



Web Smart Multimedia Manager (WSMM)

MCU Built-in Full HD Video Conference Codec

AddPac

AddPac Technology

2014, Sales and Marketing

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- Video Conference & H.239 Document Sharing Management (40 Page)



Overview

What`s New in WSMM (Web based Smart Multimedia Manager)

- Simple Menu and Easy Configuration
- Provides Built-In IVR Scenario Editor and Service Configuration
- Provides easy-to-user IP-PBX System API Services and ways to integration with 3rd party systems
- Integrated voice line management such as FXS, FXO, E1, GSM, 3G
- User portal to configure personal information, call forwarding
- Diagnostic tool to analyze SIP Call flow, current status and problems for terminal and trunk
- Video Conference & H.239 Document Sharing

System Requirement

WSMM (Web based Smart Multimedia Manager)

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

Login

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

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

Help

Smart Multimedia Manager
www.addpac.com

Add an User Extension

Extension *
First Name
Last Name *
Voice Mail Password *
User Password *
Department
Title
Email
Home Phone
Mobile Phone
User ID
Photo
Routing Access List
Routing Access List: internal

Help :: User Extension

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

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


Related Links

Smart Multimedia Manager
www.addpac.com

Modify the User Extension

Apply Cancel Advanced Options

User Extension

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

Routing Access List

Routing Access List: internal

Advanced Options

Terminal Profile: default

General Settings

Security Profile: default

Use RTP Proxy:

Back Tone at:

Representation: Default

Description

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

Related Links

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

Related Links

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

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

Diagnostic

Smart Multimedia Manager
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Terminal Diagnostic 1009 (172.16.18.100)

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

Step 1.

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

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

1005 Start Outbound Test

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

2012-06-12 20:15:36 deviceId: 70 caller: 1009 callee: 1005 Call Test Start.
----- From 1009 (172.16.18.100:5060) -----

SIP/2.0 200 OK
Via: SIP/2.0/UDP 172.16.17.30:5060;branch=z9hG4bKd84f0b0fa411
From: <sip:dial-service@172.16.17.30>;tag=d84f0b0fa4
To: <sip:1009@172.16.18.100>;tag=dc4fa2c5a4
Call-ID: dca3d74f-519d-a2e8-80c5-0002a4038e2c@172.16.18.100
CSeq: 11 INVITE
Session-Expires: 1800;refresher=uac
User-Agent: AddPac SIP Gateway
Contact: sip:1009@172.16.18.100
Require: timer
Content-Type: application/sdp
Content-Length: 179

v=0
o=1009 1339532254 1339532254 IN IP4 172.16.18.100
s=AddPac Gateway SDP
c=IN IP4 172.16.18.100
t=1339532254 0

/AVP 0
0000/1
(172.16.18.100:5060) -----

172.16.17.30:5060;branch=z9hG4bKd84f0b0fa411
service@172.16.17.30>;tag=d84f0b0fa4
2.16.18.100>;tag=dc4fa2c5a4
f-519d-a2e8-80c5-0002a4038e2c@172.16.18.100

1800;refresher=uac
ic SIP Gateway
9@172.16.18.100

lication/sdp
179

Step 2.

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Diagnostic

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

Step 1.

- Network Connection Test
- SIP Aware Test

Step 2.

- Outgoing Call Test

Built-in IVR Scenario Editor

Smart Multimedia Manager
www.addpac.com

IVR Scenarios

Apply Save Cancel

IVR Scenario Properties

Name: addpac
Description:

IVR Scenario sequence

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

Menu

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

Name * AddPac
File Path: hello_full .Open
 Cancelable
If this option is enabled, you can stop the sound by pressing any key.
Initial Timeout: 10
Allowable Count: 5
Single Digit: Add Single Digit Event of 3
Multi Digit: Add Multi Digit Event with Inter Digit Timeout 1 Sec and Max Digit Length 4

Description

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

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Built-in IVR Scenario Editor

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

IVR Scenario Sequence

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

Main

Smart Multimedia Manager
www.addpac.com

Extensons
Trunks
PBX Services
System Admin
Summary

Welcome to AddPac IP-PBX
root
Last Login at June 08 11:29:56AM (172.16.30.41)

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

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

Status

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

System
 Memory Storage: 0%
 Storage: 7%
 Network: [Icons]
 Call Manager: 0/100
 MCU: 0/2
 Presence: 0/100
 IVR: 0/100
 Media: 0/100
 UMS: 0/100
 RtpProxy: 0/100

Trunks
 Internal Trunk Gateway (0/0)
 SKN_TG (0/0)
 Dacom_Trunk (0/0)

FXS (1) [0, 1, 2, 3]
 E1 (0) [0, 1]
 FXO (1) [4, 5, 6, 7]
 GSM (2) [0, 1, 2, 3]

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

Alarm Message
It displays IP-PBX system errors

Short Cut
A short cut link.

Status
It displays current IP-PBX system major status

Main - Alarm History

The screenshot shows the Smart Multimedia Manager interface. The top section displays a welcome message for 'root' and an 'Unread Alarm Message' for 'login user authentication' on 2012-06-01 07:51:12. A red dashed box highlights this message, with a blue arrow pointing down to the 'Alarm History' page below. The 'Alarm History' page features a table with columns for Level, Messages, and DateTime, listing various system events such as NTP time sync service started, ftp service disabled, and disk upper quota limit exceeded.

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

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

Main – Quick Menu

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

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

Main – Follow Us

The screenshot displays the 'Smart Multimedia Manager' interface. On the left, a sidebar contains navigation options: Extensions, Trunks, PBX Services, System Admin, Summary, and a 'Getting Started' section with a 'GO' button. Below this is a 'Follow Us' section with icons for LinkedIn, Facebook, and YouTube. The main content area is divided into several sections: a 'Start' section with a 'Message failed' notification, a 'AddPac Technology' section with a 'facebook' logo and a 'AddPac Technology' profile card, and a 'Trunks' section. A red dashed box highlights the 'Follow Us' section and the social media links. A yellow box at the bottom left contains the text: 'Follow Us You may check AddPac product information, solution and etc. through Linked, Facebook, YouTube.' A copyright notice 'Copyright © AddPac 1999-2012 All Rights Reserved' is visible at the bottom of the interface.

Main – Status Monitoring

Smart Multimedia Manager
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Start

Welcome to AddPac IP-PBX

root
Last Login at June 08 11:29:56AM (172.16.30.41)

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

Quick Menu

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

Status

User Extensions

System

Memory Storage: 0%
Network: 7%

Call Manager
MCU Manager
Presence
IVR
Media
UMS
RtpProxy

FXS (1) E1 (0)
FXO (1) GSM (2)

Trunks

Voice Lines

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

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

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

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

Extension - Extensions

The screenshot displays the 'Smart Multimedia Manager' web interface. The left sidebar contains navigation options: Extensions, Trunks, PBX Services, System Admin, and Summary. The main content area is titled 'Smart Multimedia Manager' and shows a 'Welcome to AddPac IP-PBX' message. Below this, there is a 'Quick Menu' and a 'Status' section with a bar chart showing extension statistics: Registered (2), Unregistered (3), Unconfigured (0), and Unused license (95).

The 'Extensions' section is highlighted with a red dashed box. It contains a table with the following data:

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

Below the table is an 'Add an Extension' section with a 'Cancel' button and a detailed description of various extension types:

- Analog Extension:** An analog extension is a kind of user extension who has FXS (Foreign eXchange Station) analog voice line. Normal analog phone is connected at this extension.
- Hunt Group:** A hunt group has members of user extensions. Within a hunt group, an available member (user extension) can receive a call to the hunt group extension. A hunt group has one of simultaneous, sequential or random call hunting mode.
- Pickup Group:** A pickup group has members of user extensions who can pick up a ringing call within the group. The pickup group extension number is used for picking up a call by other group member.
- Conference Room:** A conference room extension is used for making a conference room. In case of dial-out conference, when a privileged user calls to conference room extension, all conference participants receive call to join. In case of meet-me conference, conference participants call to conference extension to join.
- IVR Extension:** An IVR (Interactive Voice Response) extension has a role of auto attendant for incoming calls from trunks. If incoming calls from trunk are routed to an IVR extension by incoming call rule, the interactive scenario will be proceed to transfer the call to a proper user extension.

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

Extension - Directory

Smart Multimedia Manager

Welcome to AddPac IP-PBX

Unread Alarm Message
No Unread Alarm Message

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

Directory

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

User Extension Status:

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

User Extensions System Trunks

Memory Storage: 1% / 8%
Network: [Icon]
Call Manager: 0/100
MCU: 0/2
Presence: 0/100
IVR: 0/100
Media: 0/100
UMS: 0/100
RtpProxy: 0/100

Trunks: Internal Trunk Gateway (0/0), SKN_TG (0/0), Dacom_Trunk (0/0)

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

Extension - Routing Access List

Smart Multimedia Manager
www.addpac.com

Start | Directory | Routing Access Lists

Welcome to AddPac IP-PBX
root
Last Login at June 11 04:38:52AM (172.16.1.50)

Unread Alarm Message
No Unread Alarm Message

Quick Menu

Status

User Extension

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

Smart Multimedia Manager
www.addpac.com

Start | Directory | Routing Access Lists

Routing Access Lists

Add a Routing Access List Refresh

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

Add a Routing Access List

Add Cancel

Routing Access List

Name *

Description

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

Outgoing Call Rules

Name

Allowed Outgoing Call Rules

Name

Description

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

Related Links

- Outgoing Call Rules

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

Extension - Terminal Profile

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with the following items: Extensions, Directory, Routing Access Lists, Terminal Profiles, Terminals, Trunks, PBX Services, System Admin, Summary, Getting Started, Follow Us, and social media links for LinkedIn, Facebook, and YouTube. A red dashed box highlights the 'Terminal Profiles' menu item, with a yellow starburst icon and a blue arrow pointing to the 'Terminal Profiles' section of the main content area.

The main content area is divided into two sections. The top section, titled 'Terminal Profiles', includes a table with the following data:

Modify	Delete	Name	Description	Date Created
1		default		2012-06-08 17:49:40

The bottom section, titled 'Global Terminal Settings', contains various configuration options:

- Calling Party Presentation: allowed, Restricted
- Language: Korean
- Call Duration Limit: 24 (1~48 Hour)
- Off-net Transfer: Off-net Transfer
- Initial Digit Timeout: 15000 (1000~100000ms)
- First Inter Digit Timeout: 3000 (1000~10000ms)
- Second Inter Digit Timeout: 3000 (1000~10000ms)
- Number of Digit(First Inter Digit Timeout): 4 (1~100)
- Internal Call: default
- External Call: default
- Internal Forwarded Call: default
- External Forwarded Call: default
- Keaplive Timeout: 30 (10~86400sec)

A 'Description' box on the right states: "Below settings are applied whole terminals in this system including trunks. Some VoIP settings can be customized to terminals by Terminal Profile."

A yellow box at the bottom left contains the following text:

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

Extension - Terminals

Smart Multimedia Manager

Welcome to AddPac IP-PBX

Unread Alarm Message

No Unread Alarm Message

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

Terminals

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

Modify the User Extension

Apply Cancel Advanced Options

Extension * 3000 3 ~ 8 digits

First Name BongYoung

Last Name * Jeong

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

User Password * 1111 For SIP registration

Department Search

Title ex) manager

Email ex) admin@addpac.com

Home Phone ex) 123-456-7890

Mobile Phone ex) 123-456-7890

User ID SIP registration ID

Photo (Maximum File Size: 100KB)

Description

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

Related Links

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

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

Trunk - Trunks

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains navigation menus for Extensions, Trunks, PBX Services, System Admin, and Summary. The main content area shows a 'Welcome to AddPac IP-PBX' message and an 'Unread Alarm Message' notification. Below this, there is a 'Quick Menu' and a 'User Extension' section with a status bar showing 2 Registered, 3 Unregistered, 0 Unconfigured, and 95 Unused licenses. The 'Trunks' section is highlighted with a red dashed box and contains a table of existing trunks:

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

Below the table is an 'Add a Trunk' section with a 'Cancel' button and three options: VoIP Trunk, SIP Proxy Server, and H.323 Gatekeeper, each with a detailed description. A yellow callout box points to the 'Trunks' menu item in the sidebar.

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

Trunk - Outgoing Call Rules

The screenshot shows the Smart Multimedia Manager interface. On the left, a navigation menu is visible with a red dashed box around the 'Trunks' section, which includes 'Trunks', 'Outgoing Call Rule', and 'Incoming Call Rules'. A yellow starburst icon is placed over the 'Outgoing Call Rule' link. A blue arrow points from this icon to the 'Outgoing Call Rules' page. The 'Outgoing Call Rules' page displays a table with one rule: 'external rule' with pattern '8T' and trunk '8T'. Below the table is the 'Add an Outgoing Call Rule' form, which includes fields for Name, Patterns, Trunks of Outgoing Call, Called Number Translation, and Number Translation. A 'Description' box on the right explains the function of an outgoing call rule.

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

Add an Outgoing Call Rule

Name *

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

Trunks of Outgoing Call *

Called Number Translation Add Rule

Number Translation

Calling Number Translation Add Rule

Description

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

Related Links

- Trunks

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

Trunk - Incoming Call Rules

Smart Multimedia Manager
www.addpac.com

Start

Welcome to AddPac IP-PBX

Unread Alarm Message
No Unread Alarm Message

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

Quick Menu

Status

User Extension

Getting Started GO

Follow Us

Linked in

facebook f

YouTube

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

Smart Multimedia Manager
www.addpac.com

Status Incoming Call Rules

Add an Incoming Call Rule

Add Cancel

Name *

Trunks of Incoming Call *

- Internal Trunk Gateway
- SM_SIP_Provider
- ss
- jschoLgk

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

Route to an extension by called number + Add Rule

Transfer Rule Modify Delete

Single Extension Routing

Route to an extension by calling number + Add Rule

Transfer Rule Modify Delete

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

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

Route DID Rule Modify Delete

Multiple Extension Routing(DID)

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

Called Pattern to delete digits from the front and adding

Trunk Routing to outgoing call rules external rule

Description

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

Related Links

- Trunks
- Outgoing Call Rules

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Incoming Call Rules

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

PBX Service - Speed Button Profiles

The screenshot displays the Smart Multimedia Manager interface. The left sidebar shows a navigation menu with 'PBX Services' expanded to 'Speed Button Profiles'. The main content area shows the 'Speed Button Profile' configuration page. A table lists one profile with ID 1, name 'button profile', and creation date '2012-04-02 10:43:18'. Below the table is a form for adding or editing speed buttons, including fields for Name, Extension, and Type. A 'Description' box on the right explains that speed buttons can be applied to IP phones and are limited by the number of buttons on the OSD or physical phone.

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

PBX Service - Announcement and Tones

Smart Multimedia Manager
www.addpac.com

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

Announcement and Tones
Language: Korean | Restore Default Ments | Global Media Setting | Refresh

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

Announcement and Tones
Apply | Cancel

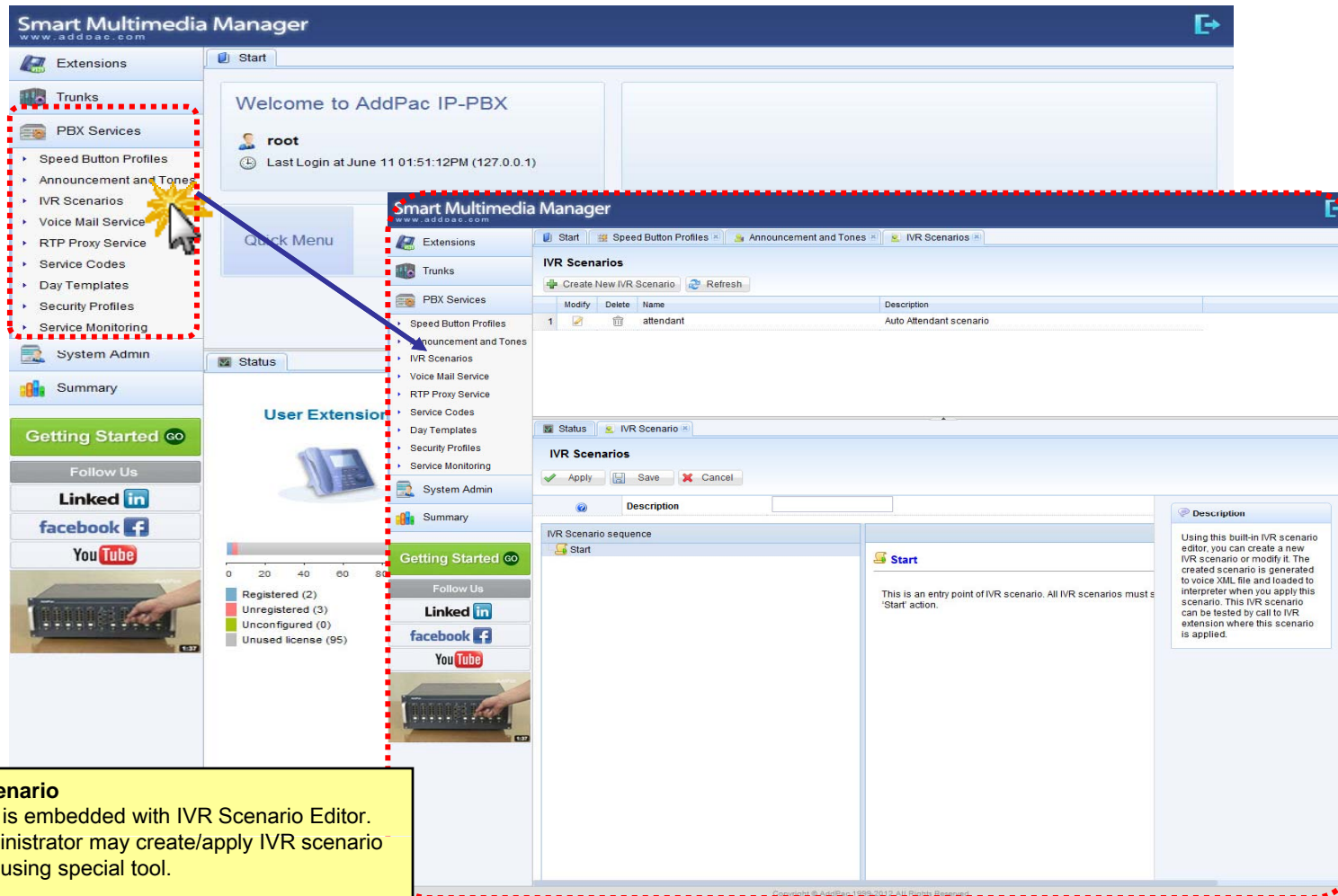
Announcement ID: 400110
Description: 연결 중 안내
Language: Korea

No.	Name	Start date	End Date	Start Time	End T
Create New Schedule					

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

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

PBX Service - IVR Scenarios



The screenshot displays the Smart Multimedia Manager (WSMM) interface. The left sidebar contains a navigation menu with 'PBX Services' expanded to show 'IVR Scenarios'. A yellow starburst highlights the 'IVR Scenarios' menu item, with a blue arrow pointing to the 'IVR Scenarios' tab in the main content area. The main content area shows the 'IVR Scenarios' configuration page, which includes a table of existing scenarios, a 'Create New IVR Scenario' button, and a detailed view of the 'attendant' scenario. The detailed view shows the 'IVR Scenario sequence' with a 'Start' action and a description box explaining the 'Start' action.

Modify	Delete	Name	Description
		attendant	Auto Attendant scenario

IVR Scenarios

Apply Save Cancel

IVR Scenario

Description

IVR Scenario sequence

Start

Start

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

Description

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

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

PBX Service - Voice Mail Services

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories like Extensions, Trunks, PBX Services, System Admin, and Summary. The 'PBX Services' menu is highlighted with a red dashed border, and a yellow starburst icon is placed over the 'Voice Mail Service' option. A blue arrow points from this icon to the 'Voice Mail Service' configuration page shown in the main content area. The configuration page includes fields for 'Retrieving Extension by Other Phone', 'Retrieving Extension by Owner Phone', and 'Leave Extension'. It also features an 'Advanced Options' section with settings for 'Audio Message Length', 'Per Extension Hdd Quota', 'Over HDD Quota', 'Use Account Blocking', 'Password Fail Count', 'Enable E-mail Notification', 'Attach File to Email', 'Delete File After Email Notification', 'Enable SMS Notification', and 'SIP Port'. A 'Description' box on the right explains the purpose of the settings.

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

PBX Service - RTP Proxy Service

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

PBX Service - Service Codes

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

The screenshot shows the 'Smart Multimedia Manager' interface. The left sidebar contains a 'PBX Services' menu with 'Service Codes' highlighted. The main content area shows the 'Service Codes' configuration page with various settings like 'Call Park', 'Call Pickup', 'Call Forwarding All Register', etc. A 'Description' box on the right explains that a service code is a special digit starting with # or *.

PBX Service - Day Templates

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories like Extensions, Trunks, PBX Services, System Admin, and Summary. The 'PBX Services' menu is expanded, and 'Day Templates' is highlighted. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root'. Below this, there's a 'Quick Menu' and a 'User Extension' section with a status bar showing 2 registered, 3 unregistered, 0 unconfigured, and 95 unused licenses. The 'Day Templates' section is active, showing a table with one entry: 'holiday' created on 2012-03-30 11:24:41. Below the table is a form to add a new day template with fields for Name and Description.

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

PBX Service - Security Profiles

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

Security Profiles
IP-PBX supports TLS Cipher Suites. User may select priority with 3 TLS Suites and may select RC4_40, RC4_128, DES_CBC, 3DES_CBC, AES_128_CBC, AES_256_CBC, SEED_CBC, ARIA_CBC in each suites.

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

Global Security Setting

Apply Cancel

TLS Cipher Suites

- N/A
- RC4_40
- RC4_128

DES_CBC
3DES_CBC
AES_128_CBC
AES_256_CBC
SEED_CBC
ARIA_CBC

Description

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

PBX Service - Service Monitoring

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

System Admin - Network Interface

Smart Multimedia Manager
www.addpac.com

Start

Welcome to AddPac IP-PBX

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

Quick Menu

System Admin

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

Summary

Getting Started GO

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User Extension

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

Network Interfaces

Apply Cancel

Interface Mode: DHCP Static IP

WAN Interface

IP Address * A.B.C.D

Subnet Mask * A.B.C.D

Default Gateway A.B.C.D

Primary DNS Server

Secondary DNS Server

IPv6 Address

IPv6 Default Gateway

LAN Interface

Interface Mode: None Bridge IP Shared NAT Static IP

IP Address

Subnet Mask

DHCP Server On Off

DHCP Range ~ A.B.C.D

IPv6 Address

Description

This PBX system can have one or two network interfaces. The WAN interface is a main network interface of this system normally has public IP address for communicating with VoIP providers and Trunk gateways in public domain. The LAN interface normally has private IP address for communicating with IP phones or user terminals in private domain.

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

System Admin - Network Services

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

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

System Admin - Administrators

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with the following items: Extensions, Trunks, PBX Services, System Admin, Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, Show Command, and Summary. The System Admin menu item is highlighted with a red dashed box, and a blue arrow points from it to the Administrators table in the main content area.

The main content area shows the Administrators management page. At the top, there is a "Welcome to AddPac IP-PBX" message and a "Click Menu" button. Below this, there is a "User Extension" section with a progress bar and a "Getting Started" button. The Administrators table is displayed with the following data:

Modify	Delete	Name	ID	Level	Description
1		root	root	Administrator	System Administrator
2		administrator	administrator	Administrator	Addpac Administrator

Below the table, there is a form for adding or modifying an administrator. The form includes the following fields:

- User name*
- Description
- ID*
- Password*
- Level (Administrator)
- Application Permission (Door Access Control Manager, Time and Attendance Manager)

A yellow box at the bottom left of the screenshot contains the following text:

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

System Admin - Licenses

The screenshot displays the 'Smart Multimedia Manager' interface. The left sidebar contains a 'System Admin' menu with options like Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, and Show Command. The main content area is titled 'Licenses' and includes a table of license settings. A yellow callout box provides additional information about the license policy.

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

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

System Admin - Voice Lines

Smart Multimedia Manager
www.addpac.com

Start

Welcome to AddPac IP-PBX

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

Click Menu

System Admin

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

Summary

Getting Started GO

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User Extension

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

Smart Multimedia Manager

Extensions Trunks PBX Services System Admin

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

Getting Started GO

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Voice Lines

Apply Cancel

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

Analog & Mobile

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

Digital

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

Extension Analog

Description

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

Related Links

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

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

System Admin - Alarm History

Smart Multimedia Manager
www.addpac.com

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

System Admin

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

Alarm History

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

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

System Status

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

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

Trunks: Internal Trunk Gateway (0/0), SKN_TG (0/0), Dacom_Trunk (0/0)

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

System Admin - Call History

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

System Admin - Show Command

The screenshot displays the Smart Multimedia Manager System Admin interface. The left sidebar contains a navigation menu with 'System Admin' expanded to show options like 'Network Interfaces', 'Network Services', 'Administrators', 'Licenses', 'Voice Lines', 'Alarm History', 'Call History', and 'Show Command'. A red dashed box highlights the 'Show Command' option, with a blue arrow pointing to the 'Show Command' tab in the main content area. The 'Show Command' tab is active, showing a 'Request Command' field with the text 'show call-manager sscp'. Below this, the system output is displayed in a code block format. A 'Description' box on the right explains that users can check system status by selecting a category or entering a command to the CLI.

System Admin - Show Command

Categories: System, VoIP, Call Manager, Command Line Interface

Request Command: show call-manager sscp

```
SSCP Timer Information.
  retry-counter = 1
  retry-timeout = 5000 (msec)
  keepalive-timeout = 30 (sec)

CM <-> CM_Servers SSCP Information.
  retry-counter = 3
  retry-timeout = 5000 (msec)
  keepalive-timeout = 3 (sec)
  keepalive-retry-timeout = 1000 (msec)
  keepalive-retry-timeout(other server) = 3000 (msec)
  binding accept status = TRUE

SSCP Policy Information.
  cm service status = on
  signaling-port = 8855
  packet-size = 1472

  client-session logging count = 0
  client-session registering count = 0

  event store time = 3(sec)
  event store count = 30
  event total store count = 0

Client Auth Session Information.
  id          ip          timerCount
  -----
  -----

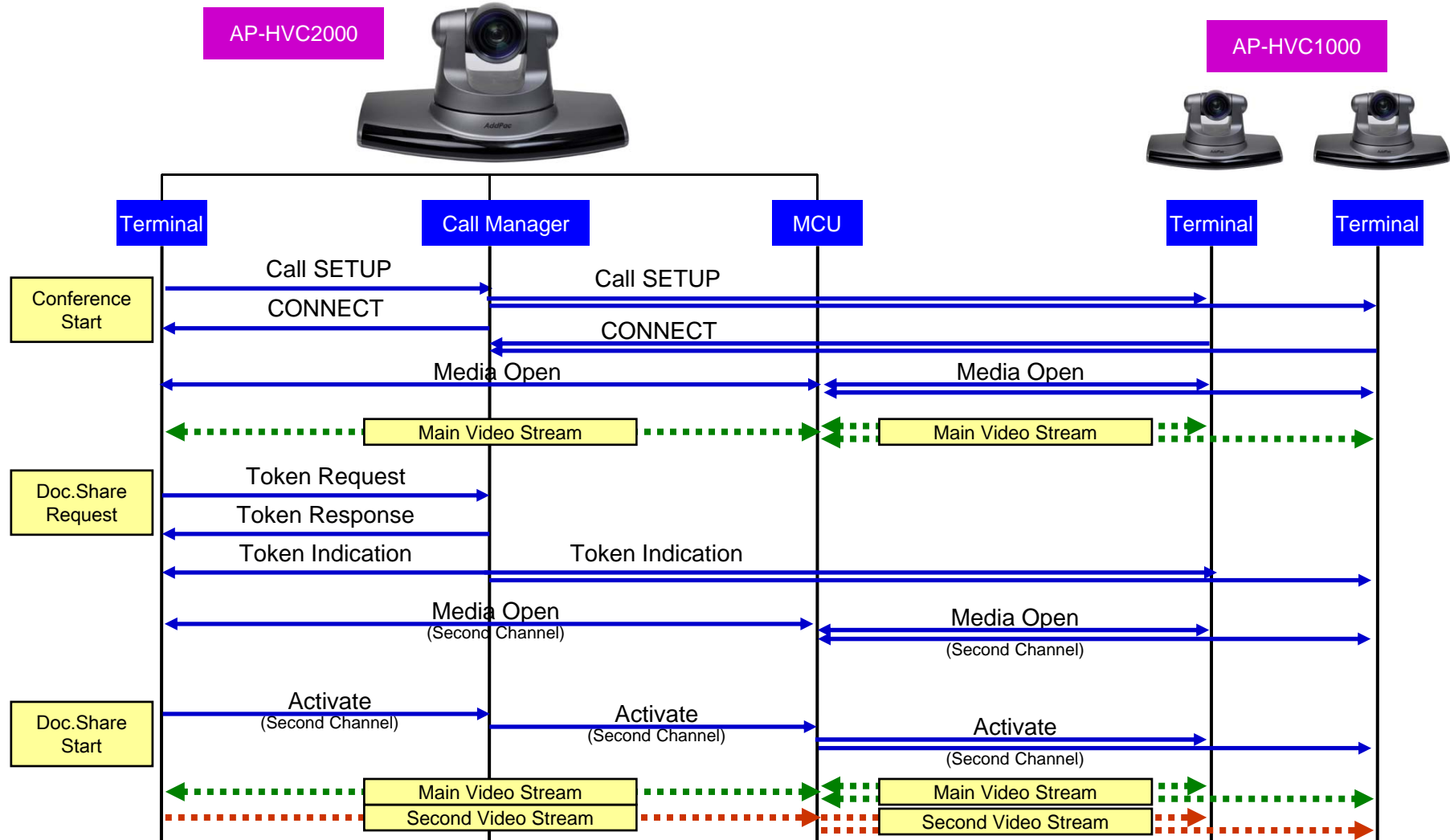
Client Sesion Information.
  session-id  user-id      ip address      port  status
  -----
  2001        1009        172.16.18.100  5060  in-service
  2002        3000        172.16.18.101  5060  in-service
  -----

Servers Information.
  server-id   binding-id   ip address      port  state
  -----
  10200000    1            172.16.17.30   5101  BIND    ums
  10100000    1            172.16.17.30   5045  BIND    rdt
  10600000    1            172.16.17.30   5021  BIND    tvr
  -----

SessionClientGroup
Group(0) sessionSize(0) :
Group(2) sessionSize(0) :
```

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

Conference & Doc. Share Call Flow



Conference - Create Room 1

The screenshot displays the 'Smart Multimedia Manager' web interface. The left sidebar contains navigation options: Extensions, Directory, Partitions, Routing Access Lists, Terminal Profiles, Terminals, Trunks, PBX Services, System Admin, Servers, Advanced, Monitoring, and Summary. The main content area shows the 'Add an Extension' configuration page. At the top, there is a table of existing extensions:

	Modify	Delete	User Portal	Extension Number	Type	Name	Date Created
1					Conference Room	Ad-Hoc Defaults	1970-01-16 10:01:34
2				0001	Voice Mail	vmail_rec	1970-01-16 10:01:38
3				0002	Voice Mail	vmail	1970-01-16 10:01:38
4				0003	Voice Mail	vmail_noauth	1970-01-16 10:01:38

Below the table, the 'Add an Extension' page is shown. It includes a 'Cancel' button and several extension types with descriptions:

- Batch Job for User Extensions:** Gives you simple and automated way to add, modify or delete one or more extensions through CSV (Comma Separated Values) file. Each CSV file can be created with your favorite text editor or Microsoft Excel.
- Hunt Group:** A hunt group has members of user extensions. Within a hunt group, an available member (user extension) can receive a call to the hunt group extension. A hunt group has one of simultaneous, sequential or random call hunting mode.
- Pickup Group:** A pickup group has members of user extensions who can pick up a ringing call within the group. The pickup group extension number is used for picking up a call by other group member.
- Park Pool:** A park pool is a set of extensions for parking calls. When a user parked an active call, an extension in this pool will be assigned. Other user can pick up the parked call using the parked extension number.
- Conference Room:** A conference room extension is used for making a conference room. The conference room can be open by WSMM or User Portal web page or by call to conference room number by privileged user (chair or operator) or by schedule. In case of dial-out participants, they receive call when conference is opening. In case of dial-in participants, they have to make a call to conference extension to join to opened conference.
- IVR Extension:** An IVR (Interactive Voice Response) extension has a role of auto attendant for incoming calls from trunks. If incoming calls from trunk are routed to an IVR extension by incoming call rule, the interactive scenario will be proceed to transfer the call to a proper user extension.
- Push-to-Talk Group:** A PTT (Push to Talk) group has members of user extensions who will receive broadcasting announcement with auto answering and also can be a floor (speaker role) by pushing the talk button. This is half-duplex two-way broadcasting.

The 'Conference Room' section is highlighted with a red dashed box. The footer of the page contains the text: 'Copyright © AddPac 1999-1970 All Rights Reserved' and 'Version 1.2.131101'.

Conference - Create Room 2

Smart Multimedia Manager
www.addpac.com

Extensions

Start | Extensions

Extensions | Directory | Partitions | Routing Access Lists | Terminal Profiles | Terminals

Trunks

PBX Services

System Admin

Servers

Advanced

Monitoring

Summary

Getting Started GO

Clustering Guide GO

Partitioning Guide GO

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Add a Conference Room

✓ Add | ✗ Cancel | ⚙ Advanced Options

Set Conference Room Number & Name

Extension * 6000 (2-12 digits) Check Extension Extension number is valid.

Conference Name * ConfRoom1

Media Type Audio Audio + Video

Audio Codec G.711U

Video Codec H.264

Picture Size 1080p (1920*1080)

Bandwidth 1024K

Frame Rate 30 fps

Participants

Extension No.	Name	Type	User Class	Dialing Direction	Media Type	Position
6001		External	Chair	Dial-out	Video	
6002		External	Participant	Dial-out	Video	
6003		External	Participant	Dial-out	Video	
6004		External	Participant	Dial-out	Video	

External Phone: 6004

Add Conference participants

Description
A conference room extension is used for making a conference room. The conference room can be opened by WSMM or User Portal web page or by call to conference room number by privileged user (chair or operator) or by schedule. In case of dial-out participants, they receive call when conference is opening. In case of dial-in participants, they have to make a call to conference extension to join to opened conference.

Related Links
• User Extension
• Partitions

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Conference - Add Gatekeeper 1

The screenshot displays the 'Smart Multimedia Manager' web interface. The left sidebar contains navigation menus for 'Extensions', 'Trunks', 'PBX Services', 'System Admin', 'Servers', 'Advanced', 'Monitoring', and 'Summary'. The main content area is titled 'Trunks' and includes a dropdown menu for 'All Trunks', an 'Add a Trunk' button (highlighted with a red dashed box), and a 'Refresh' button. Below this is a table with columns: Modify, Delete, Diagnose, Name, Type, IP Address, State, Description, and Date Created. The 'Add a Trunk' modal window is open, showing a 'Cancel' button and a list of trunk types: 'VoIP Trunk', 'SIP Proxy Server', 'H.323 Gatekeeper' (highlighted with a red dashed box), and 'Call Manager Trunk'. Each type has a brief description. A 'Description' box on the right explains that trunks allow users to communicate with remote users in public telephone networks or other VoIP networks.

Smart Multimedia Manager
www.addpac.com

Start Extensions Trunks

Trunks
Outgoing Call Rules
Incoming Call Rules

PBX Services
System Admin
Servers
Advanced

Monitoring
Summary

Getting Started GO
Clustering Guide GO
Partitioning Guide GO

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VoIP Trunk
This is a generic VoIP Trunk which can register to this PBX or communicate without registration. The VoIP Trunk could be VoIP gateway which has analog FXS, FXO, E&M line, digital E1, T1 line or mobile GSM line, or IP-PBX or other SIP / H.323 Trunk.

SIP Proxy Server
This could be VoIP service provider who operates SIP Proxy Server and provides VoIP service to public telephone network or mobile network or other VoIP network. Also, this could be an IP-PBX who provides SIP server features. This PBX should register to the SIP Proxy Server for receiving incoming calls and sending outgoing calls.

H.323 Gatekeeper
This could be VoIP service provider who operates H.323 Gatekeeper and provides VoIP service to public telephone network or mobile network or other VoIP network. Also, this could be an IP-PBX who provides H.323 Gatekeeper features. This PBX should register to the H.323 Gatekeeper for receiving incoming calls and sending outgoing calls.

Call Manager Trunk
This is a trunk between Call Managers of IPNext PBX. In case of center PBX WSMM, add branch PBXs as this trunk, and in case of branch PBX WSMM, add center PBXs as this trunk. This Call Manager Trunk can be assigned at Call Manager Preferences of a Device Pool menu.

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

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Conference - Add Gatekeeper 2

The screenshot displays the 'Add a H.323 Gatekeeper' configuration page in the Smart Multimedia Manager. The interface includes a sidebar with navigation options like 'Trunks', 'PBX Services', and 'System Admin'. The main content area is titled 'Add a H.323 Gatekeeper' and contains several sections:

- Form Fields:** Name (Gatekeeper), Description, IP Address / Hostname (172.17.63.56), Gatekeeper ID (1719), Secondary IP Address / Hostname, Secondary Gatekeeper ID, H.323 ID (Conference), H.323 Password, and Register (checked).
- Advanced Options:** Device Pool (default), Location (N/A), DTMF Relay Mode (RTP-2833), G.729 Codec Variant (Standard), H.245 Tunneling (unchecked), Keep GK on RRJ (unchecked), Network Domain (public), Use RTP Proxy (unchecked), Call Priority (4), and Called Number Translation (Add Rule).
- General Settings:** A section with a lock icon, currently empty.
- Description:** A text area with a placeholder: 'This could be VoIP service provider who operates H.323 Gatekeeper and provides VoIP service to public telephone network or other VoIP network. Also, this could be an IP-PBX who provides H.323 Gatekeeper features. This PBX should register to the H.323 Gatekeeper for receiving incoming calls and sending outgoing calls.'
- Related Links:** A list of links: 'Outgoing Call Rules', 'Security Profiles', 'Device Pool', and 'Location'.

A red dashed box highlights the main configuration fields, and a yellow callout box contains the text: **Set Gatekeeper use local gatekeeper of HVC2000**

Conference - Add Gatekeeper 3

Smart Multimedia Manager
www.addpac.com

Extensions | Start | Extensions | Trunks

Extensions
All Extensions | Input an Extension | Search | Advanced Search | Add an Extension | Refresh

Status | Add an Extension | Add a Trunk | H.323 Gatekeeper

Add a H.323 Gatekeeper
Add | Cancel | Advanced Options

Purpose of Trunking: Inter-Site Call
Do Not Generate CDR:

Tech Prefix

Register Extensions

Extension	Name	Type
6000	ConfRoom1	conference

Register Conference Room to GK

Description
This could be VoIP service provider who operates H.323 Gatekeeper and provides VoIP service to public telephone network or mobile network or other VoIP network. Also, this could be an IP-PBX who provides H.323 Gatekeeper features. This PBX should register to the H.323 Gatekeeper for receiving incoming calls and sending outgoing calls.

Related Links
• Outgoing Call Rules
• Security Profiles
• Device Pool
• Location

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Conference – Outgoing Call Rules

The screenshot displays the 'Smart Multimedia Manager' web interface. The top navigation bar includes 'Start', 'Extensions', 'Trunks', 'Outgoing Call Rules', and 'Incoming Call Rules'. The left sidebar contains various system management options. The main content area is titled 'Outgoing Call Rules' and features a table with columns for 'Modify', 'Delete', 'Name', 'Pattern', 'Trunk', and 'Date Created'. Below this is the 'Add an Outgoing Call Rule' form, which includes fields for 'Name' (set to 'Out_Rule'), 'Patterns' (set to '600'), and 'Trunks of Outgoing Call' (set to '1'). A red dashed box highlights the 'Add an Outgoing Call Rule' button and the 'Trunks of Outgoing Call' table. A yellow box highlights the text 'Set Outgoing Call Rule to GK'.

Smart Multimedia Manager
www.addpac.com

Start | Extensions | Trunks | Outgoing Call Rules | Incoming Call Rules

Outgoing Call Rules
Add an Outgoing Call Rule Refresh

Modify	Delete	Name	Pattern	Trunk	Date Created
--------	--------	------	---------	-------	--------------

Status | Add an Extension | Add a Trunk | Outgoing Call Rules

Add an Outgoing Call Rule
Add Cancel Advanced Options

Name * Out_Rule

Patterns * 600 (([0-9#*])|TF)

Trunks of Outgoing Call

Modify	Delete	Select a Trunk
1		Gatekeeper

Called Number Translation Add Rule Translation Rule

Calling Number Translation Add Rule Translation Rule

Advanced Options

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

Related Links
• Trunks

Set Outgoing Call Rule to GK

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Conference – Incoming Call Rules

Smart Multimedia Manager
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Start | Extensions | Trunks | Outgoing Call Rules | Incoming Call Rules

Incoming Call Rules

Status | Add an Extension | Add a Trunk | Incoming Call Rules

✓ Add ✗ Cancel

Add an Incoming Call Rule

Name * In_Rule

Trunks of Incoming Call * Gatekeeper

Set Incoming Call Rule From GK

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

Related Links
• Trunks
• Outgoing Call Rules

Getting Started GO
Clustering Guide GO
Partitioning Guide GO

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Thank you!

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