

GSM SOHO Gateway Solution



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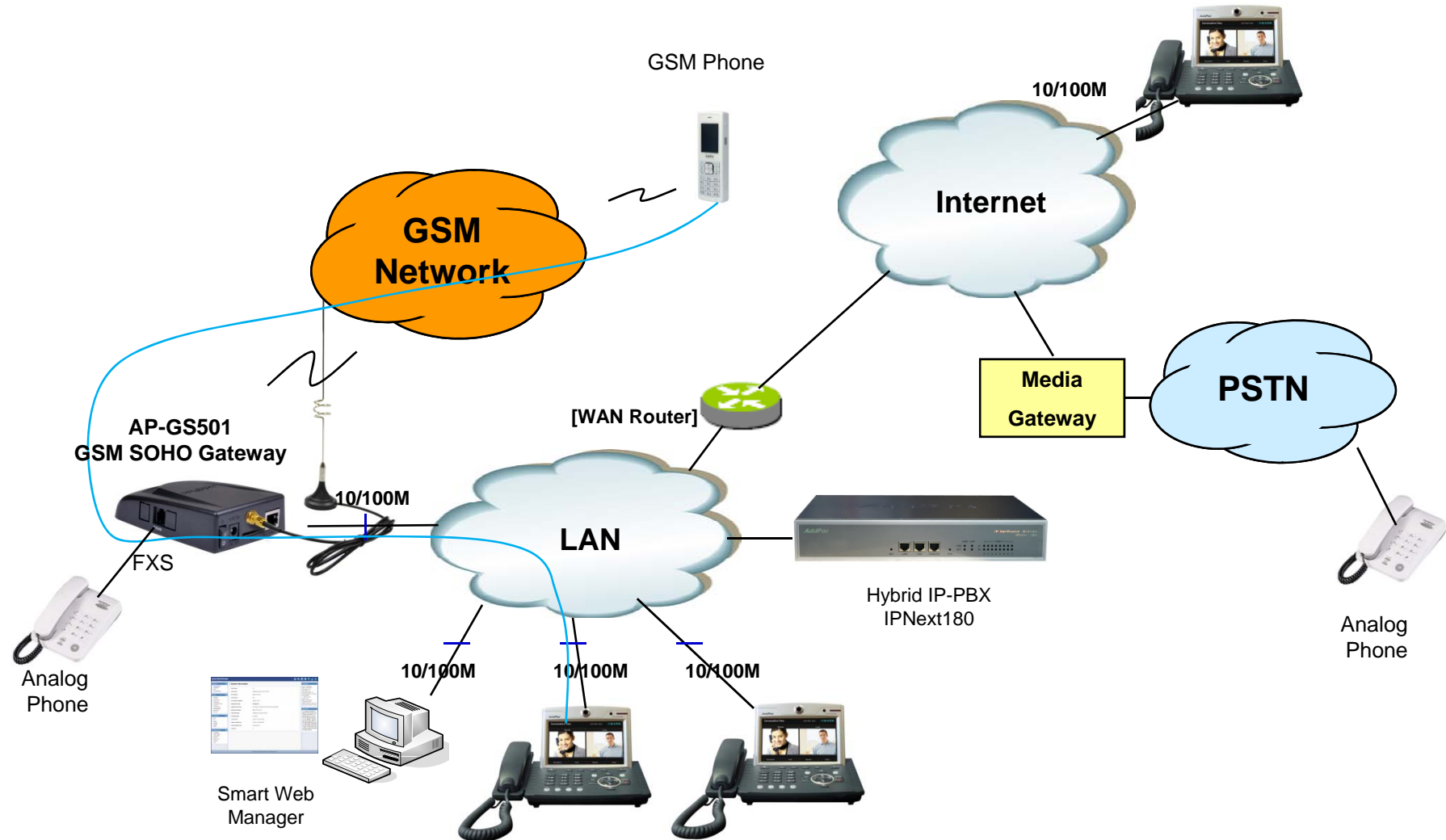
AddPac Technology

2012, Sales and Marketing

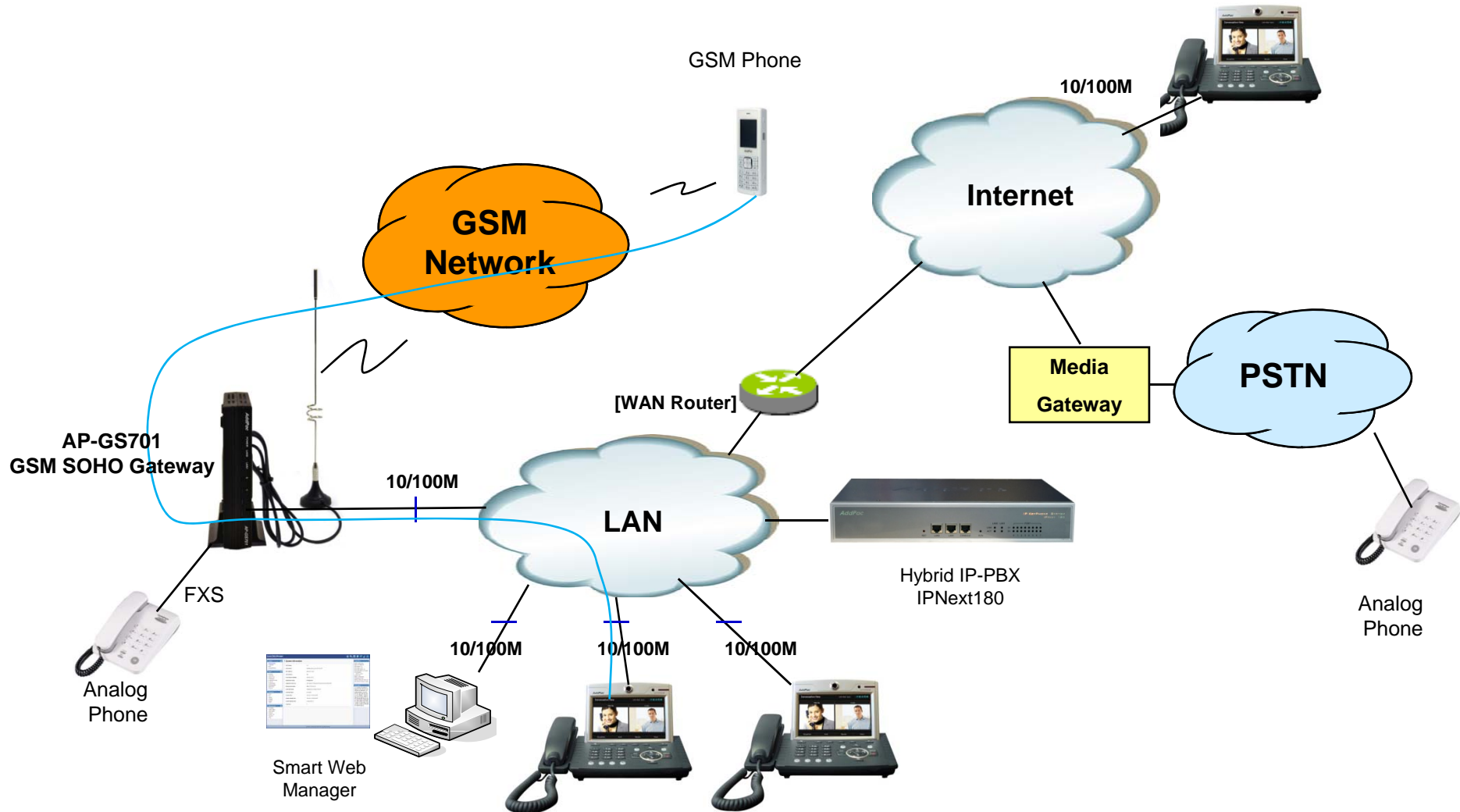
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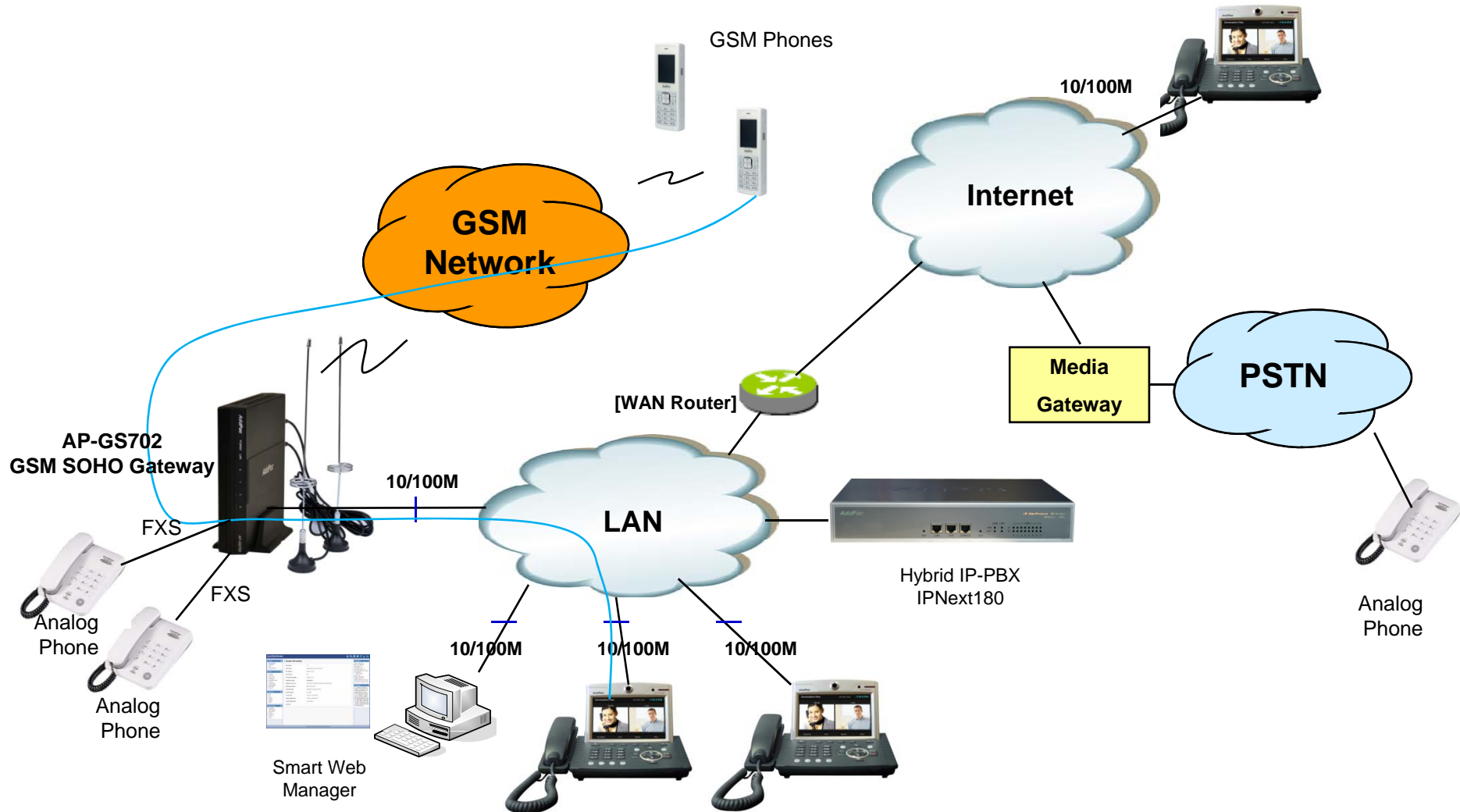
GSM Gateway Service Diagram (AP-GS501)



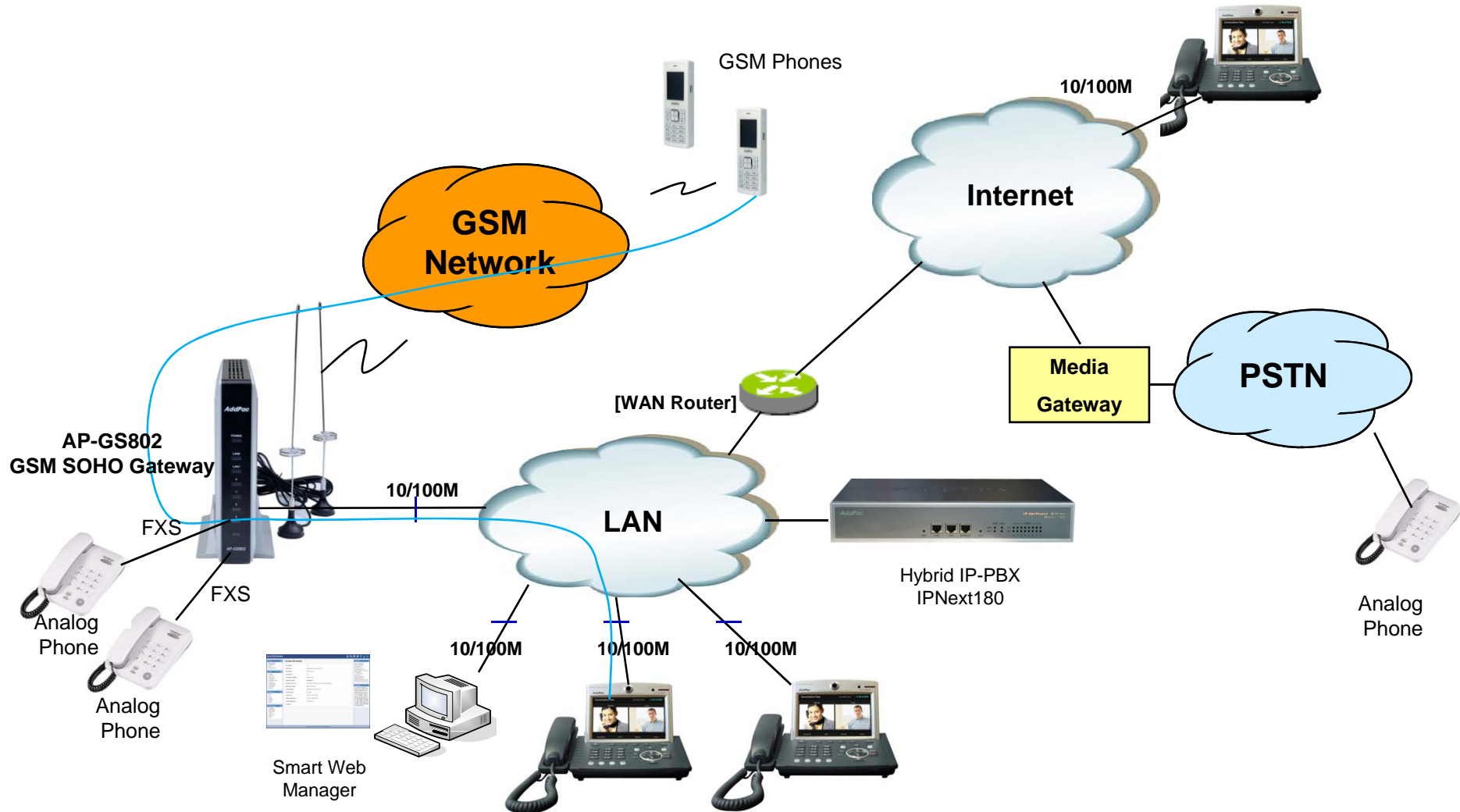
GSM Gateway Service Diagram (AP-GS701)



GSM Gateway Service Diagram (AP-GS702)

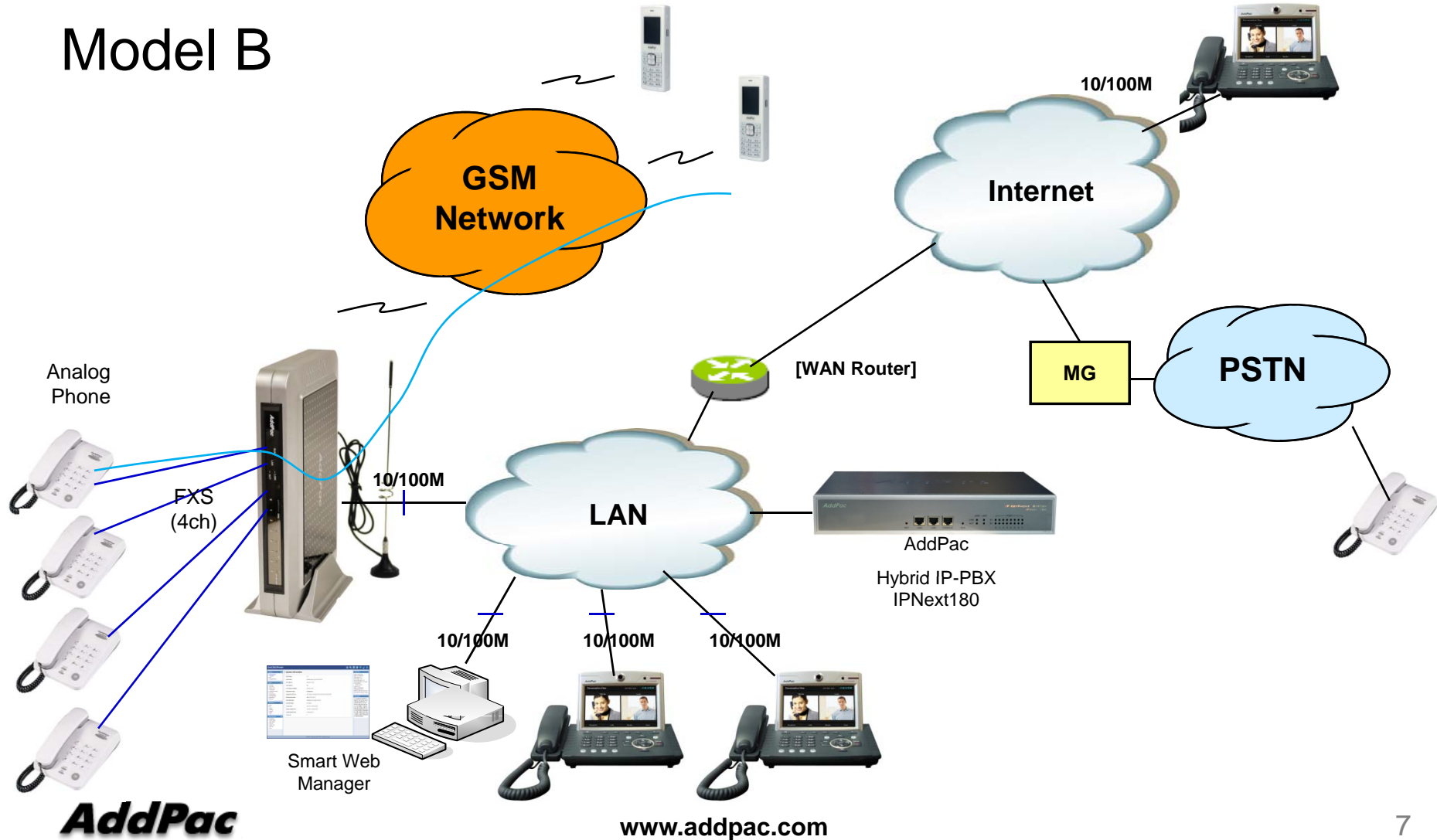


GSM Gateway Service Diagram (AP-GS802)

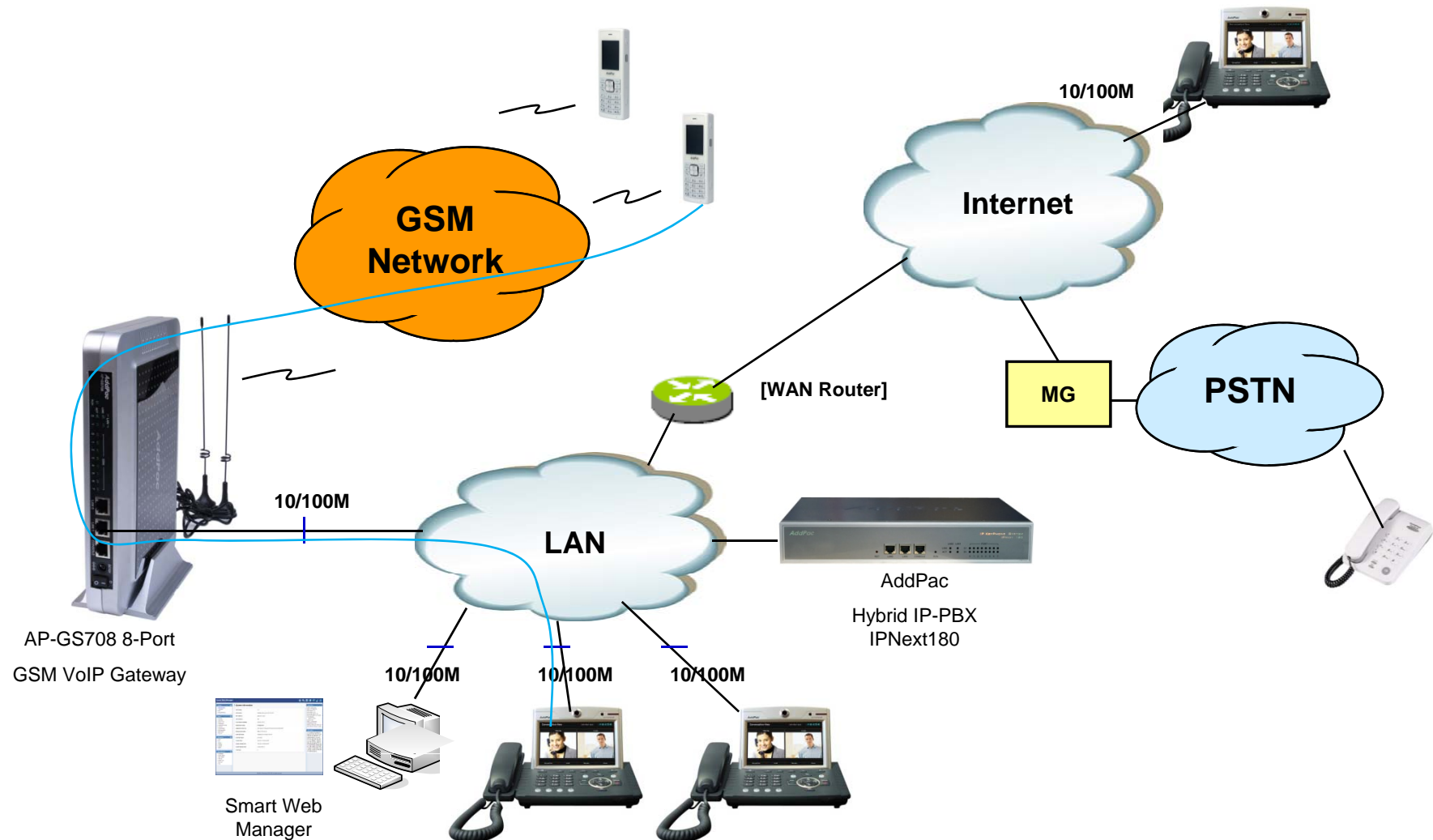


GSM Gateway Service Diagram (AP-GS804B)

Model B



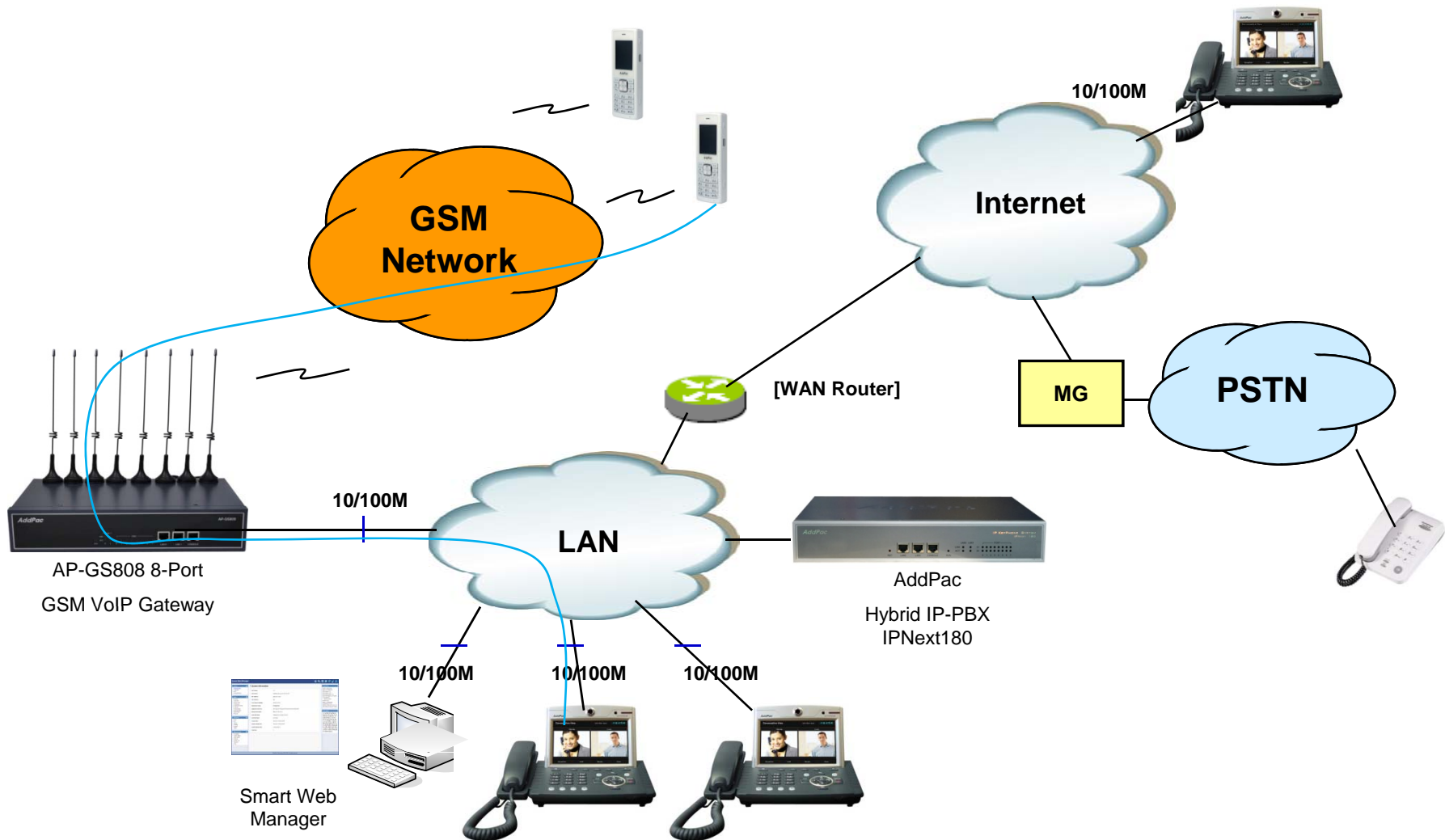
GSM Gateway Service Diagram (AP-GS708)



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GSM Gateway Service Diagram (AP-GS808)










GSM SOHO Gateway Series

GSM SOHO Gateway Comparison Table

	AP-GS501		AP-GS701		AP-GS702		AP-GS802	
								
Model	Type	Analog	Type	Analog	Type	Analog	Type	Analog
	B	1FXS	A	None	A	None	A	None
			B	1FXS	B	2FXS	B	2FXS
GSM Channel	1		1		2		2	
GSM Antenna	1		1		2		2	
SIM Card Slot	1		1		2		2	
LAN Port 10/100Mbps	1		2		1		2	
Console(RJ45)	N/A		N/A		N/A		1	

GSM SOHO Gateway Comparison Table

	AP-GS804		AP-GS708		AP-GS808		AP-GS816	
								
Model	Type	Analog	Type	Analog	Type	Analog	Type	Analog
	A	None		None		None		None
	B	4-Port FXS						
	C	4-Port FXO						
GSM Channel	4		8		8		16	
GSM Antenna	4		8		8		16	
SIM Card Slot	4		8		8		16	
LAN Port 10/100Mbps	1		2		2		2	
Console(RJ45)	1		1		1		1	
Power	External Power Adaptor		External Power Adaptor		Internal Power		Internal Power	



AP-GS1001 GSM SOHO Gateway

Main Features

AP-GS501 One(1) Port GSM SOHO Gateway

- One(1) Port GSM SOHO Gateway Service
- Analog Interface (FXS)/VoIP Interface(LAN) Both Support
- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- One(1) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Performance LAN-to-LAN Routing Capability
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with External Power Supply

Hardware Specification

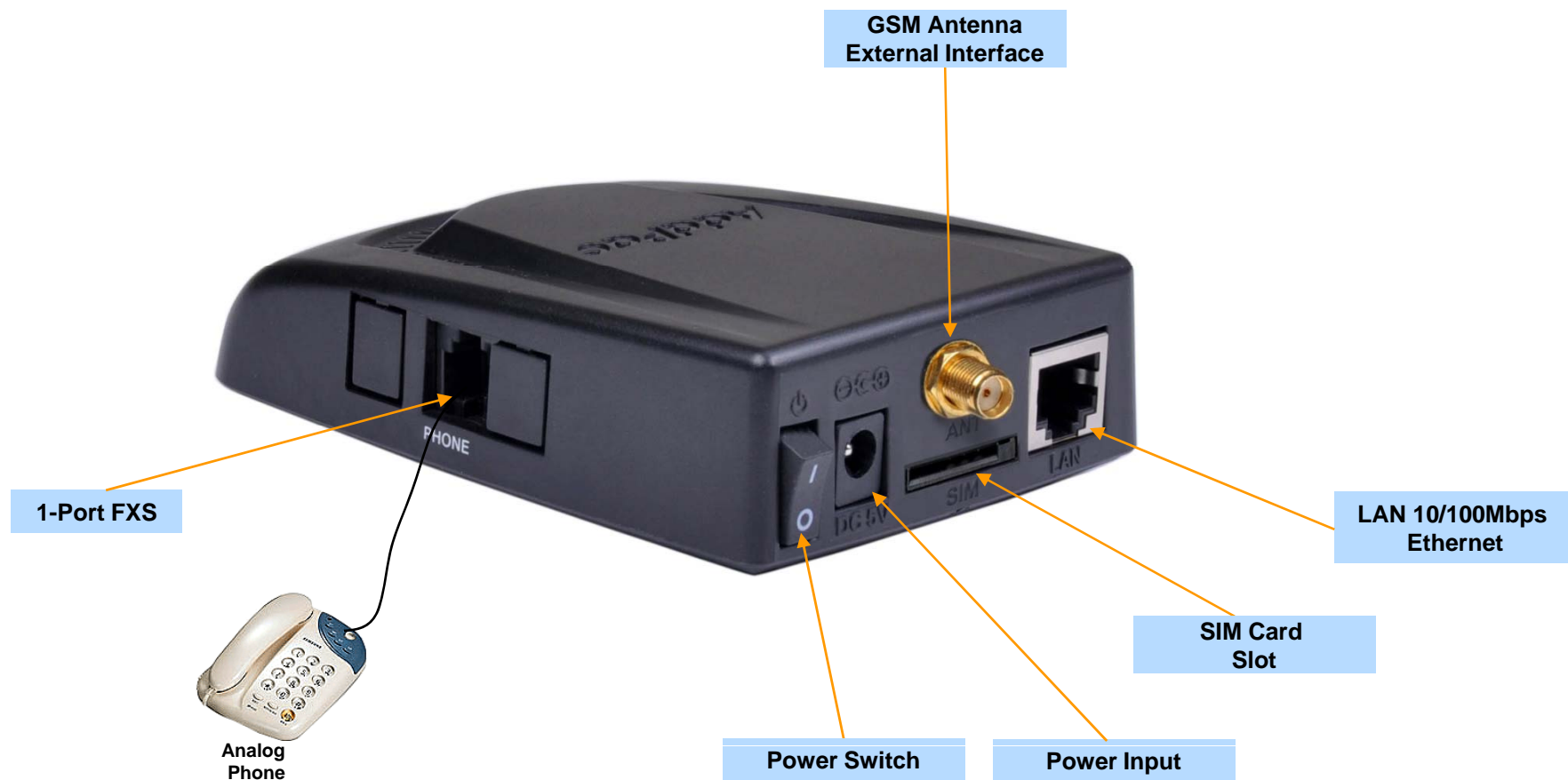
AP-GS501 One(1) Port GSM SOHO Gateway

- RISC Microprocessor Computing Power
- 1-Port GSM Gateway
- 1-Port SIM Card Slot
- 1-Port GSM Antenna Interface
- VoIP Gateway Interface : One(1) FXS Port
- Network Interface for VoIP Direct Interface
 - One(1) 10/100Mbps Fast Ethernet (RJ45)
- Run LED, LAN LED, Port LEDs
- External Power Supply



Hardware Specification

AP-GS501 One(1) Port GSM SOHO Gateway





AP-GS701 GSM SOHO Gateway

Main Features

AP-GS701 One(1) Port GSM SOHO Gateway

- One(1) Port GSM SOHO Gateway Service
- Analog Interface (FXS)/VoIP Interface(LAN) Both Support
- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Performance LAN-to-LAN Routing Capability
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with External Power Supply

Hardware Specification

AP-GS701 One(1) Port GSM SOHO Gateway

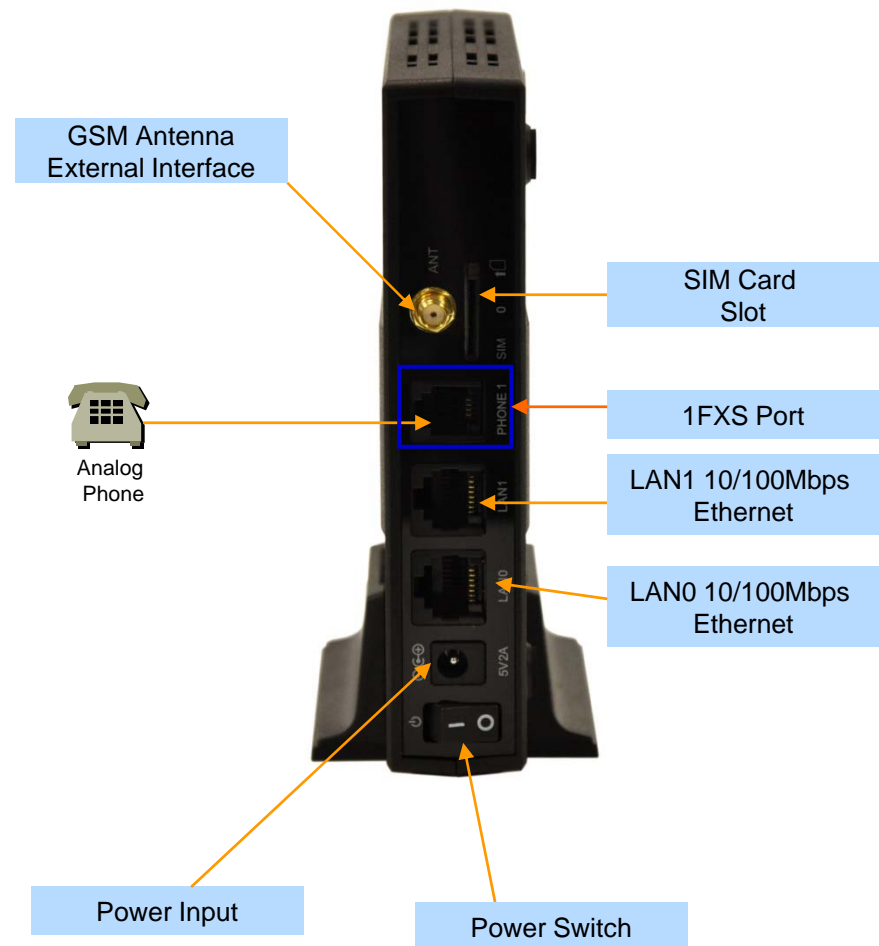
- RISC Microprocessor Computing Power
- 1-Port GSM Gateway
- 1-Port SIM Card Slot
- 1-Port GSM Antenna Interface
- VoIP Gateway Interface
 - AP-GS701 Model A: Basic Configuration
 - AP-GS701 Model B: One(1) FXS Port
- Network Interface for VoIP Direct Interface
 - Two(2) 10/100Mbps Fast Ethernet (RJ45)
- Run LED, LAN LED, Port LEDs
- External Power Supply

Hardware Specification

AP-GS701 One(1) Port GSM SOHO Gateway



Network interface Configurations





AP-GS702 GSM SOHO Gateway

Main Features

AP-GS702 Two(2) Port GSM SOHO Gateway

- Two(2) Port GSM SOHO Gateway
- Analog Interface (FXS)/VoIP Interface(LAN) Both Support
- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- One(1) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Performance LAN-to-LAN Routing Capability
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with External Power Supply

Hardware Specification

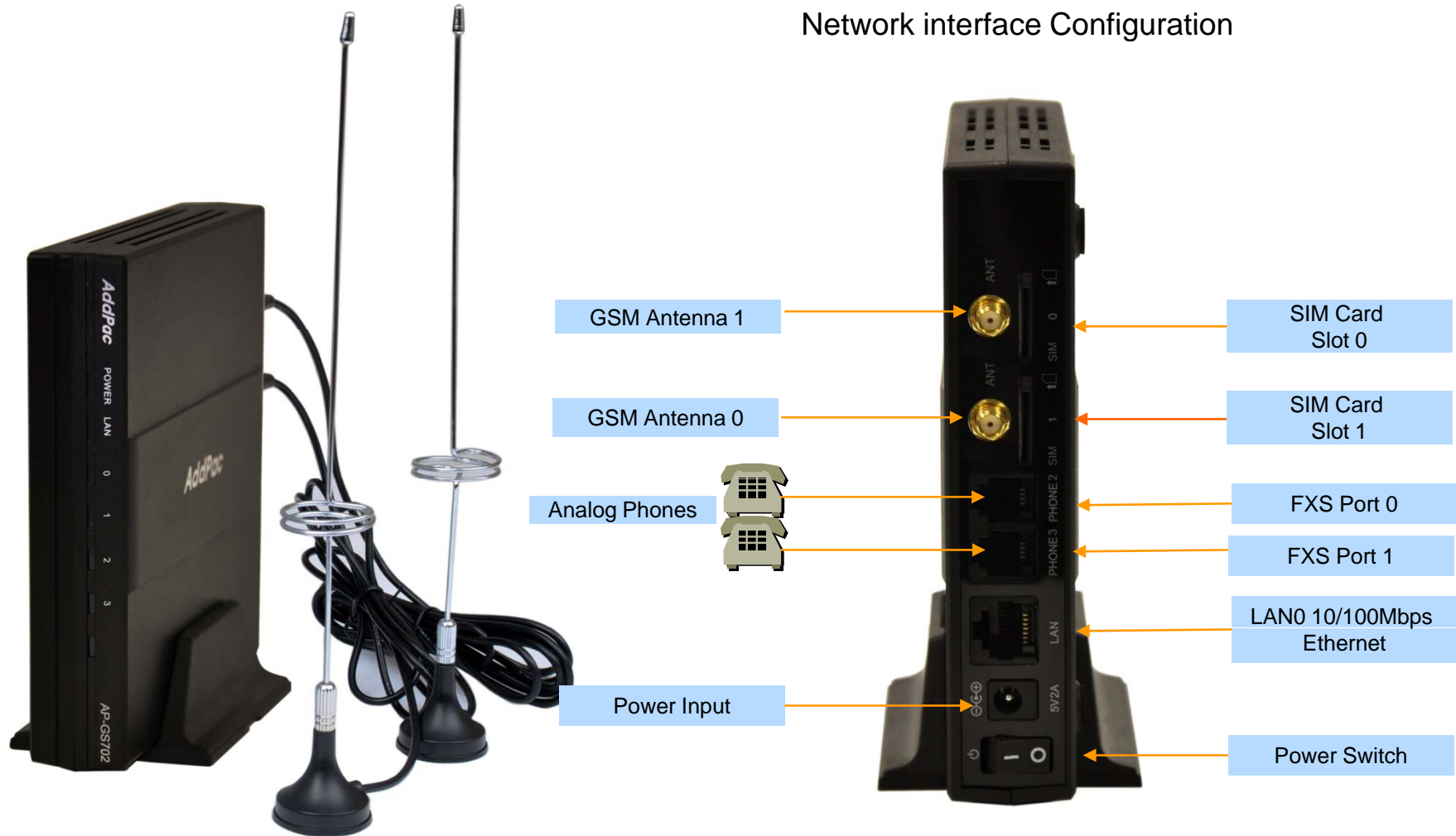
AP-GS702 Two(2) Port GSM SOHO Gateway

- RISC Microprocessor Computing Power
- 2-Port GSM Gateway
- 2-Port SIM Card Slot
- 2-Port GSM Antenna Interface
- VoIP Gateway Interface
 - AP-GS702 Model A: Basic Configuration
 - AP-GS702 Model B: Two(2) FXS Port
- Network Interface for VoIP Direct Interface
 - One(1) 10/100Mbps Fast Ethernet (RJ45)
- Run LED, LAN LED, Port LEDs
- External Power Supply

Hardware Specification

AP-GS702 Two(2) Port GSM SOHO Gateway

Network interface Configuration





AP-GS802 GSM SOHO Gateway

Main Features

AP-GS802 Two(2) Port GSM SOHO Gateway

- Two(2) Port GSM SOHO Gateway
- Analog Interface (FXS)/VoIP Interface(LAN) Both Support
- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- High Performance LAN-to-LAN Routing Capability
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with External Power Supply

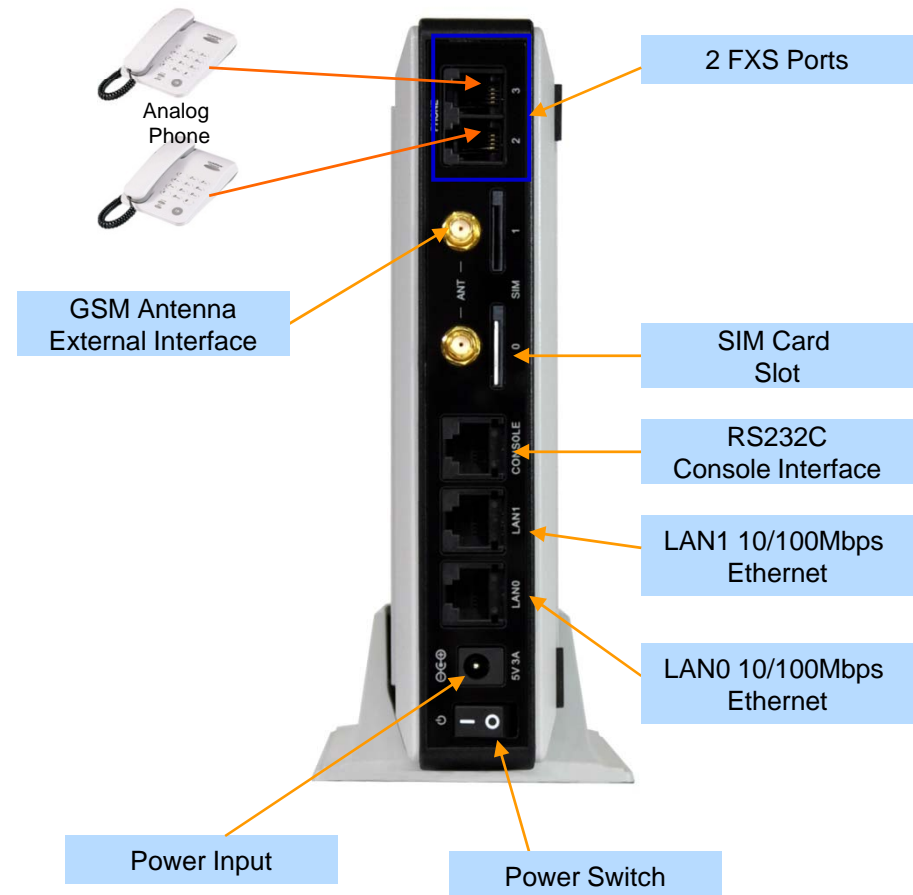
Hardware Specification


AP-GS802 Two(2) Port GSM SOHO Gateway

- RISC Microprocessor+DSP Computing Power
- 2-Port GSM Gateway
- 2-Port SIM Card Slot
- 2-Port GSM Antenna Interface
- VoIP Gateway Interface
 - AP-GS1002 Model A: Basic Configuration
 - AP-GS1002 Model B: Two(2) FXS Port
- Network Interface
 - Two(2) 10/100Mbps Fast Ethernet (RJ45)
- RS232C Console Port for CLI (RJ45)
- Run LED, LAN LED, Port LEDs
- External Power Supply

Hardware Specification

AP-GS802 Two(2) Port GSM SOHO Gateway





AP-GS804 GSM SOHO Gateway

Main Features

AP-GS804 4-Port GSM SOHO Gateway

- Analog Interface (FXS, FXO)/VoIP Interface(LAN) Both Support
- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- One(1) 10/100Mbps Fast Ethernet (IP Share ,etc)
- One(1) RS-232C Port for Command Line Interface
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with External Power Supply

Hardware Specification

AP-GS804 4-Port GSM SOHO Gateway

- RISC Microprocessor Computing Power
- 4-Port GSM Gateway
- 4-Port SIM Card Slot
- 4-Port GSM Antenna Interface
- VoIP Interface
 - None : Model A
 - 4-Port FXS Interface : Model B
 - 4-Port FXO Interface : Model C
- Network Interface for VoIP Direct Interface
 - One(1) 10/100Mbps Fast Ethernet (RJ45)
- RS232C Console Interface for CLI
- Run LED, LAN LED, Port LEDs
- External Power Supply

Hardware Specification

AP-GS804 4-Port GSM Gateway

RISC
CPU

High-end
DSP



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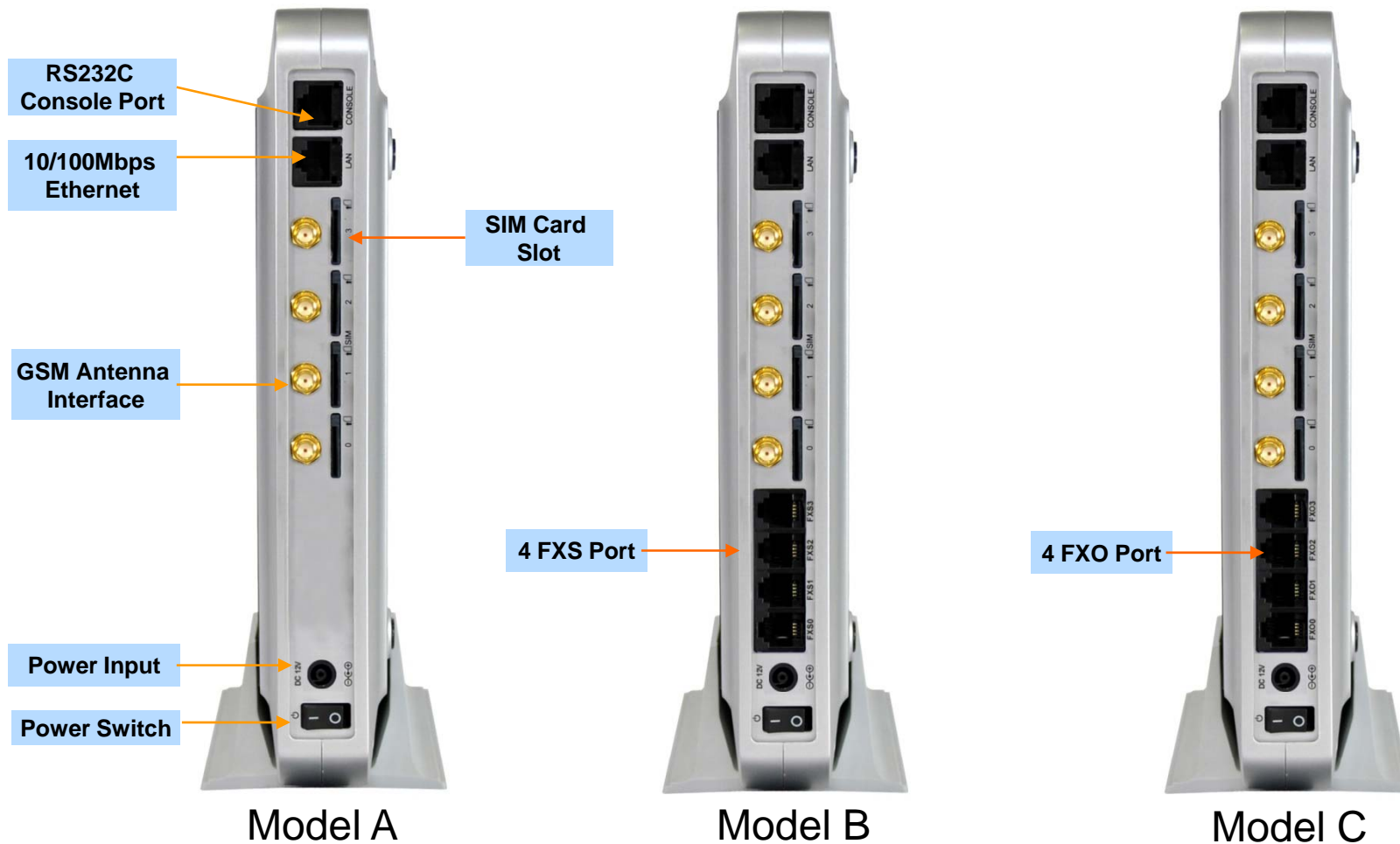
Hardware Specification

AP-GS804 4-Port GSM Gateway

RISC
CPU

High-end
DSP

Back Side View



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AP-GS708 GSM SOHO Gateway

Main Features

AP-GS708 8-Port GSM VoIP Gateway

- 8-Port GSM VoIP Gateway Solution
- 8-Port SIM Slots, 8-Port Antenna Interface
- GSM VoIP Interface(LAN) Support
- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- One(1) RS-232C Port for Command Line Interface
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with External Power Supply

Hardware Specification

AP-GS708 8-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

- RISC Microprocessor Computing Power
- Powerful High-End DSP for VoIP Interface
- 8-Port GSM VoIP Gateway
- 8-Port SIM Card Slots
- 8-Port GSM Antenna Interface
- Network Interface for VoIP Direct Interface
 - Two(2) 10/100Mbps Fast Ethernet (RJ45)
- RS232C Console Interface for CLI
- Run LED, LAN LED, Port LEDs at Front Side
- Compact and Light Design with External Power Supply

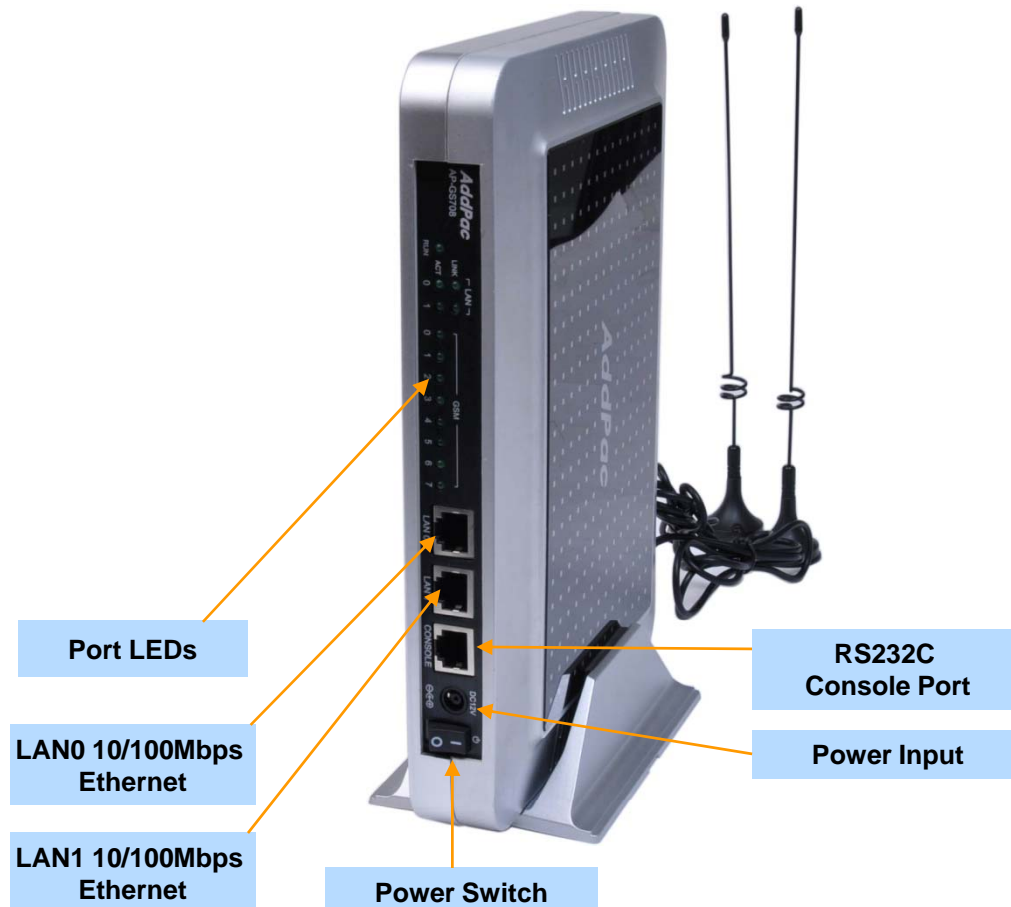
Hardware Specification

AP-GS708 8-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

Front Side View



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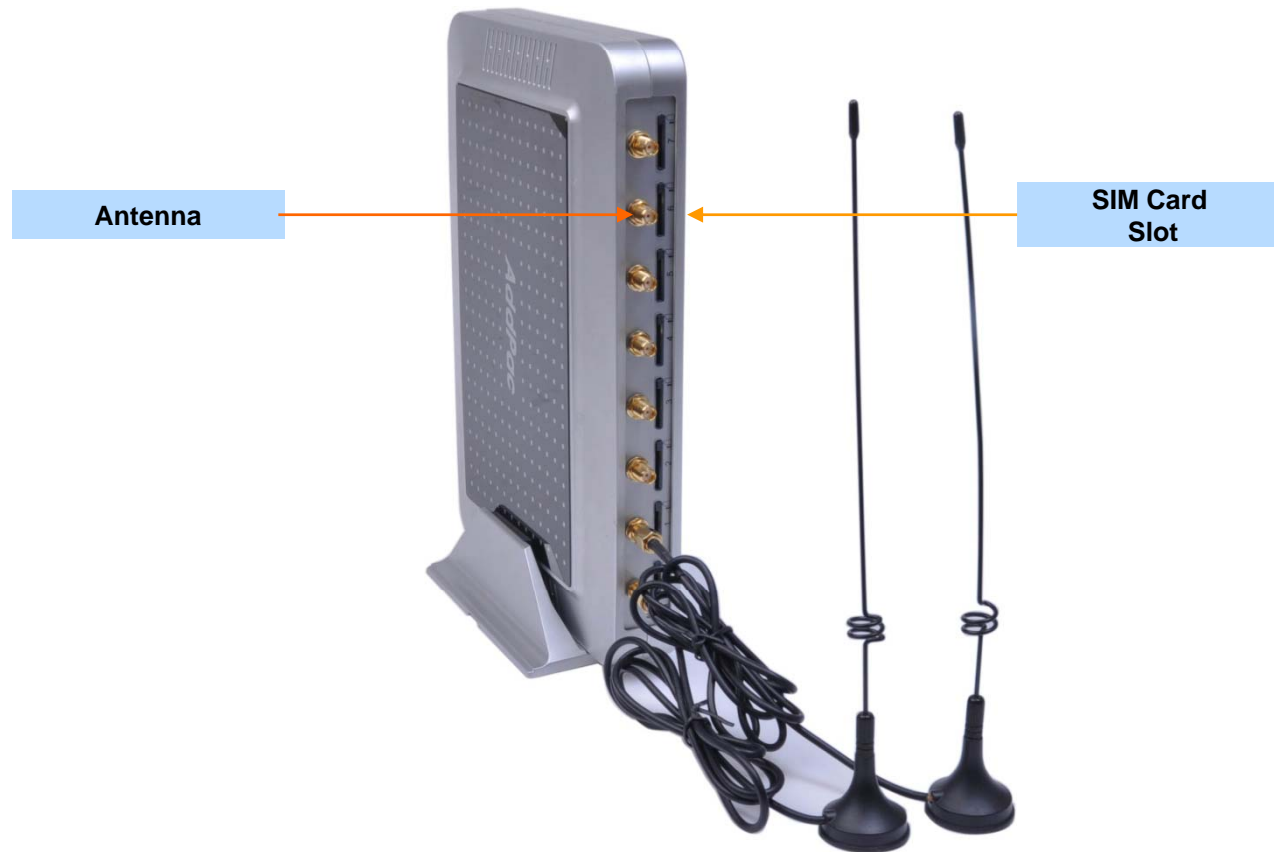
Hardware Specification

AP-GS708 8-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

Back Side View





AP-GS808 GSM SOHO Gateway

Main Features

AP-GS808 8-Port GSM VoIP Gateway

- 8-Port GSM VoIP Gateway Solution
- 8-Port SIM Slots, 8-Port Antenna Interface
- GSM VoIP Interface(LAN) Support
- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- One(1) RS-232C Port for Command Line Interface
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with Internal Power Supply

Hardware Specification

AP-GS808 8-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

- RISC Microprocessor Computing Power
- Powerful High-End DSP for VoIP Interface
- 8-Port GSM VoIP Gateway
- 8-Port SIM Card Slots
- 8-Port GSM Antenna Interface
- Network Interface for VoIP Direct Interface
 - Two(2) 10/100Mbps Fast Ethernet (RJ45)
- RS232C Console Interface for CLI
- Run LED, LAN LED, Port LEDs at Front Side
- Compact and Light Design with Internal Power Supply

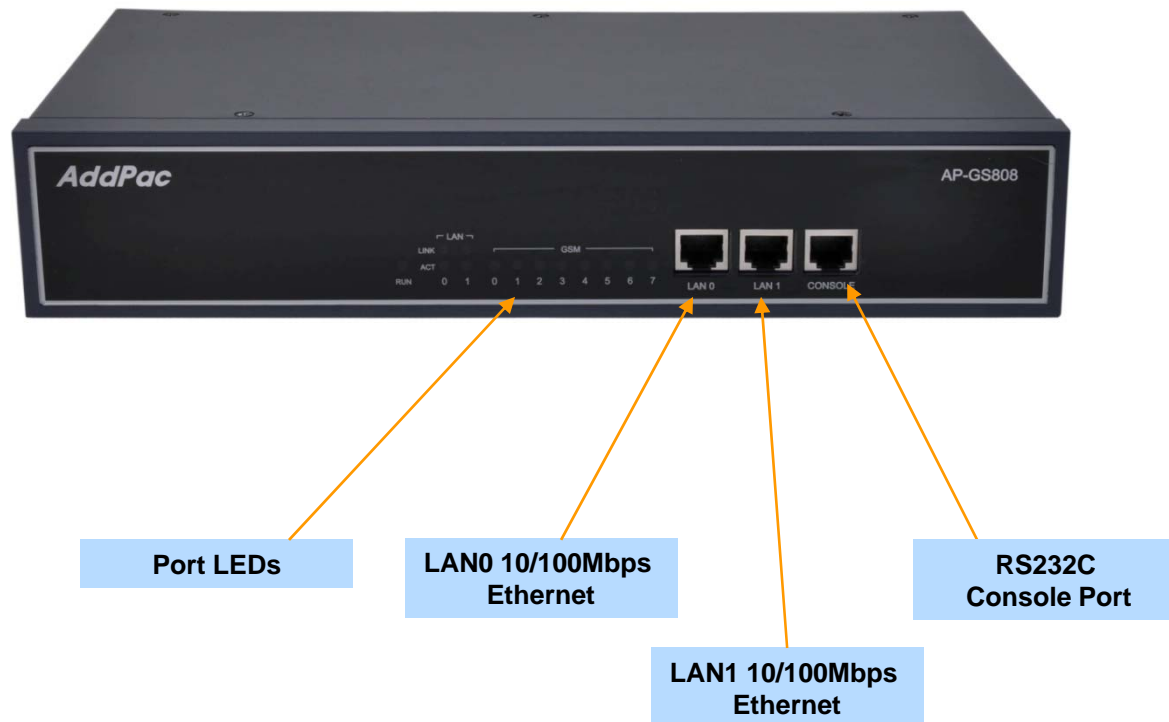
Hardware Specification

AP-GS808 8-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

Front Side View



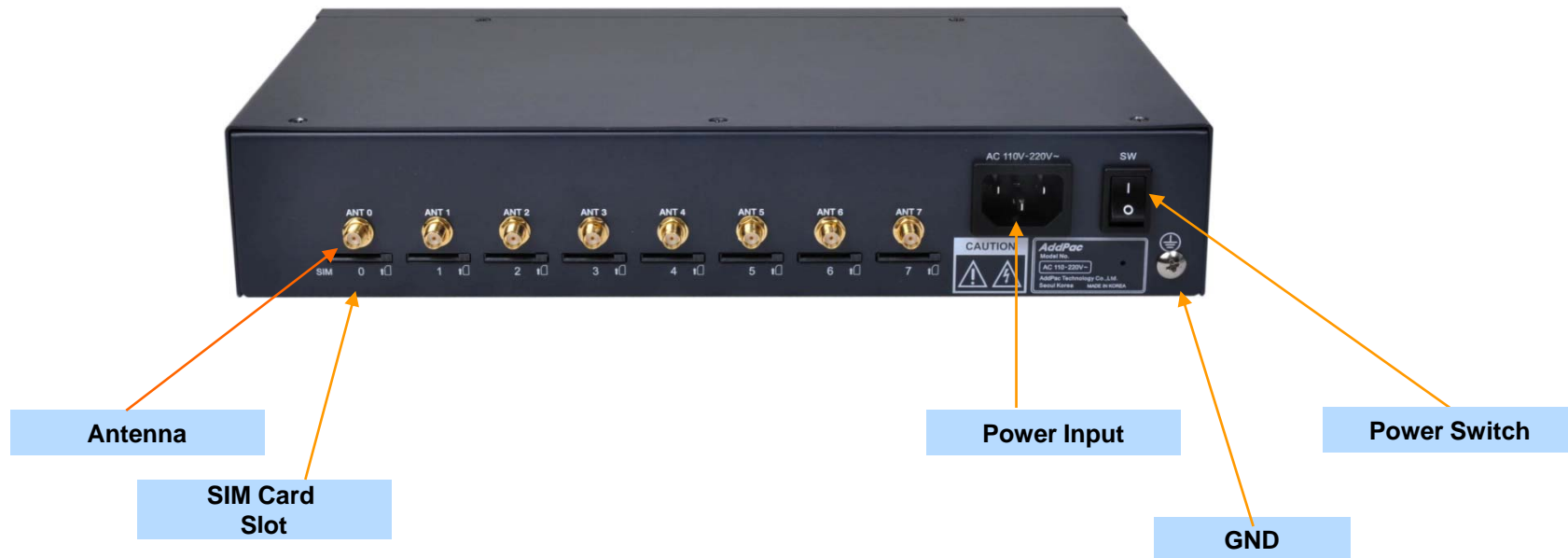
Hardware Specification

AP-GS808 8-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

Back Side View





AP-GS816 GSM SOHO Gateway

Main Features

AP-GS816 16-Port GSM VoIP Gateway

- 16-Port GSM VoIP Gateway Solution
- 16-Port SIM Slots
- 16-Port Antenna Interface
- GSM VoIP Interface(LAN) Support
- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- G.711/G.726/G.723/G.729, T.38 Fax , VAD, etc
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Two(2) 10/100Mbps Fast Ethernet (IP Share ,etc)
- One(1) RS-232C Port for Command Line Interface
- Firmware Upgradeable Architecture
- VPMS (VoIP Plug&Play Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with Internal Power Supply

Hardware Specification

AP-GS816 16-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

- RISC Microprocessor Computing Power
- Powerful High-End DSP for VoIP Interface
- 16-Port GSM VoIP Gateway
- 16-Port SIM Card Slots
- 16-Port GSM Antenna Interface
- Network Interface for VoIP Direct Interface
 - Two(2) 10/100Mbps Fast Ethernet (RJ45)
- RS232C Console Interface for CLI
- Run LED, LAN LED, Port LEDs at Front Side
- Compact and Light Design with Internal Power Supply

Hardware Specification

AP-GS816 16-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

Front Side View



LAN0 10/100Mbps
Ethernet

LAN1 10/100Mbps
Ethernet

RS232C
Console Port

Port LEDs

GND

Power Input

Power Switch

Hardware Specification

AP-GS816 16-Port GSM VoIP Gateway

RISC
CPU

High-end
DSP

Back Side View



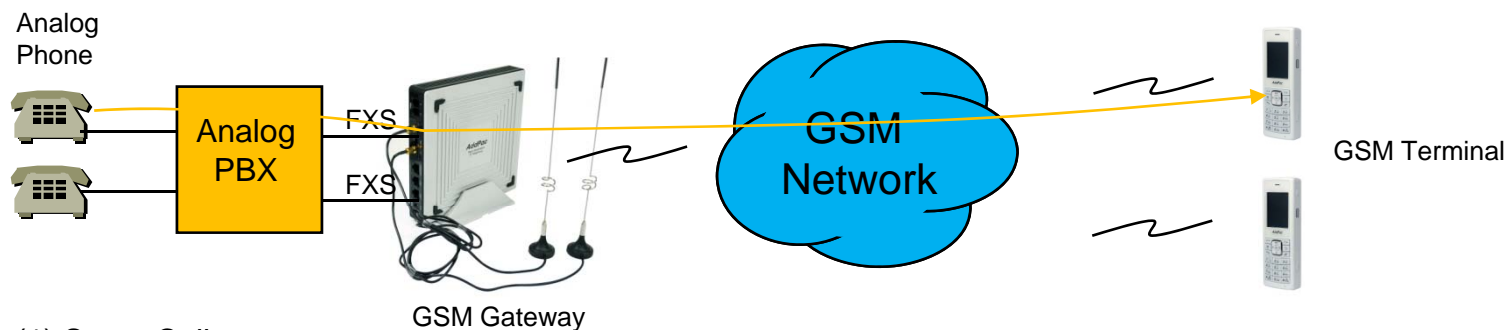


GSM Gateway Function List

Contents

- GSM Outbound Call
- GSM Inbound Call
- VoIP to GSM Outbound Call
- VoIP to GSM Inbound Call
- GSM Inbound Black / White list
- VoIP to GSM Black / White list
- WEB Callback Service
- Callback Service
- LCR(Least Cost Routing)
- GSM Messaging Service
- Radius Server Interoperability

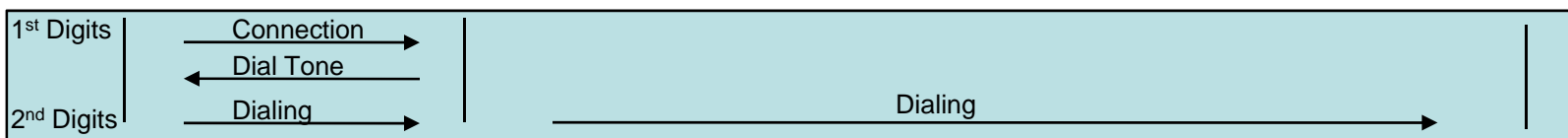
GSM Outbound Call



One(1) Stage Call



Two(2) Stage Call



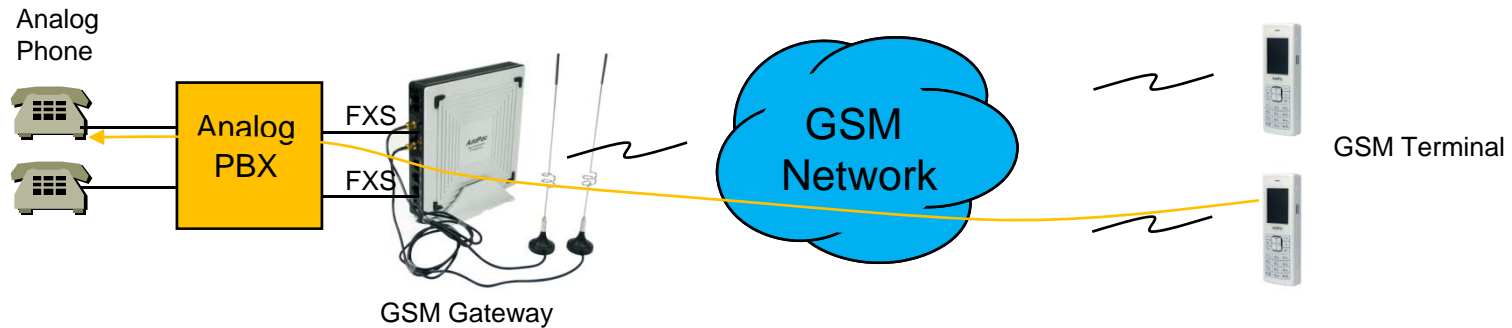
Outbound Call (1 Stage)

: Making call to mobile phone from analog phone connected to FXS directly.

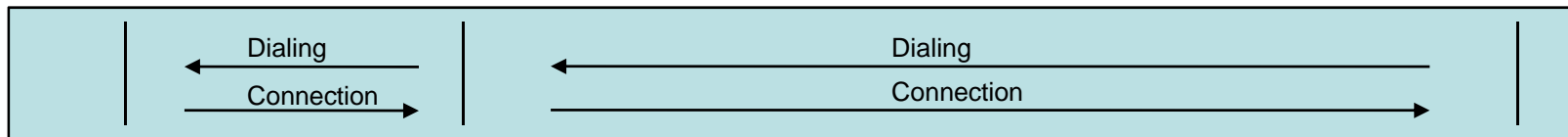
Outbound Call (2 Stage)

: Making call to mobile phone from analog phone connected to FXS after hearing of 2nd dial tone from AddPac GSM Gateway

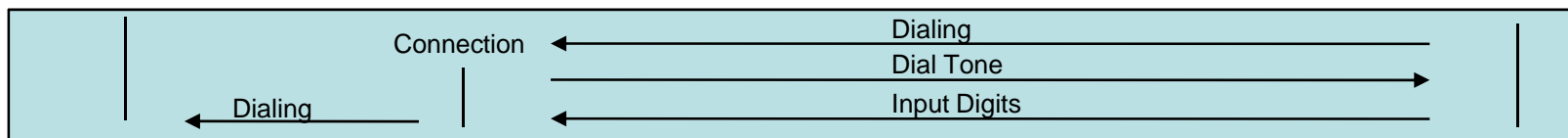
GSM Inbound Call



1 Stage Call (Baby Call)



2 Stage Call



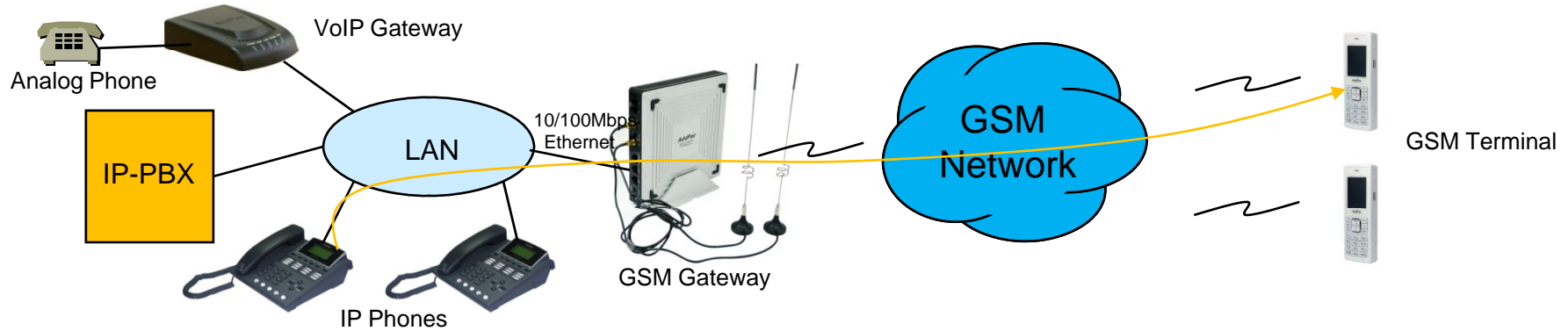
Inbound Call (1 Stage) – Baby Call

: Making call to analog phone connected to FXS directly

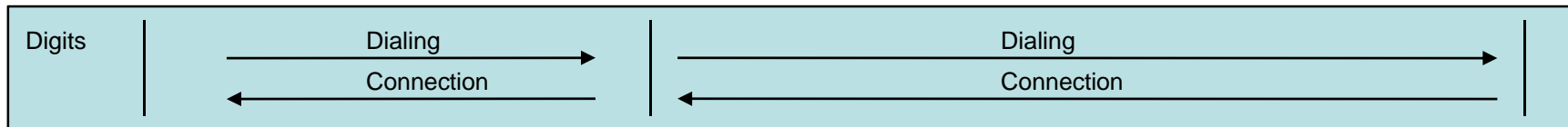
Inbound Call (2 Stage)

: Making call to analog phone connected to FXS after hearing of 2nd dial tone from AP-GS1002

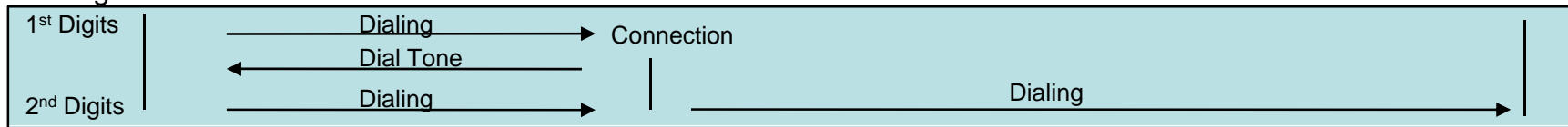
VoIP to GSM Outbound Call



1 Stage Call



2 Stage Call



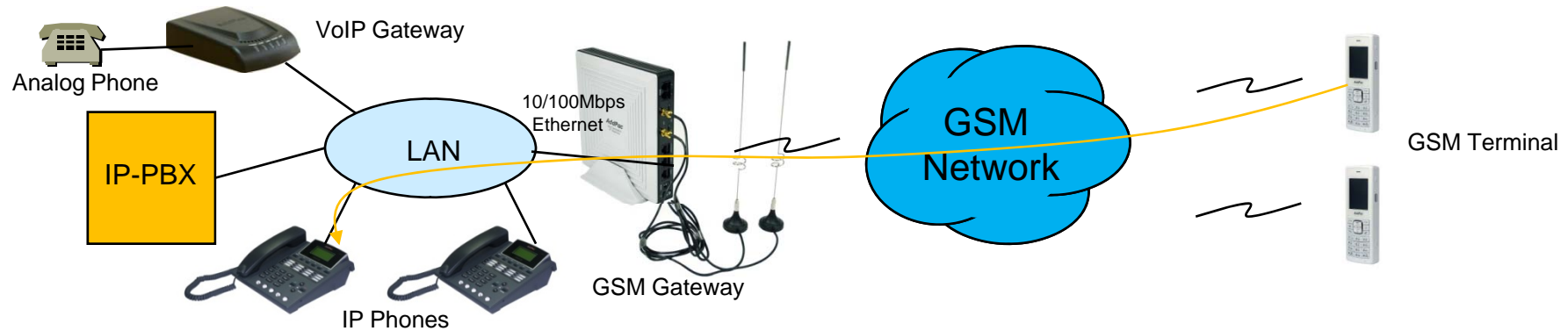
Outbound Call (1 Stage)

: Making call to mobile phone from analog phone connected to VoIP gateway or IP Phone directly

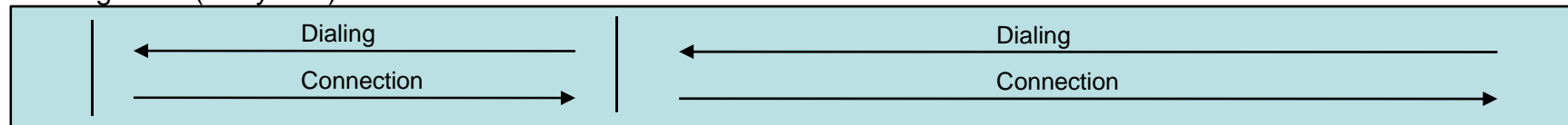
Outbound Call (2 Stage)

: Making call to mobile phone from analog phone connected to VoIP gateway after hearing of 2nd dial tone from GSM Gateway

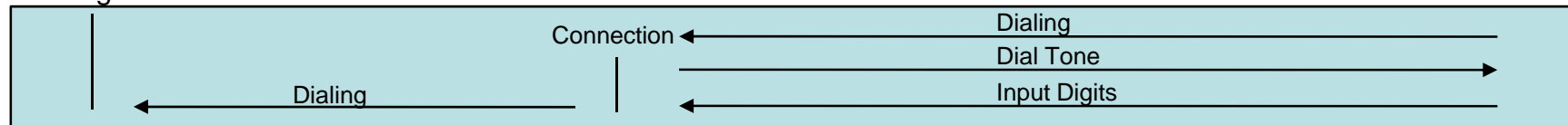
VoIP to GSM Inbound Call



1 Stage Call (Baby Call)



2 Stage Call



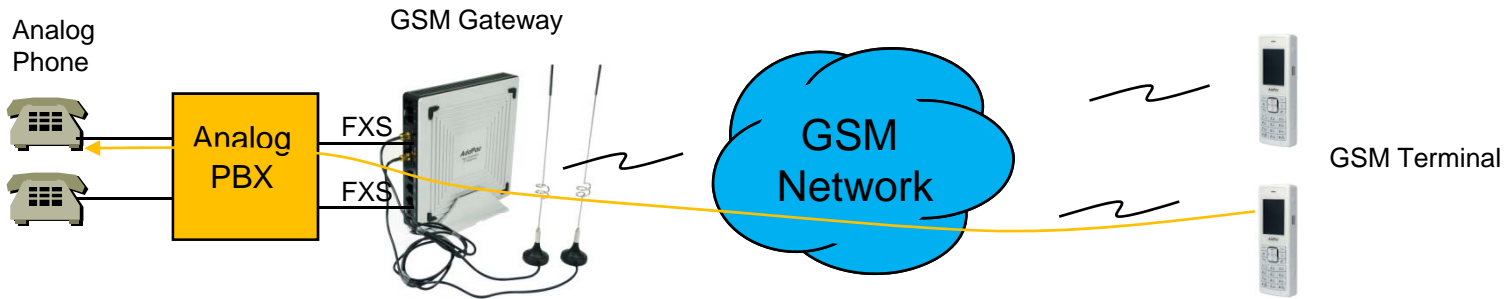
Inbound Call (1 Stage) – Baby Call

: Making call to IP phone in VoIP network directly.

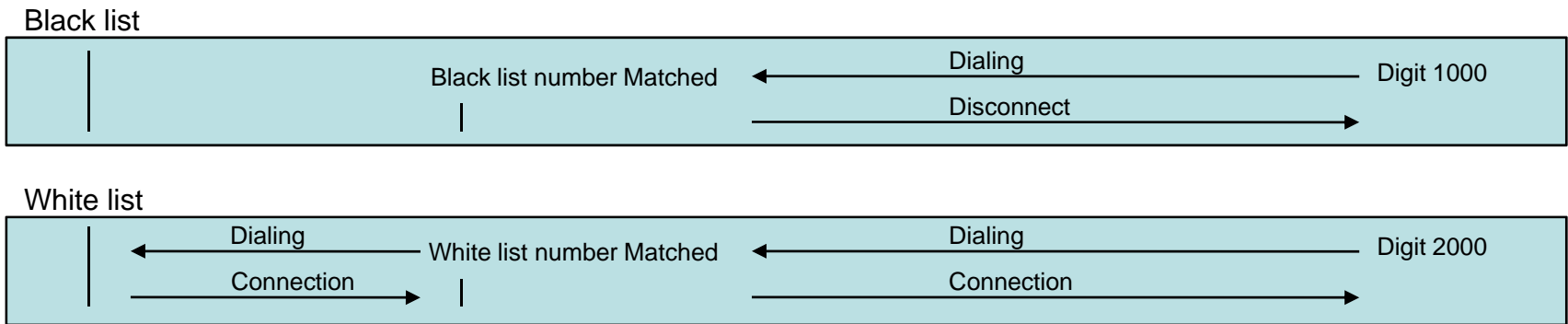
Inbound Call (2 Stage)

: Making call to IP phone in VoIP network after hearing of 2nd dial tone from GSM Gateway

GSM Inbound Call Black / White list



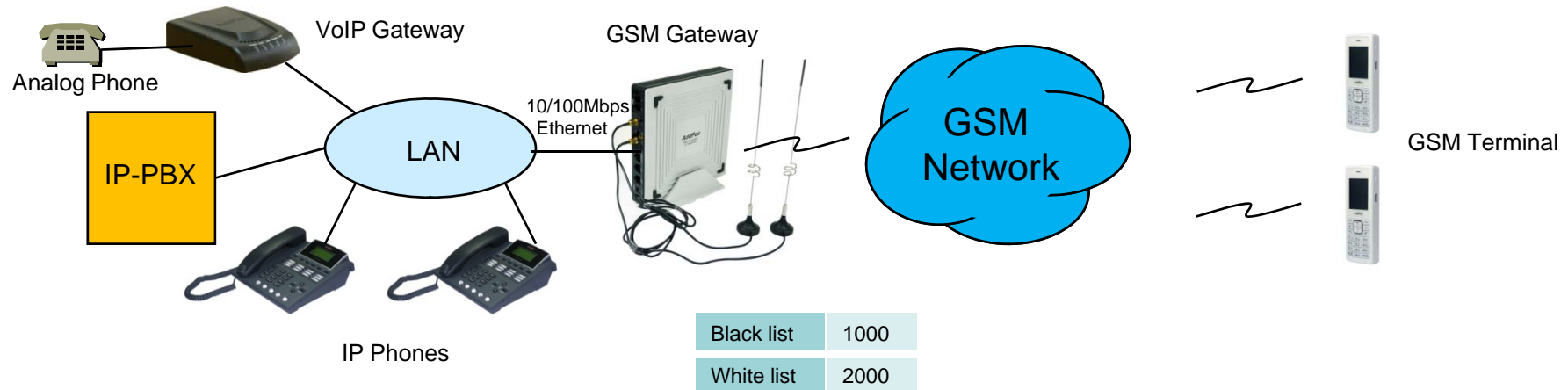
Black list	1000
White list	2000



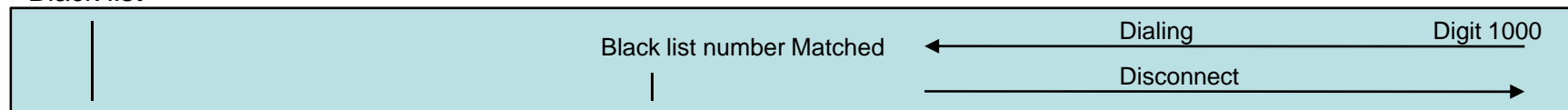
Black list
: The number on black list is restricted to receive call.

White list
: The only number on white list is allowed to receive call

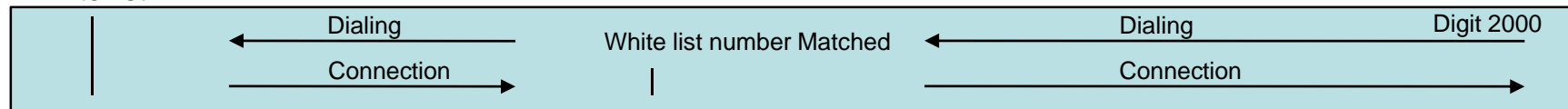
VoIP to GSM Black / White list



Black list



White list



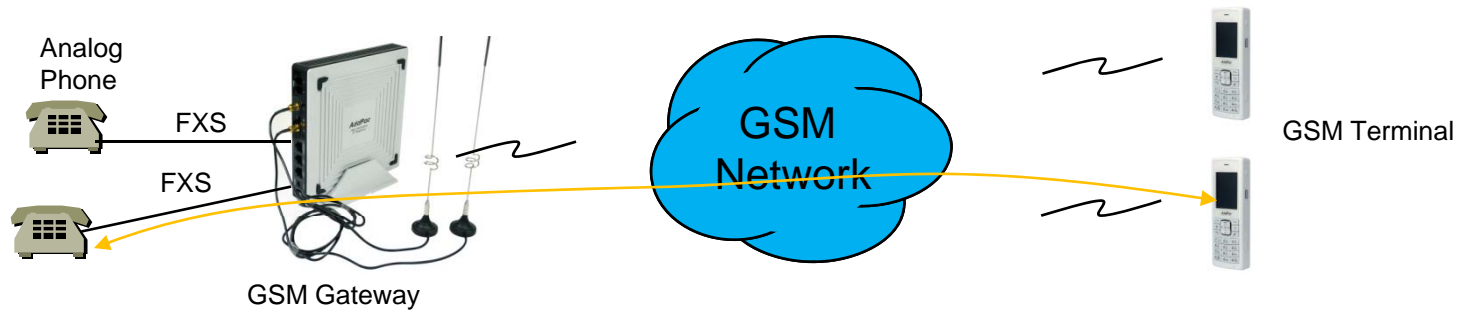
Black list

: The number on black list is restricted to receive call.

White list

: The only number on white list is allowed to receive call

WEB Callback Service



Origination number(1000)

WEB Call back White list	1000
--------------------------	------

Destination number(2000)

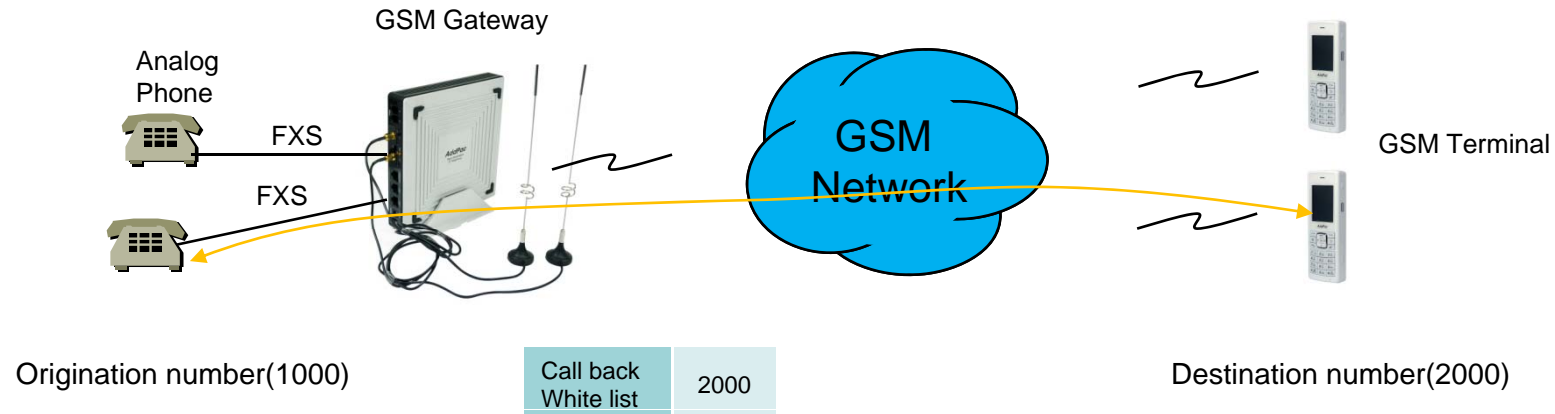
WEB Callback Service



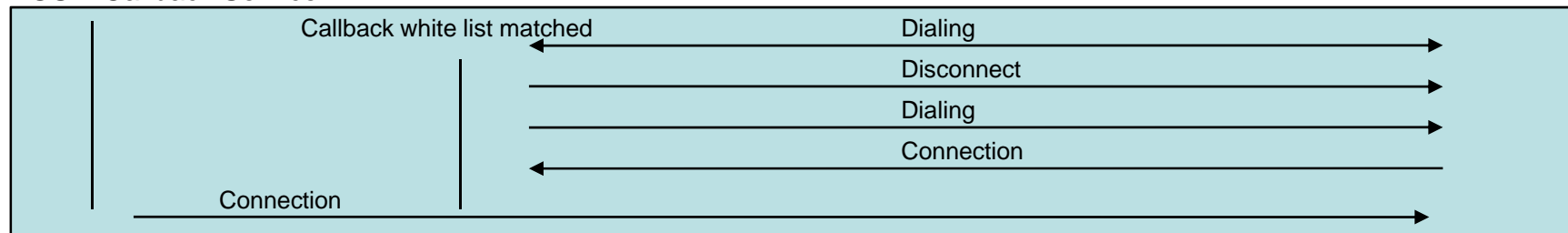
WEB Callback Service

- : The remote call is made by user's control by WEB Interface.
- The WEB callback number on white list must be the same of source number.

GSM Callback Service



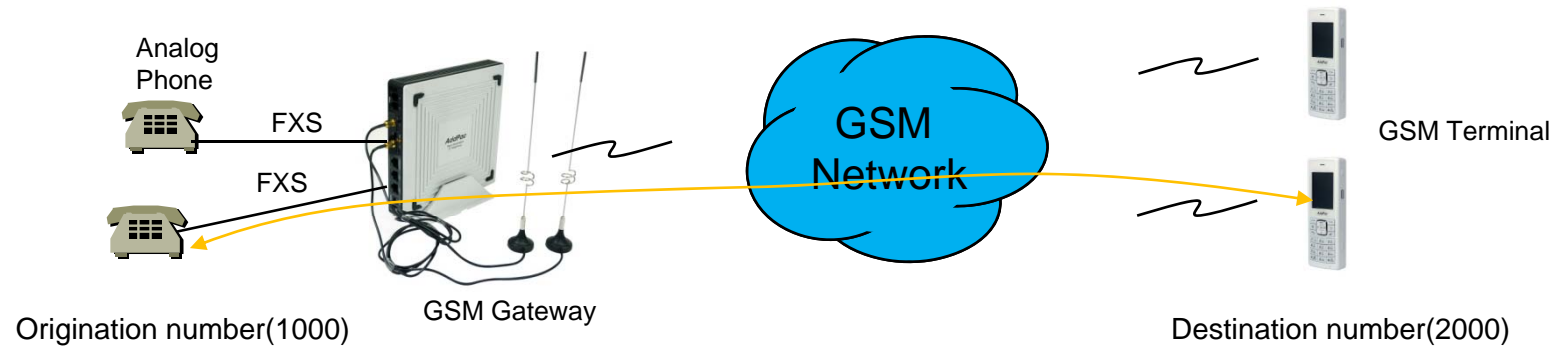
GSM Callback Service



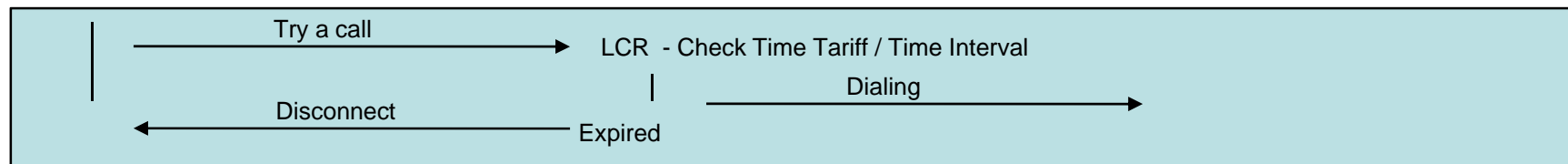
GSM Callback Service

: When the user on the callback white list makes call, GSM Gateway disconnects it and makes call back to the user

LCR(Least Cost Routing)



LCR(Least Cost Routing)



GSM LCR Time Interval

: The only registered user is allowed to use GSM call in the rule of date, week, and time

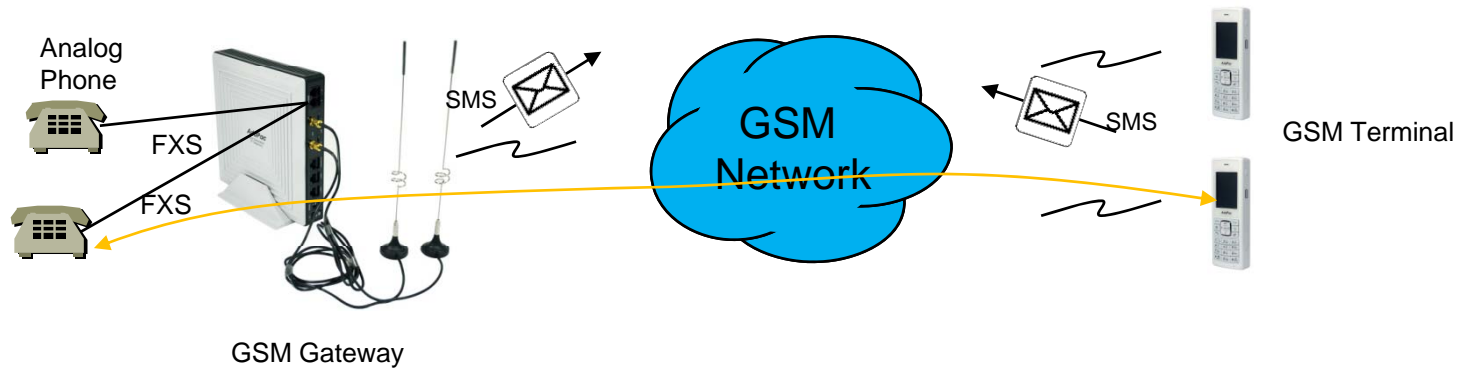
GSM LCR Time Tariff

: User is able to check remained time, used time listed on LCR, etc

GSM LCR Simulator

: GSM Gateway supports virtual call simulation used on WEB

GSM Messaging Service



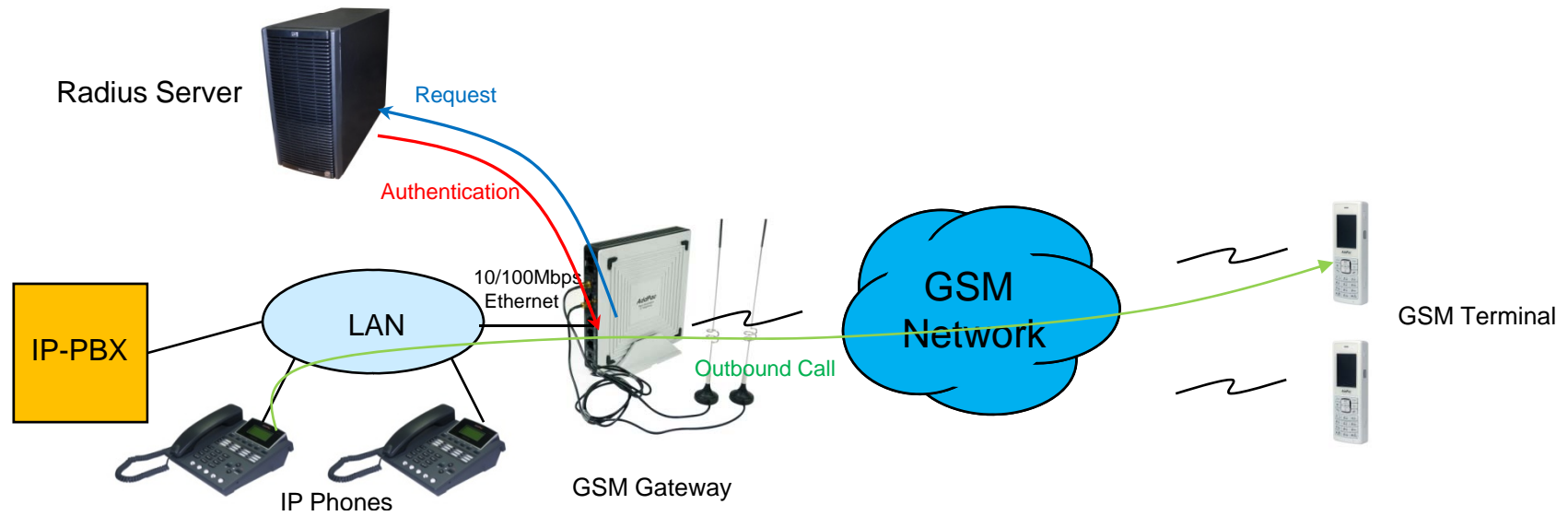
GSM Messaging Service

- : SMS is able to send and receive by GSM Gateway's WEB Interface
- : English, Korean, Spanish, Russian, Portuguese

USSD

- : In case of using Pre-paid SIM card, checking and recharging is allowed by GSM Gateway

Radius Server Interoperability



Radius Server Interoperability

: When billing system is required, GSM gateway supports radius server interoperability



Smart Web Manager for GSM Gateway

Contents

- Main Page Layout
- System Configuration
 - Network Setup, Language, NAT, PPTP, NTP
- Basic Configuration
 - Protocol, SIP Server , FXS Extension, GSM Extension
 - DTMF/CODEC, VoIP Dial Plan, GSM Dial Plan, Static Routing, Hot Line
- Advanced Configuration
 - Gain/CID, GSM PINs, FAX, Service, Filtering, Security
 - GSM Web Callback, GSM Callback
- Miscellaneous Configuration
 - Call Status, System Status, Alarm Status, GSM Status
 - Call Log, System Log, Ping, BTS Selection, GSM BTS Info
- LCR(Least Cost Routing)
 - Black & White List, Time Interval, Tariff Group, LCR Test
- SMS
 - Inbox, SMS New Message

Main Page Layout

Main Menu
For easy system setup, provide the various menu and category

- System
 - Network Setup
 - Language
 - NAT
 - PPTP
 - NTP
- Basic
 - Protocol
 - Server SIP
 - SIP Registration
 - FXS Extension
 - GSM Extension
 - DTMF/CODEC
 - VoIP Dial Plan
 - GSM Dial Plan
 - Static Route
 - Hot Line
- Advanced
 - Gain & CID
 - GSM PINs
 - Fax
 - Service
 - Filtering
 - Security
 - SNMP
 - WEB Callback
 - GSM Callback

Tool Bar
Provide frequently used tools like as System Update, Configuration Backup, Initialization, Restart, Telnet

Information
Display the current system version and status summary

H/W Version	2.0
SW Version	8.00d
MAC Address	0002.a400.0000
VoIP Protocol	SIP
Voice Interface Module	G(2)S(2)
Registration Status	Registered
Supported Codec List	
Network Information	Static 172.16.9.16
WAN LINK Status	100Mbps FULL Duplex Link UP
LAN LINK Status	Link Down
Current Time	Fri Jan 1 01:49:57 2010
System Startup Time	Fri Jan 1 00:00:00 2010
	0 days 01:49:57
	0

Workspace
Workspace for detailed action

Description
Display the help message if you move mouse over main menu

Information
AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface : G(2)S(2)
Protocol : SIP
Status : Registered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

System - Language

Smart Web Manager
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System

- Network Setup
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs

Configure Language

한국어

English

Apply

Information

AddPac Technology
Model : GS1002_G2
HW Version : 2.0
SW Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Registered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Choose the basic language to be applied. English is set at default.

Configure Language
English, Korea

System - NAT

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System

- Network Setup
- Language
- **NAT**
- PPTP
- NTP

Basic

- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security

NAT Static Table

IP Protocol	Global Port	Local Address	Local Port	Control
tcp	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

When many PCs are connected to LAN, create a table for delivering TCP/UDP port to PC.

NAT Static Table
When many PCs are connected to LAN, create a table for delivering TCP/UDP port to PC

System - PPTP

The screenshot shows the 'Smart Web Manager' interface for configuring PPTP. The main content area is titled 'Tunneling' and contains three sections: 'Mode', 'Source & Destination', and 'Service'. Three yellow callout boxes provide additional context:

- Tunneling Mode:** Configure Tunneling mode. This points to the 'Mode' section where 'None(Disable Tunneling, default)' is selected.
- Tunneling source & destination:** Source LAN port & destination IP. This points to the 'Source' dropdown (FastEthernet0/0) and the 'Destination' text input field.
- Tunneling Service:** Service mode. This points to the 'Service' section where 'Voice and Data Use Tunnel Interface (default)' is selected.

The 'Information' sidebar on the right provides system details:

- AddPac Technology
- Model : GS1002_G2
- H/W Version : 2.0
- S/W Version : 8.00d
- Smart Web Version : 0.4
- Smart Web Build : Mar 24 2010
- Voice Interface: G(2)S(2)
- Protocol : SIP
- Status : Unregistered
- CurrentCalls : 0 Call
- Network : Static 172.16.9.16
- Mac Address : 0002.a400.0000
- Unread Message: P0:0(0), P0:1(0)

System - NTP

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- PPTP
- **NTP**

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route

NTP

Enable Disable

Primary Server (Domain Name or IP Address)

Secondary Server (Domain Name or IP Address)

Interval (1~72 hours)

Hours Offset : (-23~23 hours) : (0~60 minute)

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

NTP
Configure NTP server (s) & Options

Basic - Protocol

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Basic

- **Protocol**
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

Protocol

SIP(Session Initiation Protocol)

Apply

Configure VoIP signaling protocol SIP , H.323 (optional)

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Configure the settings of the protocol to be used for VoIP communication

Basic – SIP Server

Smart Web Manager
www.addpac.com

SIP (Session Initiation Protocol)

Use SIP Server Yes No

Primary SIP Server Server address (IP or Domain Name) and Port (default 5060)

Secondary SIP Server Server address (IP or Domain Name) and Port (default 5060)

Local Domain name (SIP userpart of authentication)

SIP Signaling Port (default 5060, between 1 to 65535)

Register Expiration (in seconds, default 60, between 10 to 86400)

Session Re-Fresh INVITE UPDATE

Session Expire Time (in seconds, default 1800, between 30 to 86400, 0 = disable)

Apply

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Configure the settings for SIP.
Contact your service provider
for the settings

SIP Server
Primary & Secondary server,
Local domain name,
SIP Signaling Port (reboot necessary)
Timer
* register expire
* session refresh
* session expire

Basic – FXS Extension

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- **FXS Extension**
- GSM Extension

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

FXS Extension

Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

FXS Extension Configuration

Index	Port	Numbers	Preference	HuntStop	Select
0	0/2	1234	0	0	<input type="checkbox"/>

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Set up for using FXS port to extension number (forwarding No)

Port Information
voice port type & physical port

FXS Extension
Configure phone-number for using inter-office Preference (0 : highest)

Basic – GSM Extension

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- PPTP

System

- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension**

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

GSM Extension

Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

GSM Extension Configuration

Index	Port	Numbers	Preference	HuntStop	Select
0	0/0	T	0	X	<input type="checkbox"/>

GSM Extension with Translation

Port	Destination Pattern	Digits to Insert	Number of Digits to Delete
P0:0	33	8	1
P0:1			0

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Set up for using GSM port to extension number (forwarding No)

Port Information
voice port type & physical port

GSM Extension
Configure GSM phone-number for receiving a call (usually 'T' is used for each port)

GSM Extension with Translation
Used to GSM callback

- The Received CID is not real serving number.
- The specified translation rule is applied.

Basic – DTMF/CODEC

The screenshot shows the Smart Web Manager interface for configuring DTMF and CODEC settings. The main content area is titled "DTMF/CODEC" and contains two sections: "Voice CODEC" and "DTMF Relay mode".

CODEC
Configure voice codec preference
(g711a, g711u, g729, g7231, g726)

Preference	Value
Preference 1	None
Preference 2	None
Preference 3	None
Preference 4	None
Preference 5	None
Preference 6	None

DTMF Relay mode

- DTMF relay by In-band voice
- DTMF relay by RTP payload defined by RFC 2833
- DTMF relay by Out-of-band signal
- DTMF relay by Cisco out-of-band signal

DTMF
Configure DTMF relay method
(in-band, RFC2833, out-of-band, CISCO type out-of-band)

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Configure the settings for GSM Dial Plan and Prefix table

Basic – VoIP Dial Plan

VoIP PLAN

Configure translation rule for VOIP Peer.
 - first, 'Number of Digits to Delete' option is applied.

- second, 'Digits to Insert' option is applied.

(ex) Origin called Number = 123456
 Number of Digits to Delete = 2
 Digits to Insert = "88"

 result = 883456

System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

VoIP Dial Plan / Prefix

Plan Table

Index	Digits to Insert	Number of Digits to Delete	Digit Pattern	Control
0	2	0	T	<input type="checkbox"/>

Prefix Table

Index	Prefix	PlanIndex	Control
0	T	2	<input type="checkbox"/>

Information

AddPac Technology
 Model : GS1002_G2
 H/W Version : 2.0
 S/W Version : 8.00d
 Smart Web Version : 0.4
 Smart Web Build : Mar 24 2010
 Voice Interface
 G(2)S(2)
 Protocol : SIP
 Status : Unregistered
 CurrentCalls: 0 Call
 Network : Static 172.16.9.16
 Mac Address: 0002.a400.0000
 Unread Message:
 P0:0(0)
 P0:1(0)

Description

Configure the settings for the outbound call of main/remote and incoming E1 and routing

Prefix Table

Configure VoIP Peer with translation rule.
 (Serviced by SIP SERVER)

Basic – GSM Dial Plan

Smart Web Manager
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GSM Dial Plan / Prefix

Port Information

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

Plan Table

Index	Digits to Insert	Number of Digits to Delete	Digit Pattern	Control
0	1	1	2T	<input type="checkbox"/>

Prefix Table

Index	Prefix	2nd Prefix	PlanIndex	Slot/Port	Control
0	33	2T	0	0/0	<input type="checkbox"/>

Information

AddPac Technology
Model : GS1002_G2
HW Version : 2.0
SW Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Hot Line Setup

System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

VoIP PLAN
Configure translation rule for GSM Peer.
- first, 'Number of Digits to Delete' option is applied.
- second, 'Digits to Insert' option is applied.

(ex) Origin called Number = 123456
Number of Digits to Delete = 2
Digits to Insert = "88"

result = 883456

Port Information
voice port type & physical port

Prefix Table
Configure GSM Peer with translation rule.

Basic – Static Route

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- **Static Route**
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM RTS Info

Static Route

Set Remote Site Call(5-digit number is set to begin *2->*2...)

No	Remote Site IP	Prefix	Insert Digit	Delete Digit	Name of Remote Site	Answer Addr	Control
0	172.16.1.1	2...	172.16.9.16	0	Factory	T	<input type="checkbox"/>
*	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
							<input type="button" value="Apply"/>

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Static Route
Configure Static VoIP Peer for using Inter-Office .
(Already, I know IP & phone-number)

Basic – Hot Line

Smart Web Manager
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Hot Line

Hot Line Configuration

Port	Hot Line Number	Digit Input Timeout <0~10 sec>
S0P0(G)	<input type="text"/>	n.a.
S0P1(G)	<input type="text"/>	n.a.
S0P2(S)	8888	5
S0P3(S)	<input type="text"/>	<input type="text"/>

Apply

Hot Line
- Used as baby-call(Connection PLAR)
- Timer (FXS port only : No Digit event is occurred for configured timer value, Auto-Dialing will be started)

Information
AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Advanced – Gain & CID

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID**
- GSM PINs
- Fax
- Service
- Filtering
- Security
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM RTS Info

Gain

Port	Port Type	InputGain	OutputGain	Caller ID
P0:0	GSM	0	0	<input checked="" type="checkbox"/>
P0:1	GSM	0	0	<input checked="" type="checkbox"/>
P0:2	FXS	0	0	<input checked="" type="checkbox"/>
P0:3	FXS	0	0	<input checked="" type="checkbox"/>

Apply

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Adjust the input voice volume from FXS/FXO/E1/E&M to DSP and the output volume from DSP to the phone or PSTN line;

Gain & CID
Configure Input-gain, output-gain and caller-ID.

Advanced – GSM PINs

Smart Web Manager
www.addpac.com

GSM PINs

PINs ⓘ

Port	PIN for SIM card
P0:0	<input type="text"/>
P0:1	<input type="text"/>

GSM PIN
Configure GSM PIN(Personal Identification Number)

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Configure GSM PINs

Advanced - Fax

Smart Web Manager
www.addpac.com

Fax

Fax Mode T.38 Inband T.38 Bypass

Fax Rate Disable 2400 4800 7200 9600 12000 14400

Apply

FAX
Configure fax mode & rate (VoIP Lines)

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Enable or disable T.38/Inband T.38, which is fax internet protocol and specify Baudrate

Advanced - Service

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax
- **Service**
- Filtering
- Security
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

Service

Applicaton Services

- Enable Telnet Server Port: 23 (default 23, 1-65535)
- Enable HTTP Server Port: 80 (default 80, 1-65535)
- Enable FTP Control Port: 21 (default 21, 1-65535)
Data Port: 20 (default 20, 1-65535)
- Enable Syslog Primary Server: [] Port: [] (default 514)
Secondary Server: [] Port: [] (default 514)
Log Level: 0-emergency
Log Command: disable

Timer

- Inter Digit Time: 3 sec (default 3, 1-600)

Call Service

- Transfer: Hook-Flash Not-assigned
- Hold: Hook-Flash Not-assigned

SIP Transfer

- Mode: blind Attended

Apply

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Enable or disable Telnet, HTTP, FTP and specify the access port and Call Hold/Transfer and Timer .

Service

- Configure application service(Telnet, HTTP, ftp, syslog)
- Configure IDT(Inter Digit Time)
- Configure Call-Transfer-Mode & Hook-Flash-Usage-Type.
- Configure Call-Transfer-Mode.

Advanced - Filtering

Smart Web Manager
www.addpac.com

System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- **Filtering**
- Security
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

Filter

FTP Filter

Network Addr	Network Mask	Control
<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

HTTP Filter

Network Addr	Network Mask	Control
<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

Telnet Filter

Network Addr	Network Mask	Control
<input type="text"/>	<input type="text"/>	<input type="button" value="Add"/>

Information

AddPac Technology
Model : GS1002_G2
HW Version : 2.0
SW Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

For FTP, HTTP, Telnet, set up one IP or IP band for allowing access

Filter
Configure application service filter with IP & Subnet mask.

Advanced - Security

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System

- Network Setup
- Language
- NAT
- PPTP
- NTP

Basic

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Advanced

- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- **Security**
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

Security

IP Filtering Enable Disable

WarDialing Filtering Enable Disable

Allow Digit Length(IP to PSTN) Min Max

SIP Shutdown Enable Disable

Apply

Information

AddPac Technology
Model : GS1002_G2
HW Version : 2.0
SW Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Configure the settings for IP authentication and protocol and incoming number

Security

- IP Filtering : Allowing only the inbound call which is registered to Call-Routing of the server by static IP.
- WarDialing : Allowing only the inbound call with the number registered to Inter-Office and phone-number.
- Digit Length : Allowing only the inbound call with the number registered to Inter-Office and phone-number
- SIP : SIP signaling packets are filtered.

Advanced – GSM Callback

GSM Callback
 The employee working at the out of office can use this function.

- The Call received from GSM network is automatically disconnected.
- GSM gateway calls to the calling number.

Calling Number White List
 The employee working at the out of office are usually registered.

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GSM Callback

Calling Number Whitelist

Group	Index	DialPattern	Control
3	0	123T	<input type="checkbox"/>

3 | 0 | [] | [Delete] | [Add]

Callback

GSM Port	My Number	WhiteList Group
P0:0		3
P0:1		N.A.

[Apply]

Information

AddPac Technology
 Model : GS1002_G2
 HW Version : 2.0
 SW Version : 8.00d
 Smart Web Version : 0.4
 Smart Web Build : Mar 24 2010
 Voice Interface
 G(2)S(2)
 Protocol : SIP
 Status : Unregistered
 CurrentCalls : 0 Call
 Network : Static 172.16.9.16
 Mac Address : 0002.a400.0000
 Unread Message:
 P0:0(0)
 P0:1(0)

Description

Execute GSM callback function and Configure callback whitelist

Callback
 The white list group is adapted to specific GSM port

Miscellaneous – Call Status

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

Call Status

Port Status (Analog)

Slot	Port Group				
	Port	0()	1()	2()	3()
SLOT 0	Status	I	I	I	I
	Select	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>

Unblock Block

Connection State : (Connected) (Disconnected || Blocked)
Call State : (Idle) (Ring || Dial) (Called) (Calling) (Blocked)

Call Status

Port	Direction	Established Time	Calling Number	Called Number	CODEC	Src/Dest. IP

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Verify port status and retrieve the present call information

Analog Port
Real-time display about analog port status (occupation, call status). Provide a specific port blocking function

Active Call Status
Real-time display about current active call status (calling party addr, called party addr. Codec, etc)

Miscellaneous – System Status

System Status

- voice port status & information
- SIP-UA status & information
- gateway status & information
- system utilization information

Smart Web Manager
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System

- Network Setup
- Language
- NAT
- PPTP
- NTP

System Status

Voice Port

Port	LineType	Status	InGain	OutGain	TieType	TieDigits	CallNum	Tcalled	Tcalling
0/ 0	GSM	Idle	0	0	none		-1	-1	-1
0/ 1	GSM	Idle	0	0	none		-1	-1	-1
0/ 2	FXS	Idle	0	0	hot-line	8888	-1	-1	-1
0/ 3	FXS	Idle	0	0	none		-1	-1	-1

SIP-UA

Proxyserver Registration Information
proxyserver registration option = e164
Proxyserver list :

Server address	Port	Priority	Domain	Status (LastFailReason)
172.17.116.215	5060	128	any	Failed(Rx:OtherMsg)

Proxy Server registration status :

E.164	UserName	Password	Port	Status
1005	1005	NONE	0/ 2	Registered
33	33	NONE	0/ 0	Failed

SIP UA Timer counters
retry counter = 10

SIP UA Timer values
 tretry (sip retry timer) = 500 msec.
 tinterval (sip retry max interval timer) = 4 sec.
 treg (sip register timer) = 60 sec.
 tregtry (sip register retry timer) = 20 sec.
 texpires (sip invite expire timer) = 180 sec.
 tsipping (sip ping timer) = 45 sec.

SIP UA Session Timer value
 Min-SE = 1800 sec.
 Session-Expires = 1800 sec.

SIP DNS SRV Query : Disable
 SIP Call Transfer Mode : Basic
 SIP Media Channel Start Mode : Default
 SIP Reliable Provisional Response Option : Supported with value <100rel>
 SIP Response Option : default
 SIP Local Domain : NULL
 SIP Special Char : NULL
 SIP Routing Method of Incoming Call : Default
 SIP Remote-Party-ID : Disabled

Information

AddPac Technology
 Model : GS1002_G2
 H/W Version : 2.0
 S/W Version : 8.00d
 Smart Web Version : 0.4
 Smart Web Build : Mar 24 2010

Voice Interface
 G(2)S(2)
 Protocol : SIP
 Status : Unregistered
 CurrentCalls: 0 Call
 Network: Static 172.16.9.16
 Mac Address: 0002.a400.0000
 Unread Message:
 P0:0(0)
 P0:1(0)

Description

Verify the present port information, Server Register status, CPU and Memory usage

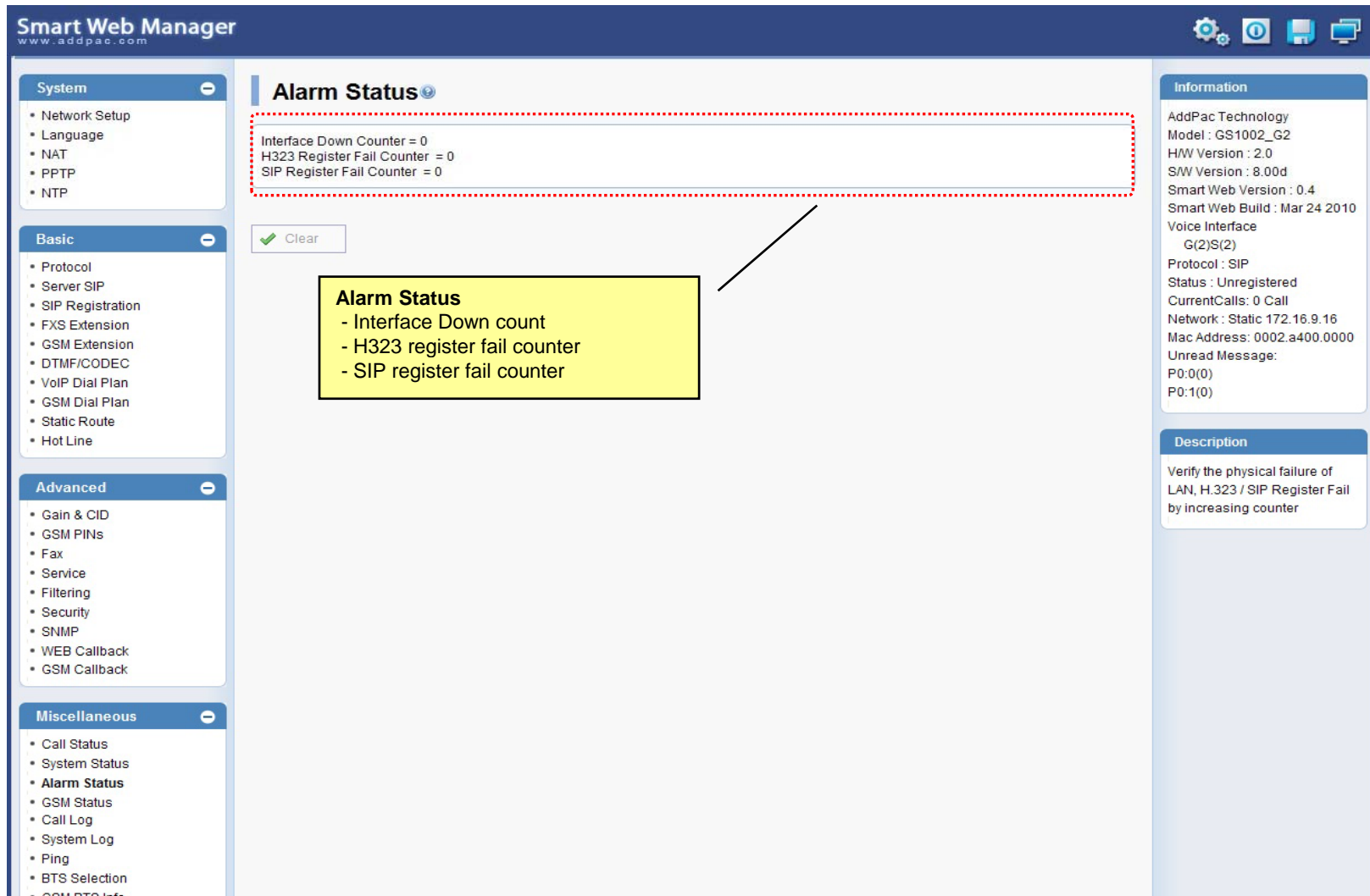
Advanced

- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line
- Gain & CID
- GSM PINs
- Fax
- Service
- Filtering
- Security
- SNMP
- WEB Callback
- GSM Callback

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- Ping
- BTS Selection
- GSM BTS Info

Miscellaneous – Alarm Status



The screenshot shows the 'Smart Web Manager' interface with the 'Alarm Status' page selected. The left sidebar contains a tree view with categories: System, Basic, Advanced, and Miscellaneous. The 'Alarm Status' page displays the following information:

- Alarm Status**
 - Interface Down Counter = 0
 - H323 Register Fail Counter = 0
 - SIP Register Fail Counter = 0
-

A red dashed box highlights the alarm status text, and a yellow box with a black border contains a summary of the alarm status:

Alarm Status

- Interface Down count
- H323 register fail counter
- SIP register fail counter

The right sidebar contains 'Information' and 'Description' sections. The 'Information' section lists system details such as AddPac Technology, Model, and Version. The 'Description' section provides a warning: 'Verify the physical failure of LAN, H.323 / SIP Register Fail by increasing counter'.

Miscellaneous – GSM Status

Smart Web Manager
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GSM Status

GSM Port Status & Information

Port	My Phone Number	Device Information		Accounting (Used/Quota/Free)	
		Register Status	Signal Strength	Voice Quota(secs)	SMS Quota(E.A.)
P0:0		REG	0dB	0 / 36000 / 36000	0 / 300 / 300
P0:1		REG	0dB	0 / -1 / -1	0 / -1 / -1

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010

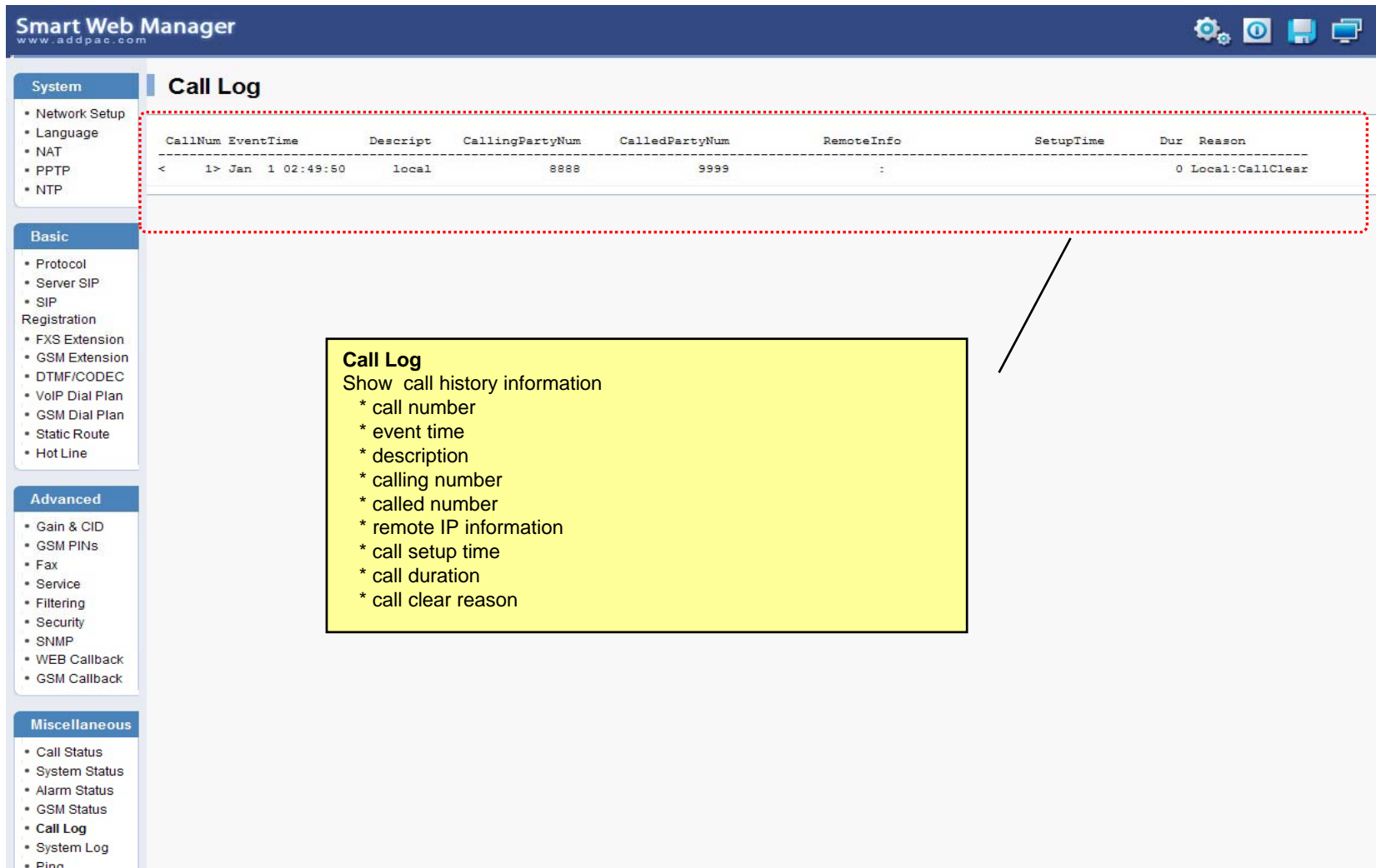
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description
Display GSM wireless status

GSM Status

- my number
- GSM register status
- GSM signal strength
- Account information
 - * voice quota (used / quota / free)
 - * SMS quota (used / quota / free)

Miscellaneous – Call Log



The screenshot shows the 'Smart Web Manager' interface with the 'Call Log' section active. The interface includes a left sidebar with navigation menus for System, Basic, Advanced, and Miscellaneous. The main content area displays a table of call log entries. A red dashed box highlights the table header and the first entry. A yellow callout box provides a list of fields shown in the call log.

CallNum	EventTime	Descript	CallingPartyNum	CalledPartyNum	RemoteInfo	SetupTime	Dur	Reason
< 1>	Jan 1 02:49:50	local	8888	9999	:		0	Local:CallClear

Call Log
Show call history information

- * call number
- * event time
- * description
- * calling number
- * called number
- * remote IP information
- * call setup time
- * call duration
- * call clear reason

Miscellaneous – System Log

Smart Web Manager
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System Log

command logging buffers (messages logged)

event logging buffers (messages logged)

System Log
- command log
- system alarm log (ex : interface down)

Information
AddPac Technology
Model : GS1002_G2
HW Version : 2.0
SW Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description
Retrieve the system log

System
• Network Setup
• Language
• NAT
• PPTP
• NTP

Basic
• Protocol
• Server SIP
• SIP Registration
• FXS Extension
• GSM Extension
• DTMF/CODEC
• VoIP Dial Plan
• GSM Dial Plan
• Static Route
• Hot Line

Advanced
• Gain & CID
• GSM PINs
• Fax
• Service
• Filtering
• Security
• SNMP
• WEB Callback
• GSM Callback

Miscellaneous
• Call Status
• System Status
• Alarm Status
• GSM Status
• Call Log
• **System Log**
• Ping
• BTS Selection
• GSM BTS Info

Miscellaneous - Ping

Smart Web Manager
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System

- Network Setup
- Language
- NAT

Advanced

- Protocol
- Server SIP
- SIP Registration
- FXS Extension
- GSM Extension
- DTMF/CODEC
- VoIP Dial Plan
- GSM Dial Plan
- Static Route
- Hot Line

Miscellaneous

- Call Status
- System Status
- Alarm Status
- GSM Status
- Call Log
- System Log
- **Ping**
- BTS Selection
- GSM BTS Info

Ping

Host address Start

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls: 0 Call
Network : Static 172.16.9.16
Mac Address: 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Verify the physical failure of LAN, H.323 / SIP Register Fail by increasing counter

PING
You can diagnose network status by PING.

PING
Show real time ping status.

LCR – Black & White List

Smart Web Manager
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GSM LCR / Black List & White List

BlackList

Index	DialPattern	Control
0	888T	<input type="checkbox"/>
		Delete
<input type="text" value="0"/> <input type="text"/>		Add

WhiteList

Index	DialPattern	Control
0	2...	<input type="checkbox"/>
		Delete
<input type="text" value="0"/> <input type="text"/>		Apply

Information

AddPac Technology
 Model : GS1002_G2
 HW Version : 2.0
 SW Version : 8.00d
 Smart Web Version : 0.4
 Smart Web Build : Mar 24 2010
 Voice Interface
 G(2)S(2)
 Protocol : SIP
 Status : Unregistered
 CurrentCalls : 0 Call
 Network : Static 172.16.9.16
 Mac Address : 0002.a400.0000
 Unread Message:
 P0:0(0)
 P0:1(0)

Description

Configure black & white list

LCR Black & White List
 Black List : The patterns are disallowed GSM outbound call.
 White List : The patterns are allowed GSM outbound call.

LCR – Time Interval

The screenshot displays the Smart Web Manager interface for configuring GSM LCR / Time Interval Groups. The main content area is titled "GSM LCR / Time Interval Group" and contains a "TimeInterval" section. This section features a table with the following data:

Group	Days	StartTime(hh:mm)	EndTime(hh:mm)	Control
0	Weekdays	00:00	23:59	<input type="checkbox"/>

Below the table, there are input fields for adding a new group: "0" for Group, "weekend" for Days, "0" for StartHour, "0" for StartMinute, "0" for EndHour, and "0" for EndMinute. There are also "Delete" and "Add" buttons.

A yellow callout box points to the table with the text: "Time Interval GSM outbound call is restricted by Time Interval".

The left sidebar shows a navigation menu with categories: System, Basic, Advanced, and Miscellaneous. The right sidebar contains "Information" and "Description" sections. The "Information" section lists system details like AddPac Technology, Model: GS1002_G2, and Smart Web Build: Mar 24 2010. The "Description" section contains the text: "Configure time interval group".

LCR – Tariff Group

Smart Web Manager
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GSM LCR / Tariff Group

Tariff Group

Group	Time Group	Restore Call Limit		Accounting Period		Free Quota		Control
Type	RestoreDay	First(sec)	Others(sec)	Voice(min)	SMS(E.A.)			
0	0	monthly	15	30	10	600	300	<input type="checkbox"/>
0	0	daily	1					Delete Add

TariffPort

Port	TariffGroup
P0:0	0
P0:1	N.A.
P0:2	N.A.
P0:3	N.A.

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

Configure tariff group

Time Interval Group
Time interval Group is adapted to this tariff group.

Restore call limit
Quota restore time

Accounting Period
Used to voice call.
(ex)
- configured as First (30 seconds) Others(10 seconds)

first connect -- 30 seconds are accounted.
after 30 seconds -- 10 seconds are additionally accounted.
after 10 seconds – 10 seconds are additionally accounted.
and so on.

Tariff Group
Tariff group is adapted to specific GSM port.

Free quota
Free quota information.
Current Usage information is supported at GSM Status.

LCR – LCR Test

The screenshot shows the 'Smart Web Manager' interface for an LCR Test. The main area contains input fields for 'Caller' (8888) and 'Called Number' (9999), a 'Start' button, and a log window showing simulation results. The log indicates a failure due to an 'Outbound White Group' mismatch. Annotations on the left describe the LCR simulator and the real-time status display.

System

- Network Setup
- Language
- NAT
- PPTP
- NTP

LCR Test
LCR simulator

- SIP Registration
- FXS Extension
- GSM Extension

LCR Test
Show real time simulation status.

Advanced

- Gain & CID
- GSM PIN

LCR Test

Caller:

Called Number:

```
< 1> LCR : =====
< 2> LCR : == GSM LCR(Least Cost Route) Simulator Start ==
< 3> LCR : =====
< 4> LCR : -- src digits : 8888(GSM) -> dst digits : 9999(GSM)
< 5> LCR : -- MatchAllProcess After Sorted
< 6> LCR : <0> id(3048) dest(T) prefer(0) selected(0)
< 7> LCR : -- Trying : <0> id(3048) dest(T)
< 8> LCR : -- Error: Outbound White Group(id:1) UnMatched
< 9> LCR : -----
< 10> LCR : -- Result : Fail
< 11> LCR : =====
< 12> LCR : == GSM LCR(Least Cost Route) Simulator End ==
< 13> LCR : =====
```

Information

AddPac Technology
Model : GS1002_G2
H/W Version : 2.0
S/W Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

LCR Test

SMS – Inbox

Smart Web Manager
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GSM SMS / InBox

number of messages are 0 P0:0 OK

Index	Sender	Received	Message	Select
-------	--------	----------	---------	--------

< > Delete

Information

AddPac Technology
Model : GS1002_G2
HW Version : 2.0
SW Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description

GSM SMS / In Box

- total message
- unread messages (Blue color)
- received time
- content

SMS – SMS New Message

The screenshot shows the 'Smart Web Manager' interface with the 'GSM SMS / New Message' page. The left sidebar contains a navigation menu with categories: System, Basic, Advanced, and Miscellaneous. The main content area features a form for sending a message, with fields for 'Phone Number', 'Message', and 'Port' (set to P0:0), and a 'Send' button. A red dashed box highlights the form fields, and a yellow callout box points to it with the text: 'New Message send a new message to the other GSM mobile phone.' The right sidebar displays system 'Information' and a 'Description' section.

Smart Web Manager
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GSM SMS / New Message

Max size is 80 characters

Phone Number

Message

Port

Send

New Message
send a new message to the other GSM mobile phone.

Information

AddPac Technology
Model : GS1002_G2
HW Version : 2.0
SW Version : 8.00d
Smart Web Version : 0.4
Smart Web Build : Mar 24 2010
Voice Interface
G(2)S(2)
Protocol : SIP
Status : Unregistered
CurrentCalls : 0 Call
Network : Static 172.16.9.16
Mac Address : 0002.a400.0000
Unread Message:
P0:0(0)
P0:1(0)

Description



Thank you!

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