



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

The screenshot displays the AddPac Smart Multimedia Manager (WSMM) web interface. The top navigation bar includes 'Extensions', 'Trunks', 'PBX Services', 'System Admin', and 'Summary'. A 'Getting Started' button is highlighted in green. Below the navigation bar, there are social media links for LinkedIn, Facebook, and YouTube. The main content area features a 'Welcome to AddPac IP-PBX' message, an 'Unread Alarm Message' notification for user 'root', and a 'Quick Menu' with options like 'Add an User Extension', 'Add an Analog Extension', and 'Add a Conference Room'. The 'Status' section shows a system diagram with 'User Extensions', 'System', and 'Trunks'. The 'System' section includes resource usage bars for Memory (0%), Storage (7%), and Network, along with a list of services and their status (e.g., Call Manager 0/100, MCU 0/2). The 'Trunks' section shows 'Internal Trunk Gateway' (0/0) and 'SKN\_TG' (0/0) with green checkmarks, and 'Dacom\_Trunk' (0/0) with a red X. A legend on the left indicates 'Registered (2)', 'Unregistered (3)', 'Unconfigured (0)', and 'Unused license (95)'. The footer contains the copyright notice: 'Copyright © AddPac 1999-2012 All Rights Reserved'.

# AddPac

## AddPac Technology

2012, Sales and Marketing

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- 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' (WSMM) interface. The main window 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, Routing Access List, Terminal Profile, Security Profile, Use RTP Proxy, Generate Ring Back Tone at PBX, and Presentation. A red dashed box highlights a help icon in the 'User Extension' section of the form, with an arrow pointing to a help window. The help window is titled 'Help :: User Extension' and contains sections for 'Analog Extension', 'User Extension', and 'First Name / Last Name'. The 'Analog Extension' section includes a sub-section for 'Analog Port' with a description: '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.' The 'User Extension' section includes a sub-section for 'Extension' with a description: '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.' The 'First Name / Last Name' section includes a sub-section for 'First Name / Last Name' with a description: '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	internal	

**Related Links**

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

**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.

**Advanced Options**

Terminal Profile	default
Security Profile	default
Use RTP Proxy	<input type="checkbox"/>
Back Tone at	<input checked="" type="checkbox"/>
Representation	Default

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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**  
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 **Succeeded**

**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
```

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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.

Getting Started

Follow Us

LinkedIn

facebook

YouTube

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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%  
 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 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, showing a list of system events. A yellow callout box provides instructions on how to access the alarm history from the main page.

**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.

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

# 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. Below this is a 'Quick Menu' section with several options: 'Add an User Extension', 'Add a VoIP Trunk', 'Extensions', 'Add an Analog Extension', 'Add an Outgoing Call Rule', 'Terminals', 'Add a Conference Room', and 'Add an Incoming Call Rule'. A red dashed box highlights the 'Quick Menu' and the 'Add an User Extension' form below it. A blue arrow points from the 'Add an User Extension' link in the Quick Menu to the form. The 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 shows the 'Smart Multimedia Manager' web interface. On the left sidebar, there is a 'Follow Us' section with icons for LinkedIn, Facebook, and YouTube. Blue arrows point from these icons to corresponding social media pages overlaid on the main interface. The LinkedIn page shows a post about AddPac Technology Hybrid IP-PBX System. The Facebook page shows the AddPac profile. The YouTube page shows a video titled 'AddPac Hybrid IP-PBX IPNext180 / IPNext187 / IPNext190 (16/24/32 Port)'. 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** 1  
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 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

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

Start

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

Getting Started

Follow Us  
Linked in  
facebook  
YouTube

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

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

Start Extensions

Extensions

All Extensions Input an Extension Search Add an Extension Refresh

	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	vmail_rec	2012-06-08 17:49:53
8			0002	Voice Mail	vmail	2012-06-08 17:49:54
9			0003	Voice Mail	vmail_noauth	2012-06-08 17:49:54

Add an Extension

Cancel

**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.

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## 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**  
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

Extensions

Directory

Routing Access Lists

Terminal Profiles

Terminals

Trunks

PBX Services

System Admin

Summary

Getting Started

Follow Us

Linked in

facebook

YouTube

Status

User Extension

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

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

Start Directory Extensions

Directory

Add Modify Delete Refresh

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	

Description

In this directory page, you can add / delete / modify departments of your organization. The users can be added at User Extension page. This directory is used for showing user profile and click to call at user portal web page.

Status

User Extensions

Registered (4)  
Unregistered (4)  
Unconfigured (0)  
Unused license (92)

System

Memory Storage 1%  
MCU 8%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)

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

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**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

### 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		Allowed Outgoing Call Rules	
Name		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 such as 'Calling Party Presentation', 'Language', 'Call Duration Limit', and 'Off-net Transfer'.

A yellow callout box at the bottom left of the screenshot 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**  
www.addpac.com

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

**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 for user 'root' and an 'Unread Alarm Message' section. Below this 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 border and contains a table of existing trunks and an 'Add a Trunk' form.

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:...

**Add a Trunk**

- 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.

**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.

**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.

# 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)

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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

The screenshot displays the 'Smart Multimedia Manager' web interface. On the left, a navigation menu is visible with 'PBX Services' expanded to show 'Speed Button Profiles'. A yellow starburst icon highlights this menu item. The main content area shows a 'Speed Button Profile' configuration page. At the top, there's a 'Speed Button Profile' section with buttons for 'Add a Speed Button', 'Global Presence Setting', and 'Refresh'. Below this is a table listing profiles:

Modify	Delete	Name	Description	Date Created
		button profile		2012-04-02 10:43:18

Below the table, there's a 'Speed Button Profiles' section with 'Add' and 'Cancel' buttons. The 'General' tab is active, showing fields for 'Profile Name\*' and 'Description'. Below these is an 'Add a Speed Button' section with a table:

Name	Extension	Type	Type	Modify	Delete
		Extension	N/A		

At the bottom of the configuration form, there are '저장' (Save) and '취소' (Cancel) buttons. A 'Description' box on the right contains text about applying speed buttons to IP phones.

**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

**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.

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		

<b>Announcement Information</b>	Announcement ID	400110																
	Description	연결 중 안내																
	Language	Korea																
<b>File List</b>	<input checked="" type="checkbox"/>	File name	File type	Media type	Version	Upload												
	<input checked="" type="checkbox"/>	400110_kr.audio.ulaw.wav	package	audio	8.50	2012												
<b>Schedule Settings</b>	<input type="button" value="Create New Schedule"/> <table border="1"> <thead> <tr> <th>No.</th> <th>Name</th> <th>Start date</th> <th>End Date</th> <th>Start Time</th> <th>End T</th> </tr> </thead> <tbody> <tr><td> </td><td> </td><td> </td><td> </td><td> </td><td> </td></tr> </tbody> </table>						No.	Name	Start date	End Date	Start Time	End T						
No.	Name	Start date	End Date	Start Time	End T													



# PBX Service - IVR Scenarios

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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  
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root  
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Quick Menu

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IVR Scenarios  
Create New IVR Scenario Refresh

Modify	Delete	Name	Description
1		attendant	Auto Attendant scenario

IVR Scenarios  
Apply Save Cancel

Description

IVR Scenario sequence  
Start

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

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

**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 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 red dashed box highlights this menu, with a yellow starburst icon and a blue arrow pointing to the 'Voice Mail Service' option. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root' and a 'User Extension' status bar. The 'Voice Mail Service' configuration page is open, showing fields for 'Retrieving Extension by Other Phone', 'Retrieving Extension by Owner Phone', and 'Leave Extension'. It also includes 'Advanced Options' for audio message length, per extension HDD quota, and notification settings. A 'Description' box on the right explains that this page manages voice mail service properties.

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

**Smart Multimedia Manager**  
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**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
----------------	---------	---------	------	--------	--------

**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.

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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. The left sidebar contains a navigation menu with 'PBX Services' expanded to show 'Service Codes'. The main content area displays the 'Service Codes' configuration page, which 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 Reject(Do Not Disturb) Activation', 'Call Waiting Activation', etc. A 'Description' box on the right explains that service codes are special digits used to activate PBX services.

# 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' section in the main content area. A blue arrow points from the sidebar link to the main content area. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root' and a 'Day Templates' table with one entry: 'holiday' created on 2012-03-30 11:24:41. Below the table is a form for adding a new day template with fields for Name and Description. A yellow callout box at the bottom left explains the Day Templates function.

**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' menu is expanded, and a red dashed box highlights the 'Security Profiles' option. A blue arrow points from this menu item to the main content area. The main content area shows the 'Security Profiles' configuration page, which includes a table with one entry: 'default' with a description 'default security profile' and a creation date of '2012-06-08 19:49:52'. Below this is the 'Global Security Setting' section, which has a dropdown menu for 'TLS Cipher Suites' showing options like 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 contains the following text:

**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

**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

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

Quick Menu

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

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Service Monitoring  
Interval : 10 sec.

Active Calls | Conference

ID	Established Time	Duration	Calling Number	Called Number	Audio Codec	Video Codec	Recording	Drop Call

System Status Dashboard:

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

System Resources:

- Memory Storage: 0%
- Network: 8%
- 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)
- ss (0/0)
- SM\_SIP\_Provider (0/0)
- JschoL\_gk (0/0)

**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

The screenshot shows the Smart Multimedia Manager System Admin interface. The left sidebar contains a menu with 'System Admin' highlighted. The main content area shows the 'Network Interfaces' configuration page. The page is divided into two sections: 'WAN Interface' and 'LAN Interface'. The 'WAN Interface' section includes fields for IP Address, Subnet Mask, Default Gateway, Primary DNS Server, and Secondary DNS Server. The 'LAN Interface' section includes fields for IP Address, Subnet Mask, DHCP Server (On/Off), DHCP Range, and IPv6 Address. A description box on the right explains the roles of WAN and LAN interfaces. A yellow box at the bottom left summarizes the configuration options for WAN and LAN interfaces.

**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

**Smart Multimedia Manager**  
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System Admin

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

**Network Services**

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)

**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 displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with 'System Admin' highlighted. The main content area shows the 'Administrators' management page, which includes a table of existing administrators and a form for adding a new one.

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

The form for adding a new administrator includes fields for User name, Description, ID, Password, Level (set to Administrator), and Application Permission (with checkboxes for Door Access Control Manager and Time and Attendance Manager).

## 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**  
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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

**Licenses**

Upload License Download License Cancel

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

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

**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  
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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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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	✓

Trunk

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

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

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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**  
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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 Resources: 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

**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)

Smart Multimedia Manager

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

System

User Extensions

System

Trunks

Memory Storage 0%  
Network 8%

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)

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

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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

**Smart Multimedia Manager**  
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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
- Summary

**Show Command**

Categories

- System (CPU, Memory, Storage, Network, Config)
- VoIP (SP User Agent, Gateway, Voice Port, Dial-Peer)
- Call Manager (SSCP, SP, Domain Cluster, Terminal Summary, Presence)
- 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 = 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  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)



# Thank you!

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