



# Web Smart Multimedia Manager (WSMM) IP-PBX Series

Smart Multimedia Manager  
www.addpac.com

Extensions  
Trunks  
PBX Services  
System Admin  
Summary

Getting Started

Follow Us  
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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 an Analog Extension  
> Add a Conference Room  
> Add a VoIP Trunk  
> Add an Outgoing Call Rule  
> Add an Incoming Call Rule  
> Extensions  
> Terminals

Status

User Extensions  
System  
Trunks

Memory 0%  
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

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

Internal Trunk Gateway (0/0)  
SKN\_TG (0/0)  
Dacem\_Trunk (0/0)

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# AddPac

## AddPac Technology

### Sales and Marketing

# Contents

- Overview
- System Requirement
- WSMM Login
- Extension Management
- Trunks Management
- PBX Services Management
- System Admin Management
- Summary
- User Portal web page



# Overview

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

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

# System Requirement

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

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



# Login

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

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

# Help

The screenshot displays the 'Smart Multimedia Manager' web interface. The main content area is titled 'Add an User Extension' and contains a form with fields for Extension, First Name, Last Name, Voice Mail Password, User Password, Department, Title, Email, Home Phone, Mobile Phone, User ID, Photo, and Routing Access List. Below the form are 'Advanced Options' and 'General Settings' sections. A red dashed box highlights a help icon (a question mark in a blue circle) next to the 'User Extension' section. An arrow points from this icon to a help window titled 'Help :: User Extension'. The help window contains the following text:

- 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**

**General Settings**

Terminal Profile	default
Security Profile	default
Use RTP Proxy	<input type="checkbox"/>
Back Tone at	<input checked="" type="checkbox"/>
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 setup page provides related link functions. Related links helps easy operation of IP-PBX by providing link.

# Diagnostic

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

**Terminal Diagnostic** 1009 (172.16.18.100)

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

**Step 1.**

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

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

1005 **Start Outbound Test**

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

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

**Step 2.**

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## Diagnostic

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

Step 1.

- Network Connection Test
- SIP Aware Test

Step 2.

- Outgoing Call Test



# 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%  
 Network: 7%  
 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 displays the Smart Multimedia Manager interface. The top section shows a welcome message for 'root' and an 'Unread Alarm Message' for 'login user authentication' on 2012-06-01 07:51:12. A red dashed box highlights this message, with a blue arrow pointing down to the 'Alarm History' page below. The 'Alarm History' page features a table with columns for Level, Messages, and DateTime, showing a list of system events.

**Alarm History**

Level: All Ack: All Period: 2012-06-01 ~ 2012-06-08 Search Refresh

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

Displaying 1 - 6 of 6

**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 (3-8 digits), First Name, Last Name, Voice Mail Password (4 digits), User Password (for SIP registration), Department (with a search button), Title (example: ex) manager), Email (example: ex) admin@addpac.com), and Home Phone (example: ex) 123-456-7890). 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 with a 'Registered (2)' status bar. 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.'

# Main – Status Monitoring

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

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

**Status**

**User Extensions**

**System**

Memory Storage: 0%  
Network: 7%

Call Manager  
MCU  
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	

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

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## 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 a navigation menu with 'Extensions' highlighted. The main content area shows a 'Welcome to AddPac IP-PBX' message and an 'Unread Alarm Message' notification. Below this is a 'Quick Menu' and a 'Status' section with a 'User Extension' overview. The 'Extensions' table lists various extension types and their details.

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

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

**Add an Extension**

- 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 - 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

FXS (1): 0, 1, 2, 3  
E1 (0): 0, 1  
FXO (1): 4, 5, 6, 7

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**

Start | Directory | Routing Access Lists

### Routing Access Lists

Add a Routing Access List Refresh

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

Status | Routing Access List

#### Add a Routing Access List

Add Cancel

**Routing Access List**

Name \*

Description

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

Outgoing Call Rules	
Name	

Allowed Outgoing Call Rules	
Name	

Allowed Outgoing Call Rules

Description

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

Related Links

- Outgoing Call Rules

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



# Extension - Terminal Profile

The screenshot displays the Smart Multimedia Manager web interface. On the left, a navigation menu is visible with a red dashed box highlighting the 'Extensions' section, which includes 'Extensions', 'Directory', 'Routing Access Lists', 'Terminal Profiles', and 'Terminals'. A yellow starburst icon is placed over the 'Terminal Profiles' link, with a blue arrow pointing to the 'Terminal Profiles' tab in the main content area.

The main content area shows the 'Terminal Profiles' configuration page. It includes a table with the following data:

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

Below the table, the 'Global Terminal Settings' section is visible, containing 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
- Keepalive 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.'

At the bottom left, a yellow box 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. On the left, a navigation menu highlights the 'Trunks' option under the 'Extensions' category. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root' and an 'Unread Alarm Message' section. Below this, a 'Trunks' table lists three entries:

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 descriptive text box. A legend at the bottom left of the interface shows: Registered (2), Unregistered (3), Unconfigured (0), and Unused license (95).

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



# Trunk - Outgoing Call Rules

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

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

**Add an Outgoing Call Rule**

Name \*

Patterns \*

Trunks of Outgoing Call \*

Called Number Translation

Number Translation

Calling Number Translation

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

**Related Links**  
• Trunks

# Trunk - Incoming Call Rules

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

**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
- jschoL\_gk

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

Route to an extension by called number + Add Rule

Transfer Rule Modify Delete

Single Extension Routing

Route to an extension by calling number + Add Rule

Transfer Rule Modify Delete

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

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

Route DID Rule Modify De

Multiple Extension Routing(DID)

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

Called Pattern to delete digits from the front and adding

Trunk Routing to outgoing call rules external rule

Description

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

Related Links

- Trunks
- Outgoing Call Rules

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# PBX Service - Speed Button Profiles

**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

The screenshot displays the 'Smart Multimedia Manager' web interface. The left sidebar menu is highlighted with a red dashed box, and a yellow starburst icon points to the 'Announcement and Tones' option. The main content area shows a table of announcements with columns for ID, Description, and Scheduled status. Below the table, there is a detailed view for a specific announcement (ID 400110) with fields for Description, Language, and File List.

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 Information	
Announcement ID	400110
Description	연결 중 안내
Language	Korea

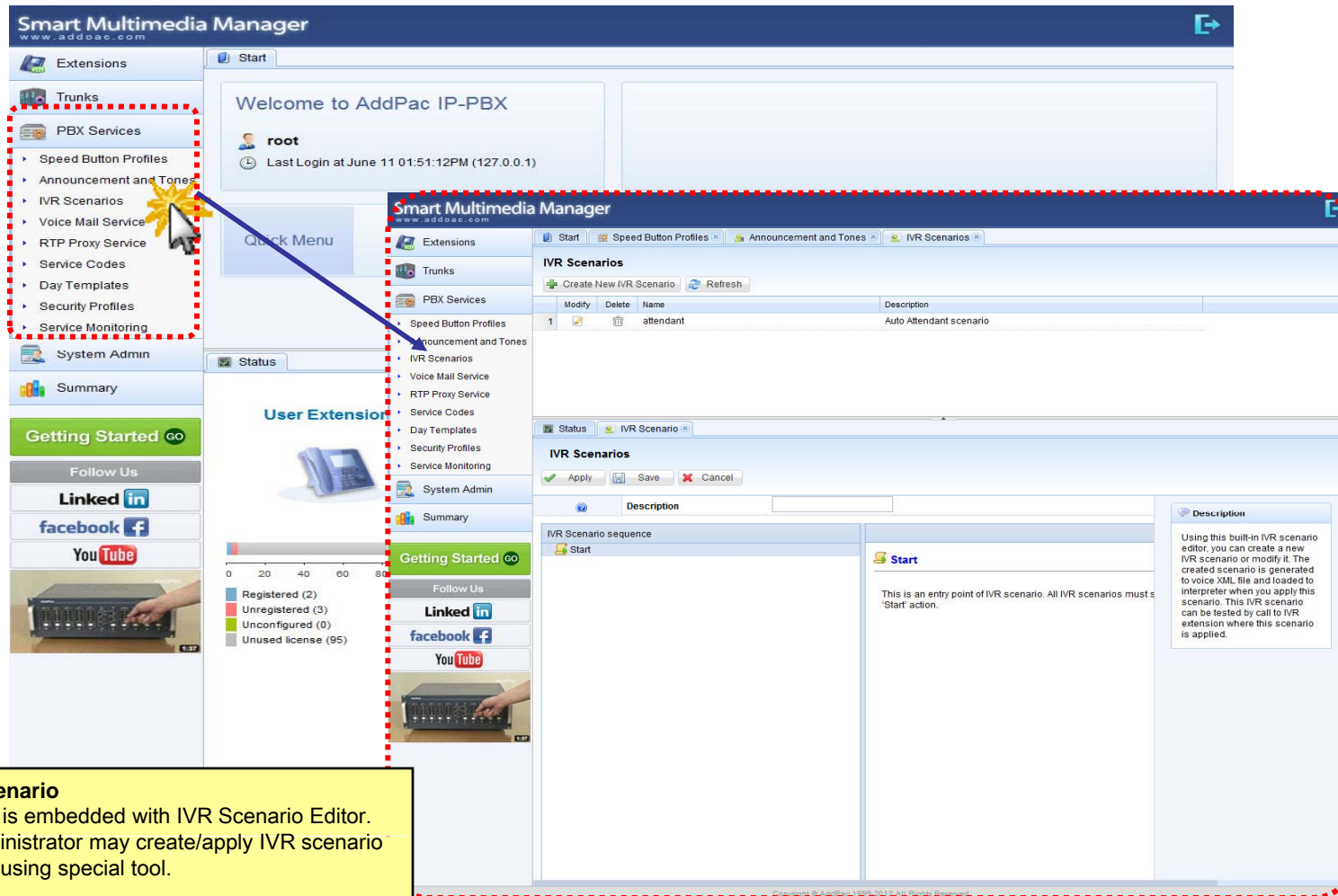
File List	File name	File type	Media type	Version	Upload
<input checked="" type="checkbox"/>	400110_kr.audio.ulaw.wav	package	audio	8.50	2012

Schedule Settings					
No.	Name	Start date	End Date	Start Time	End T
Create New 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 red dashed box highlights this menu item, with a yellow starburst icon and a blue arrow pointing to the 'IVR Scenarios' section of the main interface. The main interface shows a 'Welcome to AddPac IP-PBX' message for user 'root'. Below this is a 'Quick Menu' and a 'User Extension' status section. The 'IVR Scenarios' section is active, showing a table with one scenario named 'attendant' and a description 'Auto Attendant scenario'. Below the table are buttons for 'Apply', 'Save', and 'Cancel', and a 'Description' field. The bottom right of the interface contains a detailed description of the IVR scenario editor.

**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 'PBX Services' highlighted. A red dashed box encloses the 'PBX Services' menu and the 'Voice Mail Service' configuration page. A blue arrow points from the 'Voice Mail Service' menu item to the configuration page. The configuration page includes fields for 'Retrieving Extension by Other Phone', 'Retrieving Extension by Owner Phone', and 'Leave Extension'. It also features 'Advanced Options' for 'Audio Message Length', 'Per Extension Hdd Quota', '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

**Smart Multimedia Manager**  
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Extensions  
Trunks  
PBX Services  
Speed Button Profiles  
Announcement and Tones  
IVR Scenarios  
Voice Mail Service  
RTP Proxy Service  
Service Codes  
Day Templates  
Security Profiles  
Service Monitoring  
System Admin  
Summary  
Getting Started GO  
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Welcome to AddPac IP-PBX  
root  
Last Login at June 11 01:51:12PM (127.0.0.1)

Quick Menu

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Status RTP Proxy Service  
Apply Cancel

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

\*IPv4  
Add Network Domain

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

\*IPv6  
Add Network Domain

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

Network Domain

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

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

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## RTP Proxy Service

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

# PBX Service - Service Codes

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

The screenshot shows the 'Smart Multimedia Manager' interface with the following components:

- Left Navigation Menu:** Includes 'Extensions', 'Trunks', 'PBX Services' (highlighted with a red dashed box), 'System Admin', and 'Summary'. Under 'PBX Services', 'Service Codes' is selected.
- Main Content Area:** Displays the 'Service Codes' configuration page. It includes a 'General Code' section with fields for 'Call Park', 'Call Pickup', 'Call Forwarding All Register', 'Call Forwarding All Activation', and 'Call Forwarding All Deactivation'. Below this is an 'Advanced Options' section with various call management settings like 'Call Reject(Absence) Activation', 'Call Forwarding Busy Register', etc.
- Right Panel:** Contains a 'Description' box explaining that a service code is a special digit (#,\*) starting digits for activating a PBX service.
- Bottom Status Bar:** Shows system statistics: Registered (2), Unregistered (3), Unconfigured (0), and Unused license (95).



# PBX Service - Day Templates

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories: Extensions, Trunks, PBX Services, System Admin, and Summary. The 'PBX Services' category is expanded, showing sub-items: Speed Button Profiles, Announcement and Tones, IVR Scenarios, Voice Mail Service, RTP Proxy Service, Service Codes, Day Templates, Security Profiles, and Service Monitoring. A red dashed box highlights the 'Day Templates' link in the sidebar and the corresponding 'Day Templates' tab in the main content area. A blue arrow points from the sidebar link to the main content area. The main content area shows the 'Day Templates' configuration page, which includes a table of existing templates, a form to add a new template, and a 'General Settings' section with fields for Name and Description. The table contains one entry: ID 1, Name 'holiday', Description, and Date Created '2012-03-30 11:24:41'. The 'General Settings' section has a 'Name' field and a 'Description' field. A tooltip for the 'Description' field reads: 'Specify period or a special day(s) to apply in schedule policy.'

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

# PBX Service - Security Profiles

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories like Extensions, Trunks, PBX Services, and System Admin. The 'PBX Services' category is expanded, showing options such as Speed Button Profiles, Announcement and Tones, IVR Scenarios, Voice Mail Service, RTP Proxy Service, Service Codes, Day Templates, Security Profiles, and Service Monitoring. A mouse cursor is positioned over the 'Security Profiles' option. The main content area shows the 'Security Profiles' configuration page, which includes a table with one entry: 'default' (default security profile) created on 2012-06-08 19:49:52. Below the table is the 'Global Security Setting' section, which includes a dropdown menu for 'TLS Cipher Suites' with the following options: N/A, RC4\_40, RC4\_128, DES\_CBC, 3DES\_CBC, AES\_128\_CBC, AES\_256\_CBC, SEED\_CBC, and ARIA\_CBC. A yellow callout box at the bottom left provides additional information 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.

# 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.

The screenshot shows the 'Smart Multimedia Manager' interface. The left sidebar contains a navigation menu with 'Service Monitoring' highlighted. The main content area displays the 'Service Monitoring' page, which includes a table for 'Active Calls' and 'Conference'. The table has columns for ID, Established Time, Duration, Calling Number, Called Number, Audio Codec, Video Codec, Recording, and Drop Call. Below the table, there is a 'System' status dashboard with various indicators and a 'Trunks' section.

# System Admin - Network Interface

**Network Interface**  
IP-PBX Network interface setup.

- WAN Interface
  - IPv4 / IPv6 Address, DNS, DHCP Client
- LAN Interface
  - IPv4 / IPv6 Address, DHCP Server

The screenshot shows the 'Smart Multimedia Manager' web interface. The left sidebar contains a 'System Admin' menu with 'Network Interface' highlighted. The main content area shows the 'Network Interfaces' configuration page, which is divided into 'WAN Interface' and 'LAN Interface' sections. The WAN interface is configured with DHCP mode, and the LAN interface is configured with 'None' mode. A 'Description' box on the right explains the roles of WAN and LAN interfaces. A yellow box at the bottom left summarizes the configuration steps for each interface type.

# System Admin - Network Services

**Smart Multimedia Manager**  
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System Admin

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

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User Extension

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

**Network Services**

Time zone: % Unknown command (show clock-http)

server enable:  On  Off

System Datetime: [ ]-[ ]-[ ]-[ ]-[ ]-[ ]-[ ]-[ ] [Apply]

Primary NTP Server: [ ]

Secondary NTP Server: [ ]

Interval: [ ] NTP time resynchronize, in hour (default: 27)

TELNET

Service Enable:  On  Off

Service Port: [ 23 ] (default:23)

Service Enable:  On  Off

Service Port: [ ] (default:161)

SNMP

Community: [ ]

Trap Service IP Address: [ ]

Trap Community: [ ]

Service Enable:  On  Off

HTTP

Service Port: [ 80 ] (default:80)

Authentication:  NONE  Basic  Digest

Service Enable:  On  Off

FTP

Control Port: [ 21 ] (default:21)

Data Port: [ 20 ] (default:20)

LDAP

Server Port: [ 389 ] (default:389)

Service Enable:  On  Off

SYSLOG

Service Port: [ ] (default:514)

Log Life Time: [ ] (1 ~ 300 Day)

Description: You can change properties of system network services such as TELNET, SNMP, HTTP, FTP, LDAP, SYSLOG, and so on.

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



# System Admin - Administrators

The screenshot shows the Smart Multimedia Manager interface. The left sidebar contains a 'System Admin' menu item, which is highlighted with a red dashed box. A blue arrow points from this menu item to the 'Administrators' page. The 'Administrators' page displays a table of administrators and a form for creating or editing an administrator.

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

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

# System Admin - Licenses

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

Summary

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User Extension

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

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Status Licenses

Upload License Download License Cancel

Licenses

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 Settings

Description

Manage licenses for Call Manager, MCU, Presence, Media, Voice Mail, IVR, RTP Proxy services.

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

# System Admin - Voice Lines

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

Start

Welcome to AddPac IP-PBX

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

Click Menu

System Admin

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

Summary

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User Extension

Registered (2)

Unregistered (3)

Unconfigured (0)

Unused license (95)

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Smart Multimedia Manager

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Status IVR Scenarios Voice Lines

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

Trunk

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

Extension Analog

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	✓

Description

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

Related Links

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

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



# System Admin - Alarm History

The screenshot shows the 'Smart Multimedia Manager' web interface. The left sidebar contains a 'System Admin' menu with options like 'Network Interfaces', 'Network Services', 'Administrators', 'Licenses', 'Voice Lines', 'Alarm History', 'Call History', and 'Show Command'. The main area displays the 'Alarm History' page, which includes a table of messages with columns for Level, Messages, and DateTime. Below the table is a 'Status' section with various system metrics and a network diagram.

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 Dashboard:**

- User Extensions: Registered (2), Unregistered (3), Unconfigured (0), Unused license (95)
- System: Memory (1%), Storage (7%), Network (OK), Call Manager (0/100), MCU (0/2), Presence (0/100), IVR (0/100), Media (0/100), UMS (0/100), RtpProxy (0/100)
- Trunks: Internal Trunk Gateway (0/0), SKN\_TG (0/0), Dacom\_Trunk (0/0)

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

# System Admin - Call History

**Smart Multimedia Manager**  
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Start | Call History

Welcome to AddPac IP-PBX

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

Click Menu

System Admin

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

Summary

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User Extension

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

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

Smart Multimedia Manager

www.addpac.com

Start | Call History

Call History

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

Summary	Total Call Duration	Total Call Count	ASR	Longest Call Duration	Call Fail Count
472	472	11	81%	90 (sec)	2

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

Page 1 of 1

Status

User Extensions

System

Trunks

Memory 0%  
Storage 8%  
Network

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

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

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

# System Admin - Show Command

The screenshot displays the Smart Multimedia Manager System Admin interface. The left sidebar contains a navigation menu with 'System Admin' selected, and a sub-menu where 'Show Command' is highlighted. A red dashed box highlights the 'Show Command' option in the sub-menu. A blue arrow points from this option to the 'Show Command' window in the main content area. The 'Show Command' window has a 'Status' tab selected, and the 'Show Command' sub-tab is active. The 'Request Command' field contains 'show call-manager sscp'. The output shows system status information for SSCP, CM, and SSCP Policy, along with Client Auth and Client Session information. A 'Description' box on the right explains that users can check system status by selecting a category or entering a CLI command.

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

Click Menu

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

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

Smart Multimedia Manager
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Status Show Command

Show Command
Cancel

Categories
System CPU Memory Storage Network Config
VoIP SIP User Agent Gateway Voice Port Dial-Peer
Call Manager SSCP SP Domain Cluster Terminal Summary Presence
Command Line Interface Show

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 = 10
  event total store count = 0

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

Client Session 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 5041 BIND rdt
  10600000 1 172.16.17.30 5021 BIND tivr
  -----

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



# Thank you!

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