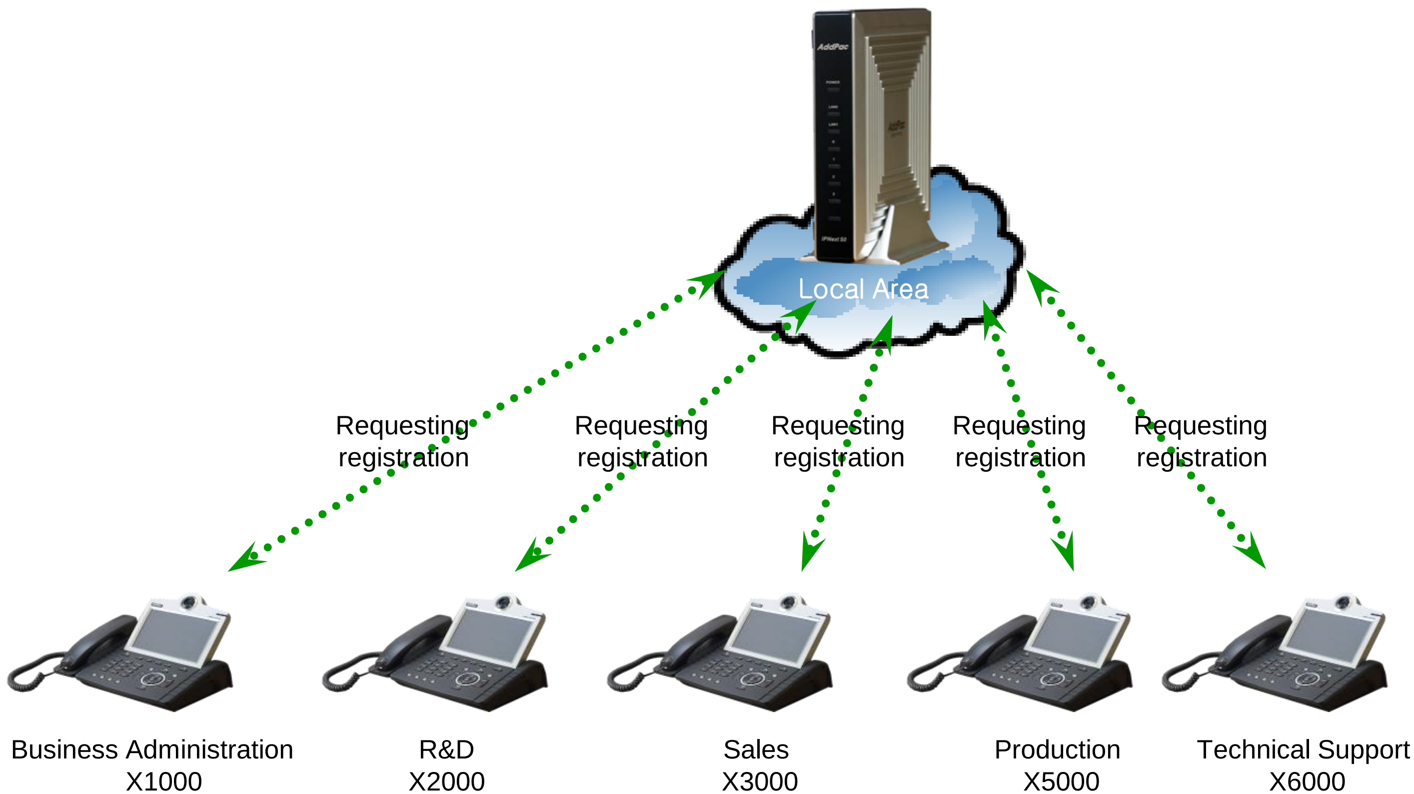
2. Assign phone number for each User
   - Select the number for user using find button

3. Set Nick name which will use in Smart Messenger



Smart Multimedia Manager Operation

1. A basic testing for call connection can be performed as to allocate the created number sources previously to 5units of AP-VP300N
2. Each IP phone can be set up by each separate IP and temporary setup files can be used for the basic setup
3. Please refer to the next page for more details of setting up IP Phone.



AP-VP300N setting

- Setup AP-VP300N

1) Go to network menu -> Internet
   select IPv4(DHCP, Static, PPPOE) or IPv6(Static, EUI-64, Auto Configuration)

2) Go to network menu -> SSCP
   Input IPNext(Master)'s IP-address in Call-manager 1 field
   change option of SSCP setup(off => on)

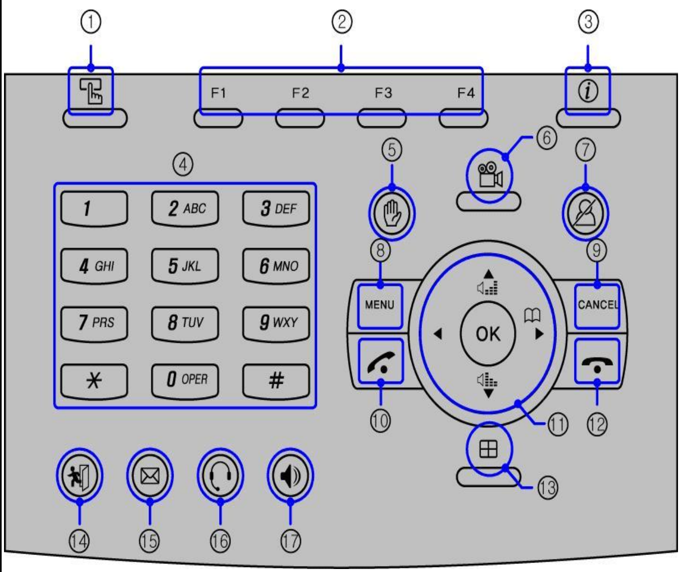
3) Go to network menu -> Presence -> Presence Server
   input IPNext's IP-address in Presence Server 1 field
   input "5051" in Server1 port

4) Go to network menu -> Presence -> Presence Setup
   input User ID, User password then set Presence Enable to "On"

AP-VP300N Button information

Creating New Subscriber Numbers (testing call connection between extension numbers-Setting up AP-VP300N by using OSD)

Description for Overall Key Features of AP-VP300N

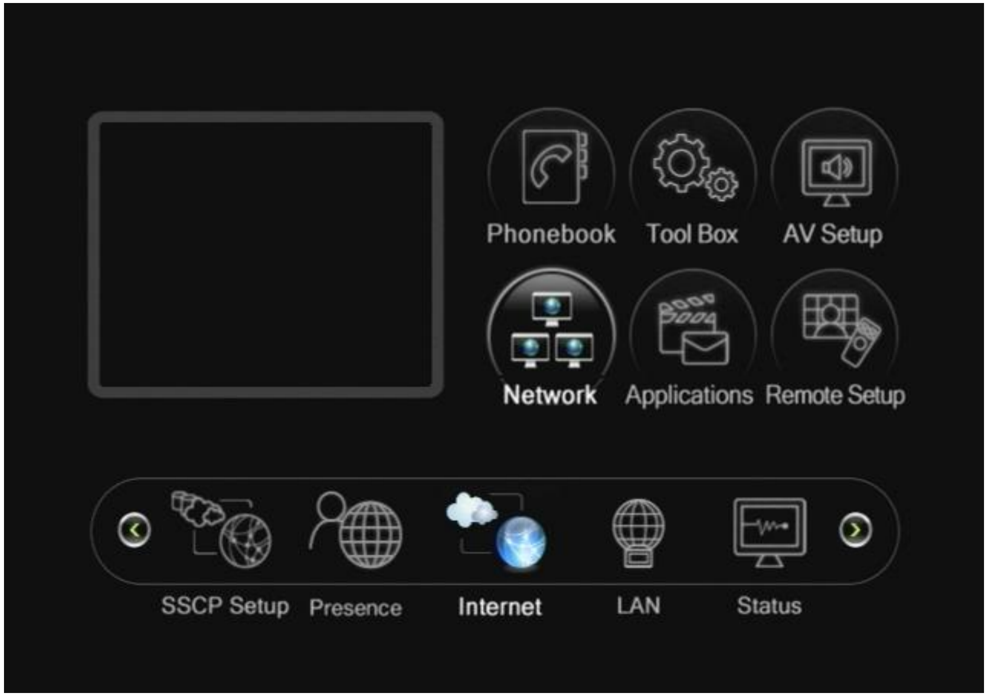


No.	Keys	Features
(1)	SpeedDial	Brining a menu for speed dial
(2)	F1~F4	When you use Phone Book or Speed Dial, these functions are assigned to the touch screen, which is located in the lower part of the screen.
(3)	INFO	While on the line, you can select whether to display information
(4)	Number Key	Dialing by using remote control. Used for entering each value of setup of OSD menu
(5)	Hook Flash	Hold key for headset or speaker phone during phone conversation
(6)	Camera	Used for turning on/off the camera screen
(7)	Privacy	While on the line, the other person is not able to see you
(8)	Menu	Brining OSD main menu
(9)	Cancel	Used for moving to a higher category in OSD menu.
(10)	Call	Used for brining call history from the latest and calling by pressing number keys
(11)	방향키, OK	Checking and moving with in OSD menus or adjusting camera movement
(12)	END	To end on line call
(13)	View	Changing conversation view/ remote view/ local view while on the line
(14)	Absence	Used during the user's absence (to be released)
(15)	Voice Mail	To listen to voice mail (to be released)
(16)	HDP Call	For using headset
(17)	SPK Call	To use imbedded speaker phone

AP-VP300N OSD setting

Creating New Subscriber Numbers (testing call connection between extension numbers-Setting up AP-VP3 00 by using OSD)

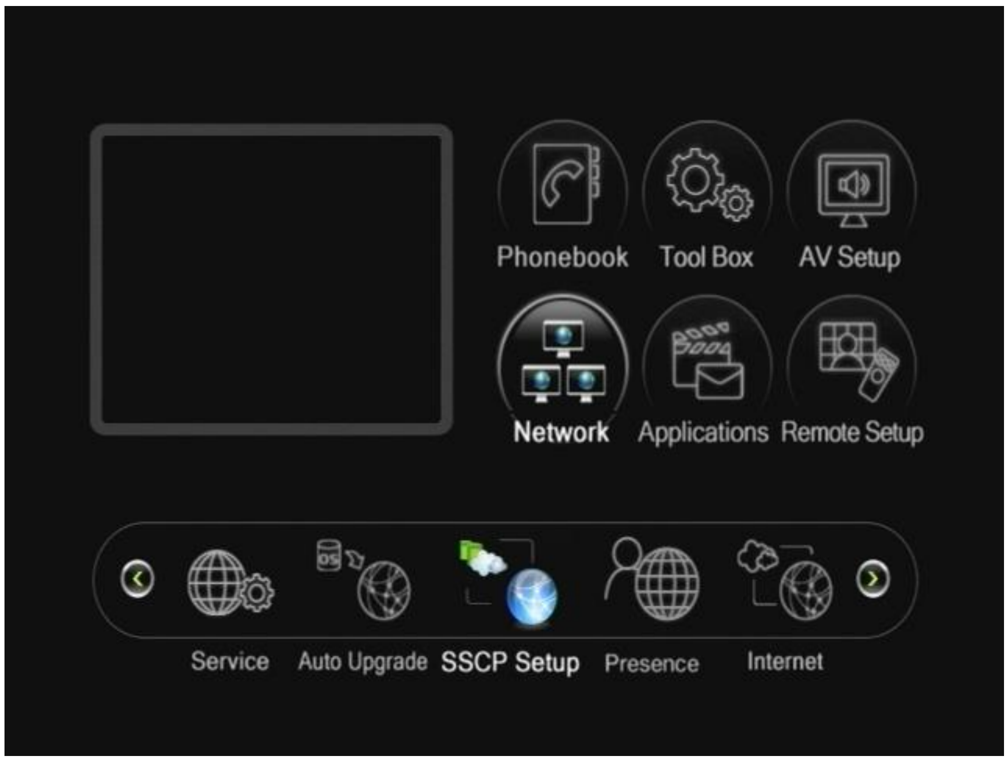
1) Entering <Main Menu> by pressing menu key
2) Moving to <Network> by using direction key.
3) Moving to <Internet> by using direction key(left, right) Pressing “OK” key to enter <Internet> sub-menu



AP-VP300N OSD setting

Creating New Subscriber Numbers (testing call connection between extension numbers-Setting up AP-VP3 00 by using OSD)

1) Moving to <SSCP Setup> by using direction key
2) Pressing OK key to enter <SSCP Setup> sub-menu



AP-VP300N OSD setting

Creating New Subscriber Numbers (testing call connection between extension numbers- Setting up AP-VP3 00 by using OSD)

1. Pressing ‘1’ to change to ‘release’ from <SSCP setup>
2. After moving to <Call-manager1>, IP addresses of Call-Manager (IP-PBX) can be entered.
3. After returning to <SSCP setup>, ‘setup’ can be changed by pressing ‘1’
4. Completing <SSCO Setup> by pressing ‘OK’ key

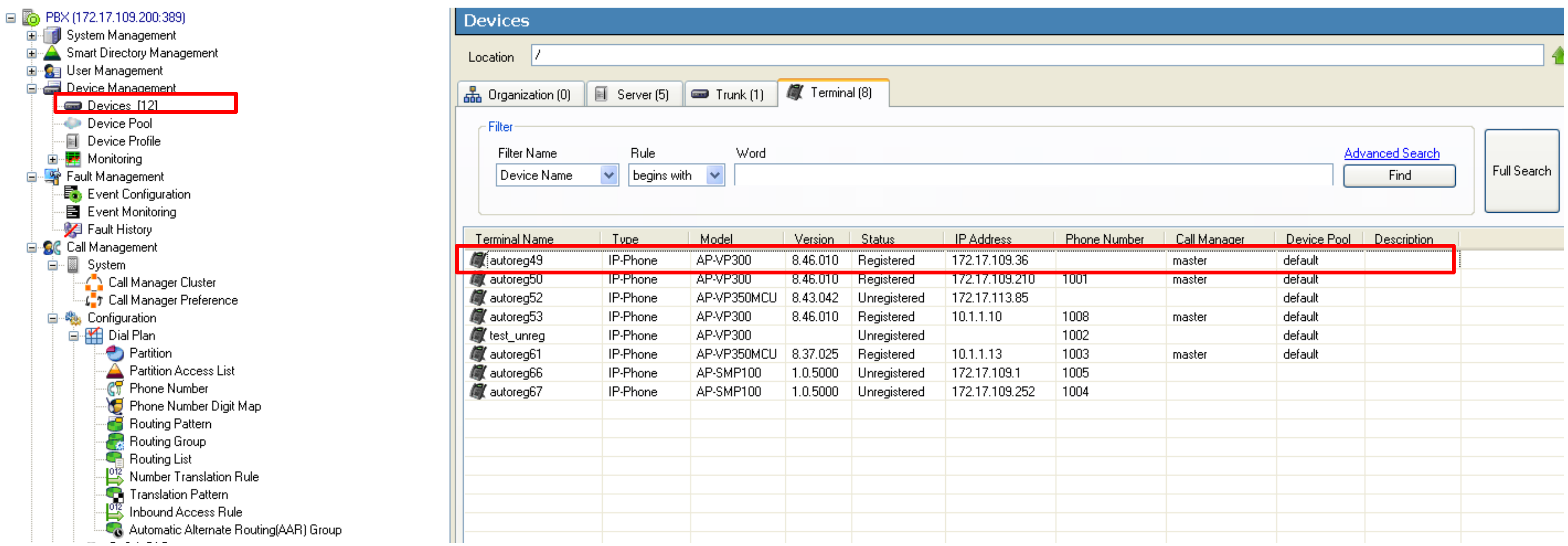
AP-VP300N OSD setting

Creating New Subscriber Numbers (testing call connection between extension numbers-Setting up AP-VP3 00 by using CLI)

```
configure terminal
 osd
  sscp enable
 sscp
  call-manager 1 10.1.1.1 5060
```

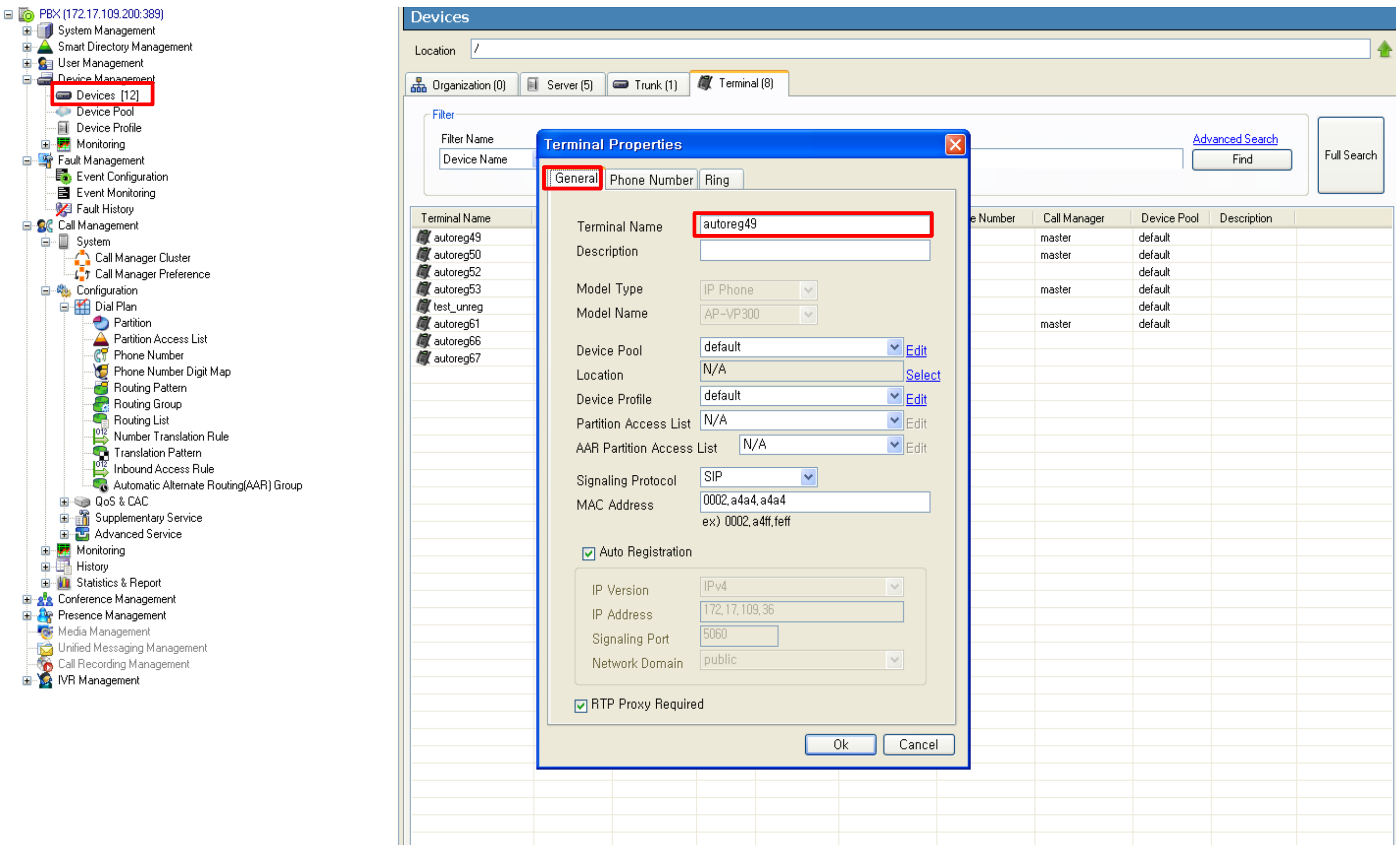
SMM - Device Management

1. You can request to register those terminals, which has been set up, to IPNext.
2. You can make reference of this registration by entering ‘Device Management > Terminals’ which is shown below:



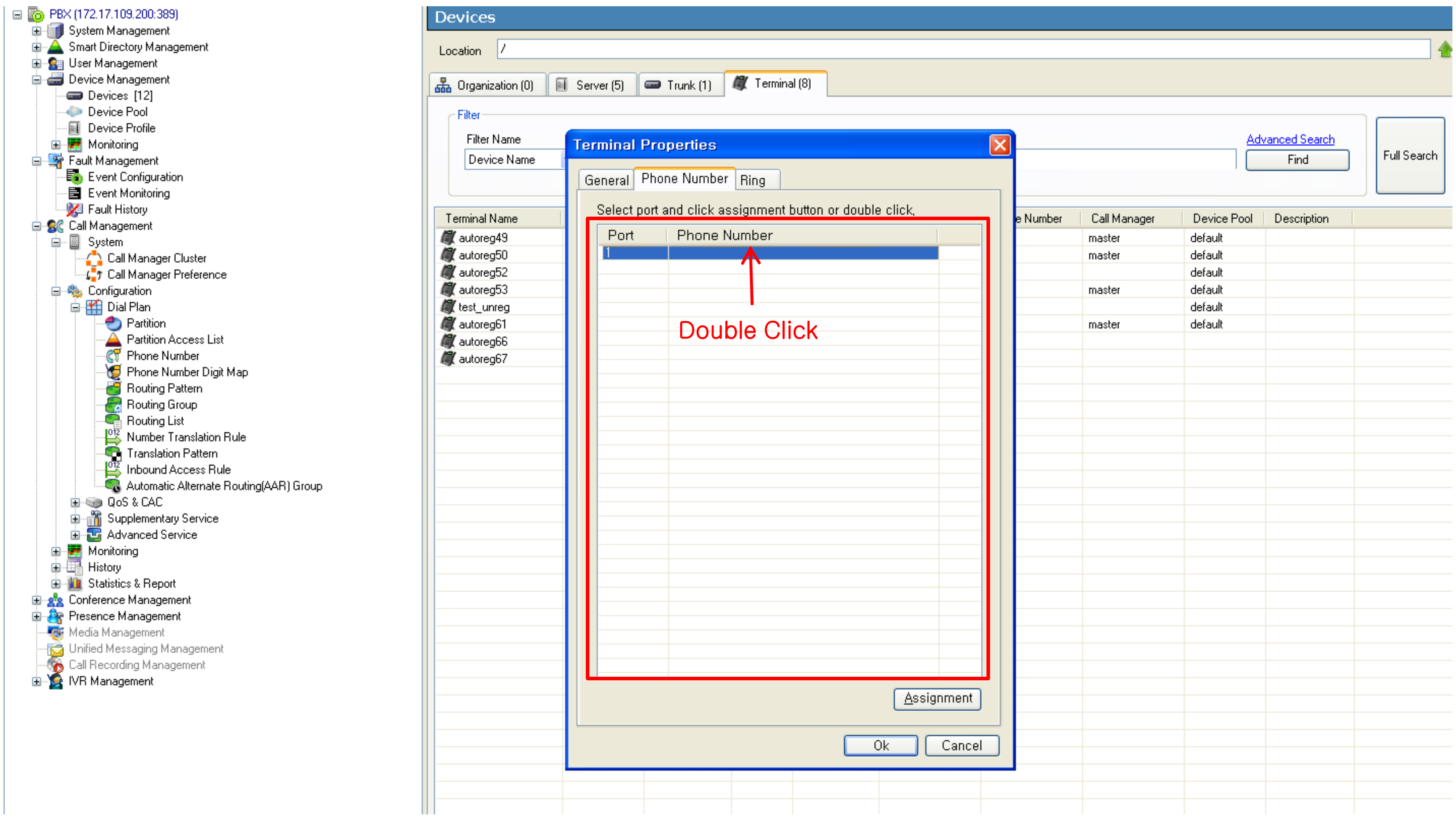
SMM - Device Management

1. If you double- click ‘Terminal Name’ you can see the window of ‘Terminal Properties’ opens up as it is shown below:
2. From ‘General’ tab, the basic setup can be made for naming the terminal and automatic registration.
3. After the setup is made in the screen below, you may double-click ‘Phone Number’ for assigning the numbers.



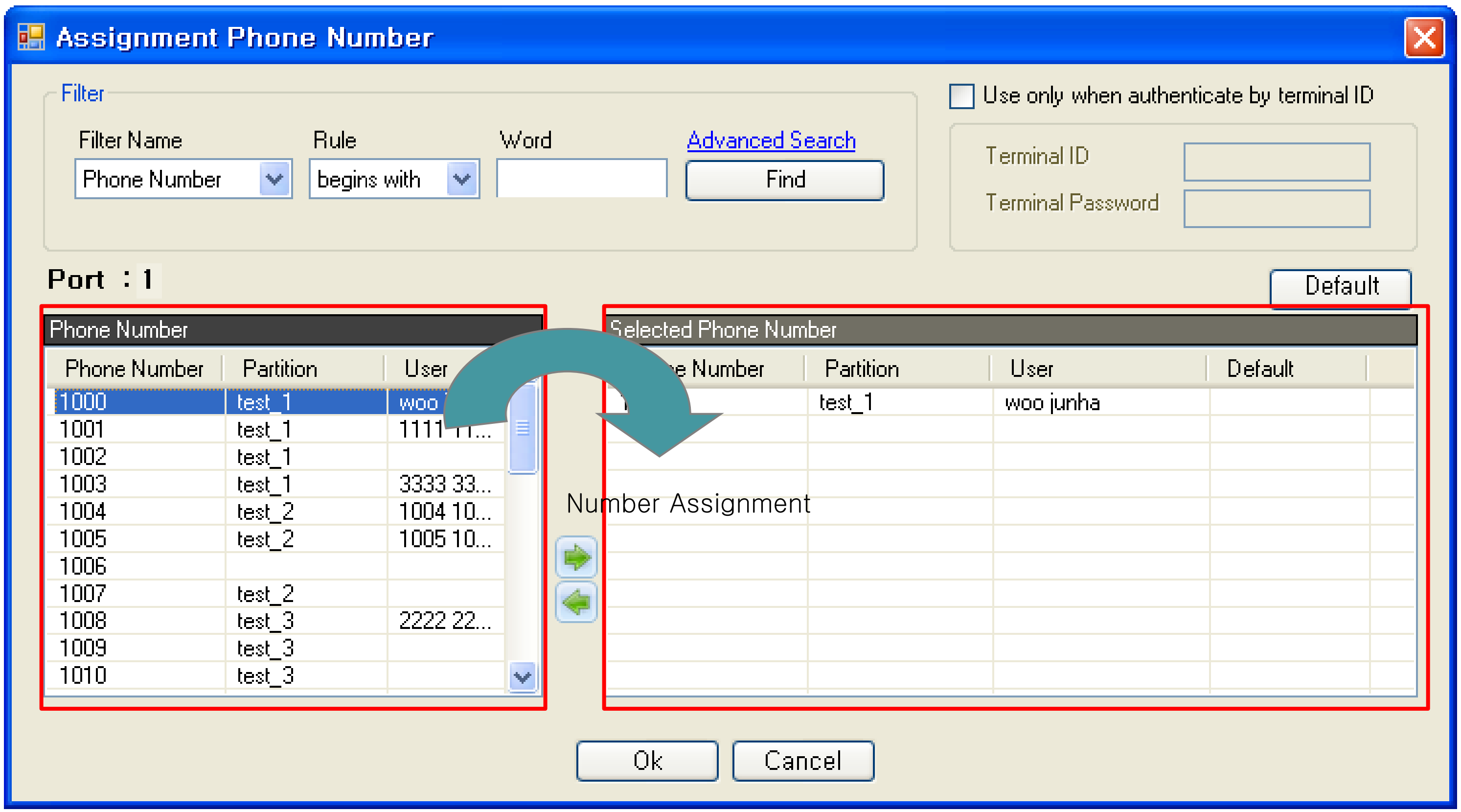
SMM - Device Management

1. The designated port of the terminal can be selected, then the number sources can be referenced by a double-click.



SMM - Device Management

1. Selecting the number to be assigned to a designated terminal as it is shown in the picture below:

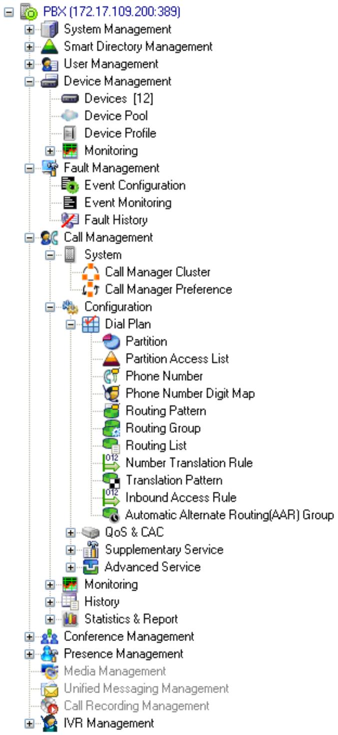


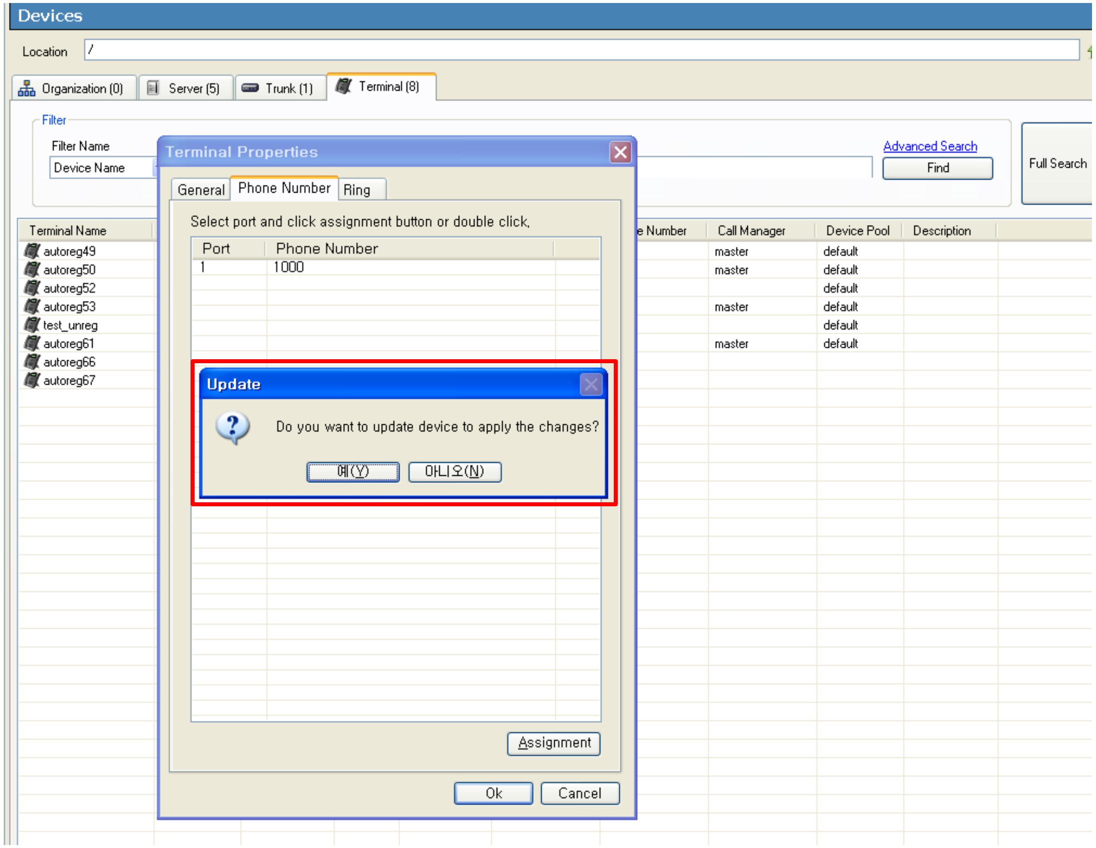
Supporting list for recently subscribed numbers

The number to be assigned to a designated terminal

SMM - Device Management

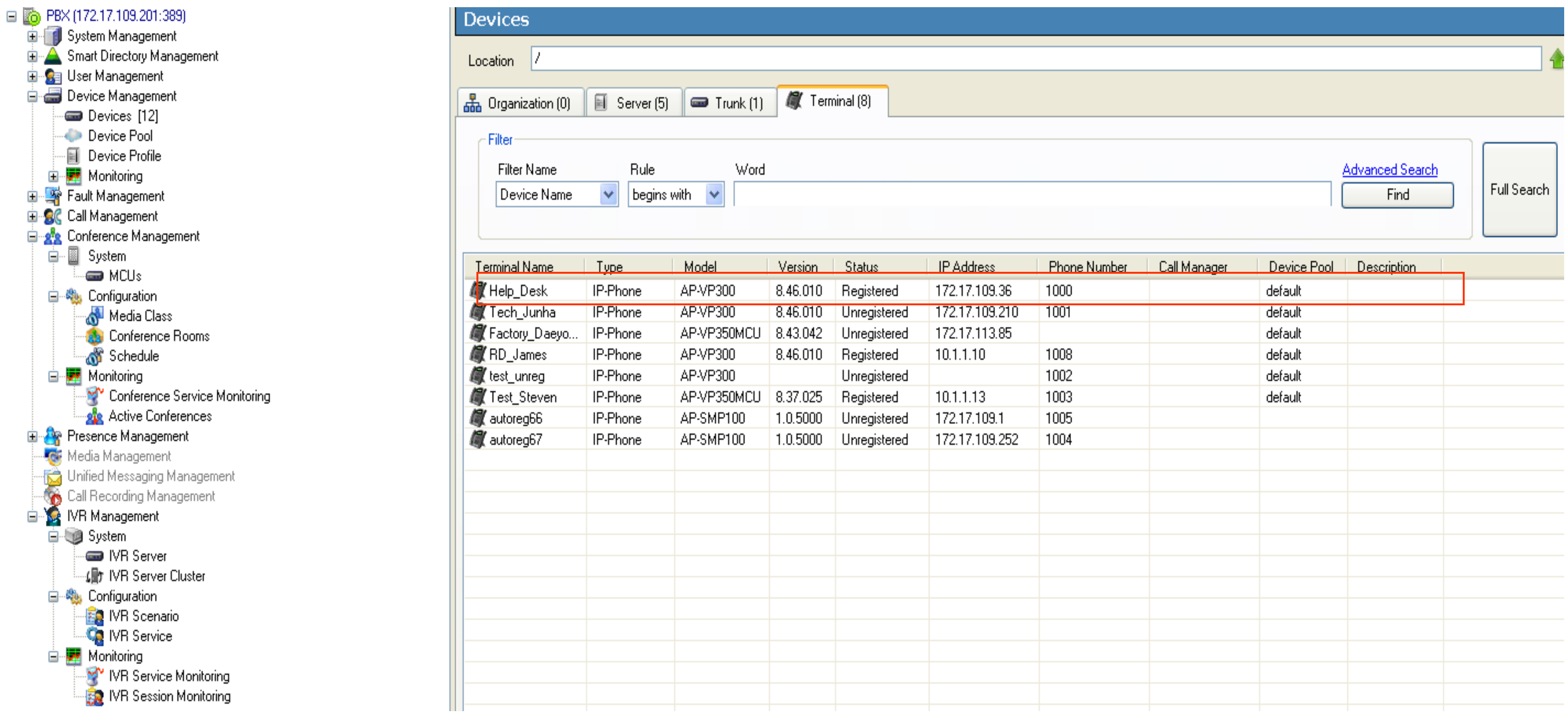
1. Confirming the assigned number as it is shown in the picture below
2. Through the procedure of confirmation (authorization granted) by the operator, the designated number can be set automatically to IP Phone





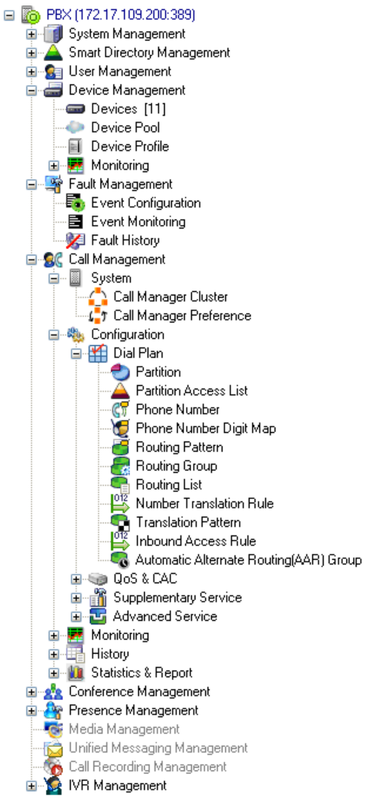
SMM - Device Management

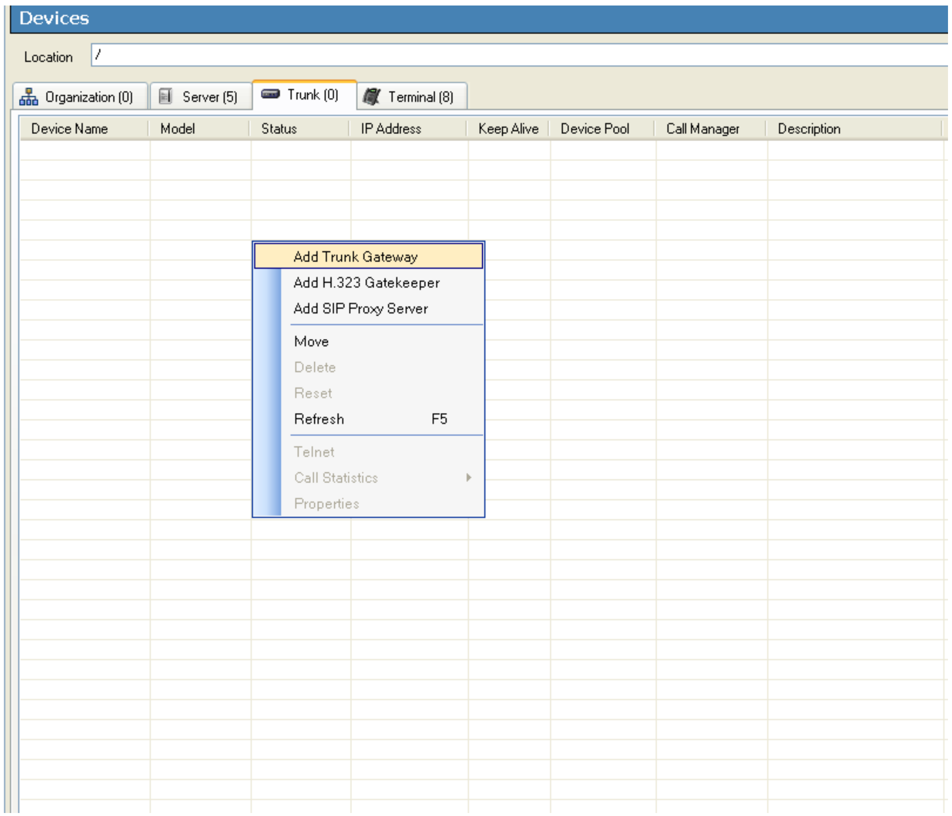
1. After finishing assignment to the designated terminal as it is shown in the picture below, it completes the process of registering the terminal.



SMM - Trunk

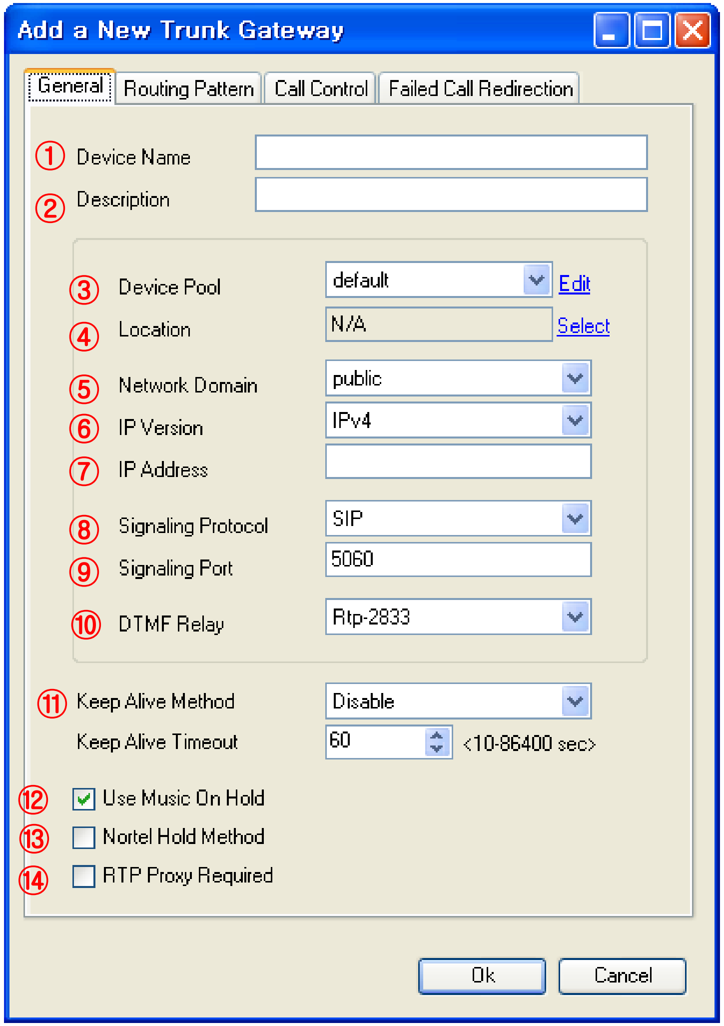
Trunk Gateway is an equipment to take the call, in case the inside user makes an outgoing call. IPNext can be used as an inbound Call Manager, making an outgoing call to outside by other equipment (VoIP, PSTN) is possible after registration to Trunk Gateway. For making an outgoing call or taking an incoming call, the setup of Trunk Gateway is mandatory. Routing Pattern needs to be set up for fixing the telephone number to be dialed for the user to be connected with the external terminal. In general, when you want to exit to outside, you may press the number 9 and this is one of the cases. The picture below shows the screen and registration information for Trunk Gateway to work.





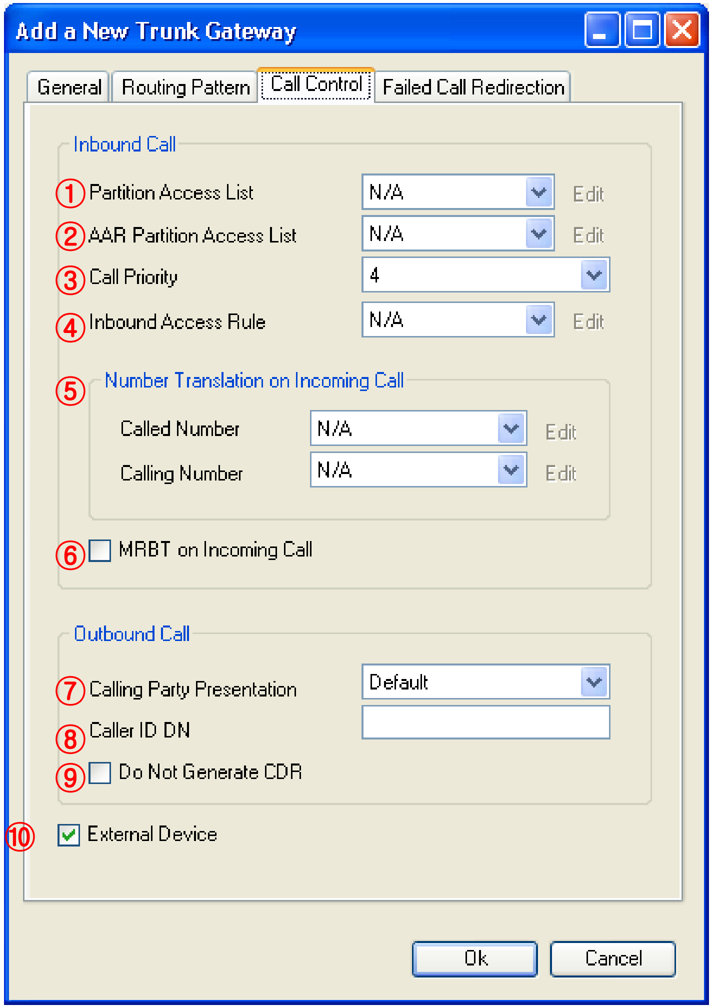
SMM - Trunk

SMM-Device Management-General Registration to Trunk Gateway



Ref.	Description
1	Registering a name of Trunk Gateway .
2	Registering a description of Trunk Gateway
3	Selecting 'Device Pool' (Basically set to default)
4	Selecting 'Location' (basically no setup is provided)
5	Selecting 'Network Domain' for Call Manager
6	Selecting IP version
7	Registering 'IP Address'
8	Selecting VoIP protocol
9	Entering a designated port for VoIP protocol
10	Setting up a type of 'DTMF-Relay' to be transmitted to Trunk -Transmitting DTMF based on Rtp-2833 : RFC-2833 standards -In-band: Transmitting DTMF tone through RTP -Transmitting DTMF by using SIP INFO, H.245 Signal
11	Keep Alive Method Option -Disable : don't check Trunk Gateway status -Register : check Trunk Gateway's status using SIP REGISTER messages from Trunk Gateway -Option : check Trunk Gateway's status using SIP REGISTER messages from Trunk Gateway
12	This is a setup to provide MOG for the terminal requesting hold
13	This is a set to provide MOH for interoperability with Nortel's soft switch
14	This is a setup by force, for using RTP Proxy while the telephone is on the line

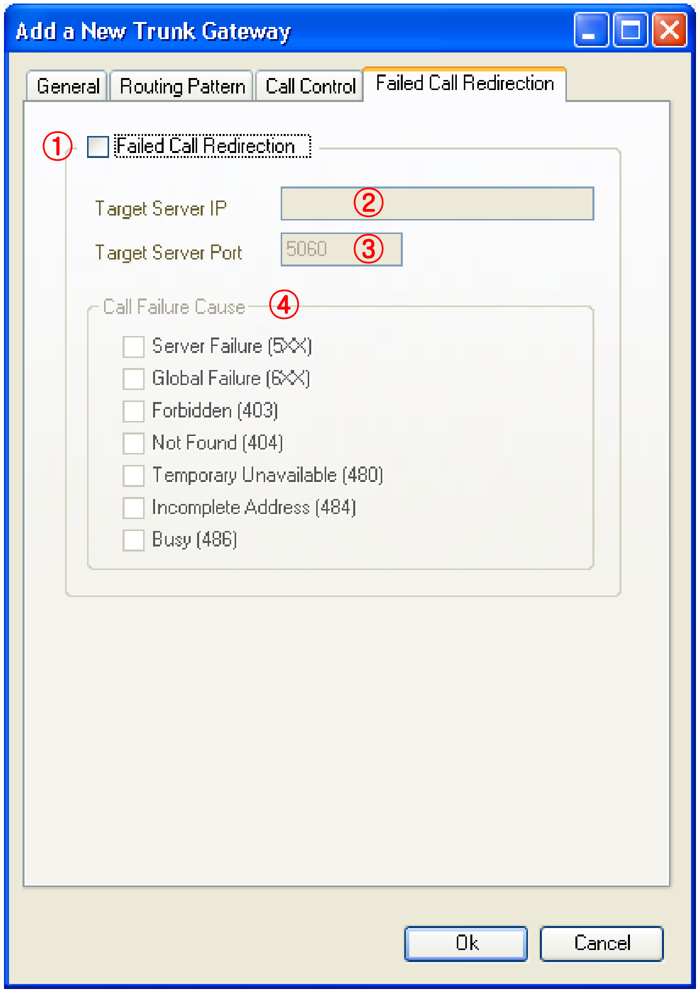
SMM-Device Management-Trunk Gateway- Registering Information of Call Control



Ref.	Description
1	Selecting Partition Access List for setting up authorization of inbound call.
2	Selecting Partition Access List for setting up AAR(Automatic Alternate Routing)
3	Setting up priority for inbound call
4	Selecting Inbound Access Rule for setting up authorization of inbound call rejection using calling number pattern
5	Applying Translation Rule for outgoing and incoming call numbers to change number for inbound call.
6	Selecting an option of provisioning MRBT(Ring Back Tone) for inbound call
7	Selecting an option of presenting a number of outgoing call to outbound call Default: Default setting of Call Manager (by following Call Manager Cluster > Options > Calling Party Presentation) Allowed: Presenting my number to the other Restricted : Not displaying my number to the other
8	Setting up a number of outgoing call by force (ie.: Representative Number)
9	Setting for don't generate CDR information, when you use more than 2 IP-PBX as Trunk, don't generate outbound call for each Trunk(IP-PBX) (if you don't set this option, the user's can be charged both of IP-PBX)
10	Setting up an option for whether Trunk is to interoperate with outbound

SMM - Trunk

SMM-Device Management-Trunk Gateway- Registering Information of Failed Call Redirection

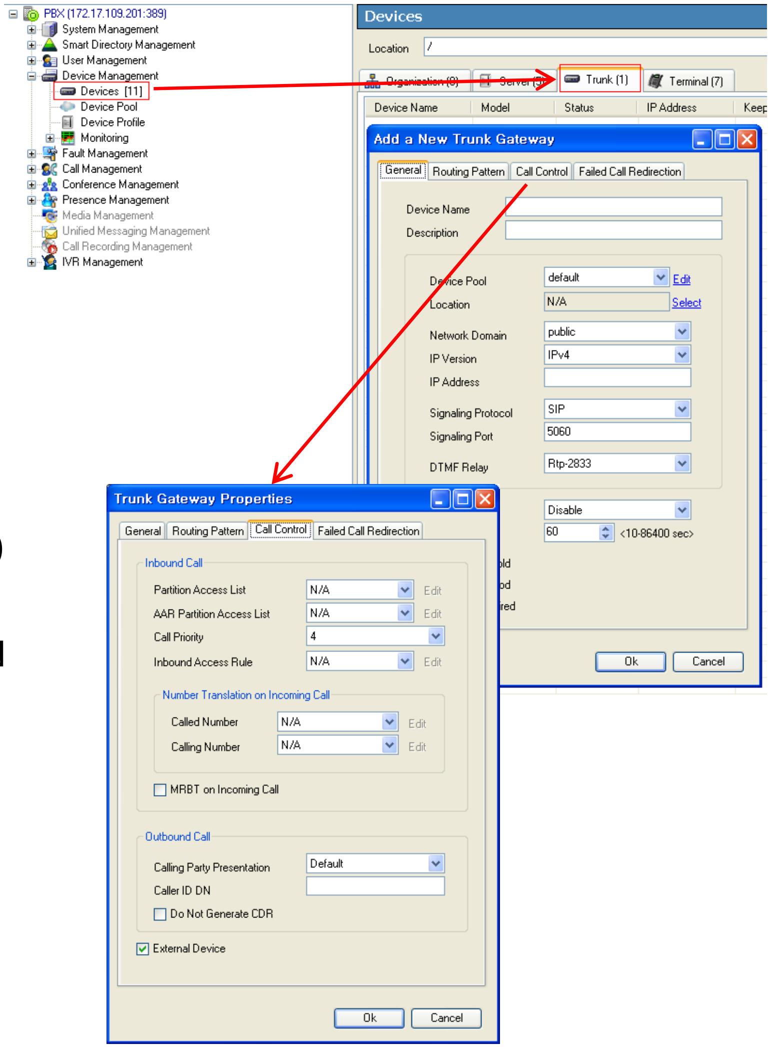


Ref.	Description
1	Set for Failed Call Redirection (SIP 302 Moved Temporary) function
2	Set IP-address for IP-PBX get happen Failed Call Redirection Messages to transfer
3	Set Server’s port information when IP-PBX send Redirection call
4	Set redirection function using specific reason

SMM – Trunk(example)

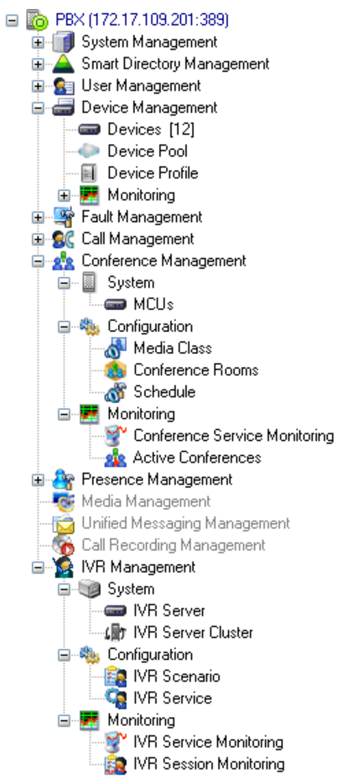
- General Tab
  a. Input Device Name
  b. Select Network Domain (Public / Private)
     - It depend on T/G's IP address.
  c. Set IP-address
  d. Select Signaling Protocol(SIP / H.323)

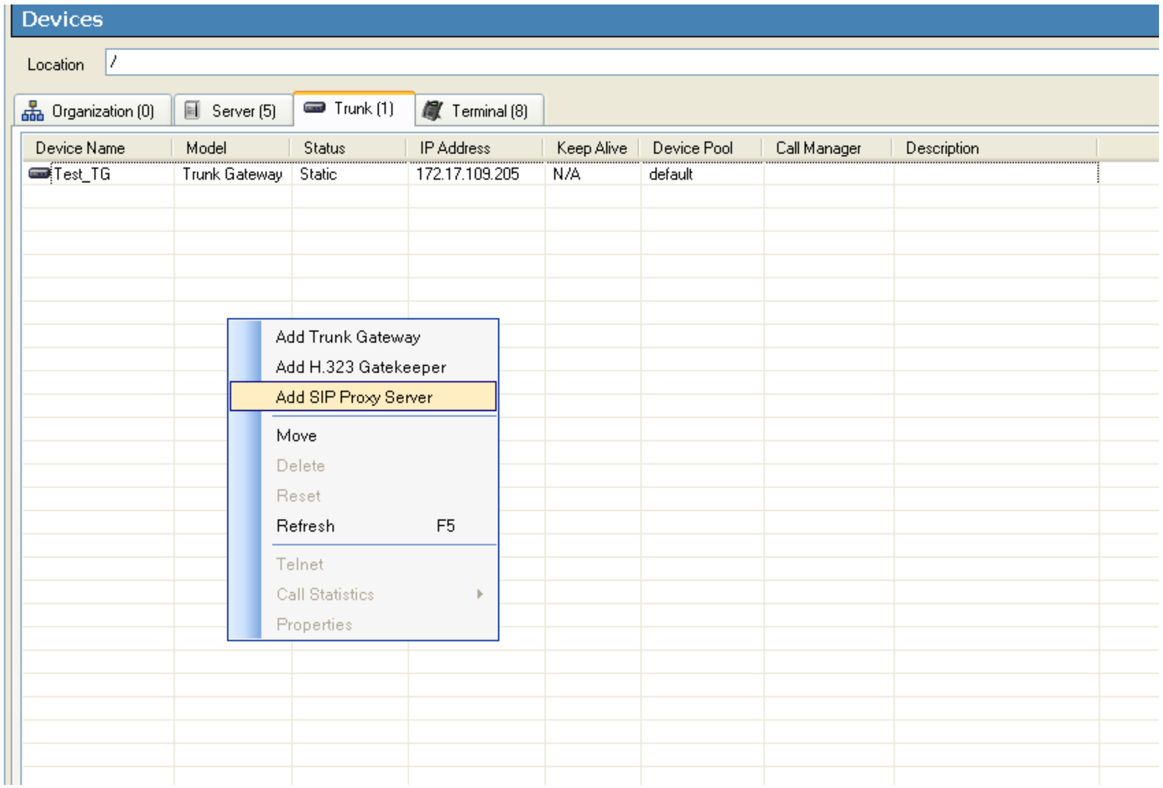
- Call Control Tab
  a. Inbound Call
     I. Select Number Translation(Incoming call)
  b. Outbound Call
     I. Calling party presentation or Caller ID DN



SMM - Trunk

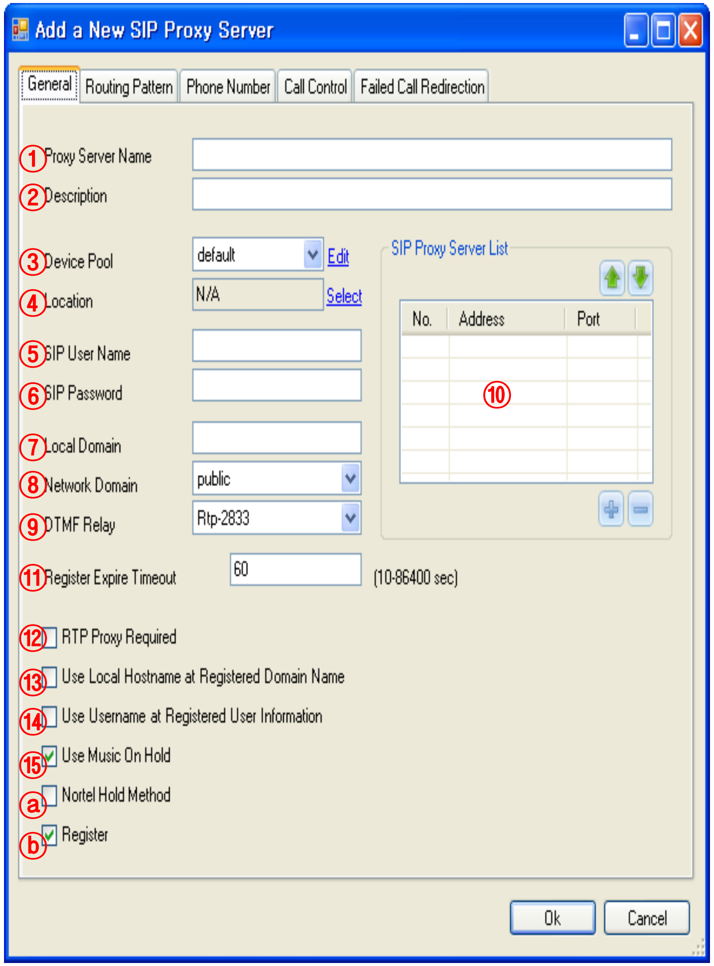
This is a function to register SIP Proxy Server for interoperating with the other SIP Proxy Server in outbound.
For example, when an outgoing call is made by registering a company telephone number to SIP Proxy Server, this function can be set up to enable the call which has been registered to SIP Proxy Server.





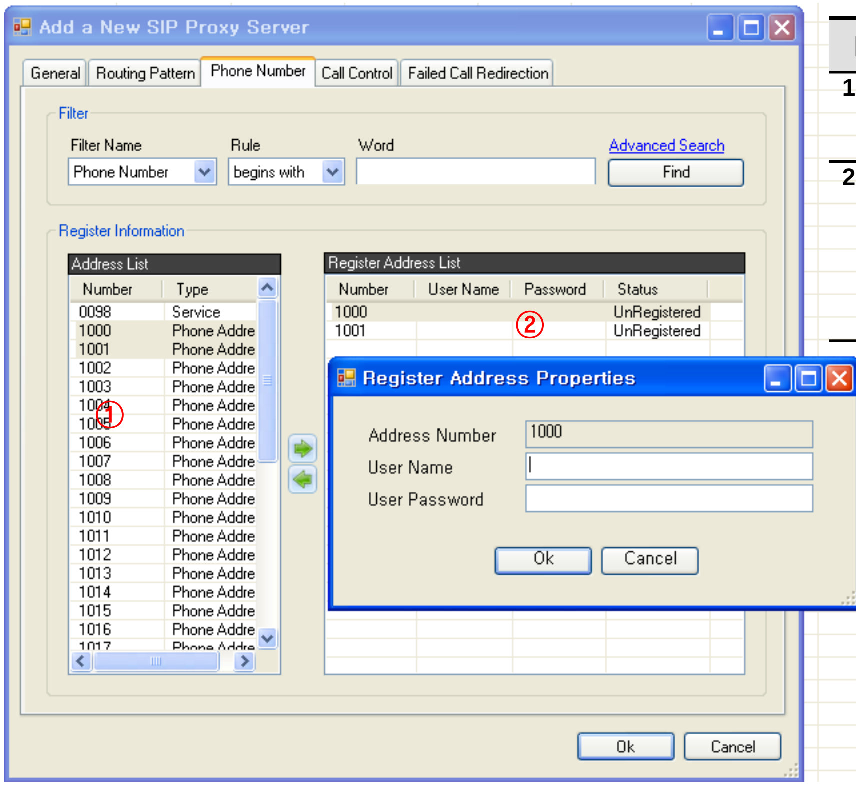
SMM - Trunk

SMM-Device Management-SIP Proxy General Registration



Ref.	Description
1	Registering a name of SIP Proxy Server
2	Registering a description of SIP Proxy Server
3	Selecting ‘Device Pool’ (Basically set to default)
4	Selecting ‘Location’
5	Registering SIP User Name
6	Registering SIP password
7	Registering ‘Local Domain’.
8	Selecting ‘Network Domain’.
9	Setting up a type of ‘DTMF-Relay’ to be transmitted to Trunk -Transmitting DTMF based on Rtp-2833 : RFC-2833 standards -In-band: Transmitting DTMF tone through RTP -Transmitting DTMF by using SIP INFO, H.245 Signal.
10	IP address and protocol list of SIP Proxy Server can be set up and priority level can placed
11	The setup value of Register Expire time to be used for registering SIP Proxy Server
12	This is a setup by force, for RTP Proxy to be used while telephone is on line.
13	Using local host name instead of domain name of Proxy Server for registration: To : 7000@local hostname From : 7000@local hostname
14	Registering 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 a setup to provide MOH for hold request from a terminal
A	This is a set up to provide MOH for interoperability with Nortel soft switch
B	This a set up whether to ‘Register’ or not for the Phone Number to be registered to Proxy Server

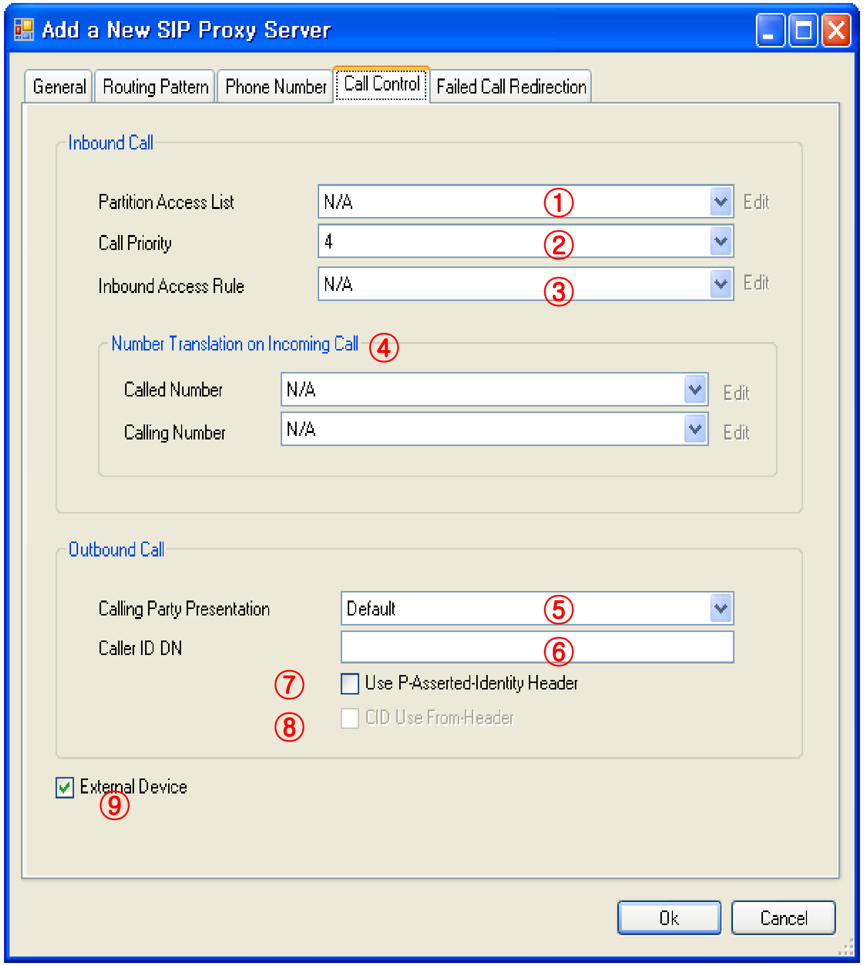
SMM-Device Management-Registering Information of SIP Proxy Phone Number



Ref.	Description
1	The list of telephone numbers which are not registered to SIP Proxy Server
2	The list of telephone numbers to be registered to SIP Proxy Server ●User Name: Registering user name for the telephone number to be authorized Password: Registering password for the telephone number to be authenticated ●Status: status of registering the telephone number

SMM - Trunk

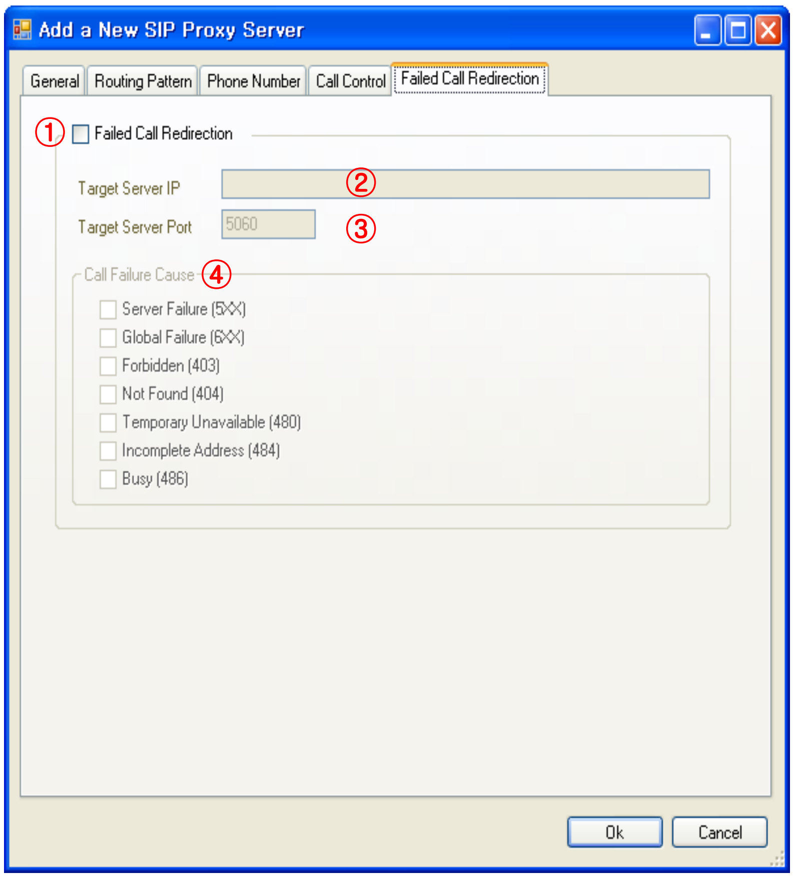
SMM-Device Management-Registering SIP Proxy Call Control



Ref.	Description
1	Selecting Partition Access List for authority setup for inbound call
2	Setting up priority level for the inbound call
3	Selecting Inbound Access Rule for setting up authorization of inbound call rejection using calling number pattern
4	Applying Translation Rule to outgoing and incoming call numbers to change the number for the inbound call.
5	Selecting an option of presenting a number of outgoing call to outbound call Default: Default setting of Call Manager (by following Call Manager Cluster > Options > Calling Party Presentation) Allowed: Presenting my number to the other Restricted : Not displaying my number to the other
6	Setting up a number of outgoing call by force (ie.: Representative Number)
7	Setting up whether to use P-Asserted-Identity header or not
8	Setting up whether to use P-Asserted-Identity in SIP From header field
9	Setting up whether Trunk is to interoperate with outbound or not

SMM - Trunk

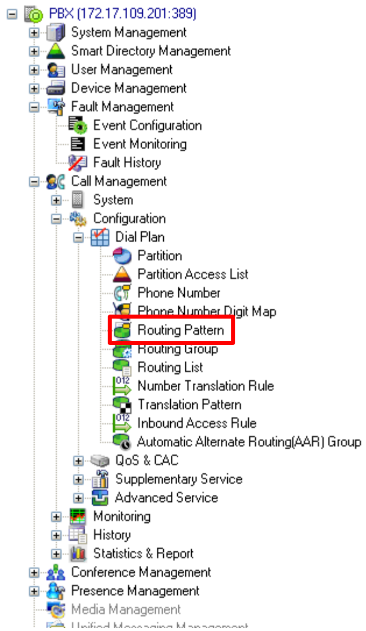
SMM-Device Management-Registering SIP Proxy Failed Call Redirection

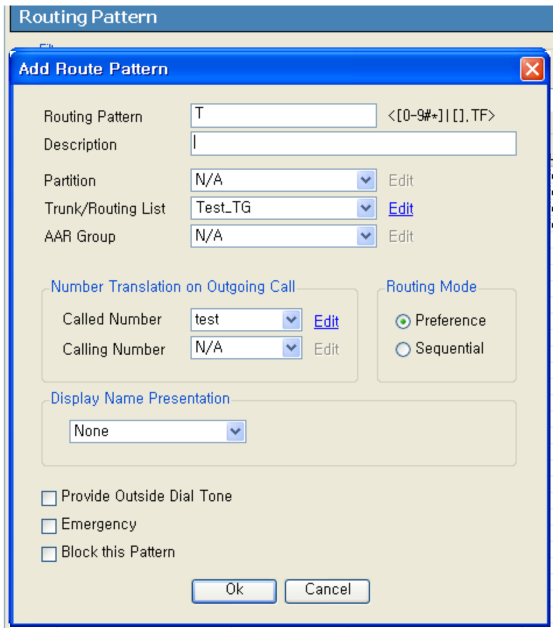


Ref.	Description
1	Set for Failed Call Redirection (SIP 302 Moved Temporary) function
2	Set IP-address for IP-PBX get happen Failed Call Redirection Messages to transfer
3	Set Server's port information when IP-PBX send Redirection call
4	Set redirection function using specific reason

SMM - Routing Pattern

- Add Routing Pattern
  - Input routing pattern
    - Lind line
    - Mobile
    - Toll
    - International
    - Etc
  - Select Trunk
  - Select number translation rule





- Provide outside Dial Tone
  - If client want to use out-side dial-tone which provide(Sip-server or G/K) need to check.
- Block this Pattern
  - Set this routing pattern for blocking
  - Users can’t use this routing pattern

SMM - Number Translation Rule

- Create Number Translation rule

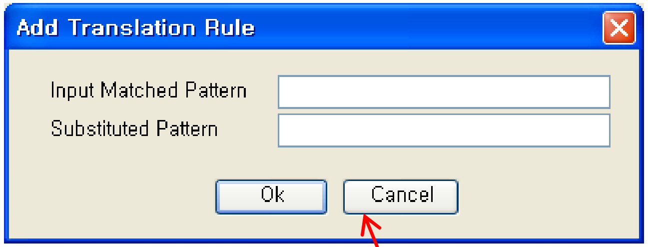
Add Translation rule

a. Input name

b. Add translation rule

I. Input Matched Pattern (a condition)

II. Substituted Pattern(a result)



Ex)

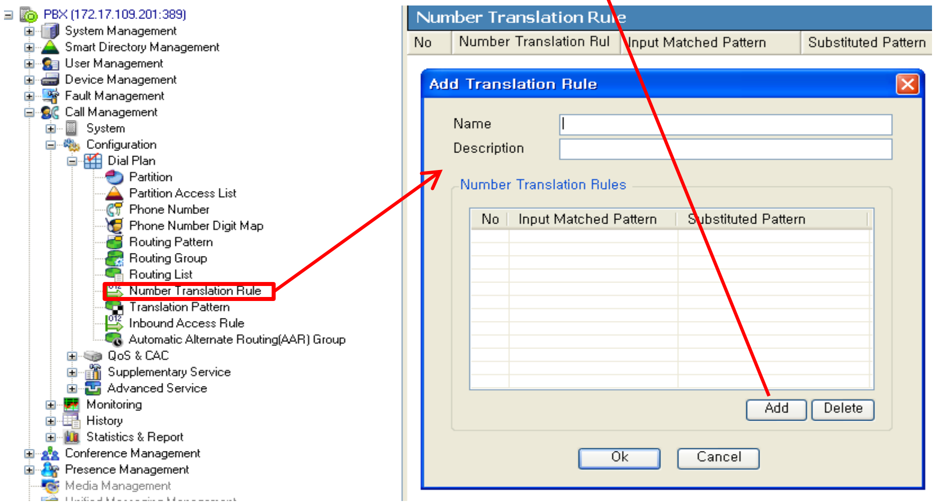
Input Matched Pattern [2-9]T

Substituted Pattern 02%01%99

User input output

2345 -> 022345

1234 -> 1234



T = no limitation(All number)

. = single digit (any number)

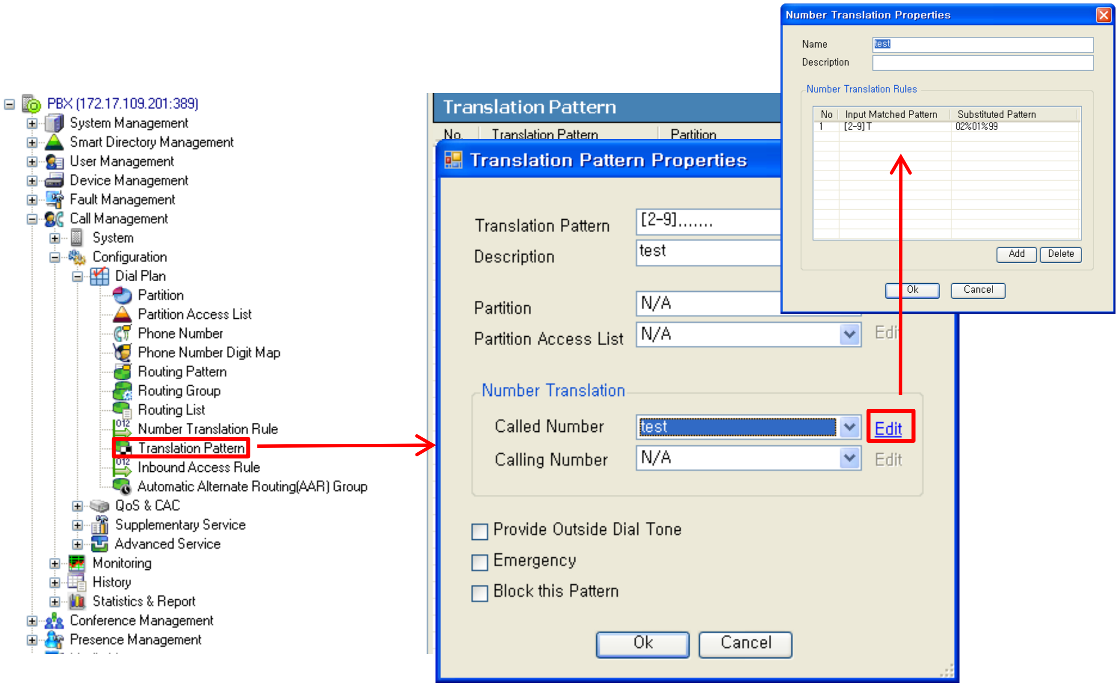
[] = range

SMM - Number Translation Rule

- Create Translation Pattern
  a. Input Translation pattern
  b. Select number translation (can select translation rule before you made it)
    I. Called Number
    II. Calling Number

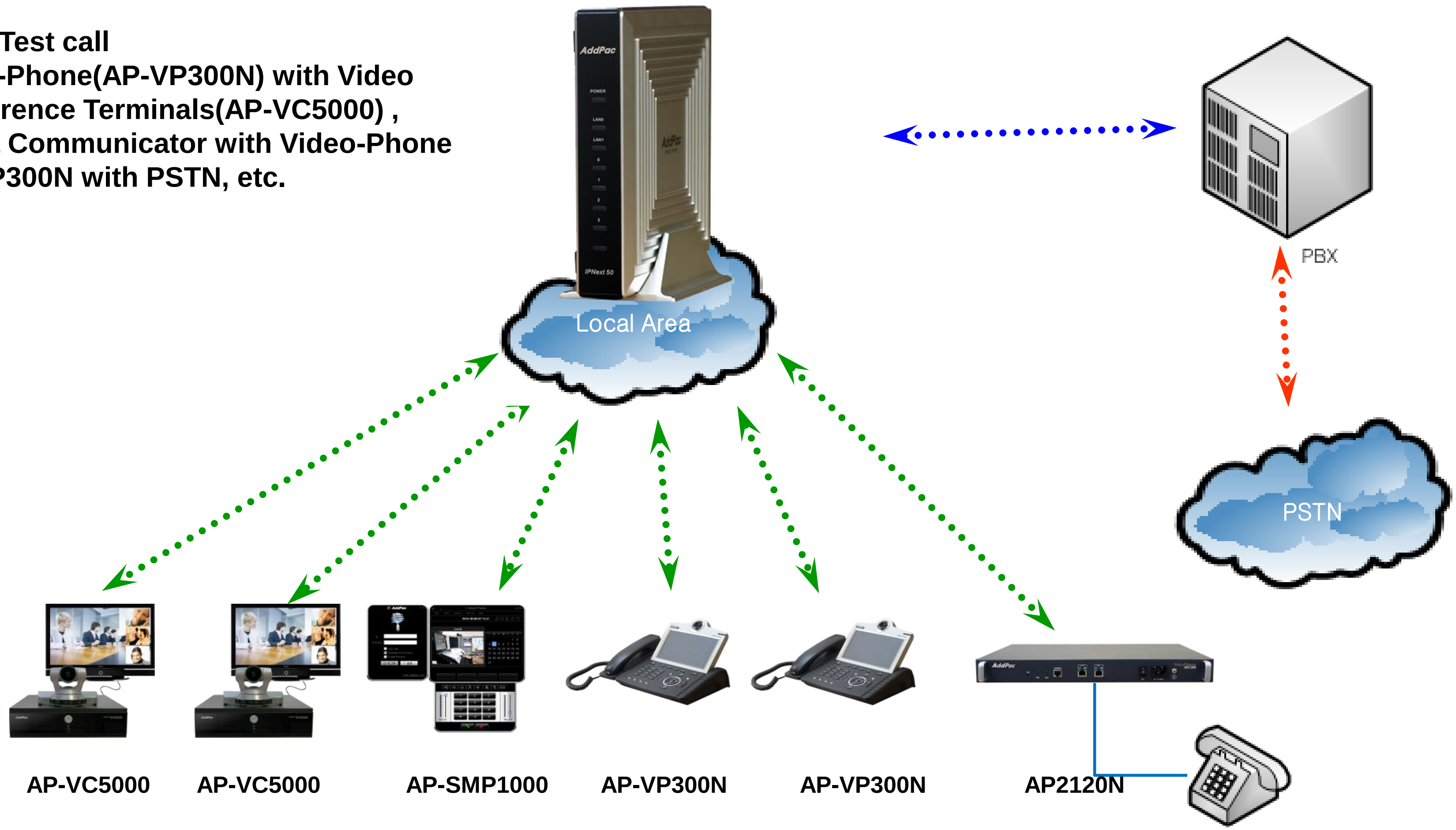
- Translation rule can set both of Called Number / Calling Number

- it can edit Translation rule using "Edit"



SMM – Test call

Make Test call
Video-Phone(AP-VP300N) with Video Conference Terminals(AP-VC5000) , Smart Communicator with Video-Phone AP-VP300N with PSTN, etc.



SMM - Setups for Additional Call Services(IPNext)

- Additional Call Services
  - Hunt Group
  - Pickup Group
  - Park
  - Park Group
  - Music Announcement
  - Auto Attendant
  - Service Code
  - Day Template
  - Monitoring

SMM - Hunt Group

Hunt Group – Priority Level - Overview

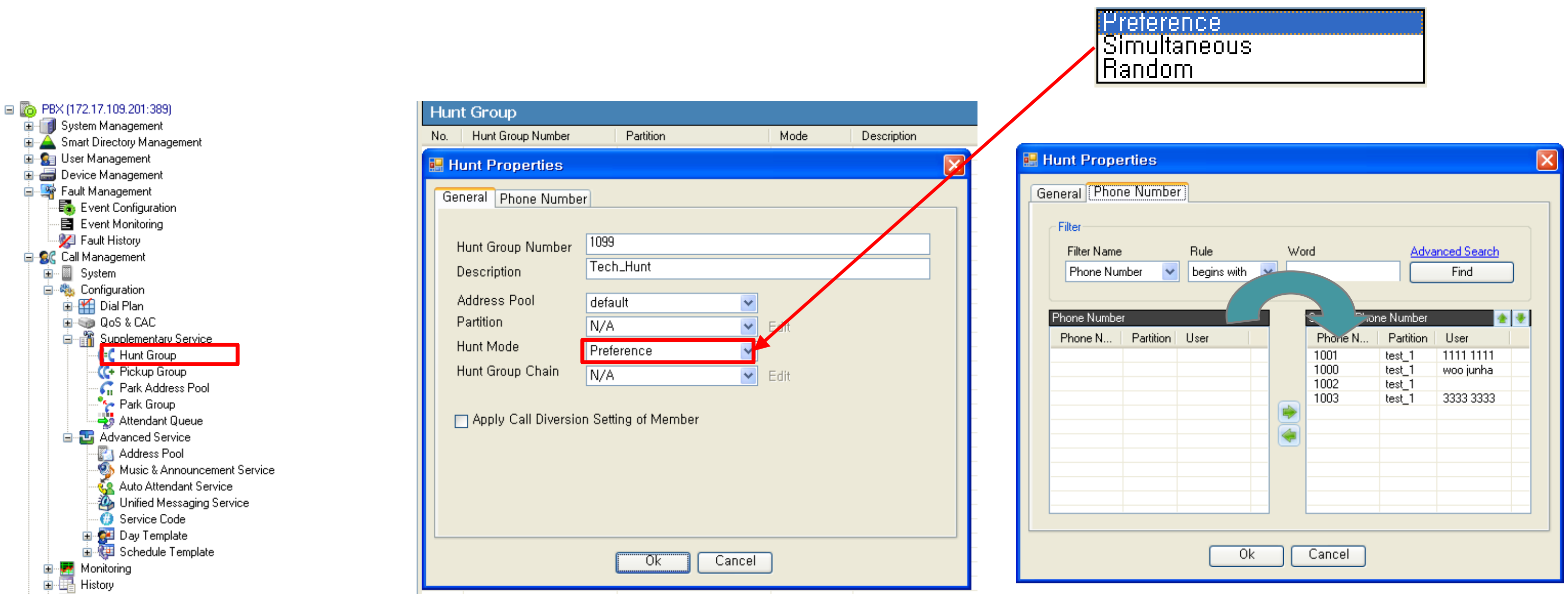
Hunt Group is a function to take a call, basing on a level of priority placed on members of this group, when the call is made to a specific and representative number. While some of the members' lines can be busy or during their absence, the call can be delivered according to the following level of priority:

1. A call is sent from 1001 to 2000.
2. The number 2000, as the number of Hunt Group 2000, it consists of each member of the numbers, 2001, 2002, 2003. Among these number, the call is delivered first to the number with the highest level of priority which is the number 2001
3. When the number 2001 is busy or absent, the call can be delivered to the 2002
4. When the number 2002 is busy or absent, the call is to be delivered to 2003
5. When the number 2003 is still busy or absent, the call can not be delivered and is to be terminated.

SMM - Hunt Group

- Set Hunt Group

1. Carrying out ‘Call Management>Supplementary Service>Hunt Group>Add Hunt Group’ of SMM Menu
2. Creating an inbound group number of Tech_Hunt (Hunt Group Number 1099)
3. Setting ‘Hunt Mode’ to ‘Preference’, ‘Simultaneous’, ‘Random’.
4. Selecting the extension number of the subject group by choosing a tab of ‘Phone Number’.
5. Checking whether the call is processed in the order of the setup by dialing from 1001 to 1003.

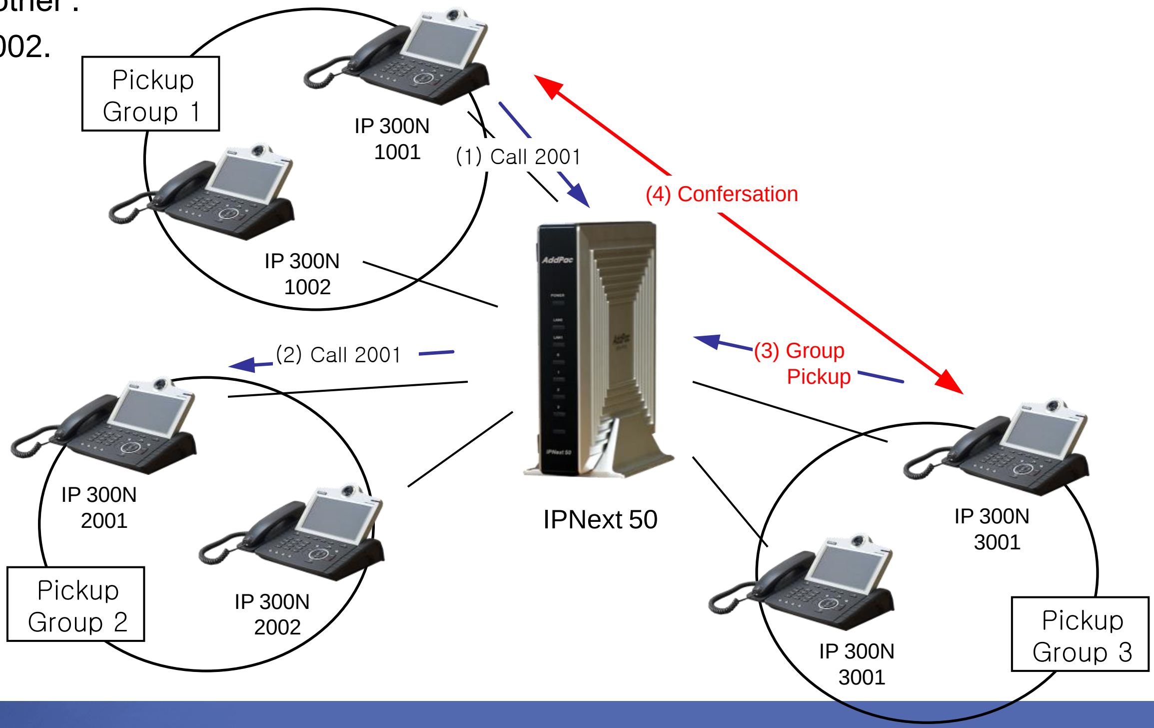


SMM – Pickup Group

Overview of Pickup Group

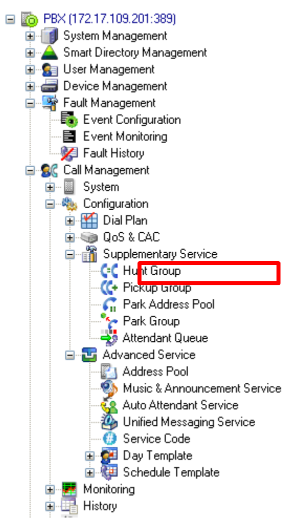
This is an enhanced function from 'Call Pickup' of the same group, which serves to pick up a call for the other group.
When a call came to the other group, you can pick up a by pressing 'Pickup' button.
The number for 'Pickup' must be created differently from the user number

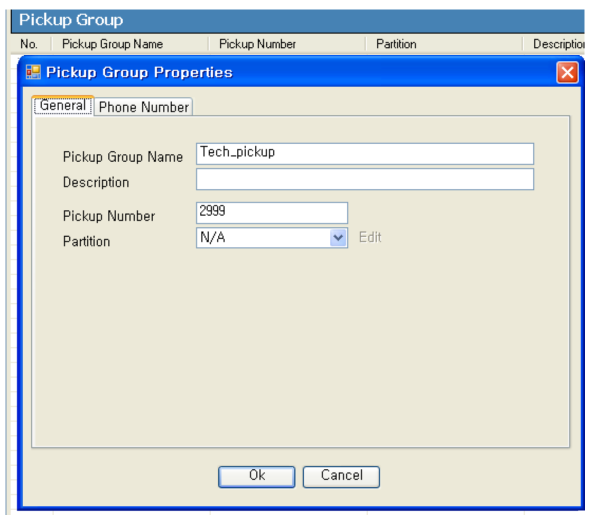
1. A call is made from 1001 to 2001.
2. Call Manager takes the message, then send it to 2001
3. When the bell rings on 2001, you can press 'Group Pickup' from 3002 (2001 and 3002 are different members of 'Pickup Group to each other'.
4. 1001 is connected to 3002.

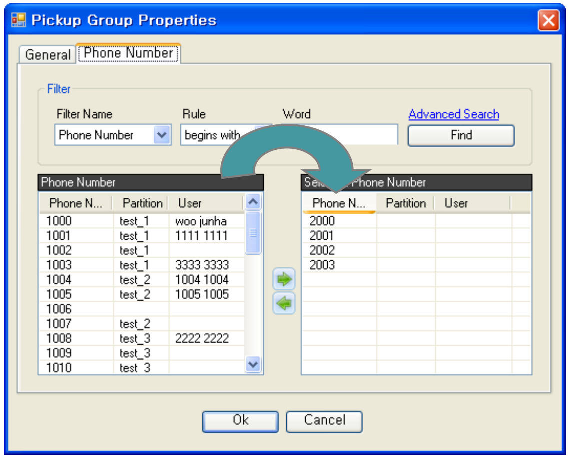


SMM - Pickup Group

1. Carrying out Call Management>Supplementary Service>Pickup Group>Add Pickup Group of SMM Menu.
2. Creating a group extension number of Technical Support division as ‘Pickup Group Number 2999’
3. Selecting an extension number of the subject group from the tab of ‘Phone Number’.
4. Completing the setup, then dialing from 1000 to 6000
5. Checking the response on 1000 by calling 6999 from an extension number 2000





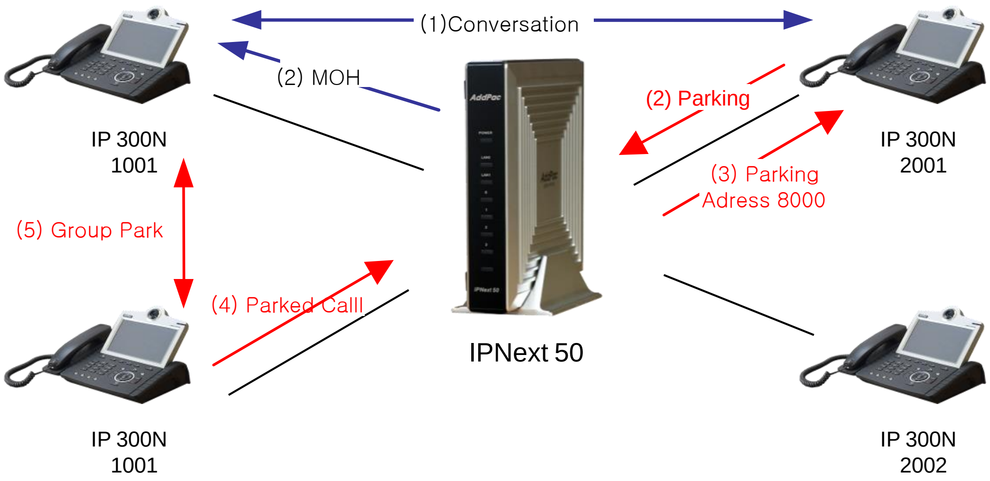


SMM - Park Address Pool

Call Park-Overview of Park Address Pool

This function is useful in a situation of which you may need to move to one place to another during phone conversation. During phone conversation, you may press ‘Call Park’ key, the ‘Park’ number is notified, then the line is off the line. Then you can take the call from the place to where you move and the call is sent to the ‘Park’ number, your last call is connected.

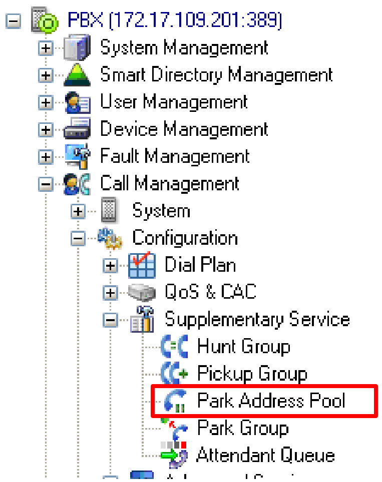
1. 1001 and 2001 are on the line.
2. 2001 pushes ‘Park’ key (1001 listens to a sound on hold while waiting)
3. Call Manager informs the ‘Park’ number.
4. After moving to 1002, you can press the ‘Park’ number
5. The call is connected to 1001.

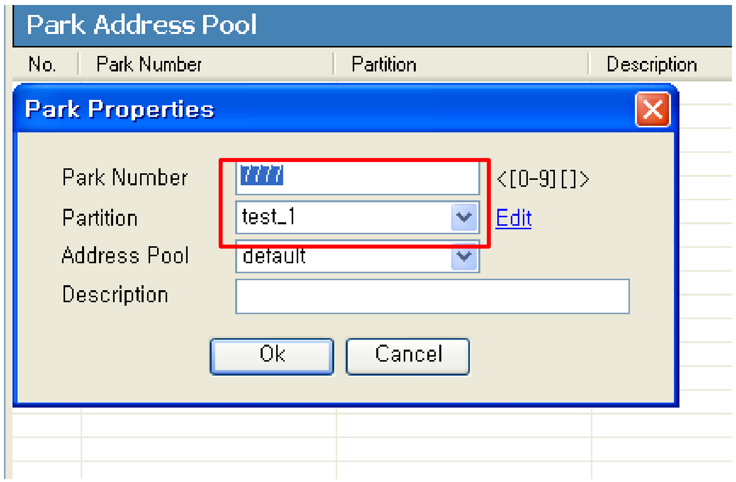


SMM – Park Address Pool

Call Park-Park Address Pool-Setup

1. Selecting ‘Call Management>Supplementary Service>Park Address Pool’.
2. Selecting Add Park and setting up ‘Park Number’.
3. Making a call from1000 to 6000.
4. 6000 reposes to the call then pressing ‘Park Number’, then pressing ‘Park Number’ from the other seat and connecting to 1000.



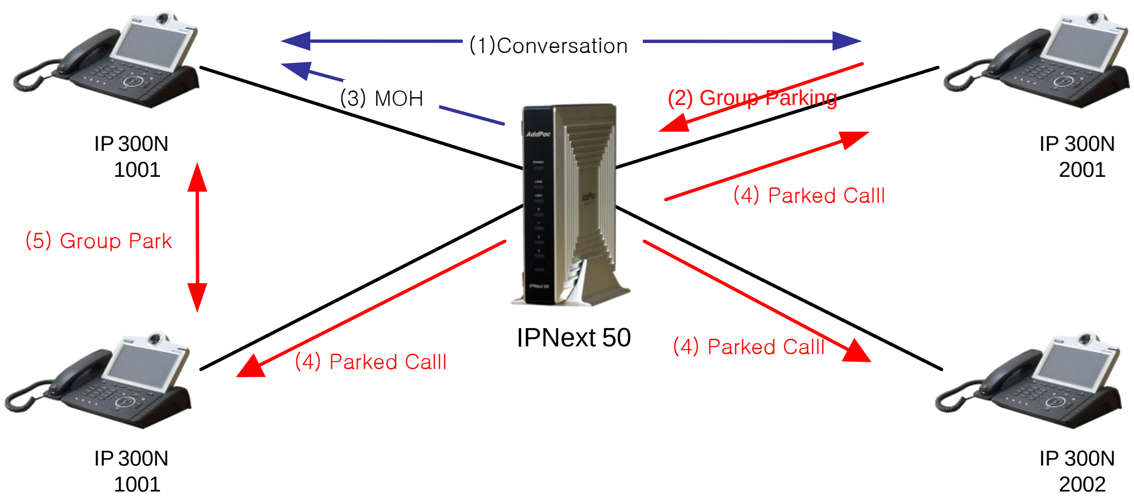


SMM - Park Group

Call Park-Overview of Park Address Pool

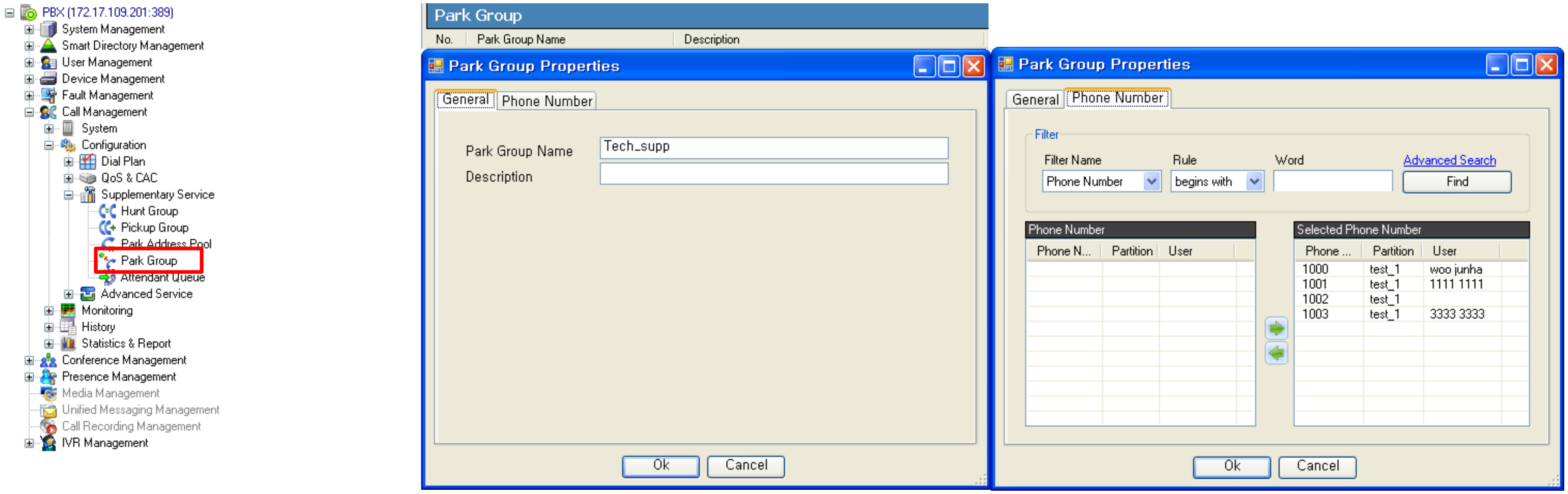
This function is useful in a situation of which you may need to move to one place to another during phone conversation. During phone conversation, you may press 'Call Park' key, the 'Park' number is notified, then the line is off the line. Then you can take the call from the place to where you move and the call is sent to the 'Park' number, your last call is connected.

1. 1001 and 2001 are on the line
2. 2001 pushes 'Group Park' key (1001 listens to a sound on hold while waiting)
3. Call Manager informs to all and each terminals included in the group
4. After moving to a seat, you can choose 'Group Park', then pick up.
5. The line connected to 1001.



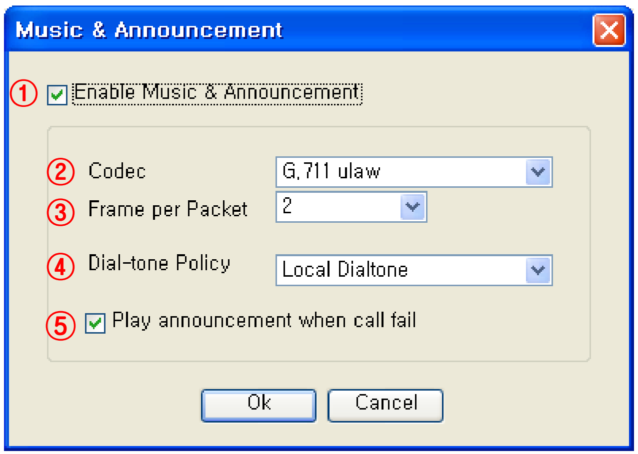
Call Park – Park Group -Setup

1. ‘Call Management>Supplementary Service>Park Group’
2. Setting ‘Park Group Name’ by choosing Add Park Group
3. Selecting the phone to be registered to ‘Park Group’.



SMM – Music & Announcement

This is a setup to enable/disable MoH (Music on Hold), messages of voice guide and dial tone policy.
The following service is shown in the picture below as to be carried out (Call Management > Advanced Service > Music & Announcement Service) .



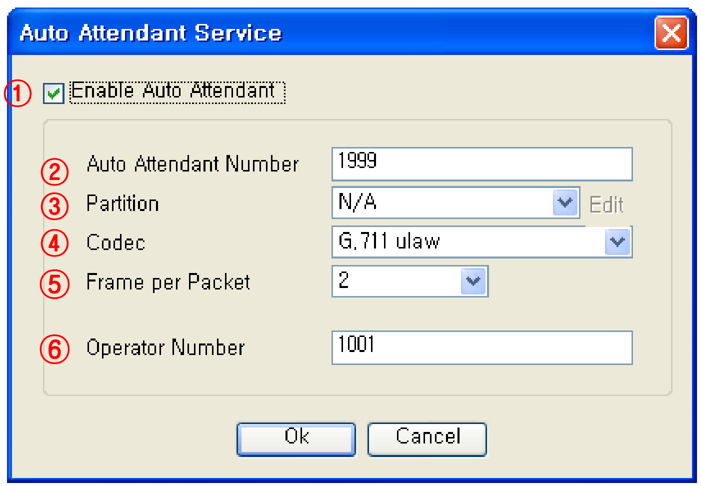
Ref.	Description
1	Enabling or disabling Music & Announcement Service. When the service is connected, Dial Tone can be furnished by Codec from IPNext200.
2	Selecting Codec which can be suitable to VoIP environment (G.711a, G.711u, G.729)
3	Setting a number of frames for each packet
4	Setting a policy of dial tone which can be applied to SSCP supporting IP Phone. -Local Dial tone: Furnishing dial tone from IP phone during Hook off -Remote Dial tone: Furnishing dial tone from Call Manager during Hook off -Local + Remote Dial tone: Dial Tone can furnished form IP Phone during Hook Off and replacing the dial tone by the one which is furnished from Call Manager
5	-Enabling or disabling Announcement when call fail.

SMM - Auto Attendant

As a process which is used in a general office environment, this is a service which allows you to listen to introduction of company by using IVR service, to make a call by entering a designated extension number and when you do not know the extension, you may press ‘0’ to be connected to information desk.

A simple type of IVR service, which is provided at default, is presented in here and for more complex type of IVR service can be used by IVR Editor which is used for editing the scenario.

The following service can be carried out (Call Management > Advanced Service > Auto Attendant Service) , as it is shown in the picture below.

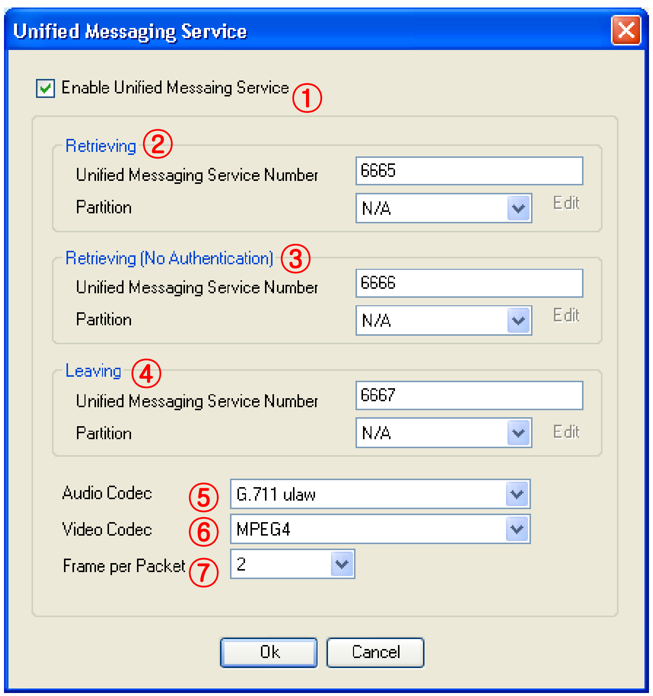


Ref.	Description
1	Enabling or disabling Auto Attendant Service
2	Setting up the telephone numbers for Auto Attendant Service
3	Selecting ‘Partition’ (options)
4	Selecting the audio codec which is to be used in Auto Attendant Service. The supporting codec is G.711A/Ilaw and G.729
5	Setting a number of frames for each port
6	Setting up a designated extension number for when ‘0’ is entered from Auto Attendant Service

SMM – Voice Mail

This service configures the settings for allowing the registered use, to the IPNext, to listen to the message at any time and any where, when the user is away from the phone or busy on the line.

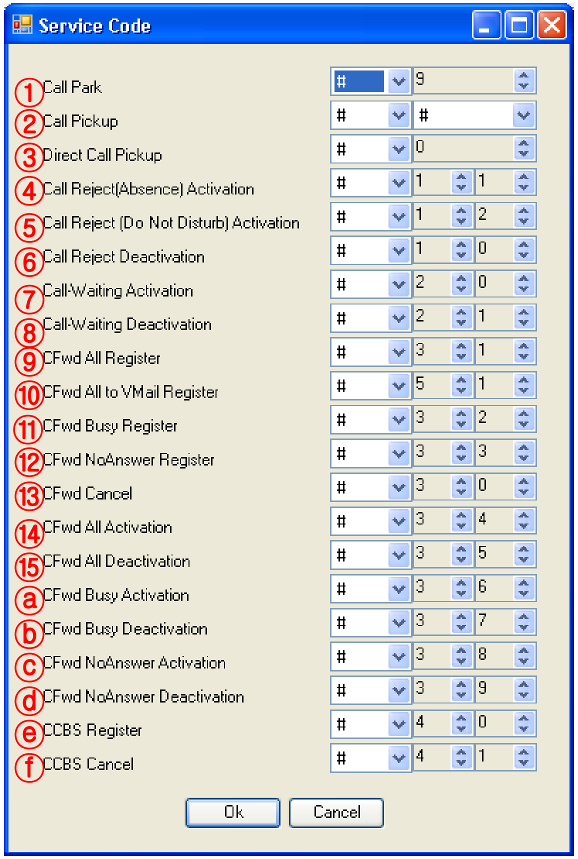
The service can be performed by locating Call Management > Configuration > Advanced Service > Unified Messaging Service as it is shown in the following figure.



Ref.	Description
1	Enable or disable the Unified Messaging Service
2	Configure the phone numbers and partition for connecting to the message box.
3	Configure the phone numbers and partition for connecting to the user's message box without authenticating the extension number.
4	Configure the leaving phone number and partition for the Unified Messaging service.
5	Choose the option for audio codec to be used for the Unified Messaging service G.711A/Ulaw and G.729 are the supporting codec.
6	Choose the option for video codec to be used for the Unified Messaging services MPEG4, H.263, H.263+ and H.264 are the supporting codec.
7	Set the number of frames for each Packet

SMM - Service Code

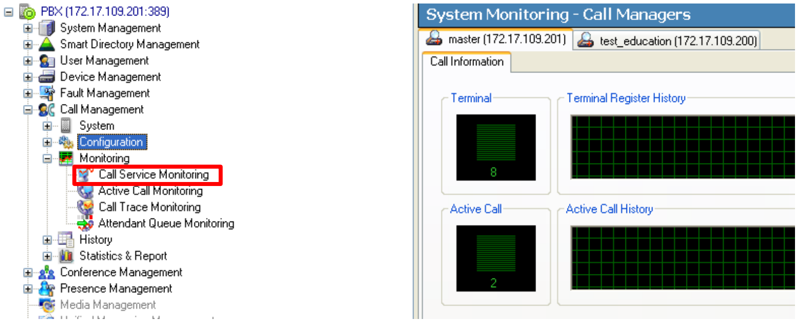
Service Code Configuration can assign 2 digit numbers at maximum starting from ‘#’ or ‘*’, sets up the supplementary service numbers, which are supported from IPNext. This configuration enables/ disables the supplementary service and more useful to the terminals not supported by SSCP, than the IP phone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



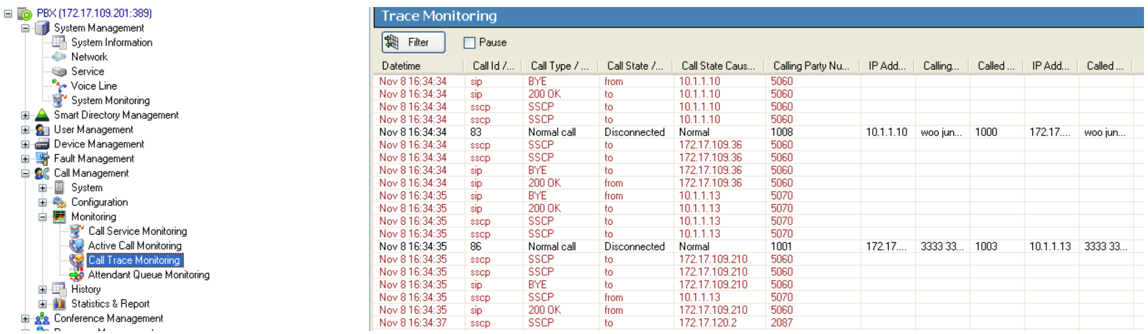
Ref.	Description
1	Set up the code for
2	Set up the code for Call Pickup
3	Set up a call to the code which does not belong to the same group for Call Pickup
4	Set up the code for the phone user’s absence
5	Set up the code for rejecting the incoming call
6	Deactivating the code from Call Reject (Absence 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 the code to deactivate Call Forwarding for absence
E	Set up CCBS
F	Cancel CCBS

SMM - Monitoring

- Monitoring
  - Call Service Monitoring
  - Active Call Monitoring
  - Call Trace Monitoring



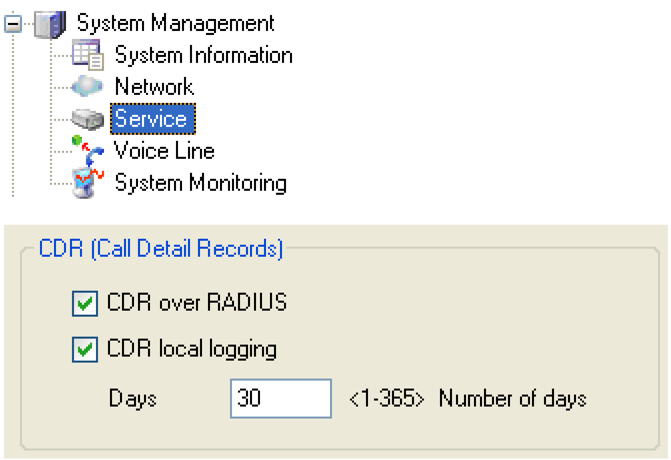
Call Service Monitoring



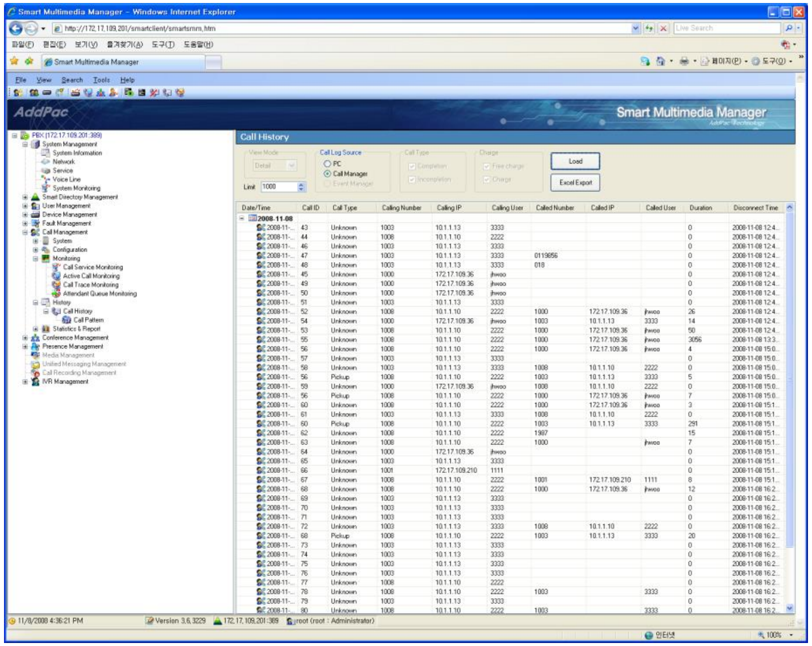
Call trace Monitoring

SMM - Monitoring

Call History searches the history of CDR log file which is saved in Smart Event Manager, Call Manager or PC.
With the information of phone number and host, this feature finds the call flow which has been processed at a glance.
To search the data, specify the event source (Menu > Tools > Preferences > Event) and operate.
To save the syslog to the SMM, use Save Event Log.



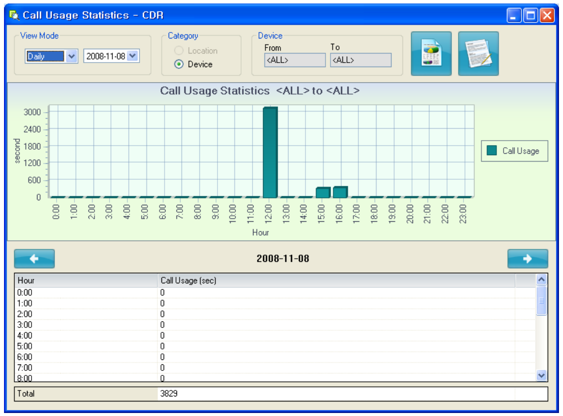
CDR enable



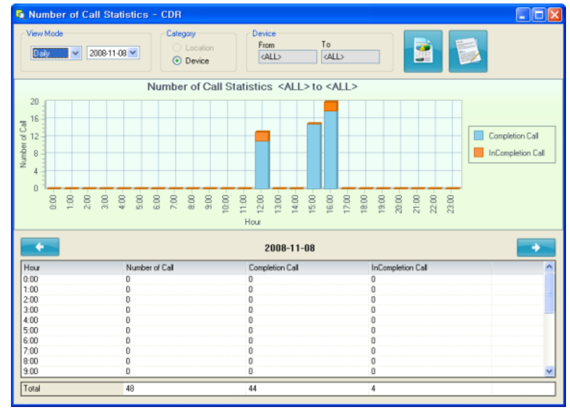
Result of Call history

SMM - Monitoring

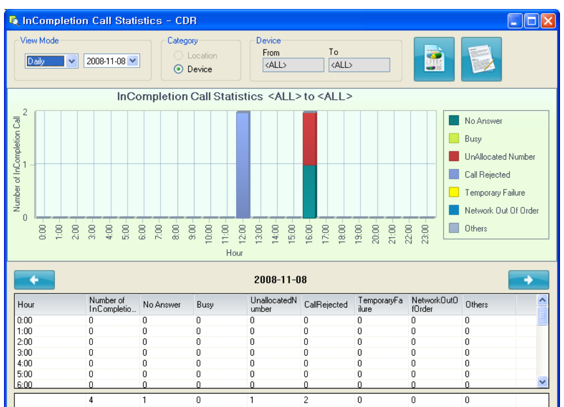
SMM provides and reporting functions and the statistical data of Call Usage, Number of Calls, Incompletion Call and Rank, basing of the basic information that IP-PBX provides.



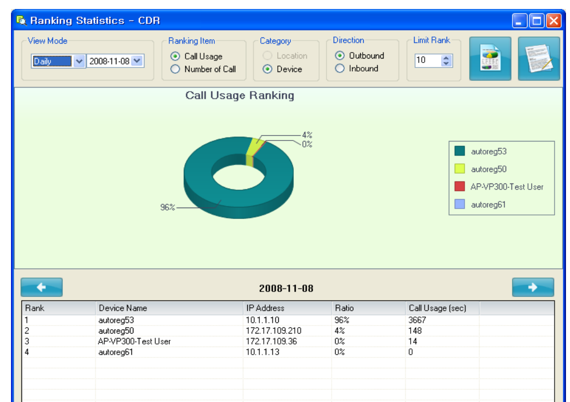
Call usage Statistics



Number of Call Statistics



InCompletion Call Statistics



Ranking Statistics

SMM Setups for Additional Call Services(Additional Servers)

1) Conference Management

2) Presence Management

3) IVR Management

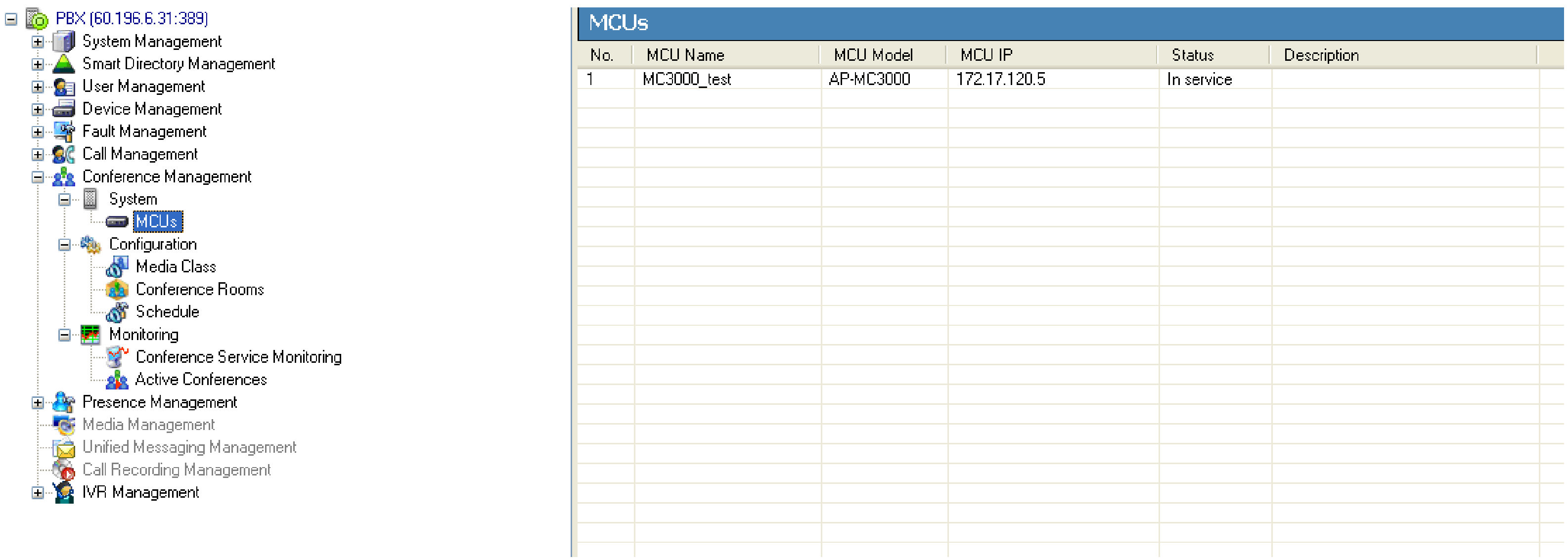
SMM - Conference Management

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.

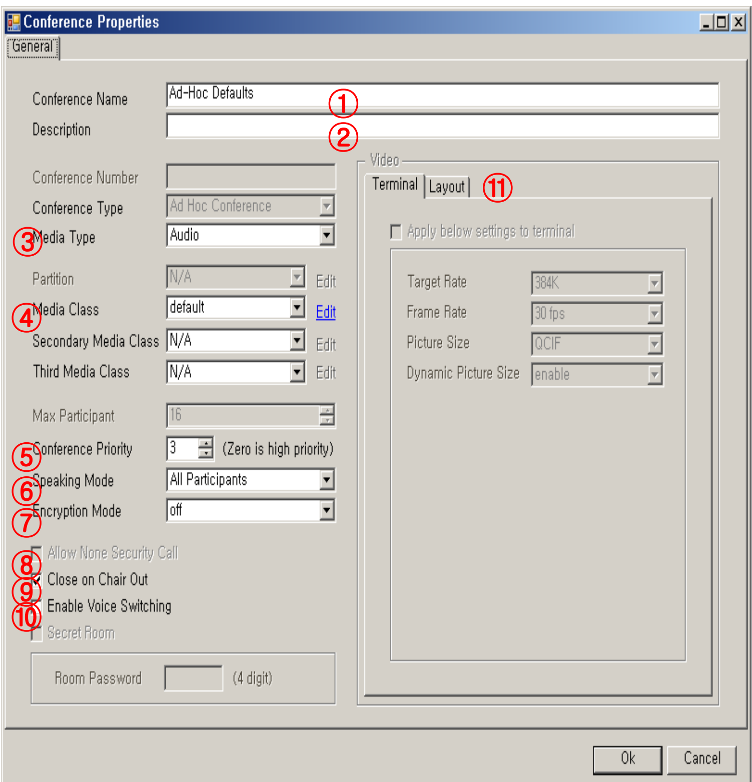
IPNext is equipped with a module for voice conference call and an additional device is required for video conferencing call.

There are MCU-specific product such as the AddPac MC3000 and MCU imbedded in terminal product such as VC2000.



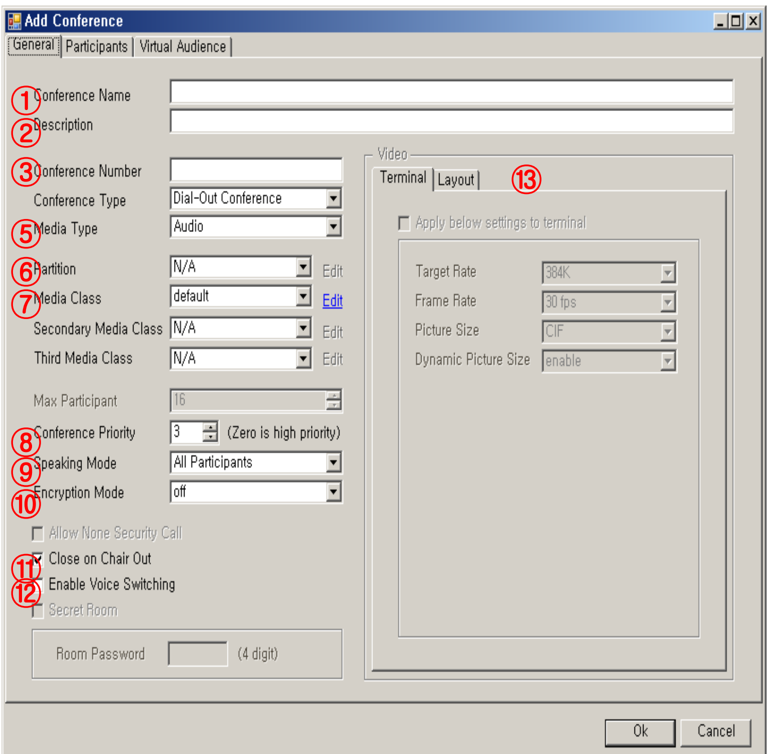
SMM - Conference Management

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.



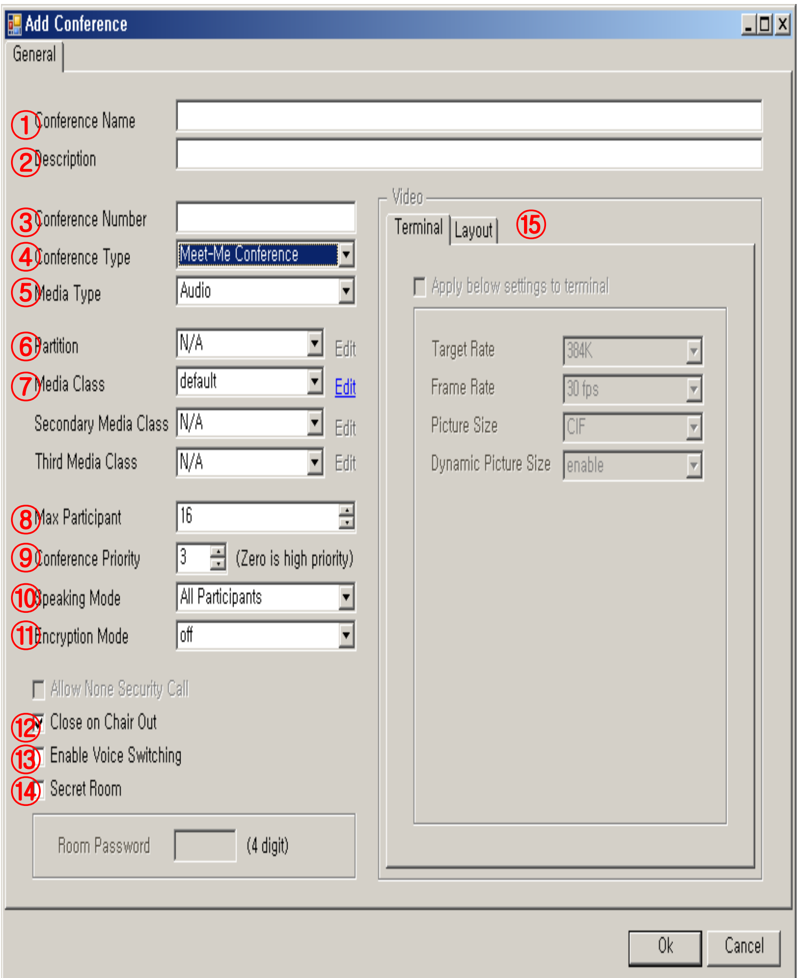
Ref.	Description
1	Register a name of Conference.
2	Register a description of Conference
3	Select a type of media (Audio, Video + Audio) for Ad-Hoc Default. According to the type of media is selected, the settings of video can beenabled or disabled.
4	Select Media Class to be applied to Conference Room. 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



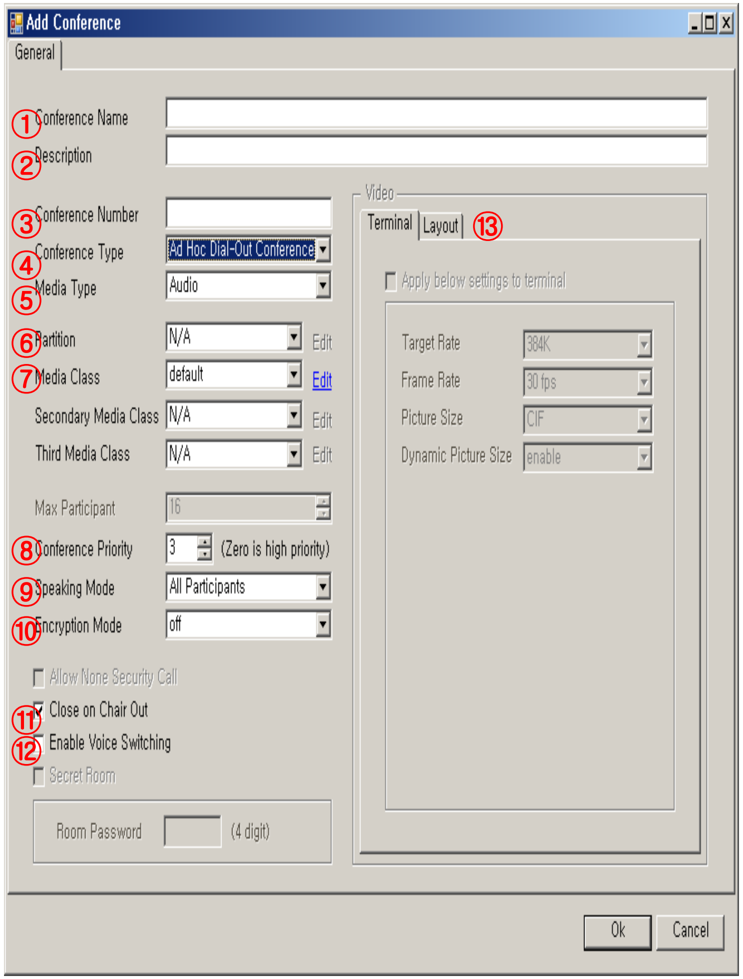
Ref.	Description
1	Register a name of Conference.
2	Register a description of Conference
3	Enter a number of Conference
4	Select a type of conference. In here, select Dial-Out Conference.
5	Select a type of media (Audio, Video + Audio) for Ad-Hoc Default. According to the type of media is selected, the settings of video can be enabled 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

- Configuring Meet Me Conference



Ref.	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)
9	Set the priority level for Conference
10	Set the range of the participants taking the floor -All Participants : All the participants can take the floor -All Visible Participants : All the participants who are displayed in the layout can take the floor -Floor + Chair : Only the participant with Floor and Chair can take the floor -Floor Only : Only the participant with Floor can take the floor
11	Set the encryption mode (AES, DES, 3-DES) for video and voice data of the Conference
12	Choose the option for terminating the conference call, 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. (4 numeric digits)
15	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 Ad-Hoc Dial-Out Conference

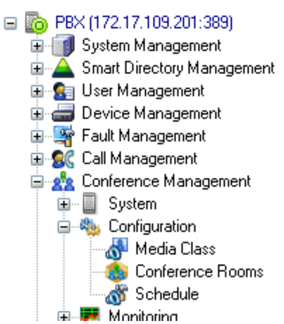


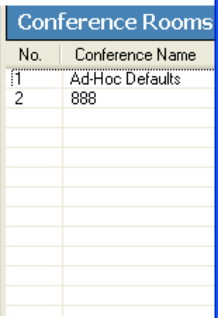
Ref.	Description
1	Enter a name of the Conference
2	Enter a description of the Conference
3	Enter the Conference Number
4	Select the type of conference. In here, select Ad-Hoc Dial-Out Conference
5	Select the type of media (Audio, Video + Audio) for Ad-Hoc Default. According to the type of media is selected, the settings of video can be enabled or disabled.
6	Choose the option for Partition (optional), Select Media Class for Conference Room
7	Choose the option for 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 priority level for the Conference
9	Set the 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	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

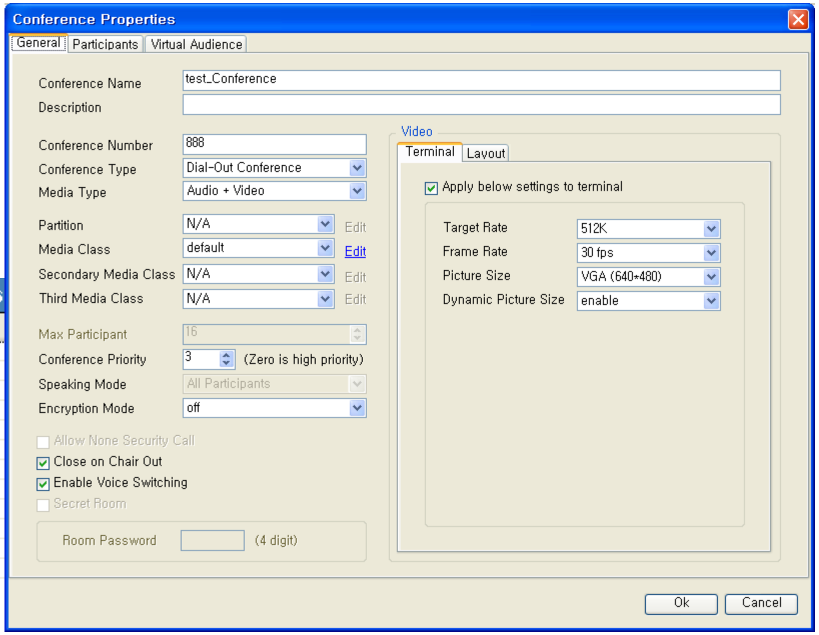
SMM - Conference Management

Make Conference room

- Select Conference number(don't use same phone number)
- Select Media type (Audio only, Audio + Video)
- Enable Voice Switching
  - If you enable this function. The speaker video will display larger than others
- Close on Chair Out
  - The Chair disconnect the conference, the room will be destroy.

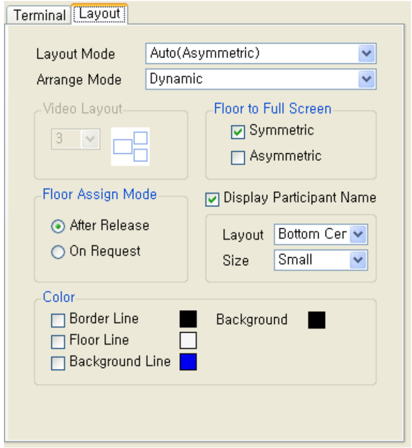


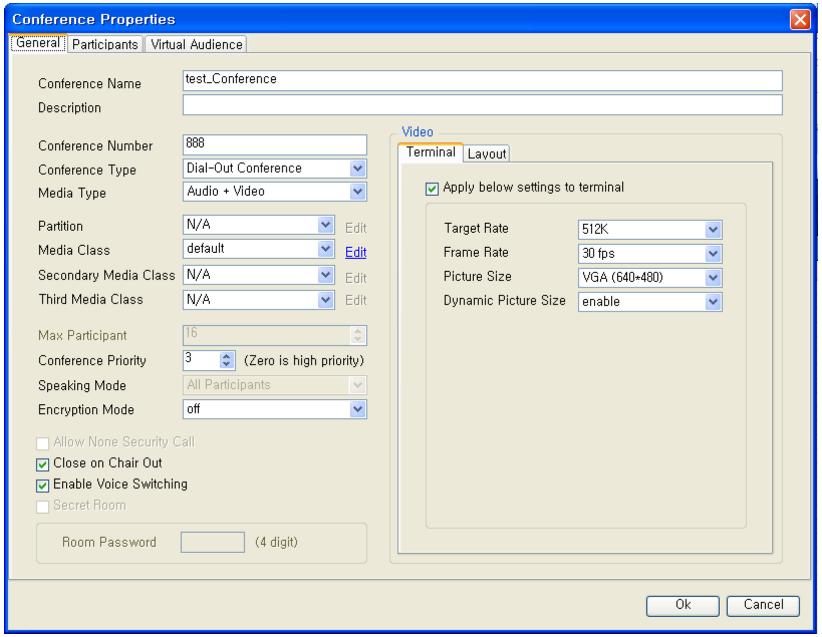




SMM - Conference Management

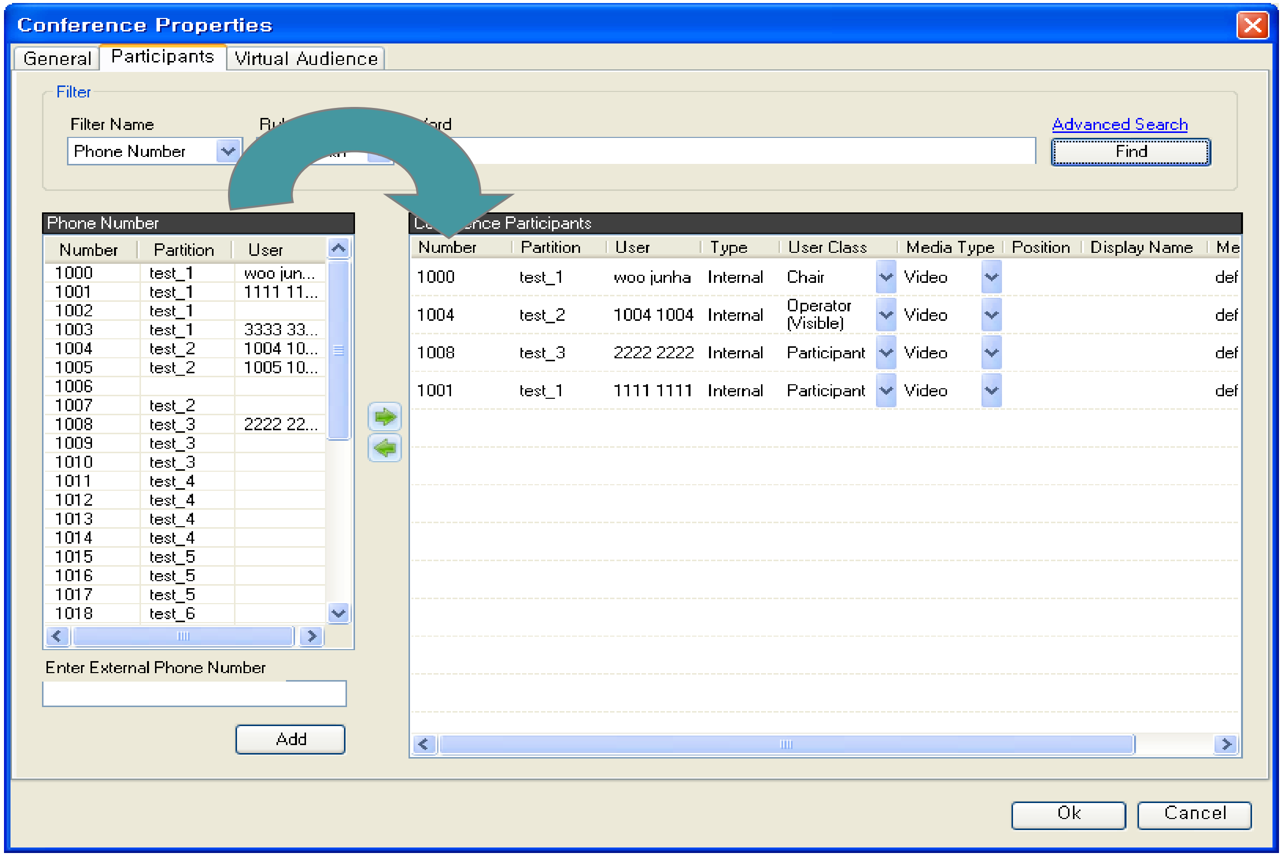
- Video
  - Terminal
    - This function can select Terminals bandwidth, frame rate, pic size.
  - Layout
    - Set layout of this conference.





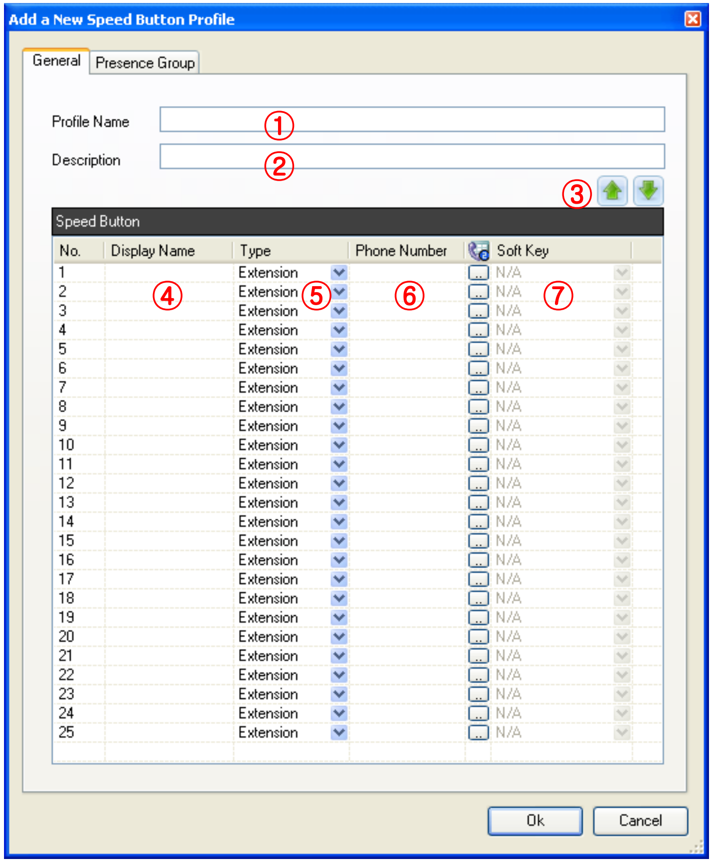
SMM - Conference Management

- Select participant for this conference room
- Set one number as a Chair(when chair disconnect this conference this conference will be destroy)



SMM - Presence Management

Speed Button assigns a telephone number to a specified button and automatically makes a call to the phone number.
It is referred as the button information in both hardware and software aspects.
AP-VP300N support this speed button. 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.



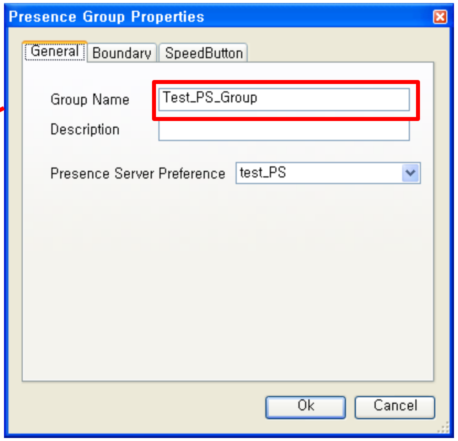
Ref.	Description
1	Enter a name for Speed Button Profile
2	Enter a description for Speed Button Profile
3	Change the position of the speed button to higher or lower in the order
4	Enter a name to be displayed for each speed button(1~25)
5	Choose the type type for each speed button (1~25) of phone number
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)
7	Use the features of Soft Key(Redial, NewCall, Hold, Transfer)on the speed button. This options is possible if Speed Button Type is set to Soft Key

SMM - Presence Management

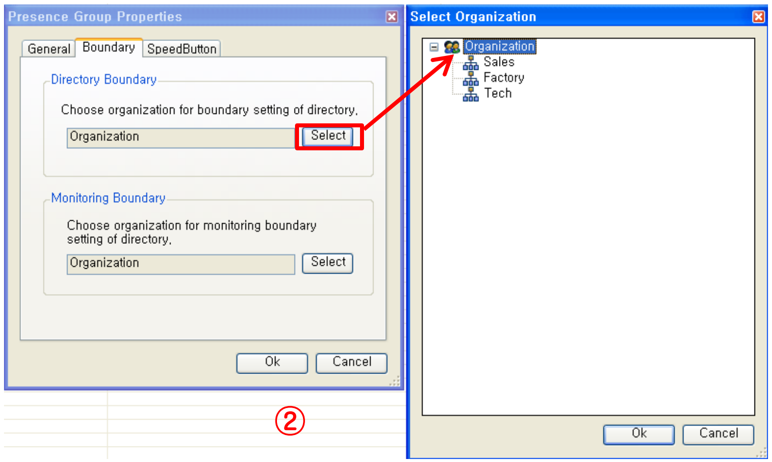
Presence group is a unit to provide presence service and is able to group the settings of speed button and a range of information sharing between the users. One user's directory can configure many presence groups, share the speed button profiles and take presence message. The same profiles are applied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

Create Presence Group

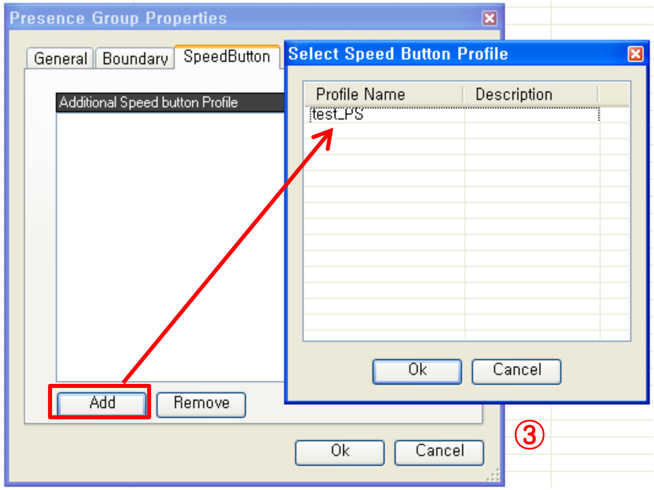
1. General
   - Input Presence Group Name
2. Boundary
   - Choose Organization or User Group
3. Speed Button
   - Add Speed Button profile



①



②



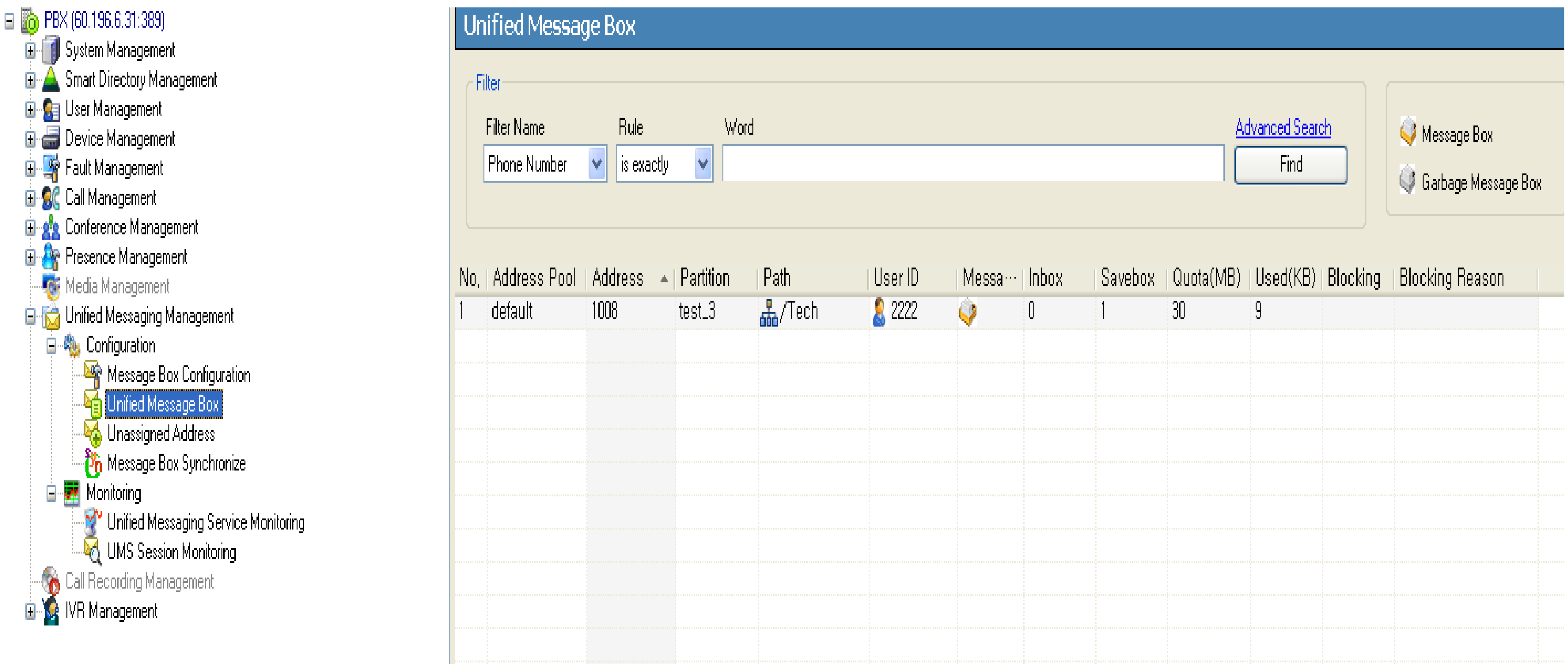
③

SMM – Unified Messaging Management

Unified messaging server provides the feature allowing the registered user to the IPNext PBX, to listen to the left message at any time and any where, when the user is away from the phone or busy on the line.

With accordance to message type, the property is categorized by voice and video and the server provides the features of message box management (resetting, creating, moving and deleting) and message management (moving and deleting).

As it is shown in the following figure, the list of unified messaging servers are displayed by choosing the menu (Unified Messaging Management > System > Unified Messaging Server):

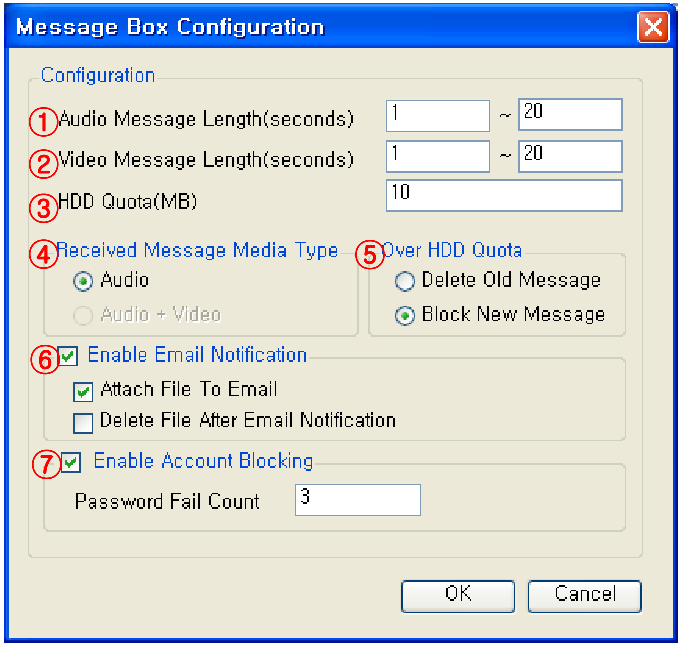


SMM – Unified Messaging Management

Message Box Configuration

Message box configuration is the overall settings related to message box and message box for leaving the voice or video message.

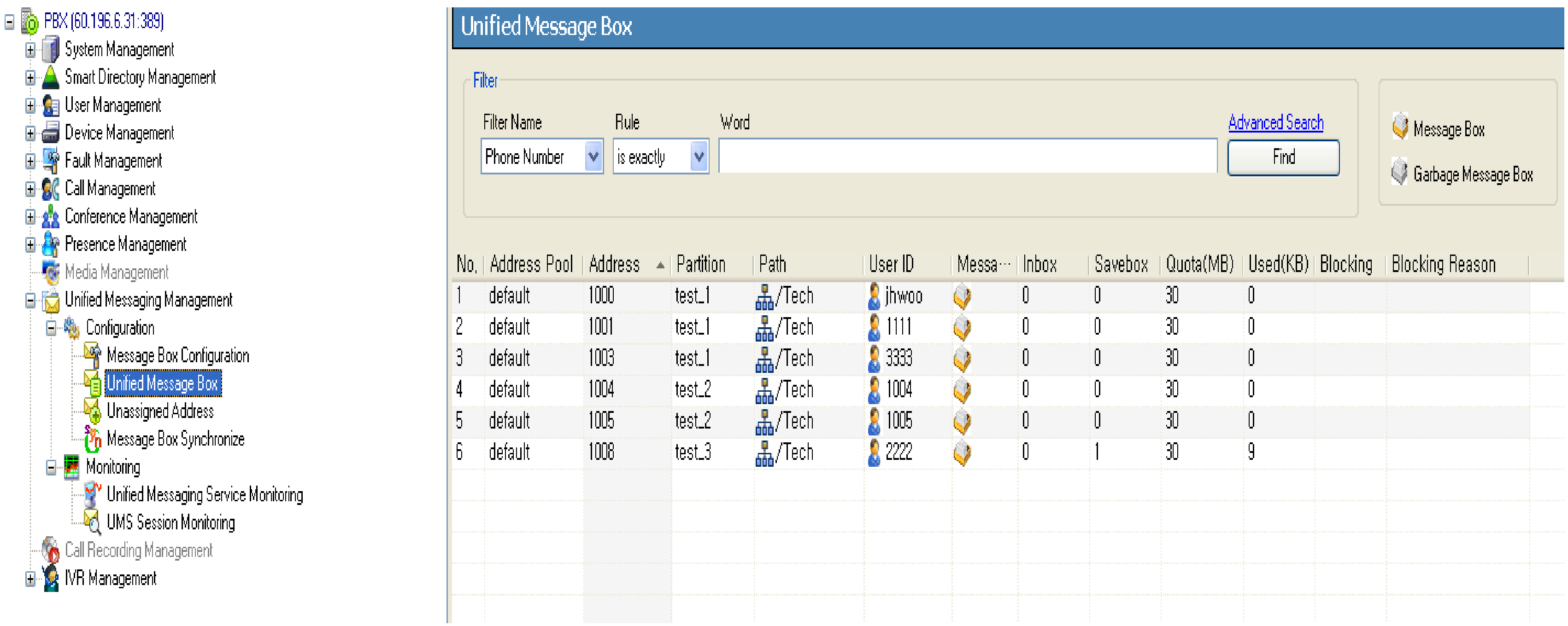
(Unified Messaging Management > Configuration > Message Box Configuration)



Ref.	Description
1	Set a length of time (seconds) for saving voice message
2	Set a length of time (seconds) for saving video message
3	Specify the maximum HDD capacity of the message box for the phone number
4	Specify a type of message to be left for the phone number (Audio or Audio + Video) * The present version does not support audio and video, but it will supported in future
5	Set the quota for exceeding the maximum capacity of HDD of the message box. The old message can be deleted or the new message can be disregarded.
6	Set Email Notification when the message is left. The recorded file can be attached to the e-mail. The server file can be deleted after notification
7	Choose the option for blocking the message box for the failed password entry. A number of password entries can be specified. If the message box is blocked, the message can not be left or verified.

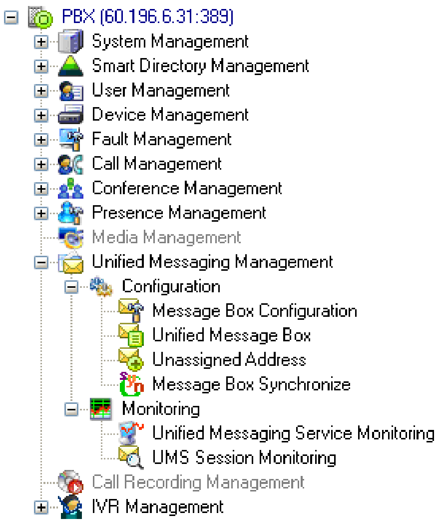
SMM – Unified Messaging Management

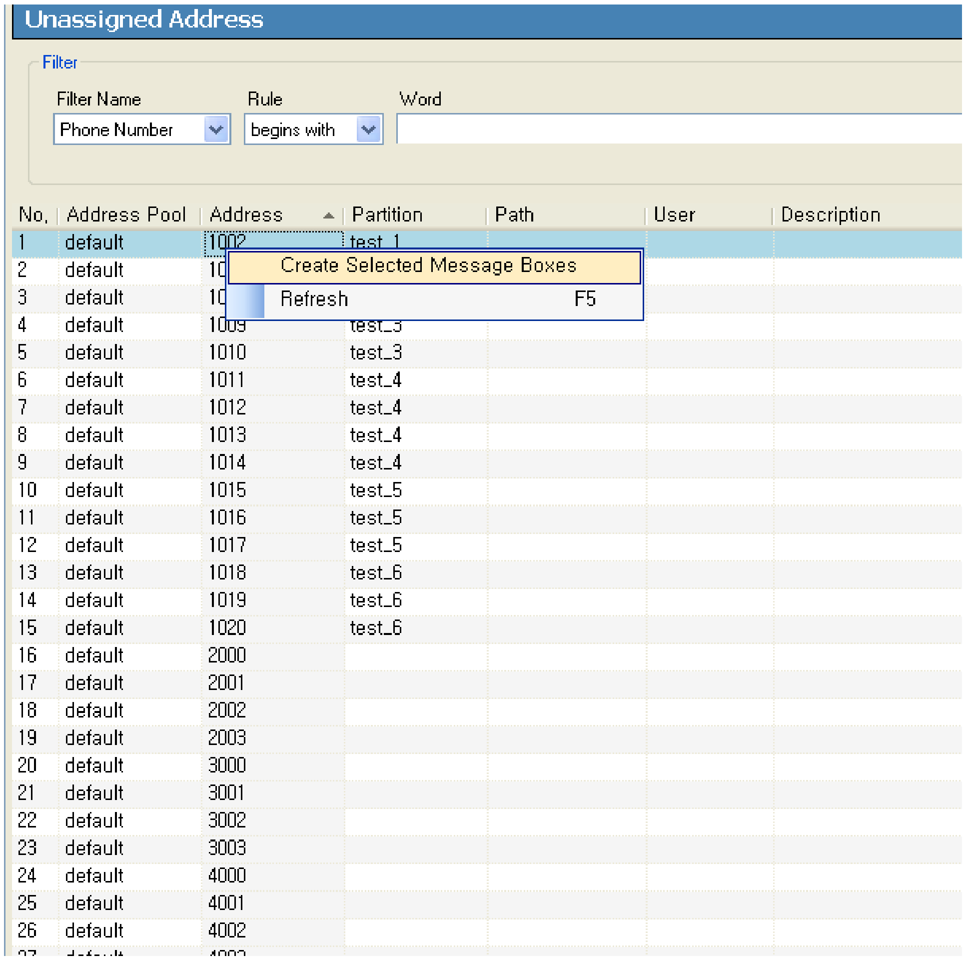
Unified message box references, searches or manages all the message boxes created on address basis.
In message box list, the assigned users or a number of messages left , HDD usage can be verified and the box provides management (initialize, move and delete) for each address of the mail boxes.



SMM – Unified Messaging Management

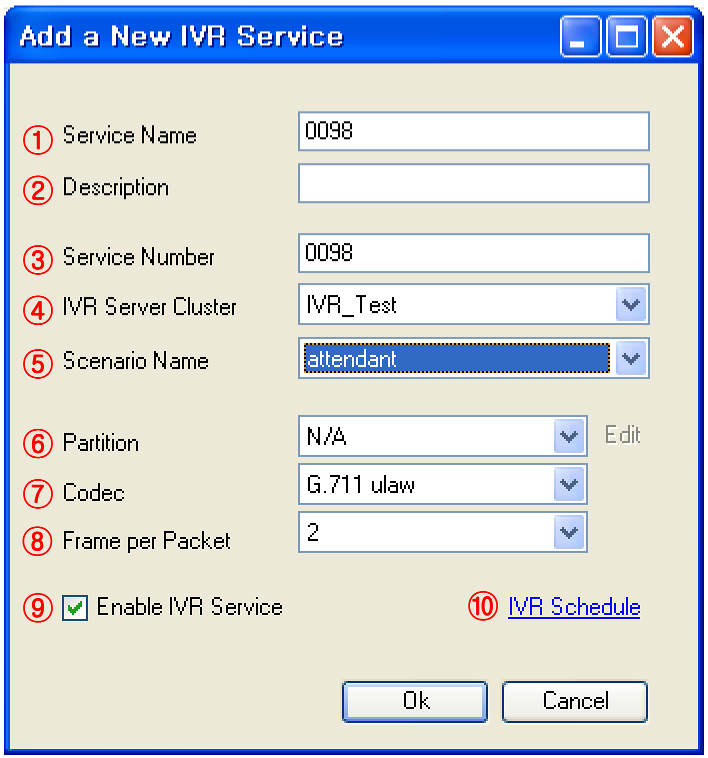
Unassigned address searches and references the address list without the message box created, and creates the message boxes collectively.





Ref.	Description
Create Selected Message Boxes	Create the message boxes for the selected address collectively
Refresh	Refresh the unassigned address list

IVR Management > Configuration > IVR Service By pressing the right button of the mouse, your can register a new IVR service.



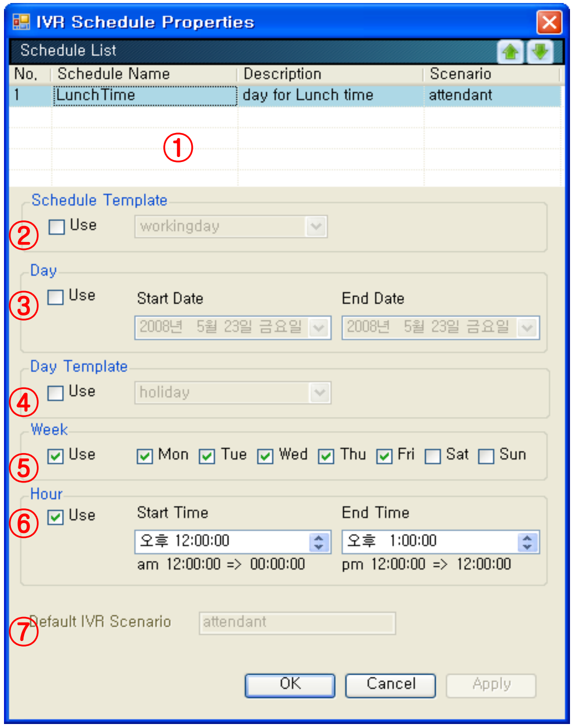
Ref.	Description
1	Create a name of the IVR service
2	Enter a description for IVR Service
3	Create a number of the IVR service
4	Choose the option for a name of IVR Server Cluster.
5	Choose the option for a name of IVR scenario. After the scenario is created by Smart IVR Editor, then IPNext 700 is configured and only the registered scenario list can be selected,.
6	Choose the option for partition (optional)
7	Choose the type of audio codec
8	Set the number of frame for each packet
9	Choose the option for enabling the IVR service
10	Go to IVR Schedule mode

The following figure displays the screen for the IVR schedule list registered to the IVR Service Number.

The IVR scenario is serviced at a specific date and time.

A schedule template can be registered for the specific date and time When they are configured, they can be serviced basing on the schedule template. The corresponding details are disabled, because there is not need to specify a particular date and time for the schedule.

The day template can be also registered (Call Management > Advanced Service > Day Template) for a specific date. After the day template is configured, the IVR scenario is serviced at the registered date in the template.



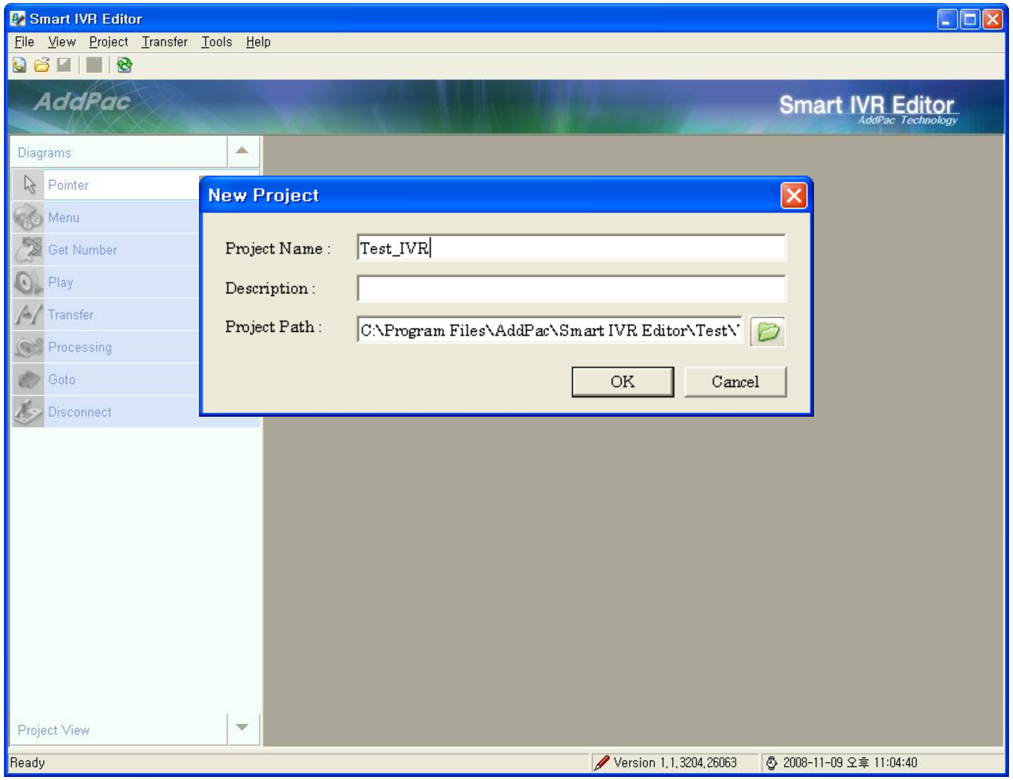
Ref.	Description
1	The registered schedule list. Many schedules can be registered for each IVR Service Number
2	Configure the registered schedule template
3	.Configure the starting date and time of the registered schedule
4	Configure the registered day template
5	Configure the days of the registered schedule
6	Configure the starting and ending time of the registered schedule
7	The basic IVR scenario of the corresponding Service Number
8	Set the priority level for the service of the registered schedule

Example of Smart IVR Editor

The main screen is displayed after executing Smart IVR Editor

1) Creating a NewProject

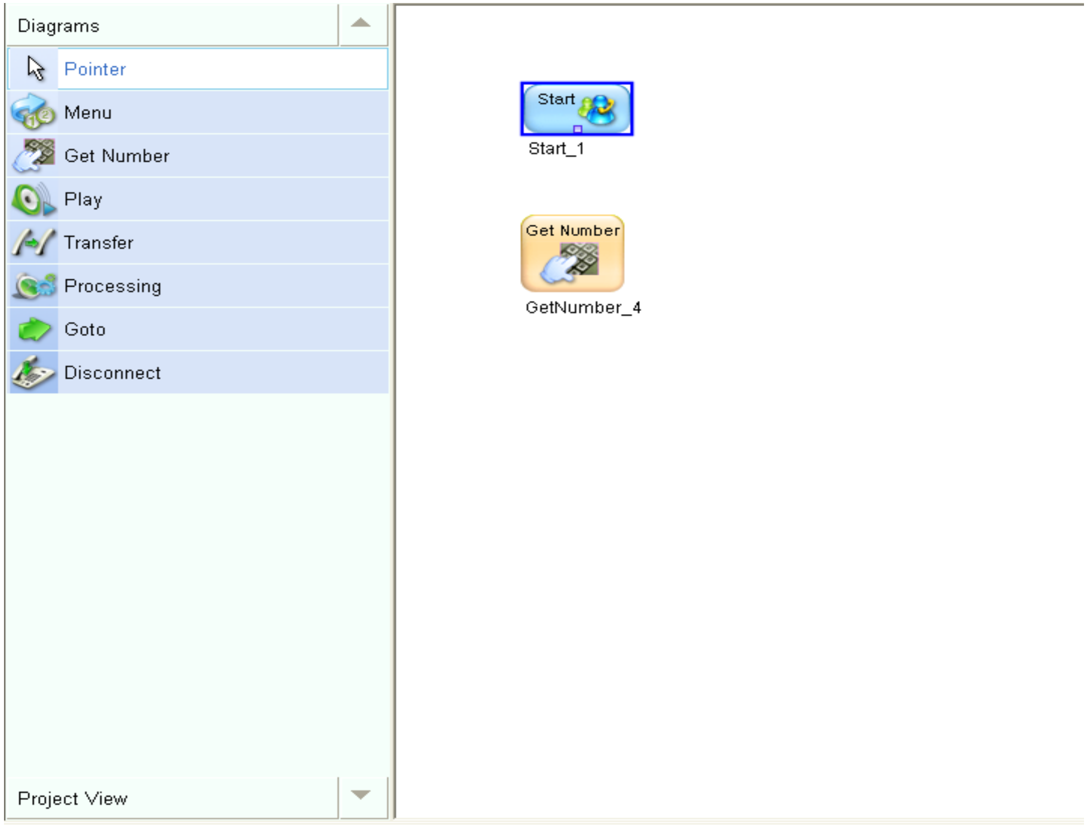
shows the New Project window for the setting of each property. Click ‘OK’, then the scenario ID with the same directory under the specified path is created and well as the basic Project within.



Example of Smart IVR Editor

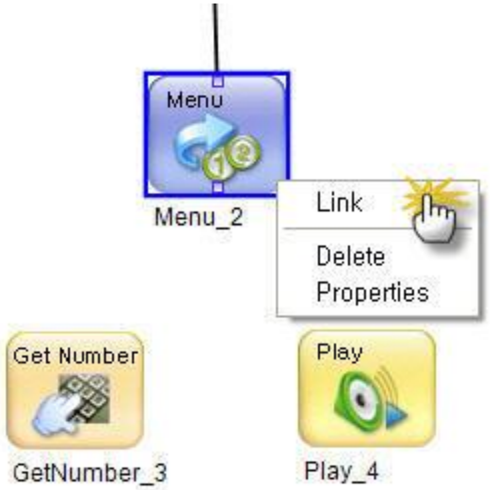
2) Add 'Get Number menu'

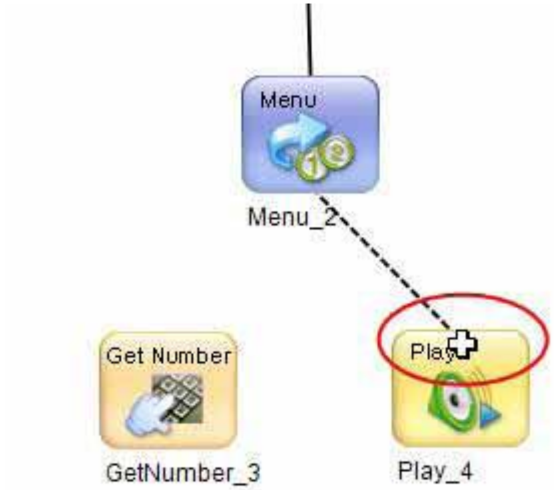
You can also create the diagram as to drag and drop from the Diagram Tool Box.



3) Connecting Diagram

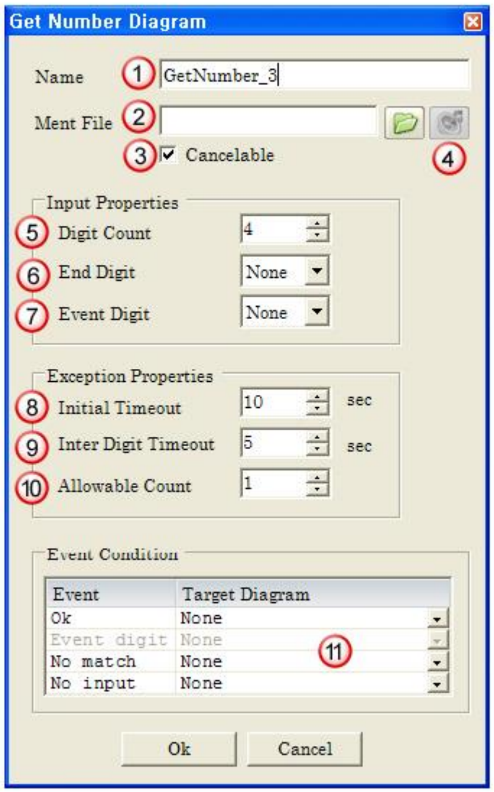
Each Diagram needs to be connected to display the order and branch conditions. All the Diagrams can have more than one connection. Depending on the types of diagram, many braches can be connected. Connecting each Diagram can be processed in the following procedure:





4) Set Get Number Menu

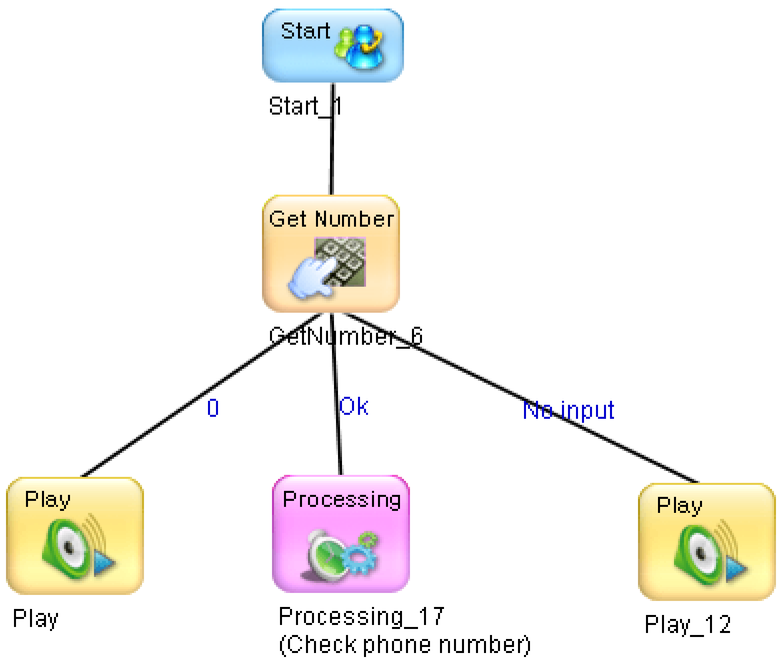
The user takes a specified number of DTMF input. Beside the input, the Diagram can be moved for the specified input like Menu Diagram, once the Event Digit is set. Each branch does not need to be connected directly and it can be connected automatically by the properties settings, like Menu Diagram.

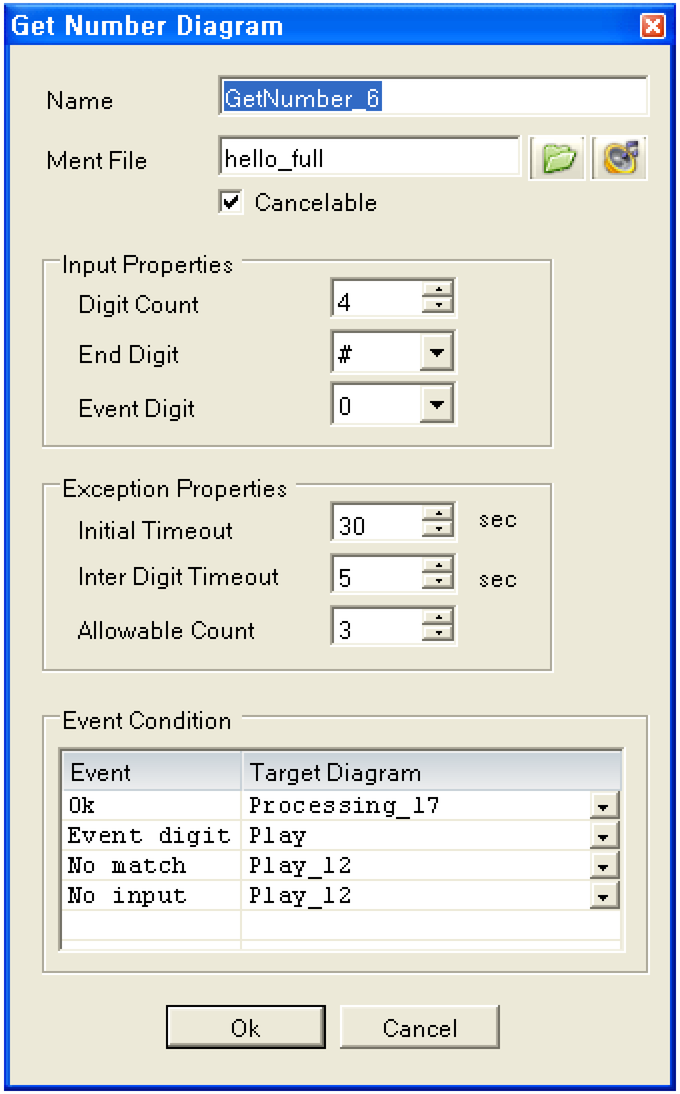


Ref.	Description
1	Name Displays a name of Get Number Diagram
2	Ment File Register the Voice File to be heard to the user before the one take the input. To cancel the registered Voice File, place the cursor on the icon, the press the right click of the mouse to can the Ment File.
3	Cancelable Cancel the Voice File which has been set up at ② When Cancelable is selected and the user makes an entry, the Voice File is stopped playing.
4	Play Plays the registered Voice File at ② .
5	Digit Count Set the number of DTMF which the user can enter
6	End Digit Set up DTMF which informs that the entry is completed. Enter End Digit DTMF for the number less than the one which has been set at ⑤.
7	Event Digit Set up the DTMF to move to another Diagram without taking the entry. When the Event Digit DTMF is entered, it moves to the Diagram for the Event Digit of the Event Condition.
8	Initial Timeout Set the timeout for the user's first entry. Display 'No input Event' if no entry is made within the time range.
9	Inter Digit Timeout Set up the Timeout between the user's entries. Display 'No match Event', if no entry is made in the range.
10	Allowable Count Set the allowable count for no match and no input in the Event Condition. When the count is less than the count of no input, no match, then request the user for more inputs. When the count is more than the one, the move to the specified Diagram
11	Event Condition Set the branch condition for occurrence of the Event by the user's input. OK is for receiving the input normally, the Event Digit is for receiving the Event Digit value. For more details of the branch condition, please refer to the branch condition of the Menu Diagram

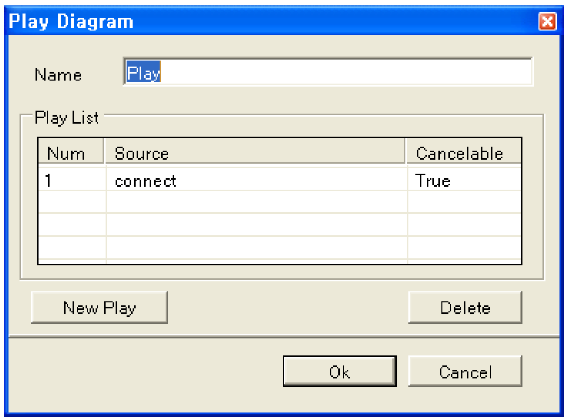
Example of Smart IVR Editor

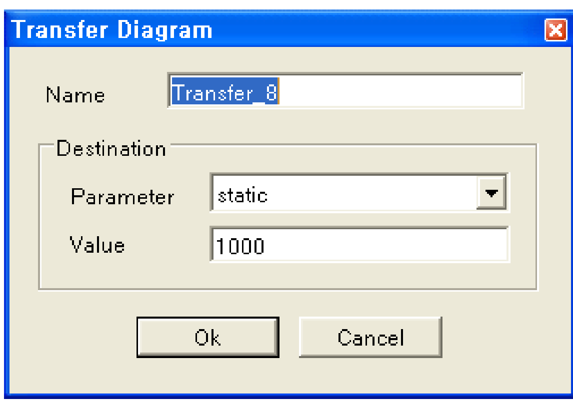
5) Add 'Processing' and two 'play' menu with make link

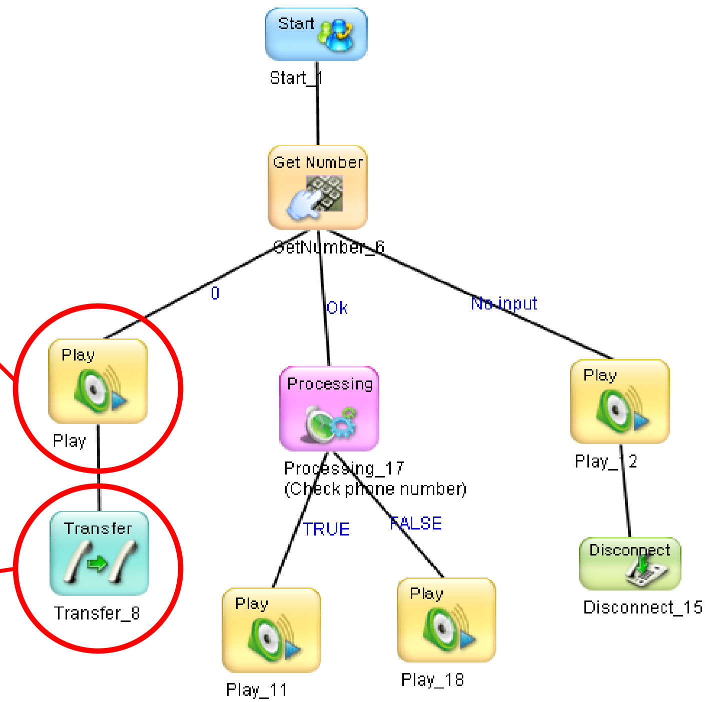




5) Add more menus which connect with high level

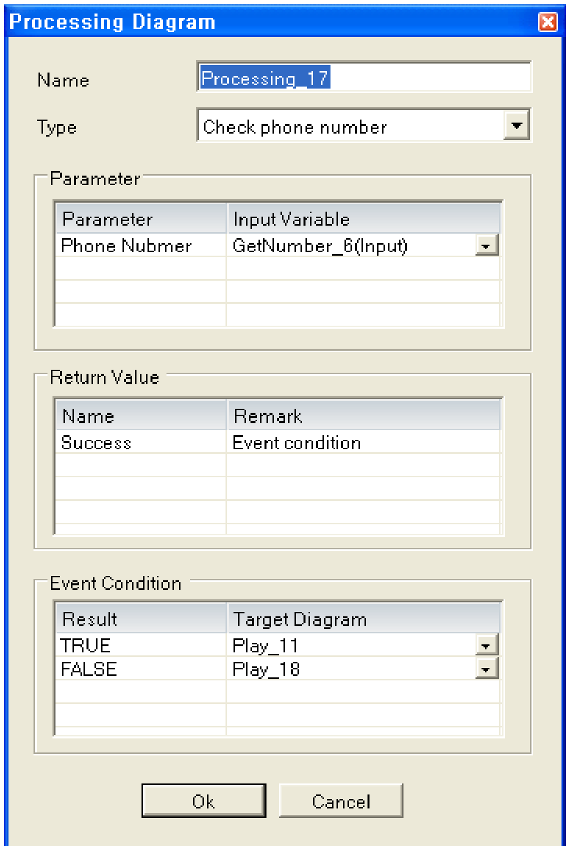






Example of Smart IVR Editor

5) Add more menus which connect with high level



Processing Diagram

Name: Processing_17

Type: Check phone number

Parameter

Parameter	Input Variable
Phone Nubmer	GetNumber_6(Input)

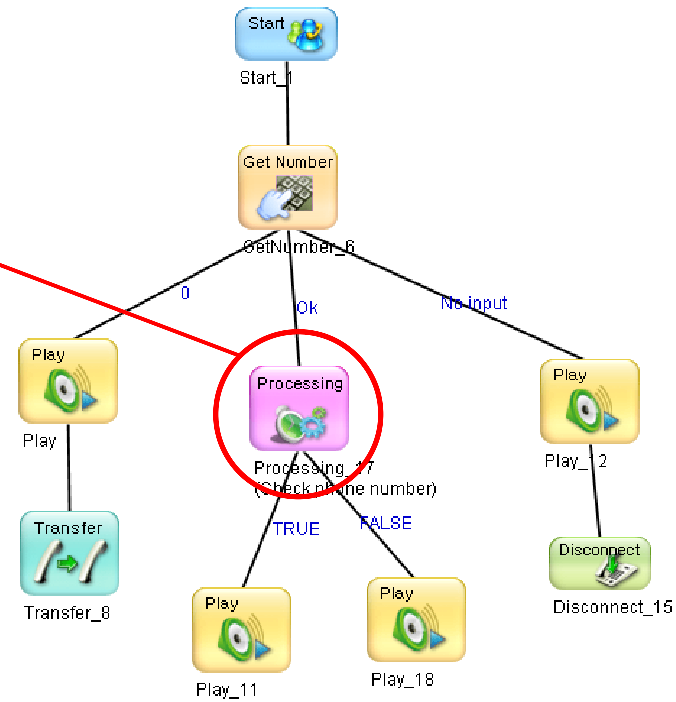
Return Value

Name	Remark
Success	Event condition

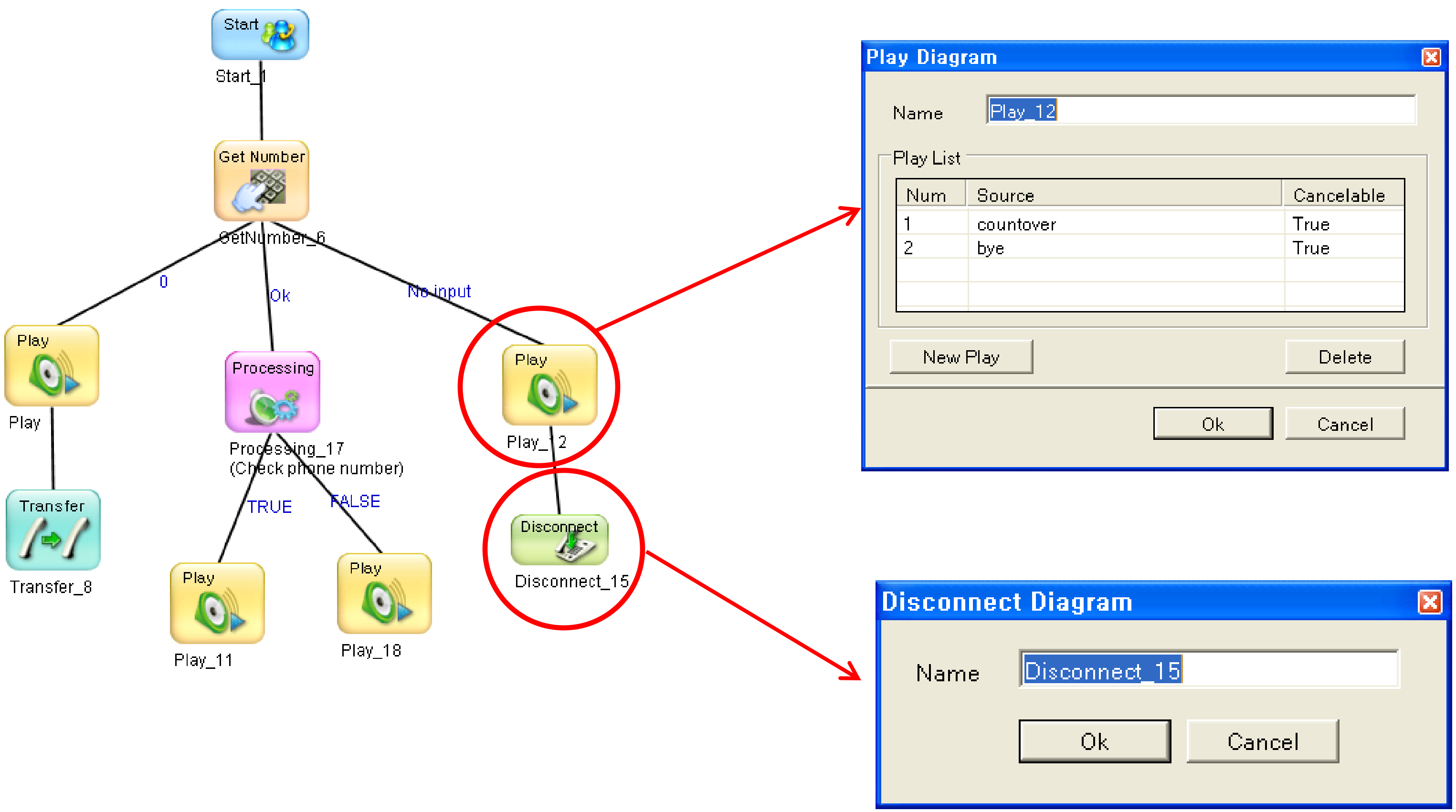
Event Condition

Result	Target Diagram
TRUE	Play_11
FALSE	Play_18

Ok Cancel

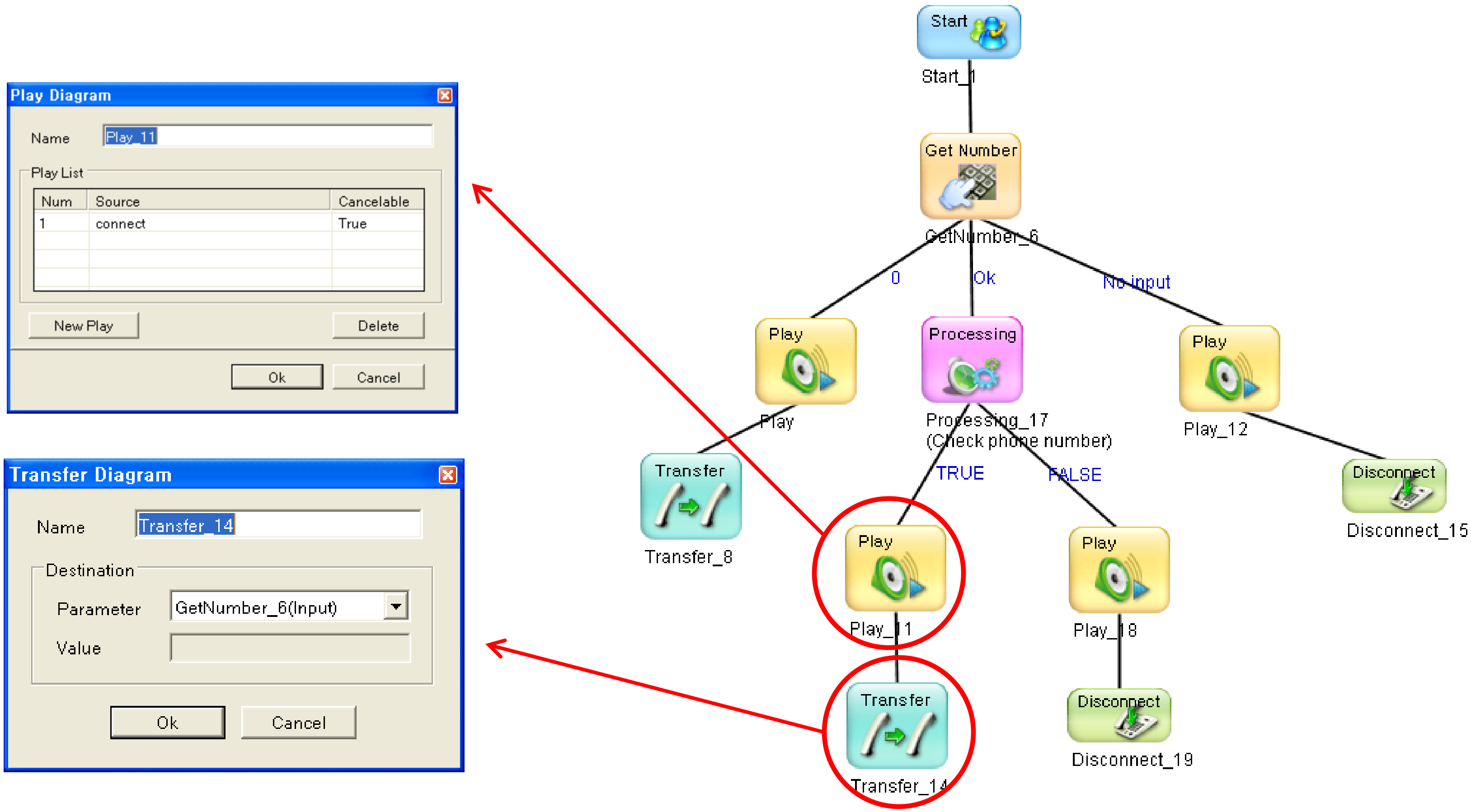


5) Add more menus which connect with high level



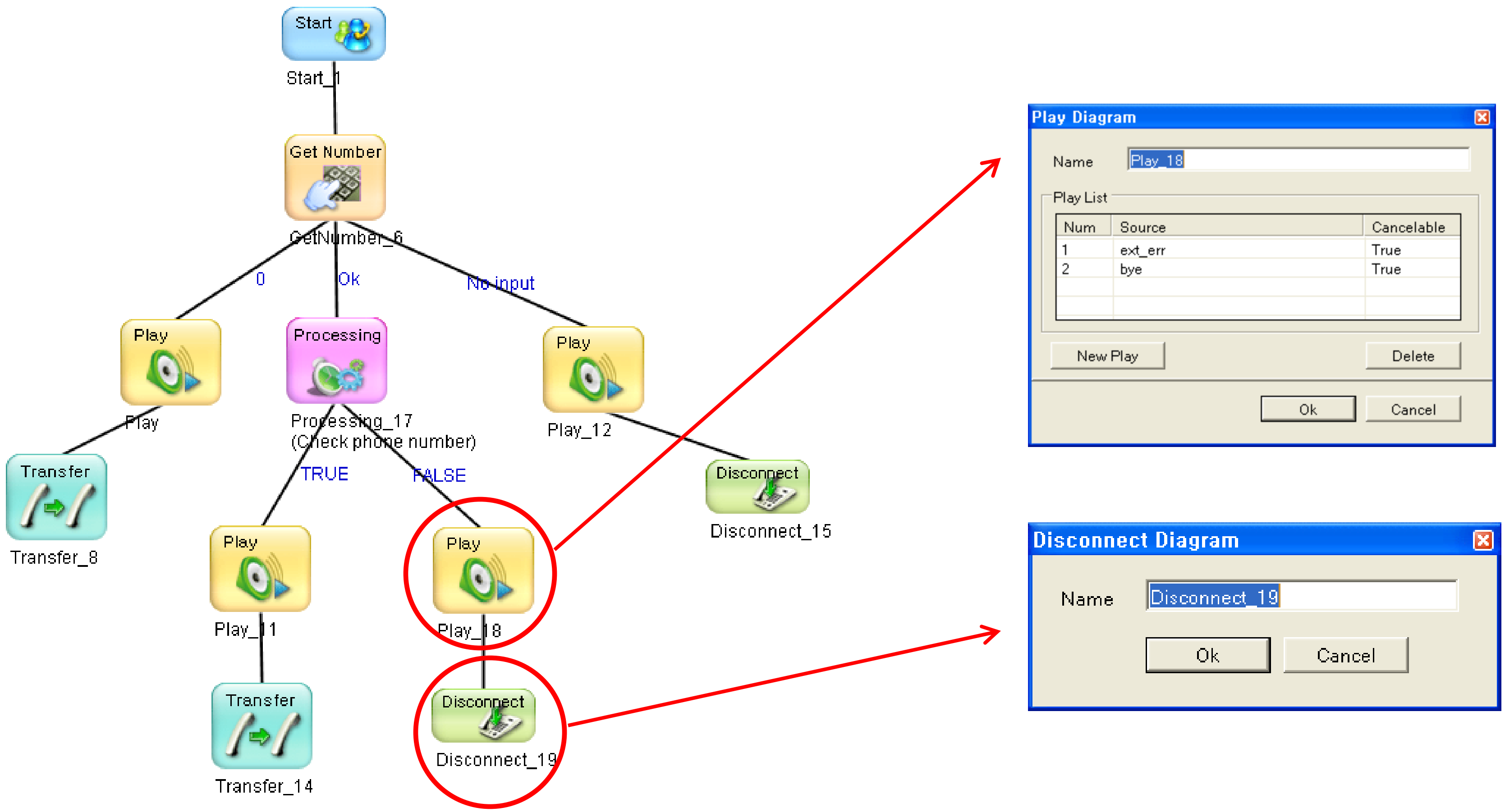
Example of Smart IVR Editor

6) Add more menus which connect with high level



Example of Smart IVR Editor

6) Add more menus which connect with high level



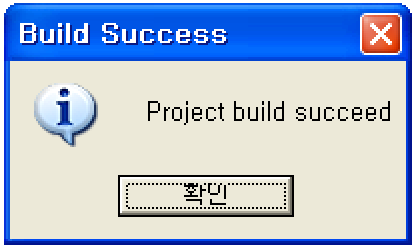
Example of Smart IVR Editor

7) Build the Project

When you are finished with writing out the scenario by using the Diagrams and Links, you can perform Project Build.

You may click Project -> Build (or Ctrl + B) to start Project Build.

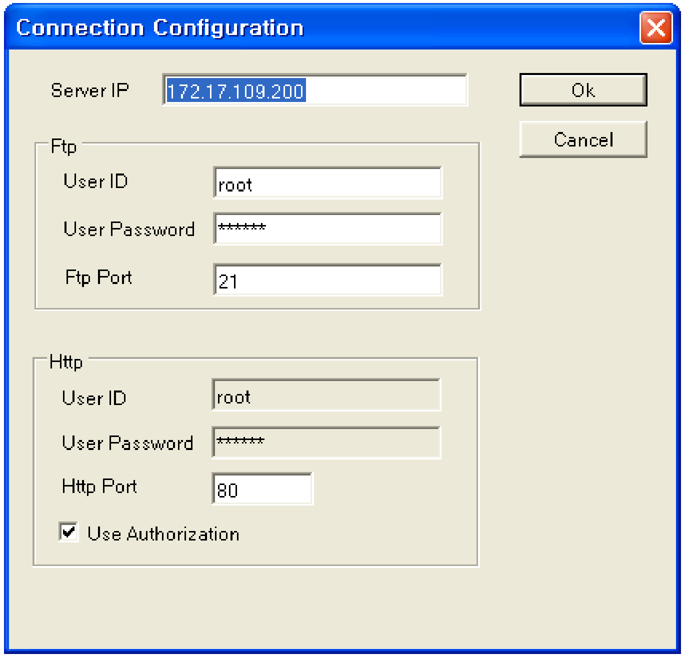
When the Project Build is finished, a message of success is displayed as it is shown



8) Set Connection Configuration

Set server information

- Server IP(IPNext)
- FTP information (User ID: root / Password : router)
- Http information (User ID: root / Password : router)
- check Use Authorization

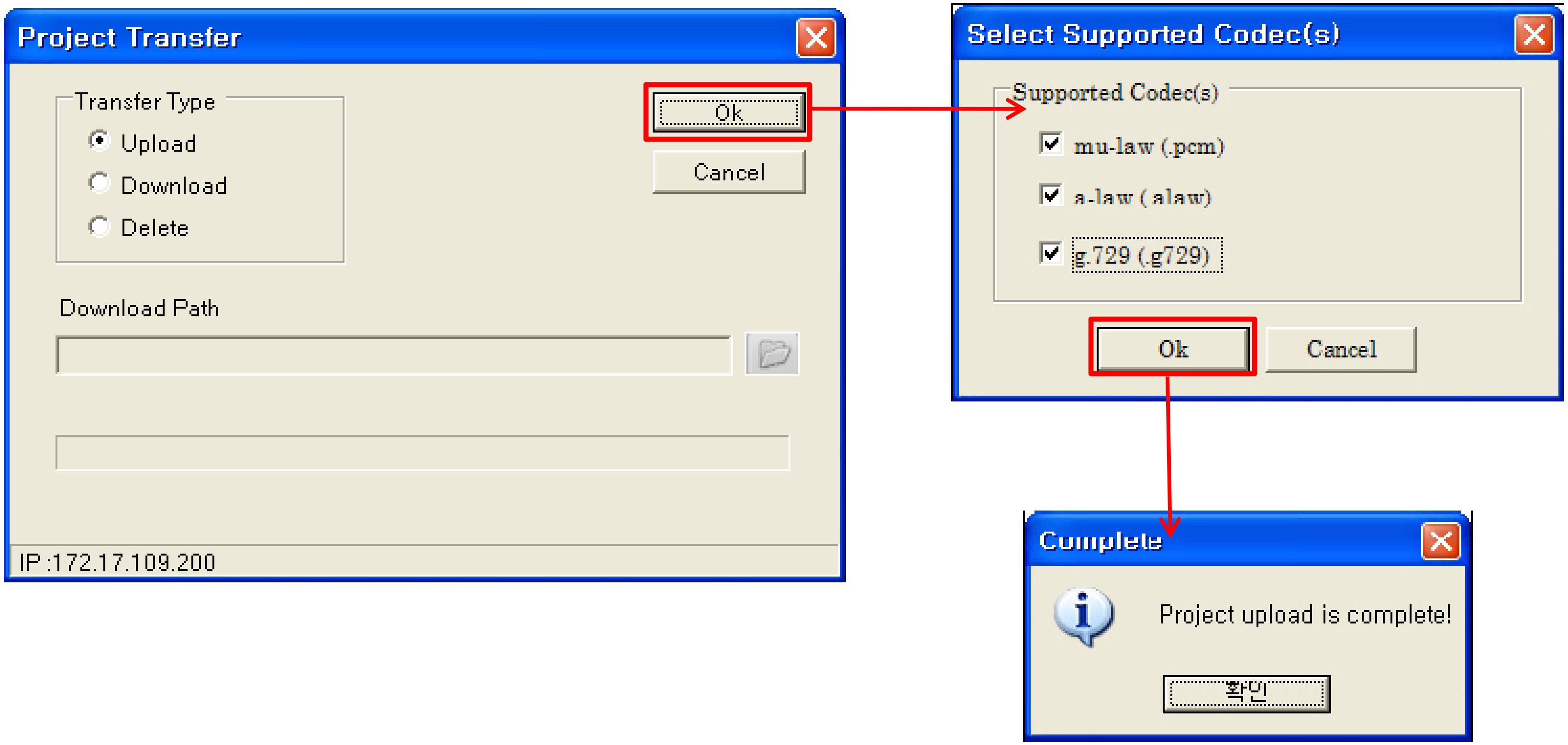


Example of Smart IVR Editor

7) Project Upload to IPNext

When the Project Transfer window opens as it is shown in below, set the Upload from the Transfer type and then click OK. You may see the window opens up for 3 types of Codec format to be selected.

when uploading the Project is completed.

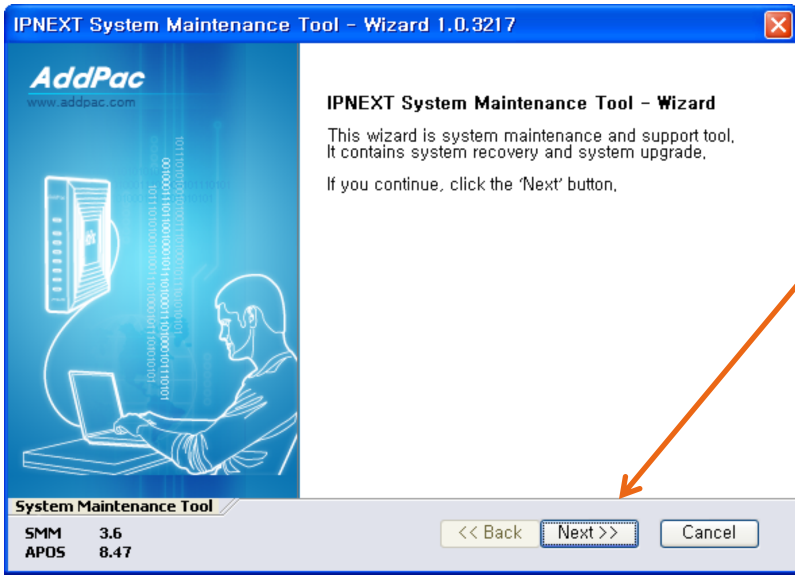


Appendix

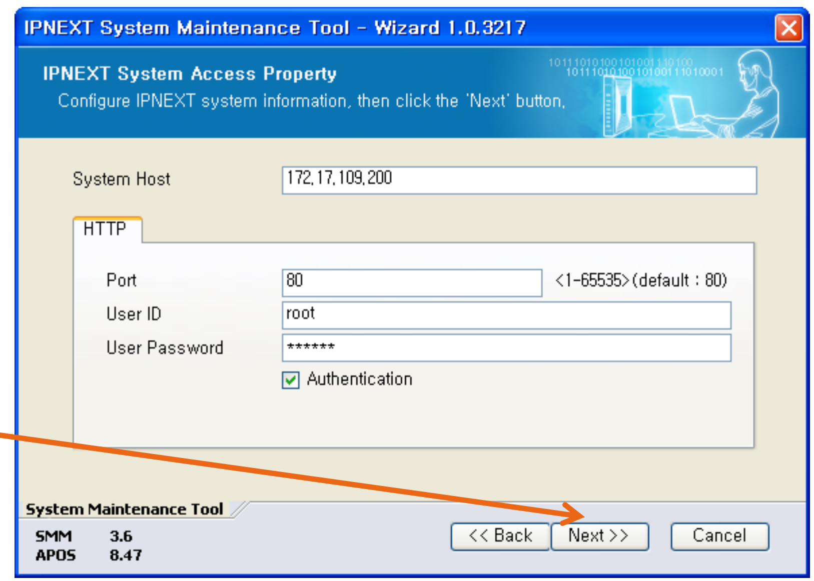
- IPNext (Initialize Ldap (SMT))

Initialize Ldap (SMT)

- Ldap install (SMT)

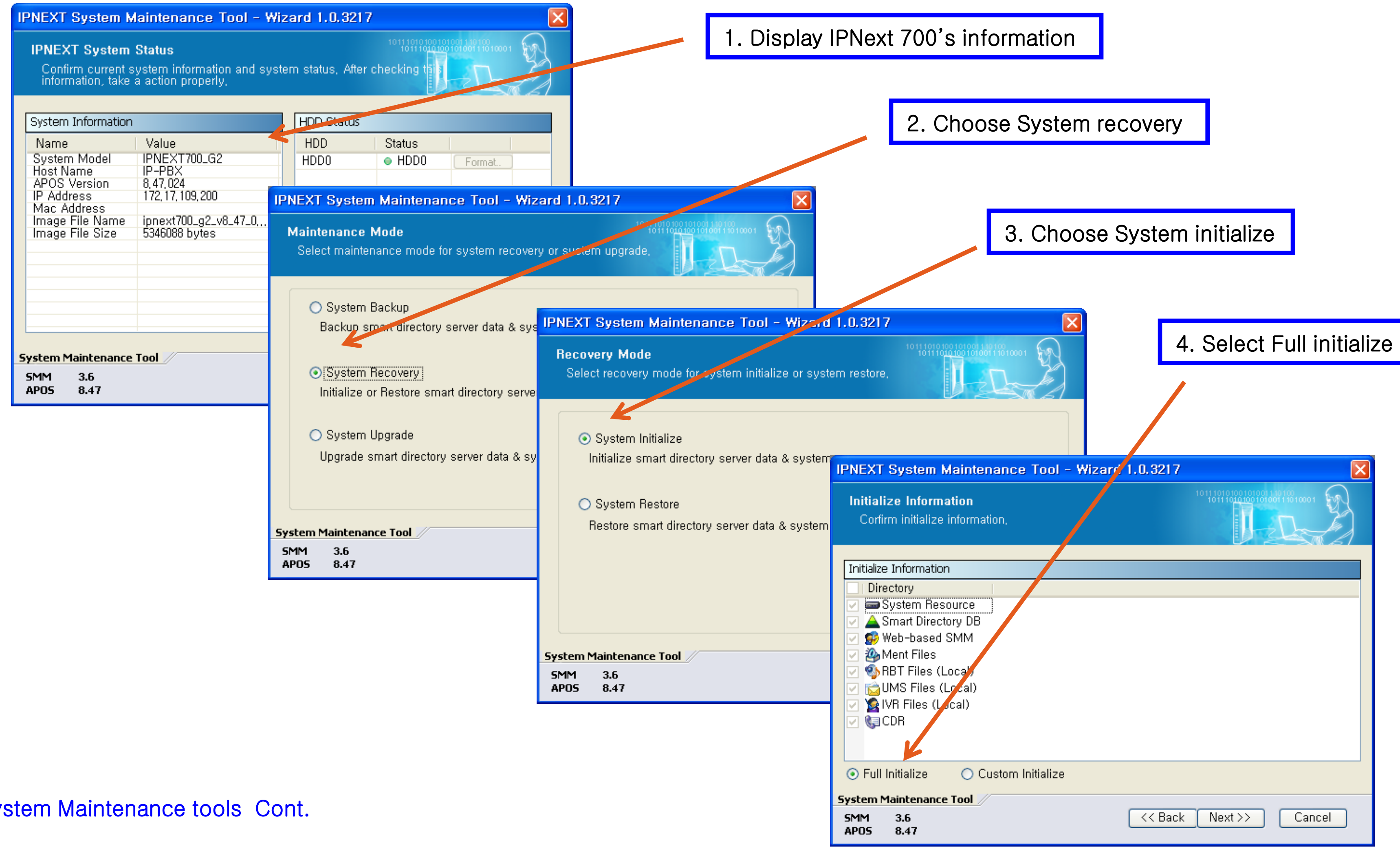


1. Main page for maintenance tool
Click to "Next"



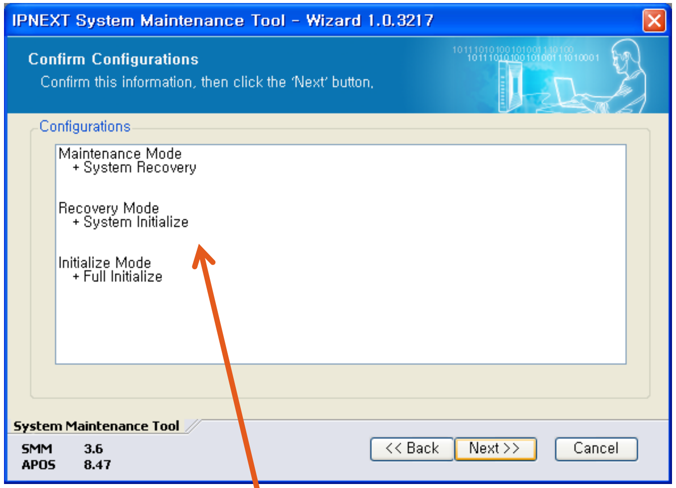
2. Set the IPNext's IP address to system host field
Don't need to change Port and user ID
(if you change IPNext's http port it need to
set same port)

Initialize Ldap (SMT)

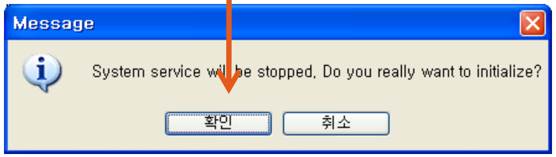


System Maintenance tools Cont.

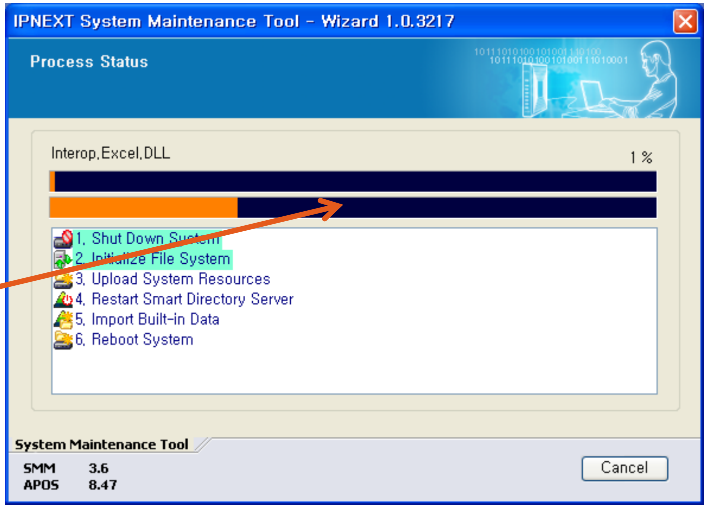
Installation guide for IPNext



2. If you already using IPPBX function, all IPPBX function be stop



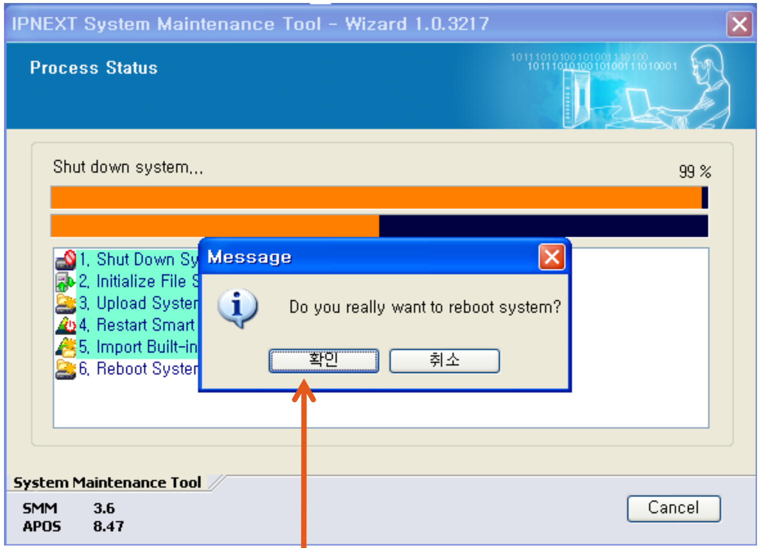
1. Display which function will do.



3. Stop all IPPBX function with restore all data to IPPBX

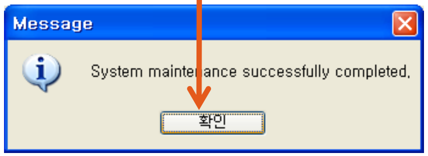
System Maintenance tools Cont.

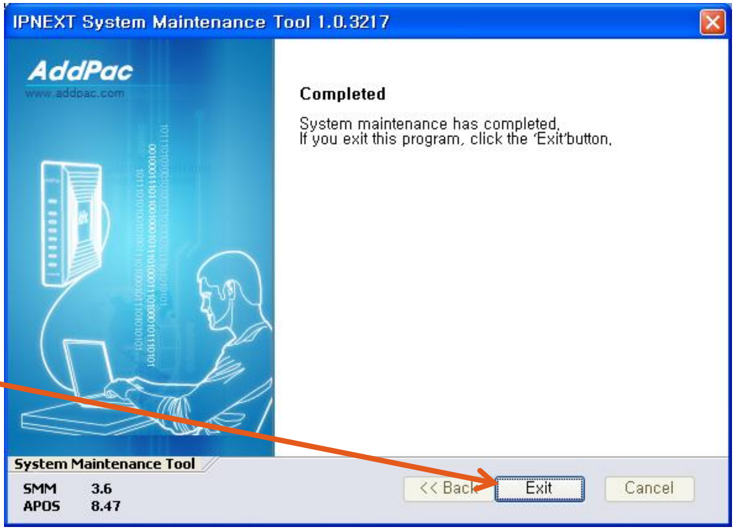
Installation guide for IPNext



1. After System initialize it need to reboot "Click OK"

2. Show all System initialize completed "Click OK"

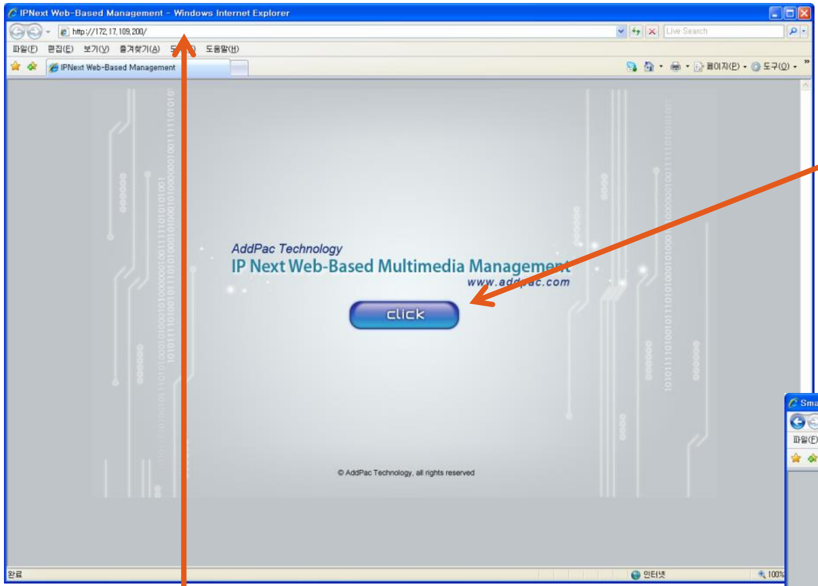




3. Display System initialize completed " Click Exit"

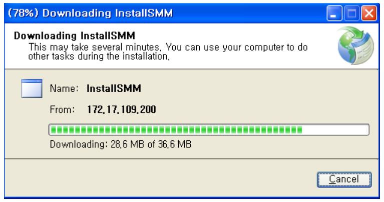
System Maintenance tools Cont.

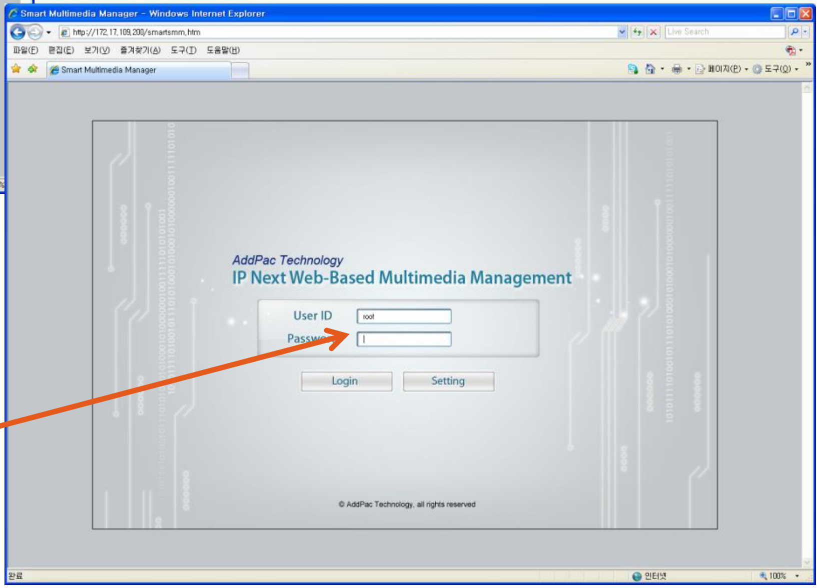
IPNext Stand alone Test



1. Access IPNext via web-browser
http://172.17.109.200:80

2. PC will download SMM
automatically when click icon.

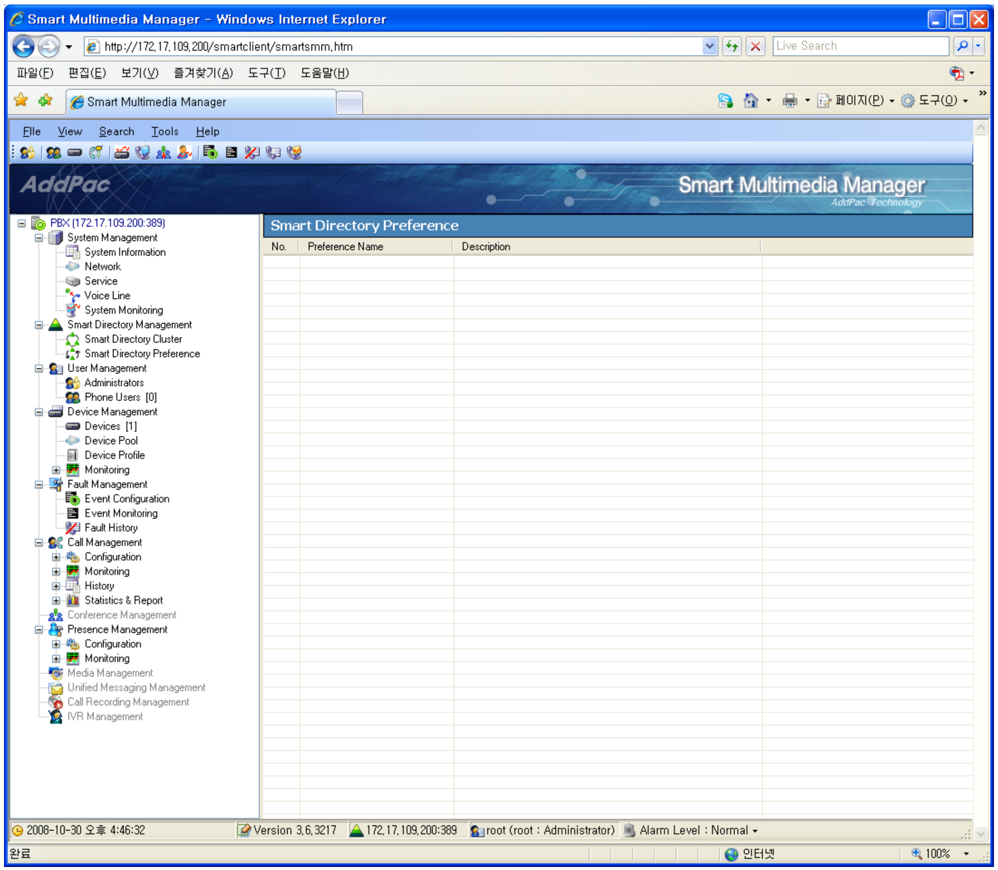




3. after download it'll display login page
login ID : root
password : router

IPNext Stand alone Test

After login Web SMM



Stand alone Simple Test Scenario.

1.Add 2~3 user datas..
2.Add 2~3 tel numbers.
3.Register 2~3 terminals.
4.Make a call each other.
5.Make backup file.(excel or ldap)
6.Add more information.

Thank you