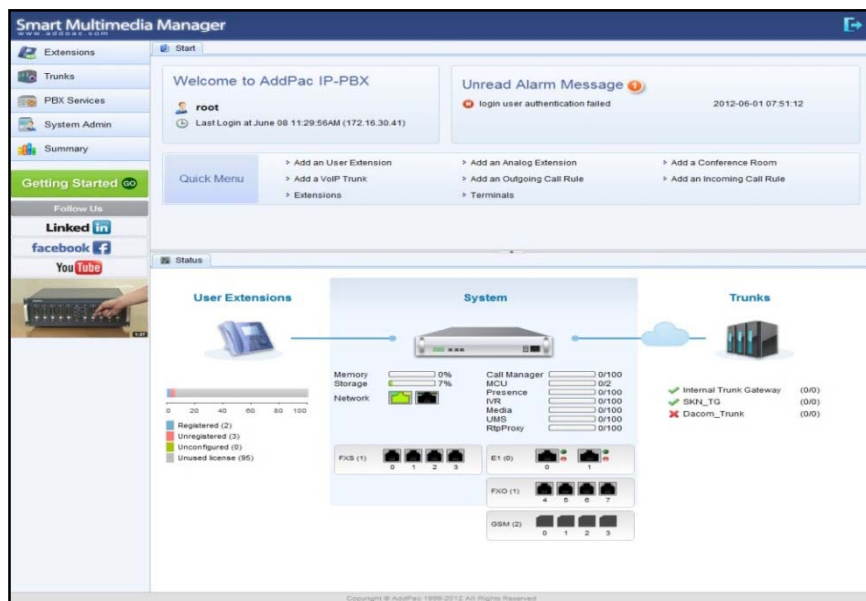




Web Smart Multimedia Manager (WSMM)

MCU Built-in Full HD Video Conference Codec



AddPac

AddPac Technology

2014, Sales and Marketing

Contents

- Overview
- System Requirement
- WSMM Login
- Extension Management
- Trunks Management
- PBX Services Management
- System Admin Management
- Video Conference & H.239 Document Sharing Management (40 Page)



Overview

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

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

System Requirement

WSMM (Web based Smart Multimedia Manager)

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

Login

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

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

Help

Smart Multimedia Manager
www.addpac.com

Add an User Extension

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

Help :: User Extension

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

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


Related Links

Smart Multimedia Manager
www.addpac.com

Modify the User Extension

Apply Cancel Advanced Options

User Extension

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

Routing Access List

Routing Access List: internal

Advanced Options

Terminal Profile: default

General Settings

Security Profile: default

Use RTP Proxy:

Back Tone at:

Representation: Default

Description

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

Related Links

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

Related Links

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

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

Diagnostic

Smart Multimedia Manager
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
ic SIP Gateway
9@172.16.18.100

lication/sdp
179

Step 2.

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Diagnostic

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

Step 1.

- Network Connection Test
- SIP Aware Test

Step 2.

- Outgoing Call Test

Built-in IVR Scenario Editor

Smart Multimedia Manager
www.addpac.com

IVR Scenarios

Apply Save Cancel

IVR Scenario Properties

Name: addpac
Description:

IVR Scenario sequence

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

Menu

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

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

Description

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

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Built-in IVR Scenario Editor

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

IVR Scenario Sequence

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

Main

Smart Multimedia Manager
www.addpac.com

Extensons
Trunks
PBX Services
System Admin
Summary

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

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

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

Status
 User Extensions
 System
 Trunks

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

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

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

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

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

Alarm Message
It displays IP-PBX system errors

Short Cut
A short cut link.

Status
It displays current IP-PBX system major status

Main - Alarm History

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

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

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

Main – Quick Menu

The screenshot displays the 'Smart Multimedia Manager' web interface. On the left is a navigation sidebar with options like 'Extensions', 'Trunks', 'PBX Services', 'System Admin', and 'Summary'. The main content area shows a 'Welcome to AddPac IP-PBX' message for user 'root' and an 'Unread Alarm Message' about 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 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. A mouse cursor is pointing at the Facebook icon. The main content area is divided into several sections: a 'Start' section with a 'Message failed' notification, a 'AddPac Technology' section with a '제품 소개' (Product Introduction) article, a 'facebook' section with the company's profile, and a 'YouTube' section with a video player showing a person presenting a device. A red dashed box highlights the 'Follow Us' section and the social media content. 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

Unread Alarm Message 1
login user authentication failed 2012-06-01 07:51:12

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

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%
7%

Network

Call Manager
MCU Manager
Presence
IVR
Media
UMS
RtpProxy

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

Trunks

Voice Lines

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

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

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

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

Extension - Extensions

The screenshot displays the 'Smart Multimedia Manager' web interface. The left sidebar contains navigation options: Extensions, Trunks, PBX Services, System Admin, and Summary. The main content area is titled 'Smart Multimedia Manager' and shows a 'Welcome to AddPac IP-PBX' message. Below this, there is a 'Quick Menu' and a 'Status' section with a 'User Extension' overview. A table lists various extensions with columns for Modify, Delete, Extension Number, Type, Name, and Date Created. A detailed view of a 'User Extension' is shown below the table, including an 'Add an Extension' form and descriptive text for different extension types.

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

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
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
IVR 0/100
Media 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

The screenshot displays the Smart Multimedia Manager interface. The left sidebar contains a navigation menu with 'Extensions' highlighted. The main content area shows the 'Routing Access Lists' configuration page. A table lists existing lists, and a form below allows adding a new one. A yellow callout box provides a definition for a Routing Access List.

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

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

Extension - Terminal Profile

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

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

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

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

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

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

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

Extension - Terminals

Smart Multimedia Manager

Welcome to AddPac IP-PBX

Unread Alarm Message

No Unread Alarm Message

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

Terminals

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

Modify the User Extension

Apply Cancel Advanced Options

Extension * 3000 3 ~ 8 digits

First Name BongYoung

Last Name * Jeong

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

User Password * 1111 For SIP registration

Department Search

Title ex) manager

Email ex) admin@addpac.com

Home Phone ex) 123-456-7890

Mobile Phone ex) 123-456-7890

User ID SIP registration ID

Photo (Maximum File Size: 100KB)

Description

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

Related Links

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

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

Trunk - Trunks

The screenshot shows the Smart Multimedia Manager interface. The sidebar on the left contains navigation options: Extensions, Trunks, PBX Services, System Admin, and Summary. The main content area is titled 'Trunks' and features a table with the following data:

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' form with a 'Cancel' button. The form includes three options with descriptions:

- 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.

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

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 Second, P-Asserted Identity Second) 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
- jschoI_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.

The screenshot shows the 'Smart Multimedia Manager' interface. The left sidebar contains a menu with 'PBX Services' expanded to show 'Speed Button Profiles'. The main content area displays the 'Speed Button Profile' configuration page, which includes a table of existing profiles and a form for adding a new one. The table shows one profile named 'button profile' created on 2012-04-02. The form includes fields for 'Profile Name*', 'Description', and a table for 'Add a Speed Button' with columns for Name, Extension, Type, and a dropdown menu. A 'Description' sidebar on the right provides additional context about the feature.

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		

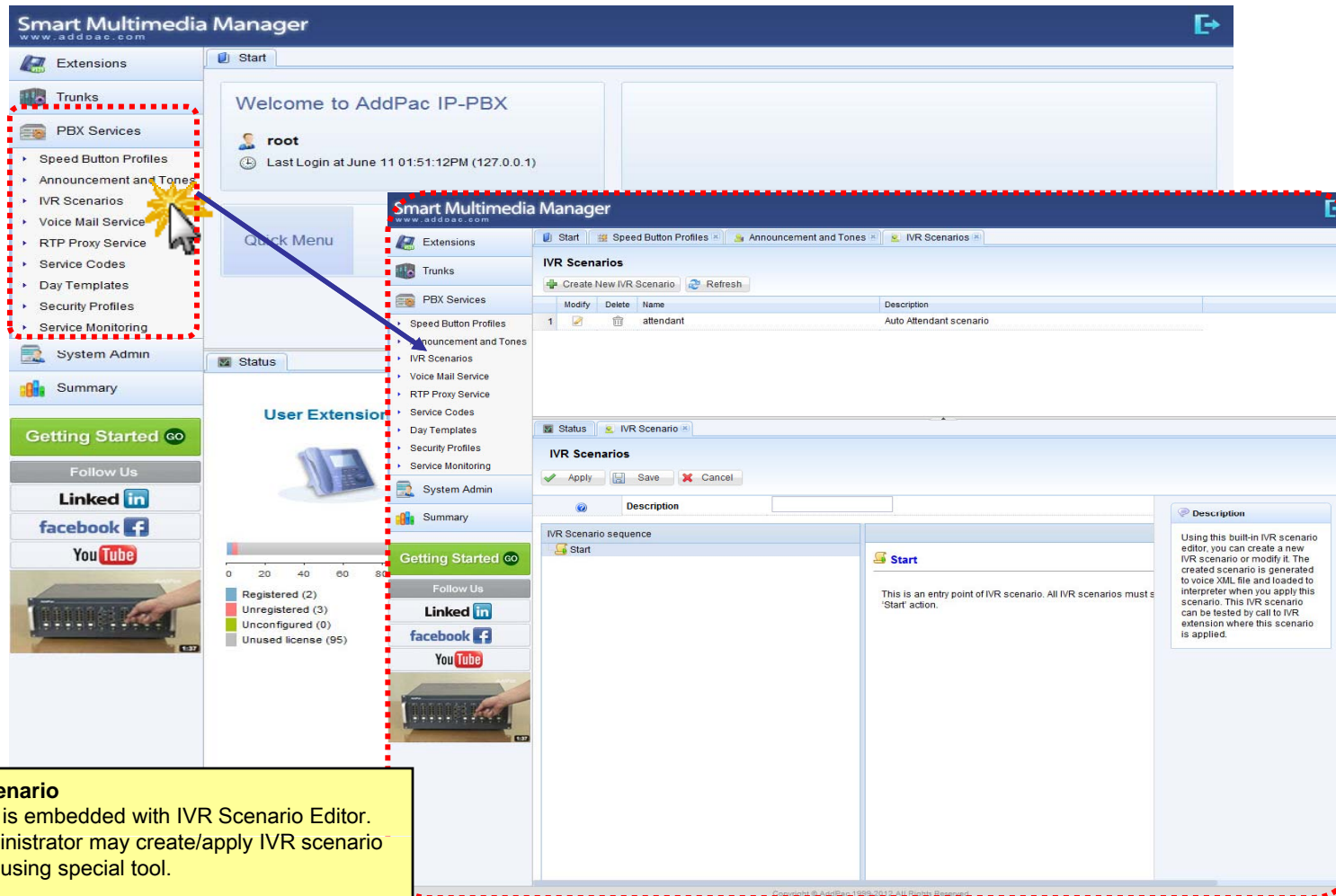
Announcement Information	
Announcement ID	400110
Description	연결 중 안내
Language	Korea

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 yellow starburst highlights the 'IVR Scenarios' menu item. The main content area shows the 'IVR Scenarios' configuration page, which includes a table of existing scenarios, a 'Create New IVR Scenario' button, and a detailed editor for the 'attendant' scenario. The editor shows a 'Start' action in the IVR Scenario sequence.

Modify	Delete	Name	Description
		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 '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. On the left, a navigation menu is visible 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

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

PBX Service - Service Codes

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

The screenshot shows the 'Smart Multimedia Manager' interface. The left sidebar contains a 'PBX Services' menu with a red dashed box around it. A blue arrow points from this menu to the 'Service Codes' configuration page. The configuration page 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 the service code is a special digit starting with # or *.

PBX Service - Day Templates

The screenshot displays the Smart Multimedia Manager web interface. The left sidebar contains a navigation menu with categories: 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', a 'Quick Menu', and a 'User Extension' status bar. The 'Day Templates' section includes a 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.

Modify	Delete	Template Name	Description	Date Created
		holiday		2012-03-30 11:24:41

Day Templates
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' menu is expanded, and 'Security Profiles' is highlighted. 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 explains that IP-PBX supports TLS Cipher Suites and lists the available options.

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

The screenshot displays the Smart Multimedia Manager interface. The left sidebar contains a navigation menu with 'Service Monitoring' highlighted. The main content area shows the 'Service Monitoring' page with a table for 'Active Calls' and 'Conference'. Below the table is a 'System' status section with various indicators and a 'Trunks' section with a table of trunk status.

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

System Health	Value
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

Trunk	Status
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

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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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Network Interfaces

Apply Cancel

Interface Mode: DHCP Static IP

IP Address * A.B.C.D

Subnet Mask * A.B.C.D

Default Gateway A.B.C.D

Primary DNS Server

Secondary DNS Server

IPv6 Address XX:XXM

IPv6 Default Gateway XX:XX

WAN Interface

Interface Mode: None Bridge IP Shared NAT Static IP

IP Address A.B.C.D

Subnet Mask A.B.C.D

DHCP Server On Off

DHCP Range ~ A.B.C.D

LAN Interface

DHCP Server On Off

DHCP Range ~ A.B.C.D

IPv6 Address XX:XXM

Description

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

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

System Admin - Network Services

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System Admin

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

Getting Started GO

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User Extension

Registered (2)

Unregistered (3)

Unconfigured (0)

Unused license (95)

Network Services

Apply Cancel

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
	Service Port	(default:161)
SNMP	Community	
	Trap Service IP Address	
	Trap Community	
	Service Enable	<input checked="" type="radio"/> On <input type="radio"/> Off
HTTP	Service Port	80 (default:80)
	Authentication	<input type="radio"/> NONE <input checked="" type="radio"/> Basic <input type="radio"/> Digest
	Service Enable	<input checked="" type="radio"/> On <input type="radio"/> Off
FTP	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)

Description

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

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 with sub-items: Network Interfaces, Network Services, Administrators, Licenses, Voice Lines, Alarm History, Call History, and Show Command. The main content area displays the 'Administrators' page, which includes a table of existing administrators and a form for creating or editing one.

Modify	Delete	Name	ID	Level	Description
		root	root	Administrator	System Administrator
		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.

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

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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

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

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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: [Progress Bar]
- 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) [Green Checkmark]
- SKN_TG (0/0) [Green Checkmark]
- Dacom_Trunk (0/0) [Red X]

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

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

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						

System Status Dashboard:

- Memory: 0%
- Storage: 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
- Internal Trunk Gateway: (0/0) ✓
- ss: (0/0) ✓
- SM_SIP_Provider: (0/0) ✗
- Jschoi_gk: (0/0) ✗

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 and the corresponding 'Show Command' button in the main content area. A blue arrow points from the 'Show Command' option in the sub-menu to the 'Show Command' button. The main content area shows the 'Show Command' dialog box with a 'Request Command' field containing 'show call-manager sscp'. Below this, the system output is displayed, including SSCP Timer Information, CM <-> CM Servers SSCP Information, SSCP Policy Information, Client Auth Session Information, Client Session Information, and Servers Information. A yellow callout box at the bottom left contains the text: 'Show Command User may check the status of IP-PBX System through category and CLI (Command Line Interface)'. The interface also shows a 'User Extension' section with a bar chart and a 'Getting Started' section with social media links.

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

```
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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

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

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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

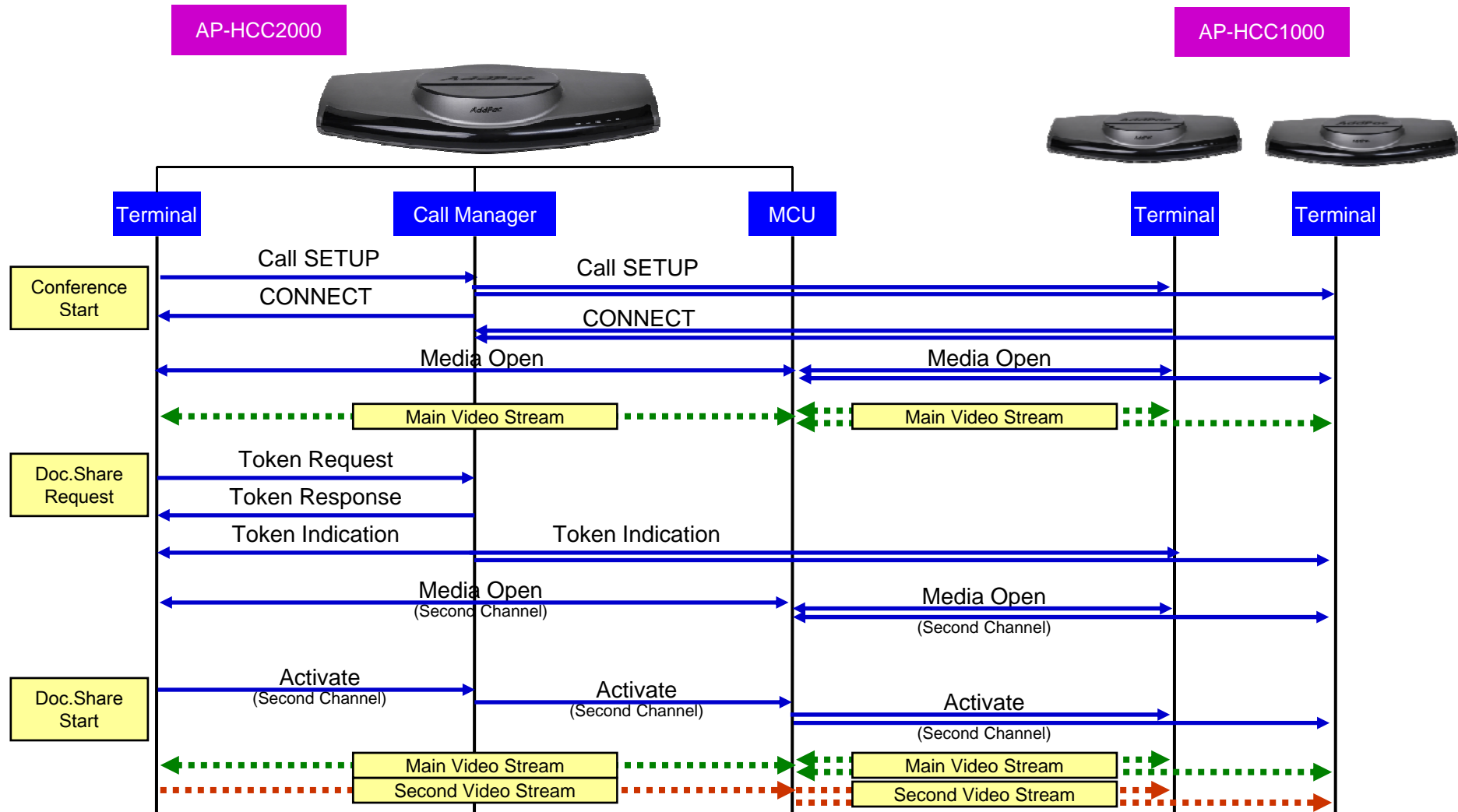
Client Auth Session Information.
  id ip timerCount
  -----
  -----

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 5045 BIND rdt
  10600000 1 172.16.17.30 5021 BIND tivr
  -----

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

Conference & Doc. Share Call Flow



Conference - Create Room 1

The screenshot displays the Smart Multimedia Manager (SMM) web interface. The left sidebar contains navigation menus for Extensions, Trunks, PBX Services, System Admin, Servers, Advanced, Monitoring, and Summary. The main content area shows the 'Add an Extension' page, which includes a table of existing extensions and a list of extension types with descriptions. The 'Conference Room' option is highlighted with a red dashed box.

Smart Multimedia Manager
www.addpac.com

Extensions

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

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

Add an Extension

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

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Conference - Create Room 2

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Extensions

Start | Extensions

Extensions

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

Status | Add an Extension | Conference Room

Add a Conference Room

Add | Cancel | Advanced Options

Set Conference Room Number & Name

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

Conference Name * ConfRoom1

Media Type Audio Audio + Video

Default Media Class

Audio Codec G.711U

Video Codec H.264

Picture Size 1080p (1920*1080)

Bandwidth 1024K

Frame Rate 30 fps

Participants

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

External Phone: 6004

Add a Schedule

Add Conference participants

Description

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

Related Links

- User Extension
- Partitions

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Conference - Add Gatekeeper 1

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

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Start | Extensions | Trunks

Trunks

All Trunks | Add a Trunk | Refresh

Modify	Delete	Diagnose	Name	Type	IP Address	State	Description	Date Created
--------	--------	----------	------	------	------------	-------	-------------	--------------

Status | Add an Extension | Conference Room | Add a Trunk

Add a Trunk

Cancel

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

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

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

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

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

Getting Started GO
Clustering Guide GO
Partitioning Guide GO
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Conference - Add Gatekeeper 2

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

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

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

Conference - Add Gatekeeper 3

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Extensions
Trunks
Trunks
Outgoing Call Rules
Incoming Call Rules
PBX Services
System Admin
Servers
Advanced
Monitoring
Summary

Getting Started GO
Clustering Guide GO
Partitioning Guide GO

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Register Extensions

Start Extensions Trunks

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

Status Add an Extension Add a Trunk H.323 Gatekeeper

Add a H.323 Gatekeeper
Add Cancel Advanced Options

Purpose of Trunking Inter-Site Call
Do Not Generate CDR
Add
Modify Delete Prefix Number

Tech Prefix

Register Conference Room to GK

All Available Extensions
All Extensions 6000 Search
Extension Name Type

Register Extensions
Extension Name Type
6000 ConfRoom1 conference

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

Related Links
Outgoing Call Rules
Security Profiles
Device Pool
Location

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Conference – Outgoing Call Rules

The screenshot displays the 'Smart Multimedia Manager' web interface. The top navigation bar includes 'Start', 'Extensions', 'Trunks', 'Outgoing Call Rules', and 'Incoming Call Rules'. The left sidebar contains various system management options. The main area is titled 'Outgoing Call Rules' and features a table with columns for 'Modify', 'Delete', 'Name', 'Pattern', 'Trunk', and 'Date Created'. Below this is the 'Add an Outgoing Call Rule' form, which includes a 'Name' field (containing 'Out_Rule'), a 'Patterns' field (containing '600.'), and a 'Trunks of Outgoing Call' table. The table has one entry with '1' in the first column and 'Gatekeeper' in the second. A red dashed box highlights the 'Add Trunk' button and the 'Gatekeeper' selection. A yellow box highlights the text 'Set Outgoing Call Rule to GK'. The form also includes sections for 'Called Number Translation' and 'Calling Number Translation'. A 'Description' panel on the right explains the function of an Outgoing Call Rule. The footer contains the AddPac logo, website URL, and version information.

Conference – Incoming Call Rules

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

Incoming Call Rules

Add an Incoming Call Rule

✓ Add ✗ Cancel

Name * In_Rule

Trunks of Incoming Call * Gatekeeper

Set Incoming Call Rule From GK

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

Related Links
• Trunks
• Outgoing Call Rules

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Thank you!

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