

# GSM SOHO Gateway Solution



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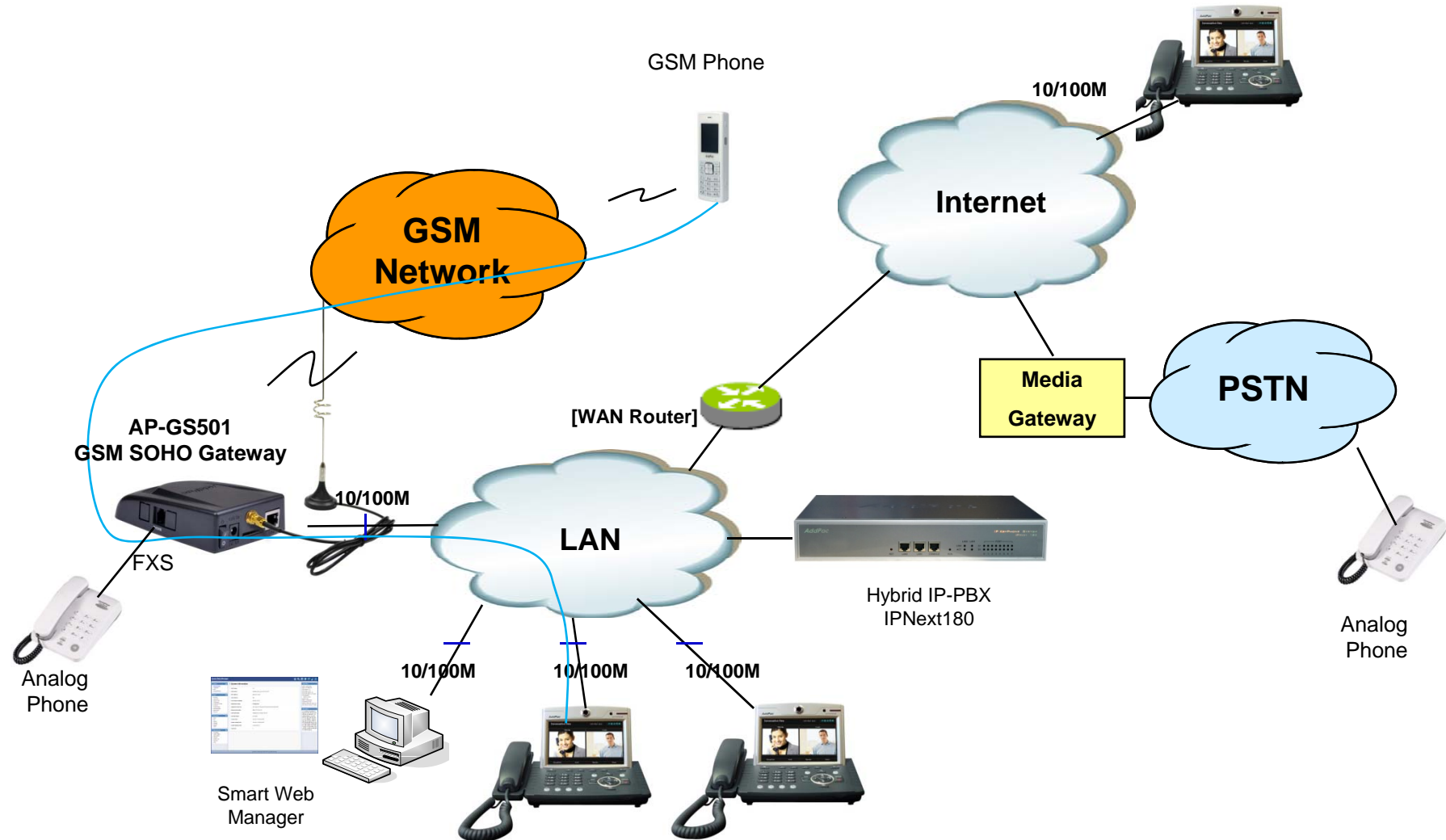
**AddPac Technology**

2012, Sales and Marketing

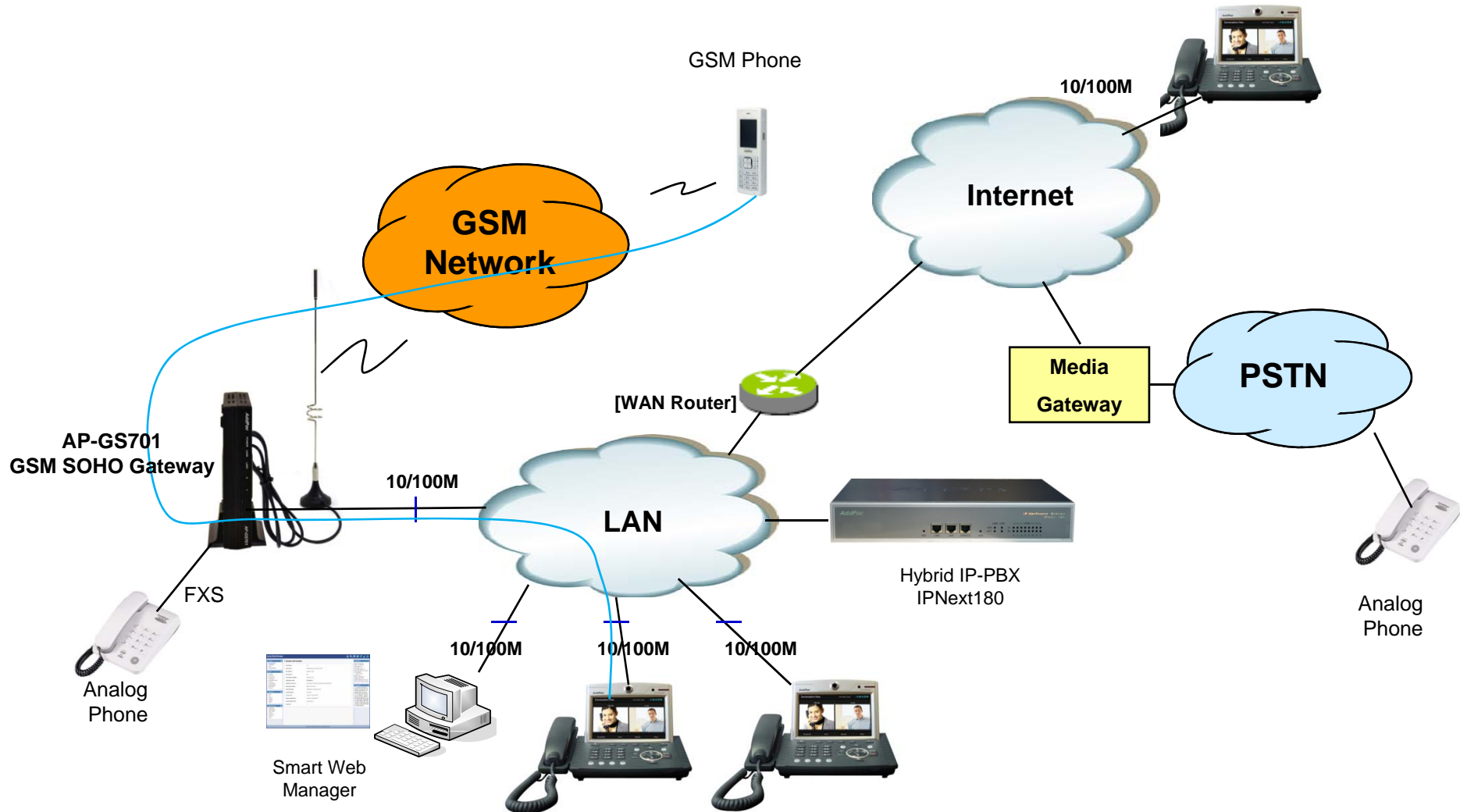
# Contents

- GSM SOHO Gateway Service Diagram
- GSM SOHO Gateway Series
  - AP-GS501(1ch)
  - AP-GS701(1ch)
  - AP-GS702(2ch)
  - AP-GS802(2ch)
  - AP-GS804(4ch)
  - AP-GS708(8ch)
  - AP-GS808(8ch)
  - AP-GS816(16ch)
- GSM SOHO Gateway Function List
- Smart Web Manager for GSM Gateway
- NMS (Network Management System) for GSM Gateway

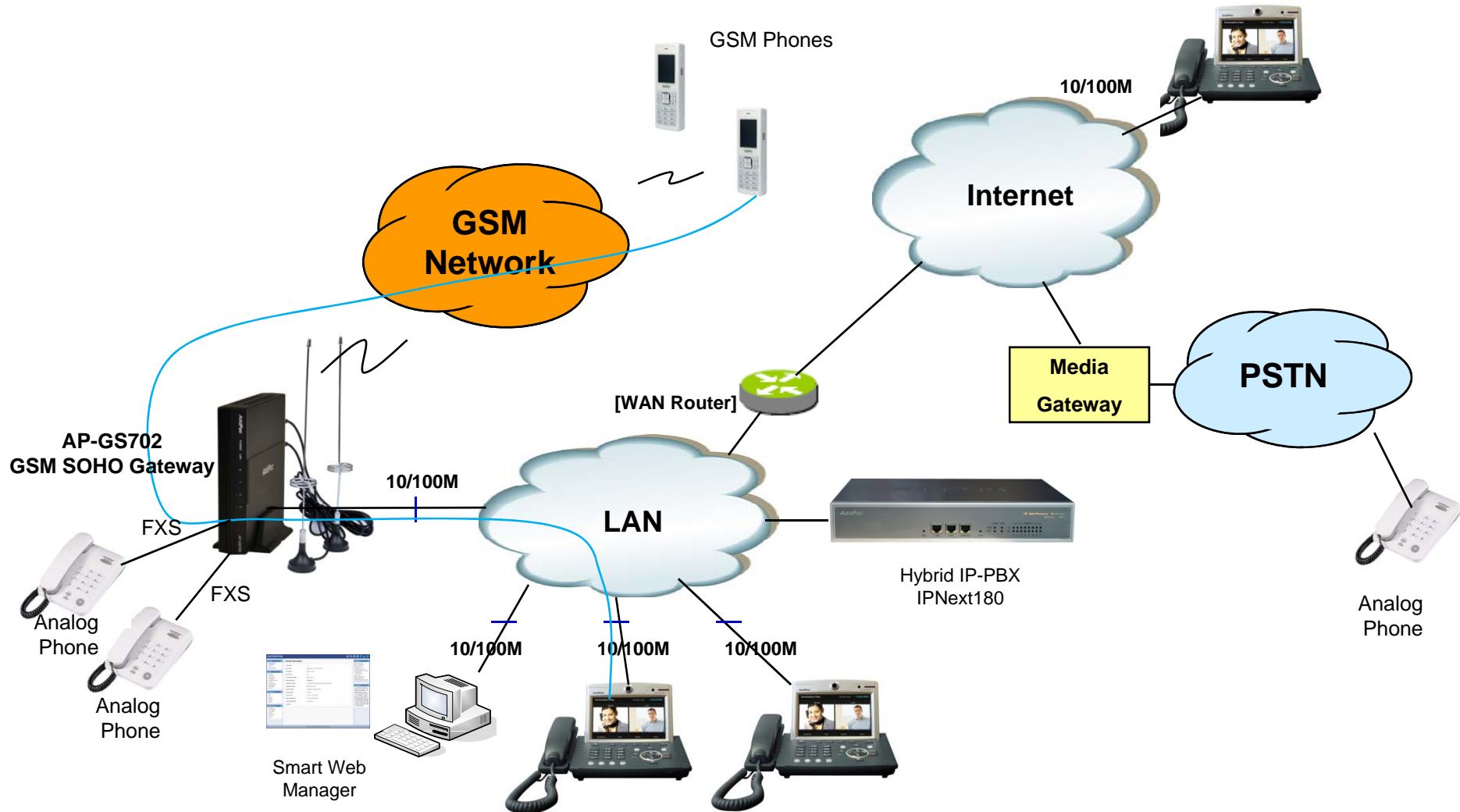
# GSM Gateway Service Diagram (AP-GS501)



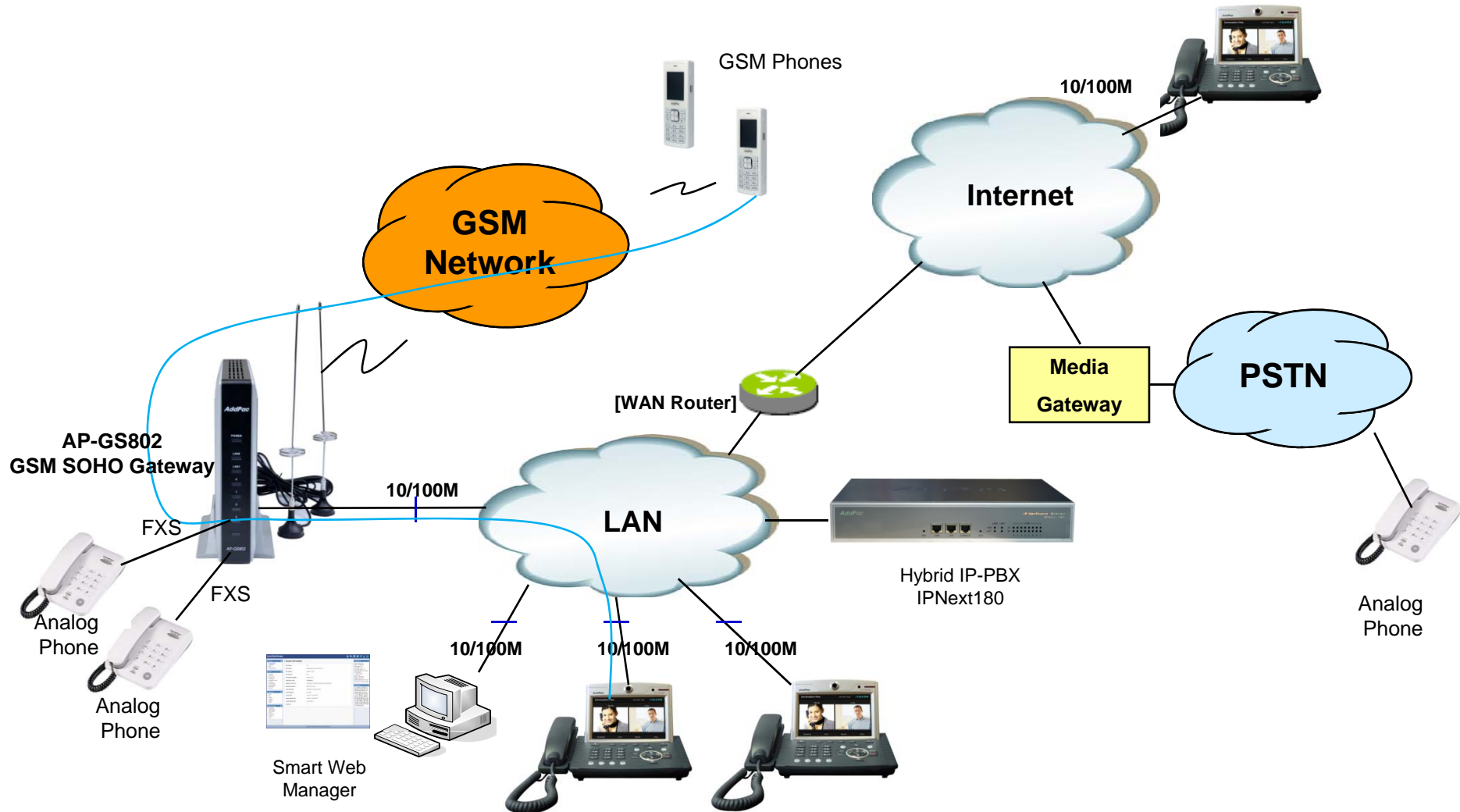
# GSM Gateway Service Diagram (AP-GS701)



# GSM Gateway Service Diagram (AP-GS702)

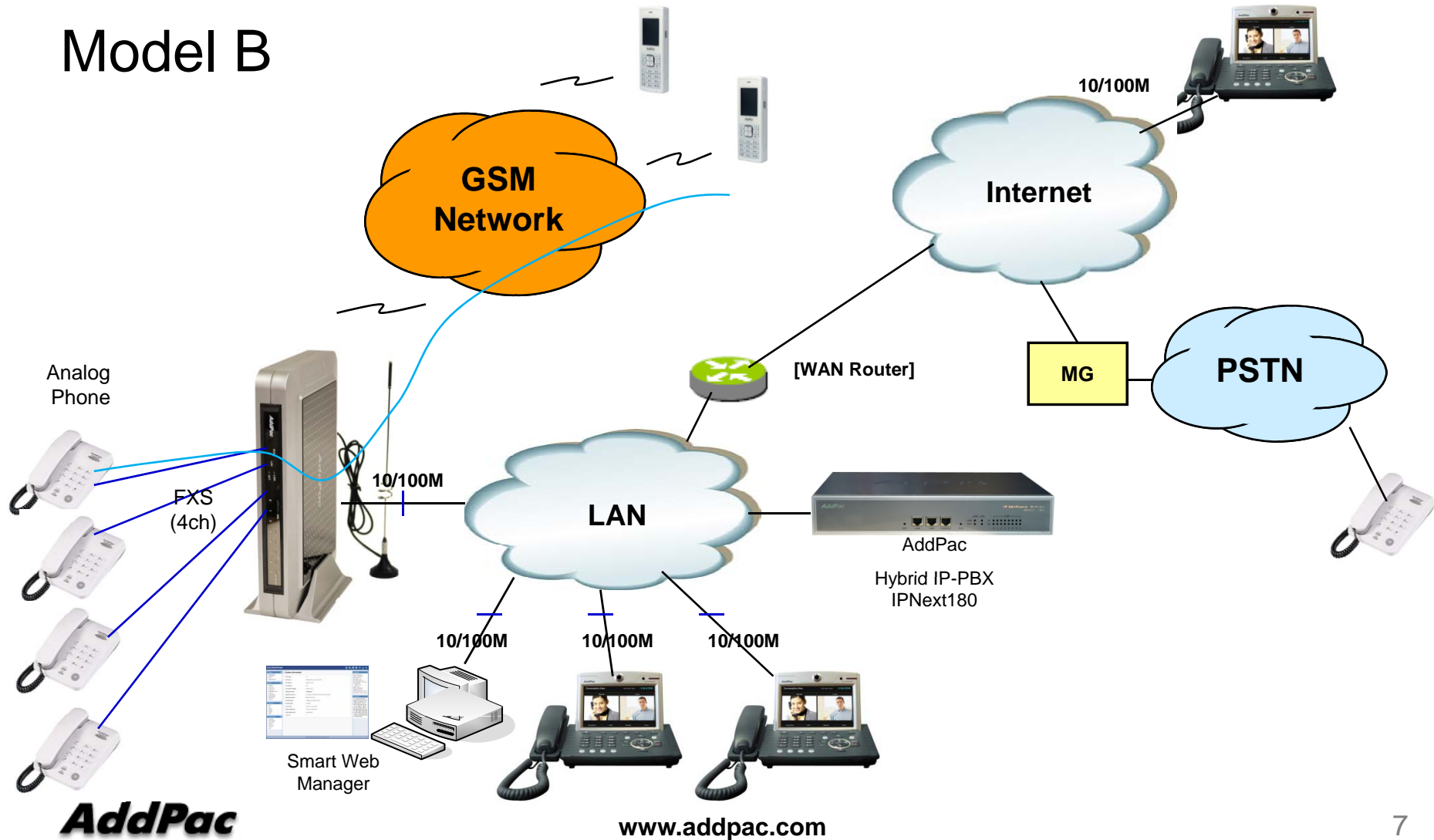


# GSM Gateway Service Diagram (AP-GS802)

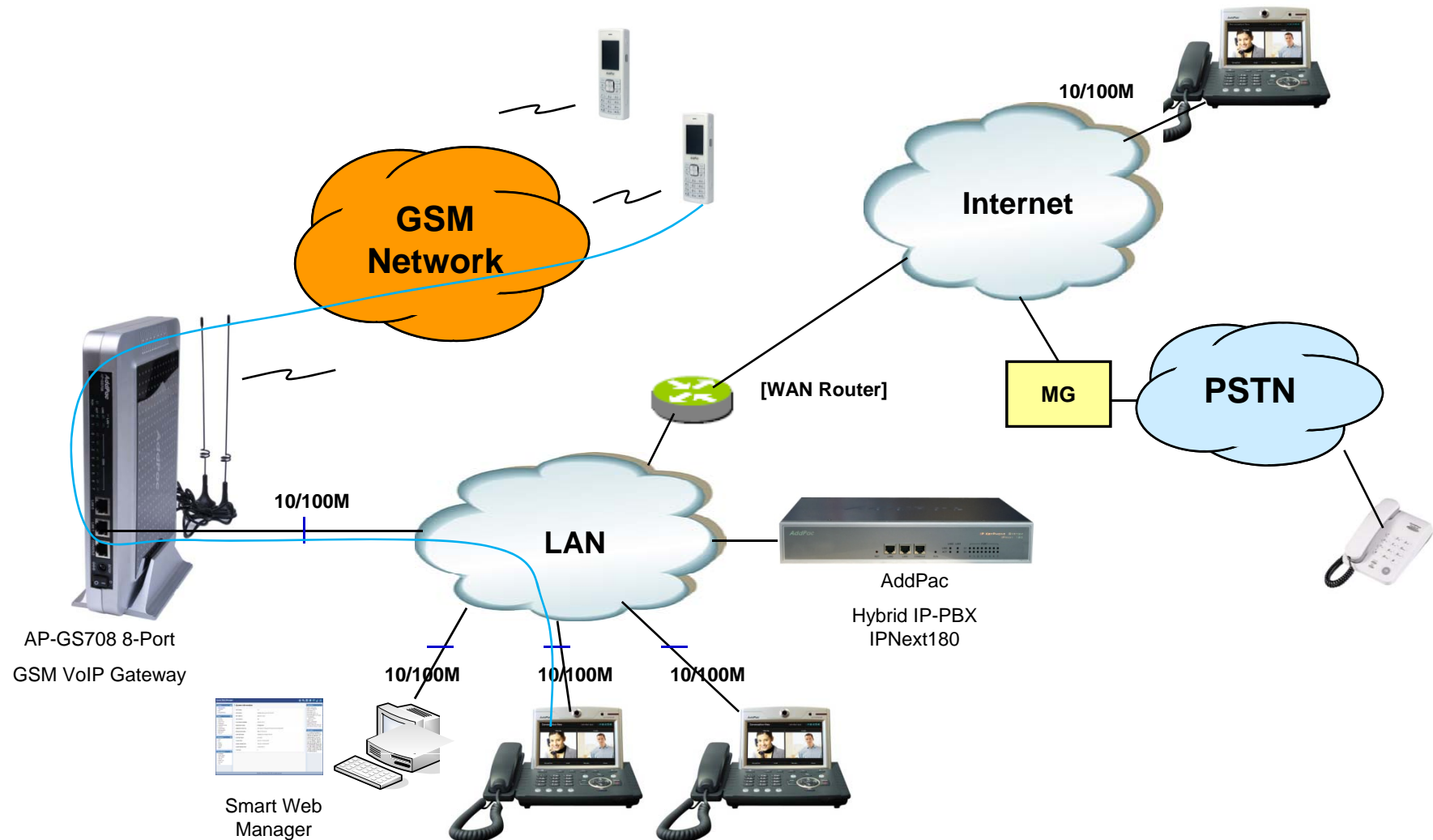


# GSM Gateway Service Diagram (AP-GS804B)

## Model B

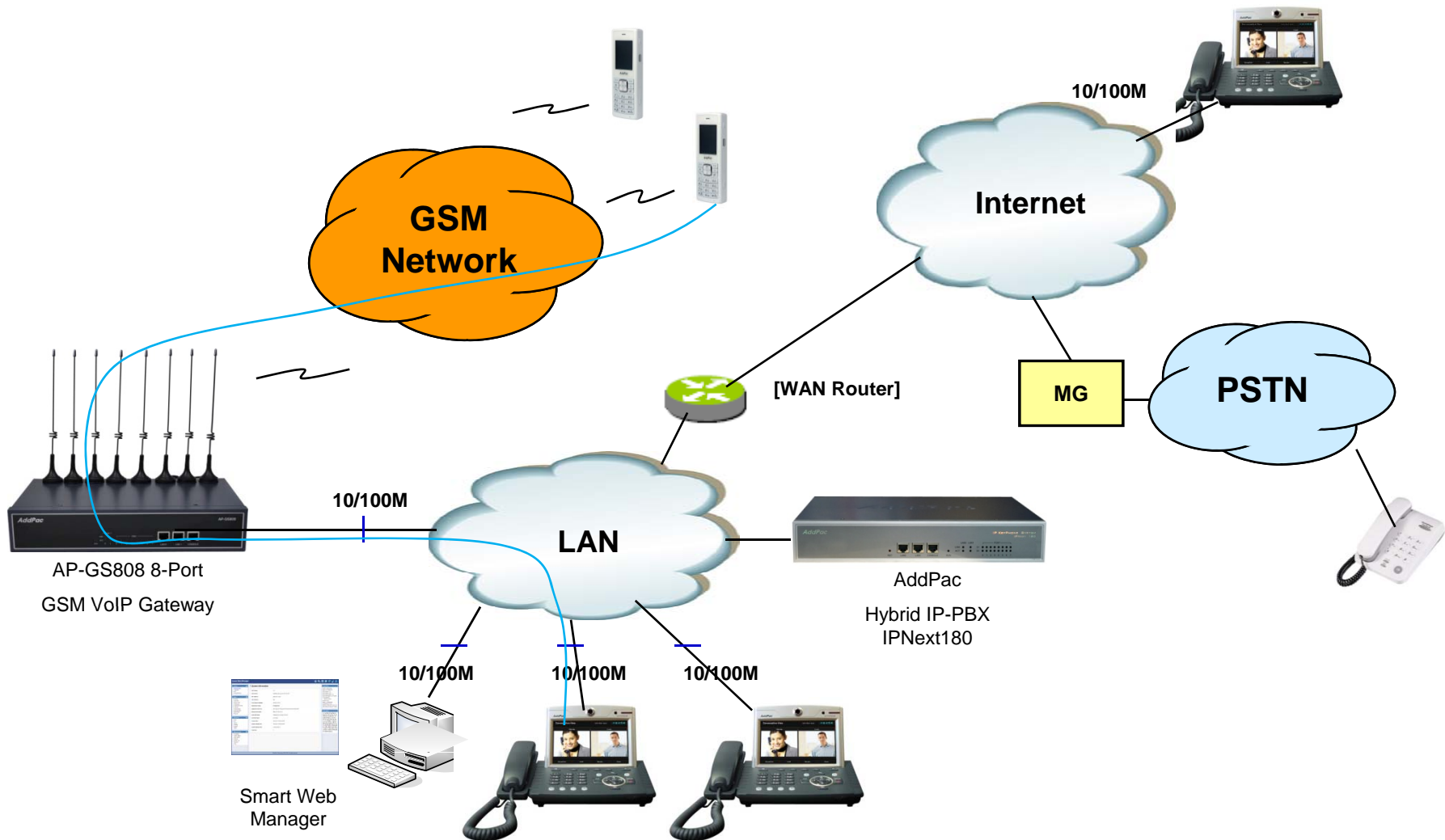


# GSM Gateway Service Diagram (AP-GS708)





# GSM Gateway Service Diagram (AP-GS808)







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# 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

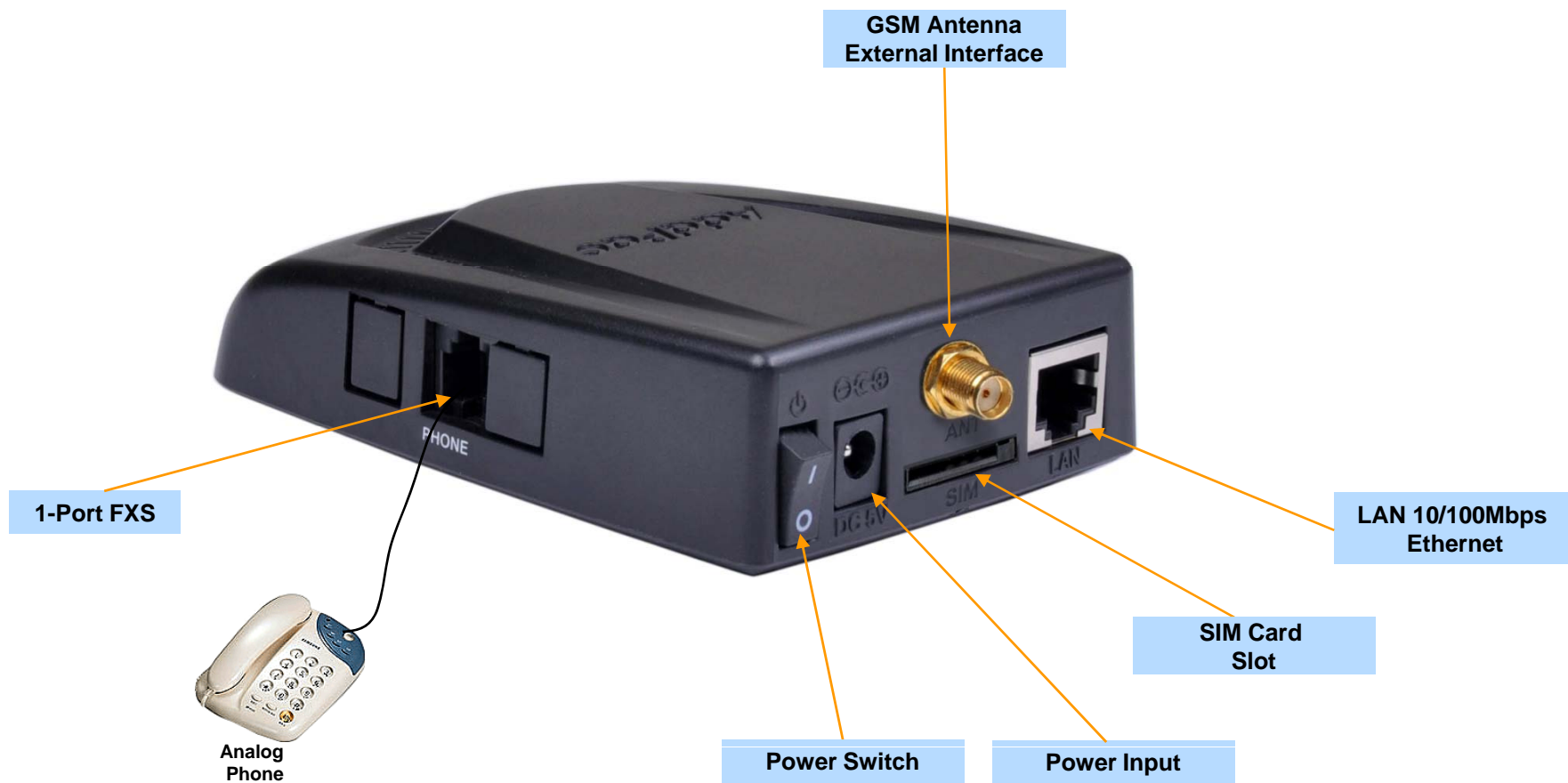
## AP-GS501 One(1) Port GSM SOHO Gateway

- 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

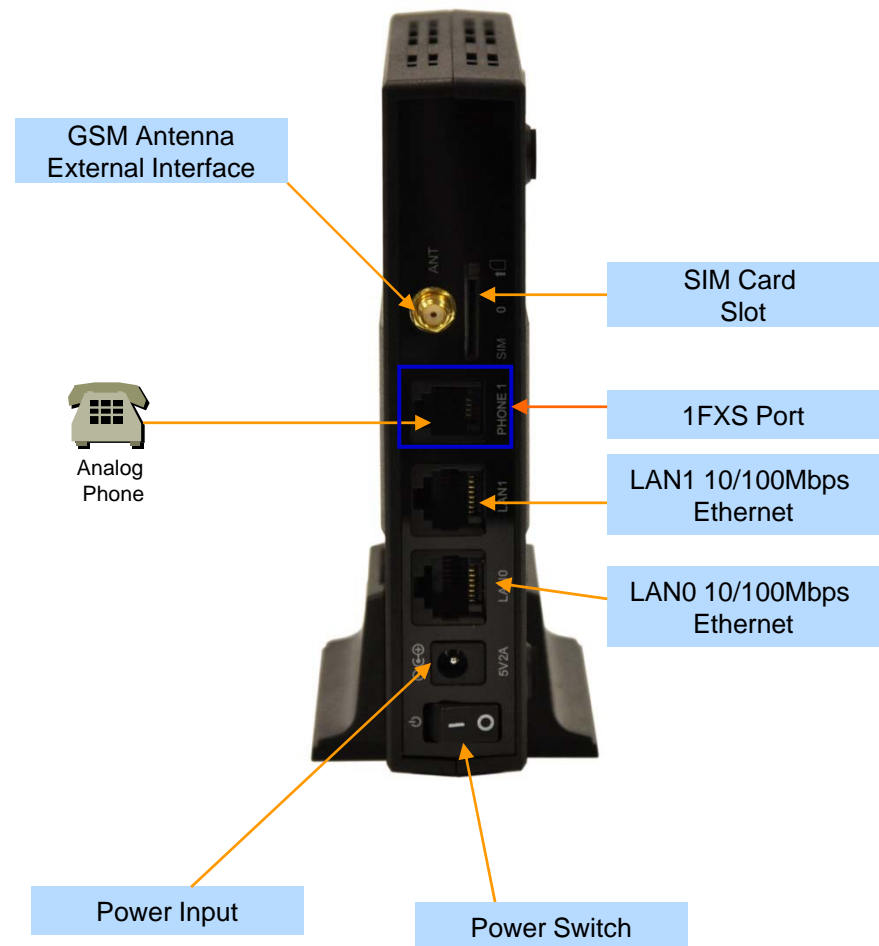
- RISC Microprocessor Computing Power
- 1-Port GSM Gateway
- 1-Port SIM Card Slot
- 1-Port GSM Antenna Interface
- VoIP Gateway Interface
  - AP-GS701 Model A: Basic Configuration
  - AP-GS701 Model B: One(1) FXS Port
- Network Interface for VoIP Direct Interface
  - Two(2) 10/100Mbps Fast Ethernet (RJ45)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

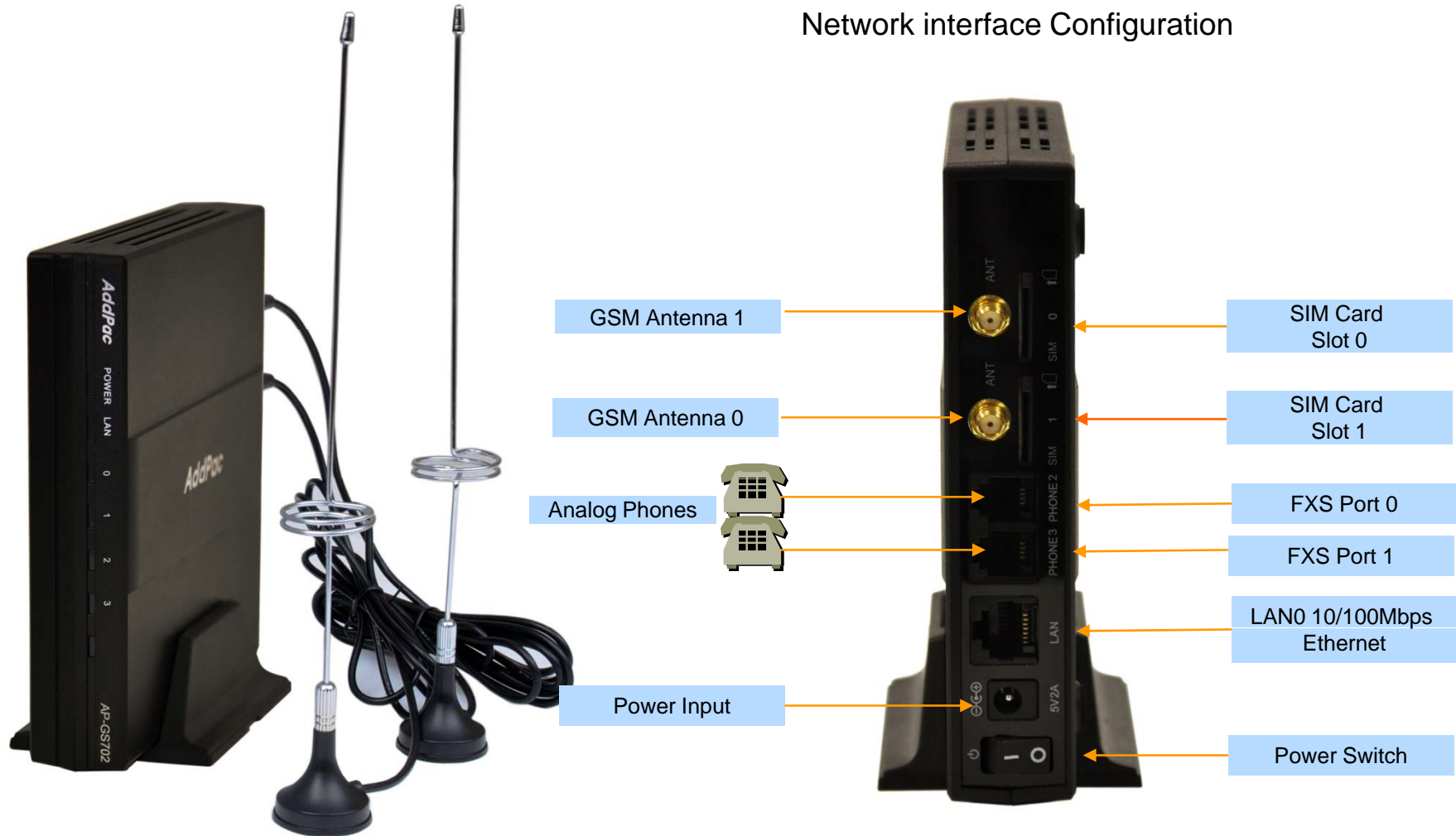
## AP-GS702 Two(2) Port GSM SOHO Gateway

- 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

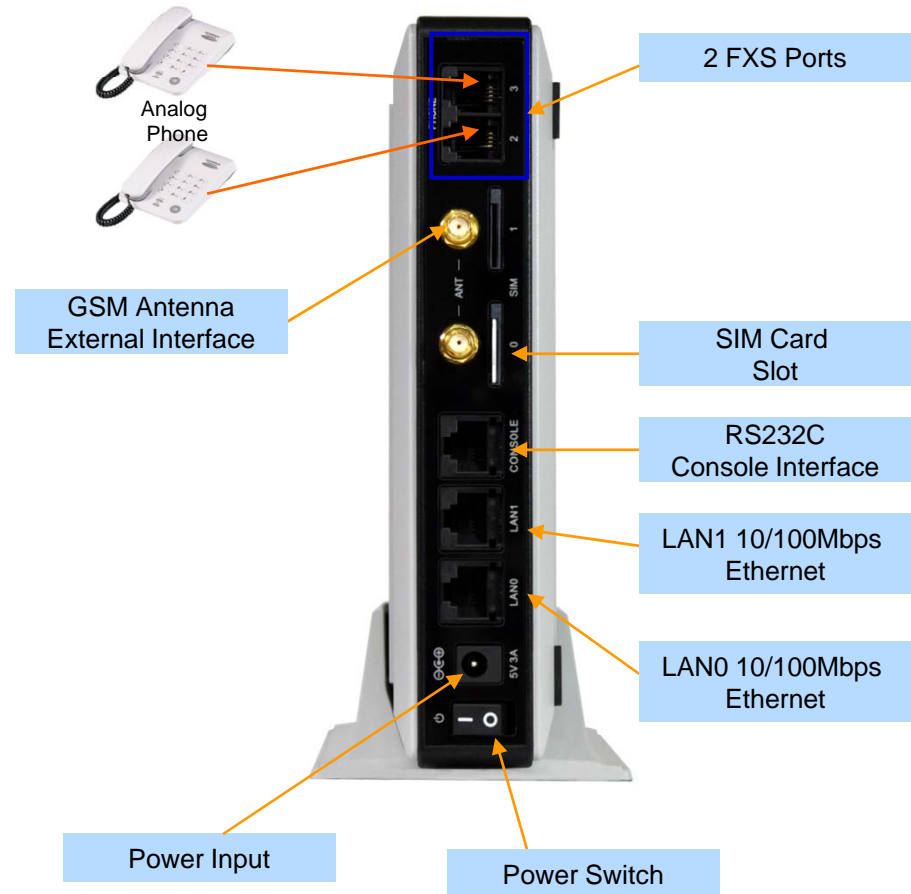
# Hardware Specification

## 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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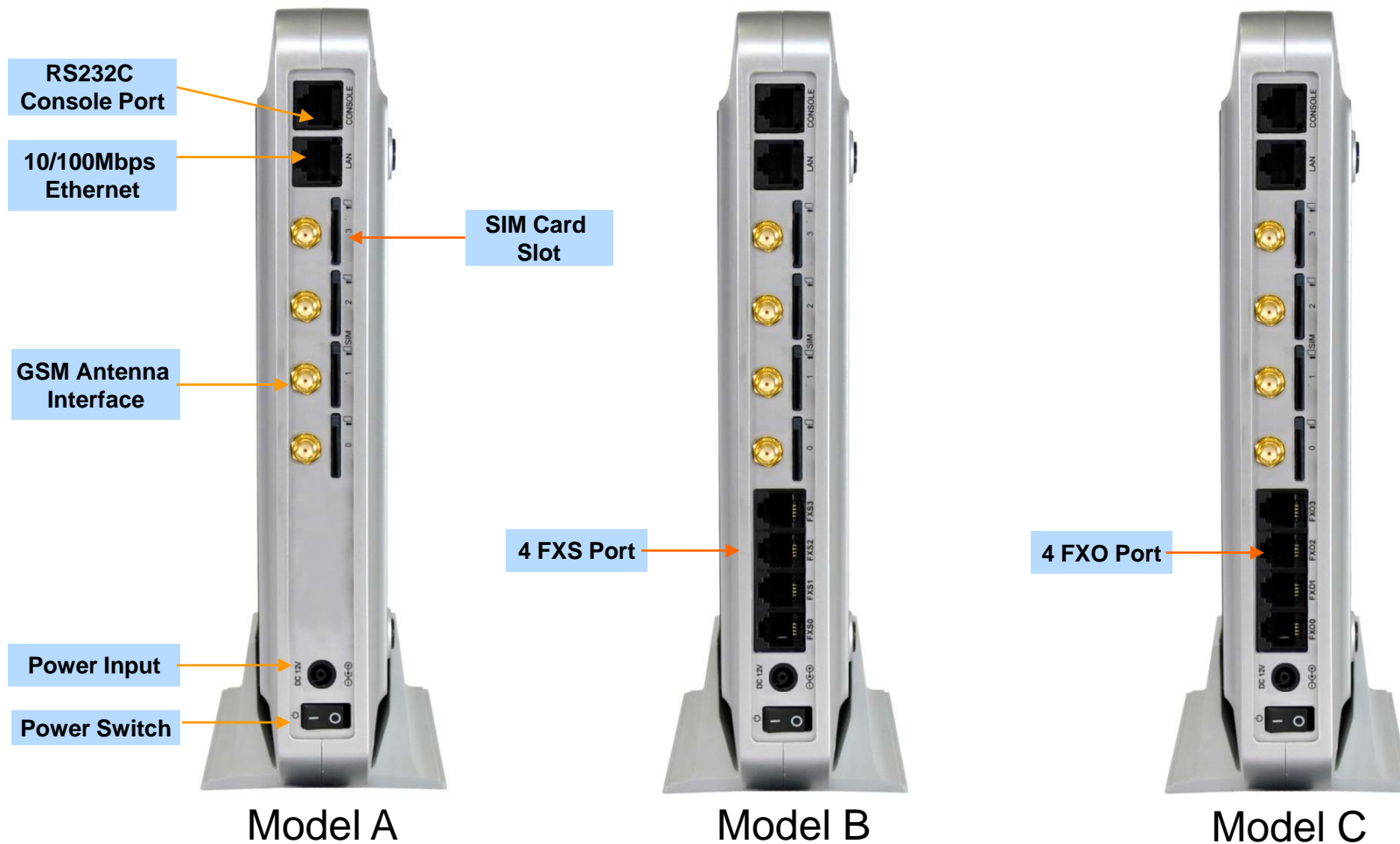
# Hardware Specification

## 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

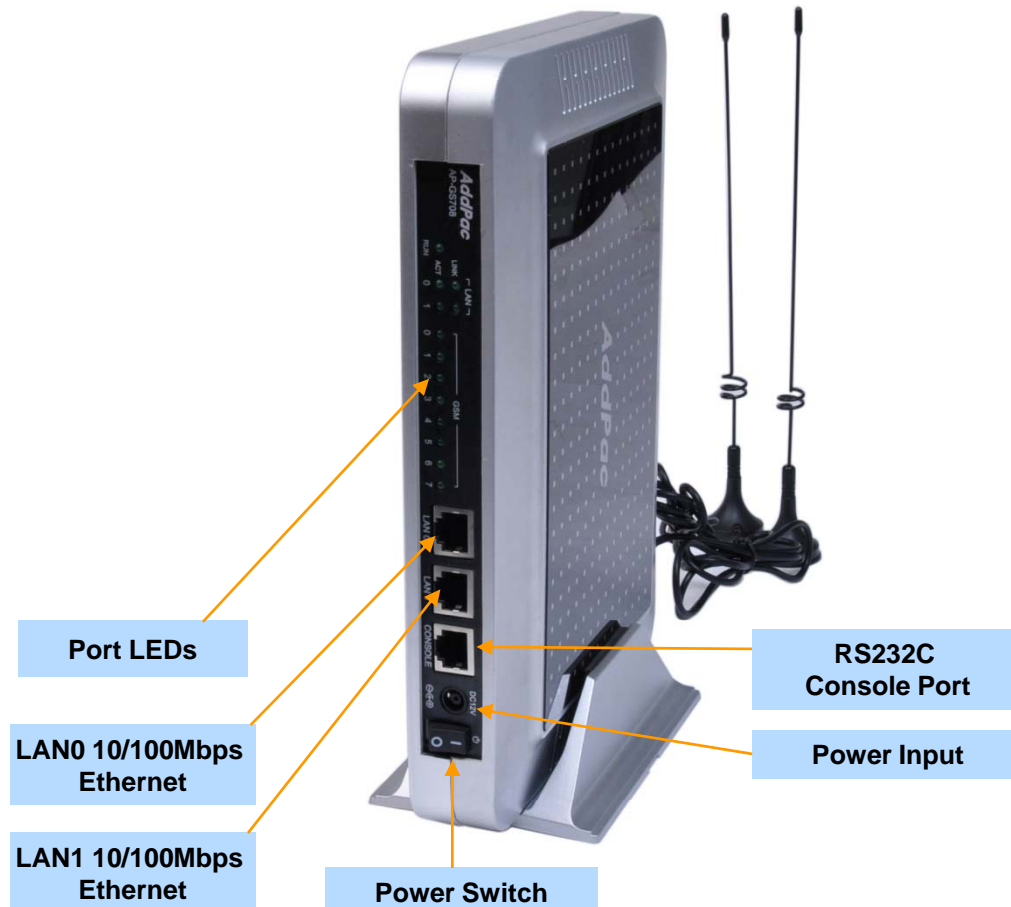
# Hardware Specification

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

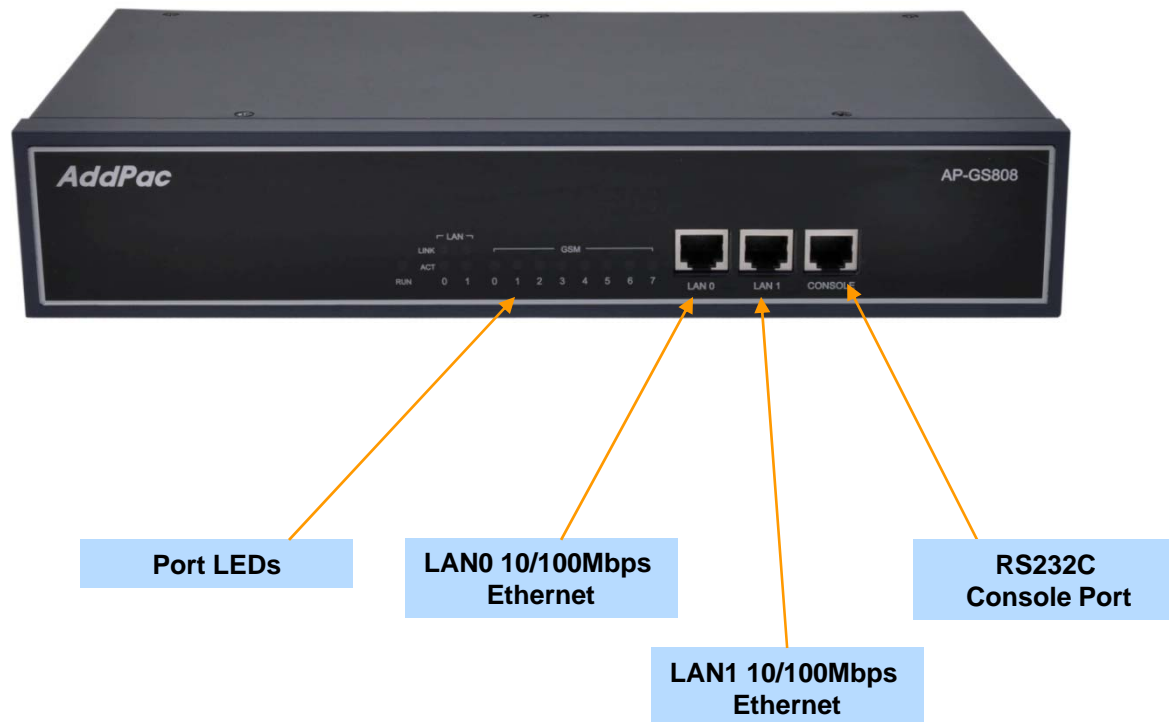
# Hardware Specification

AP-GS808 8-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## Front Side View



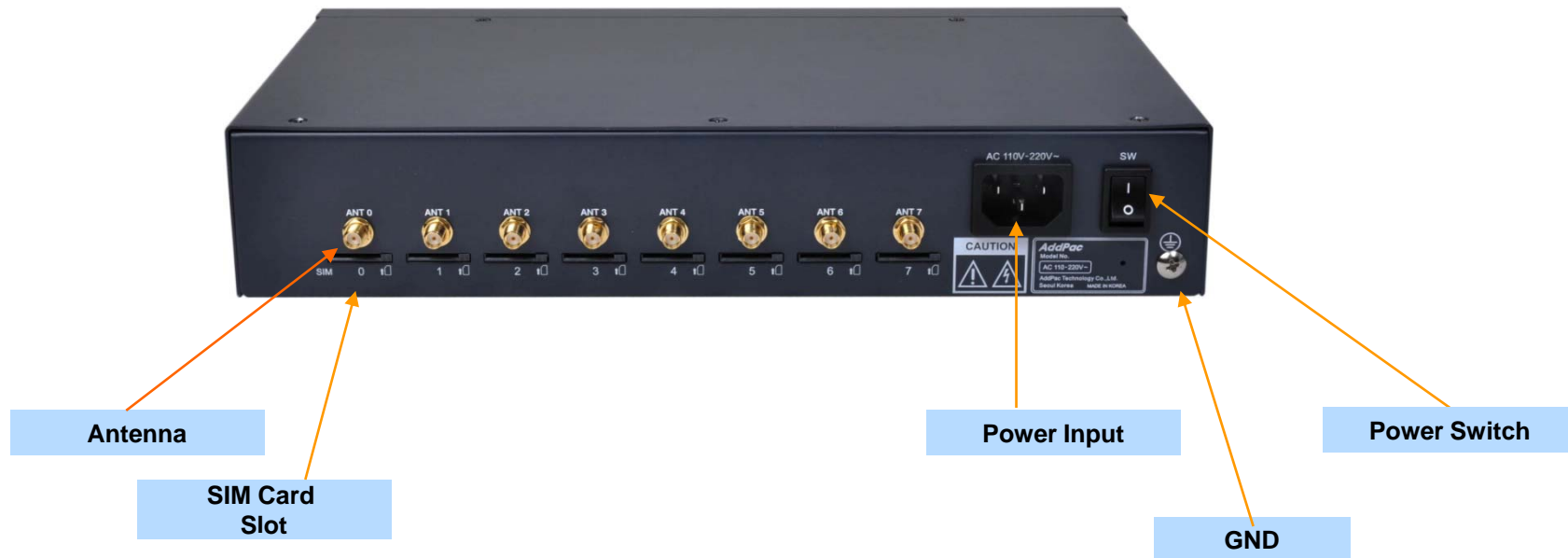
# Hardware Specification

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





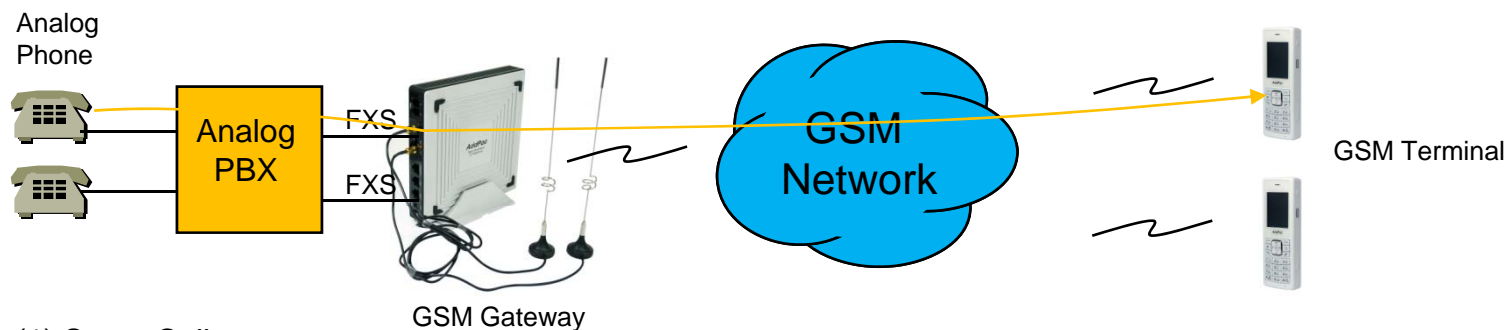


# GSM Gateway Function List

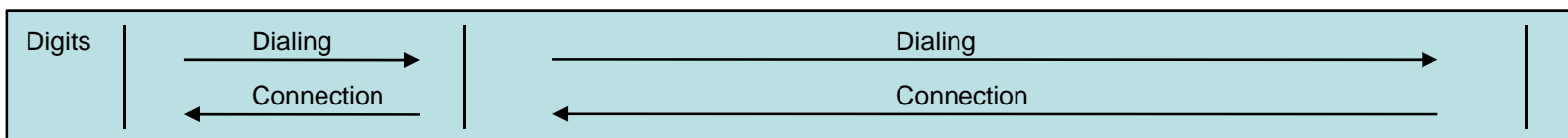
# Contents

- 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



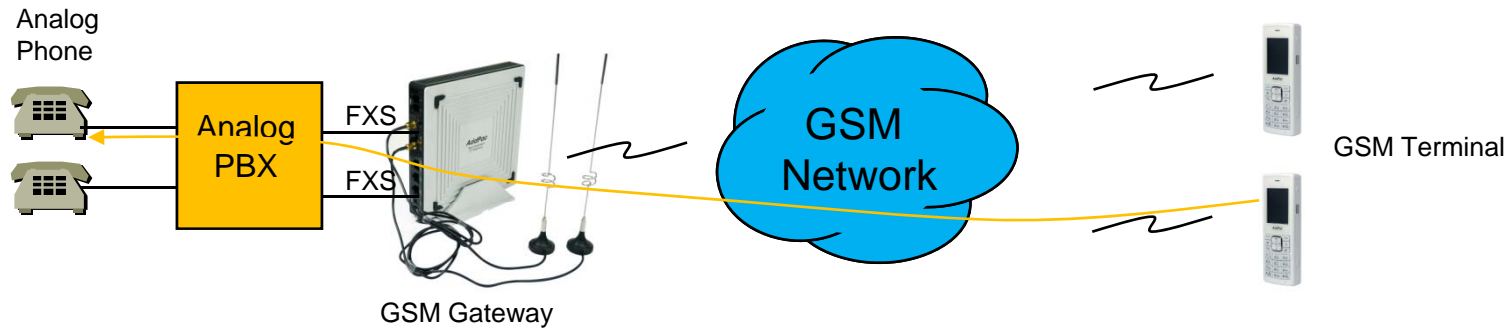
Outbound Call (1 Stage)

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

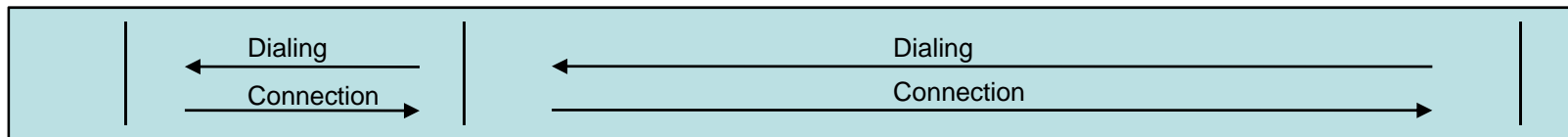
Outbound Call (2 Stage)

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

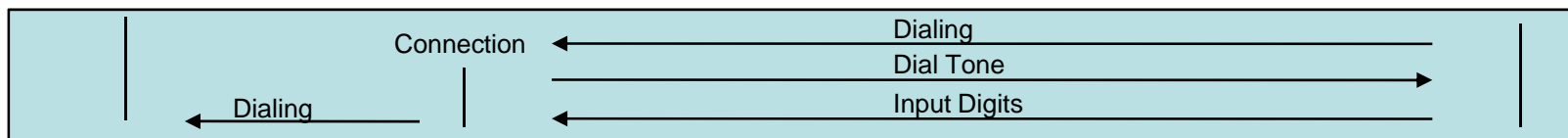
# GSM Inbound Call



## 1 Stage Call (Baby Call)



## 2 Stage Call



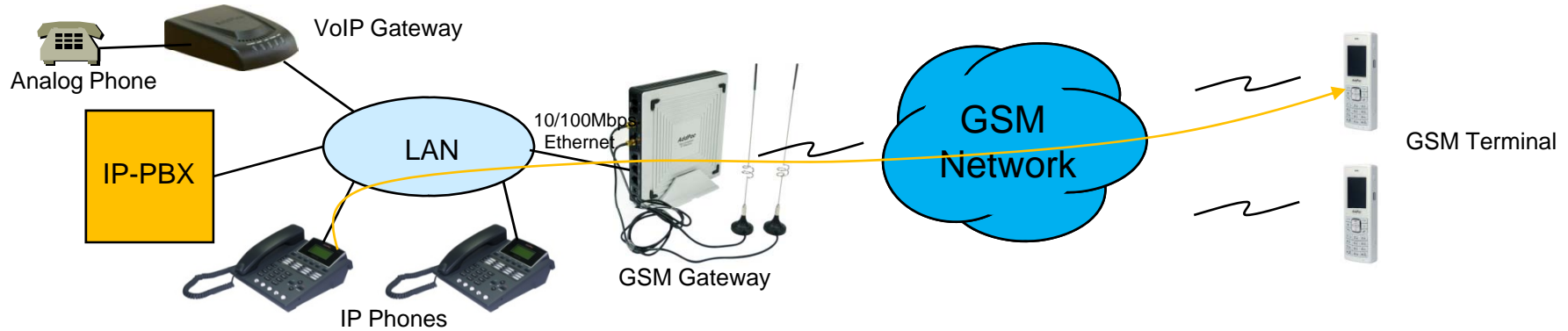
Inbound Call (1 Stage) – Baby Call

: Making call to analog phone connected to FXS directly

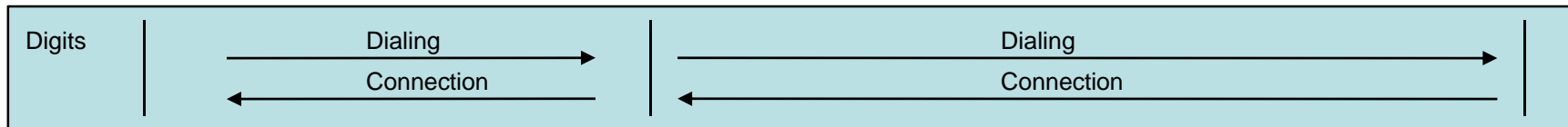
Inbound Call (2 Stage)

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

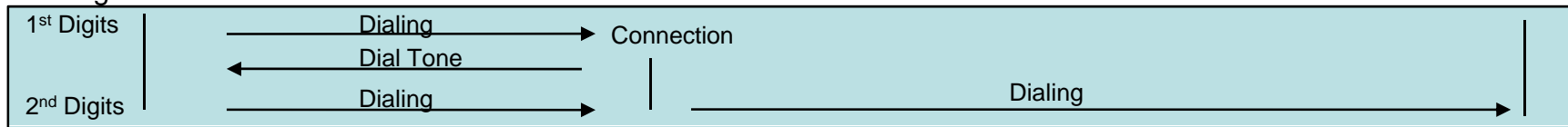
# VoIP to GSM Outbound Call



1 Stage Call



2 Stage Call



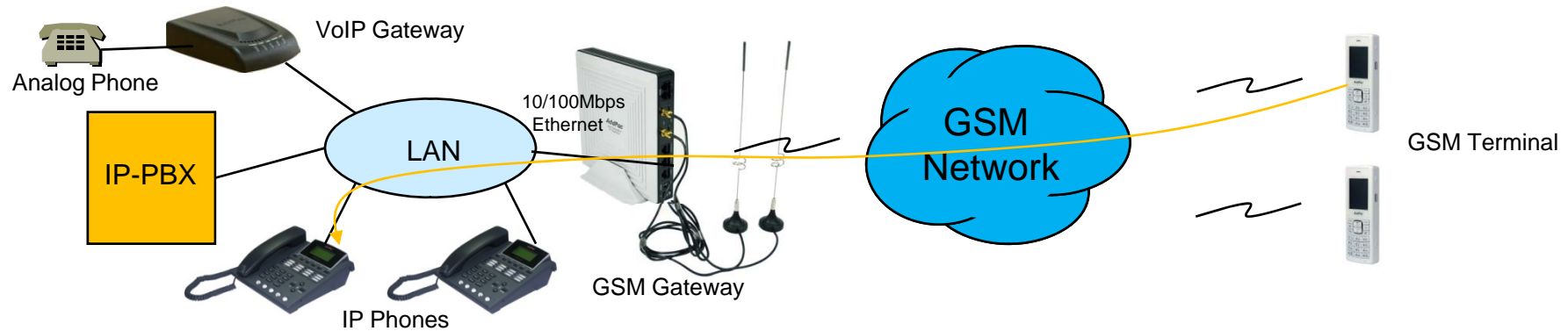
Outbound Call (1 Stage)

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

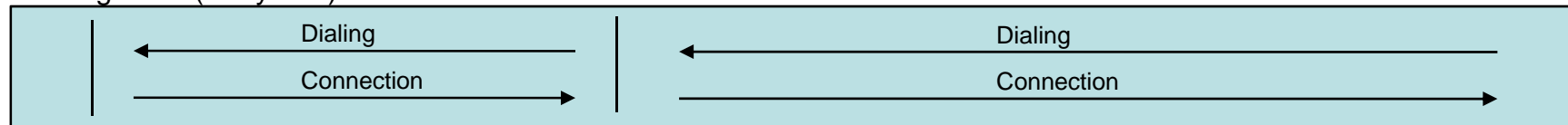
Outbound Call (2 Stage)

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

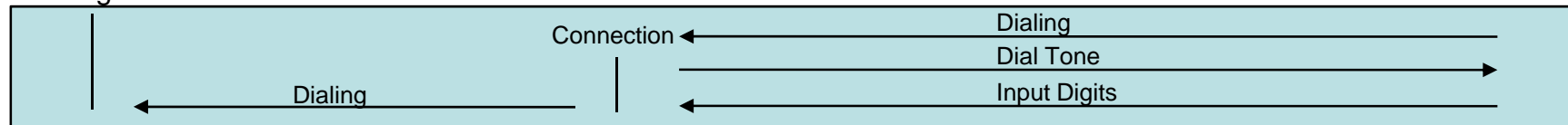
# VoIP to GSM Inbound Call



## 1 Stage Call (Baby Call)



## 2 Stage Call



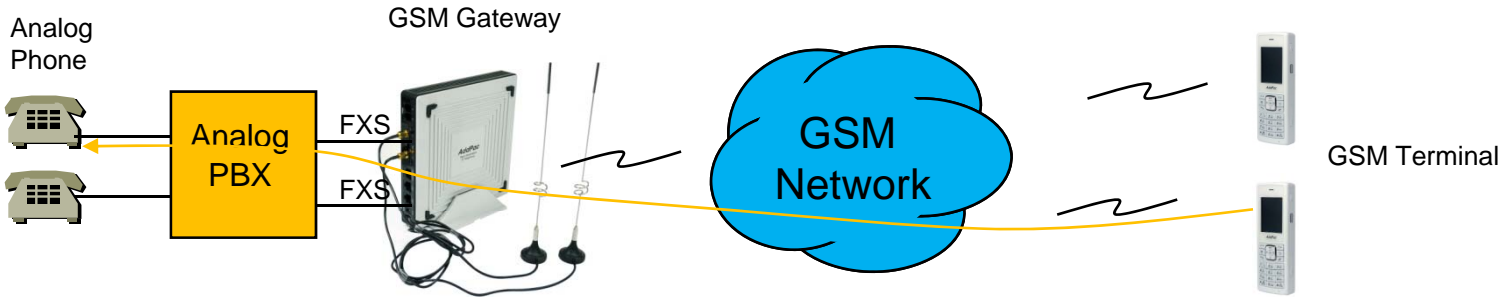
Inbound Call (1 Stage) – Baby Call

: Making call to IP phone in VoIP network directly.

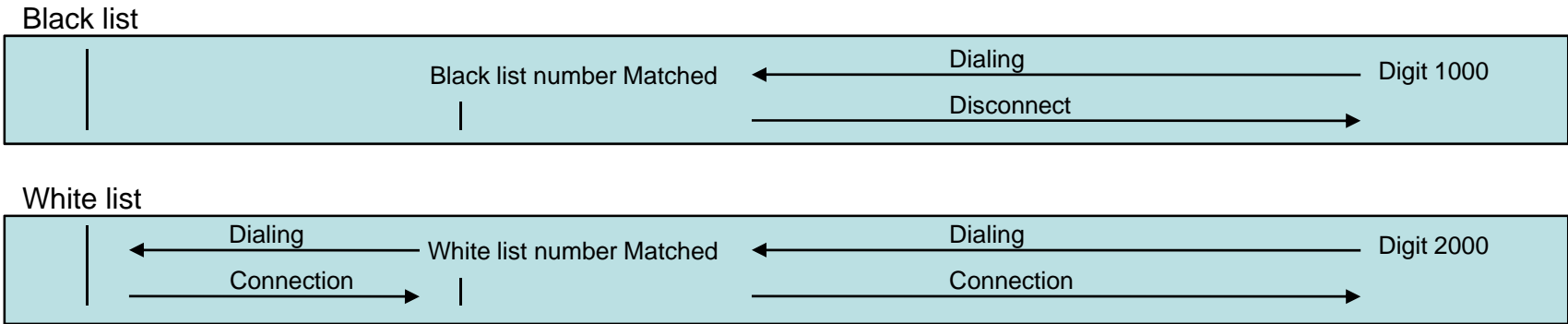
Inbound Call (2 Stage)

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

# GSM Inbound Call Black / White list



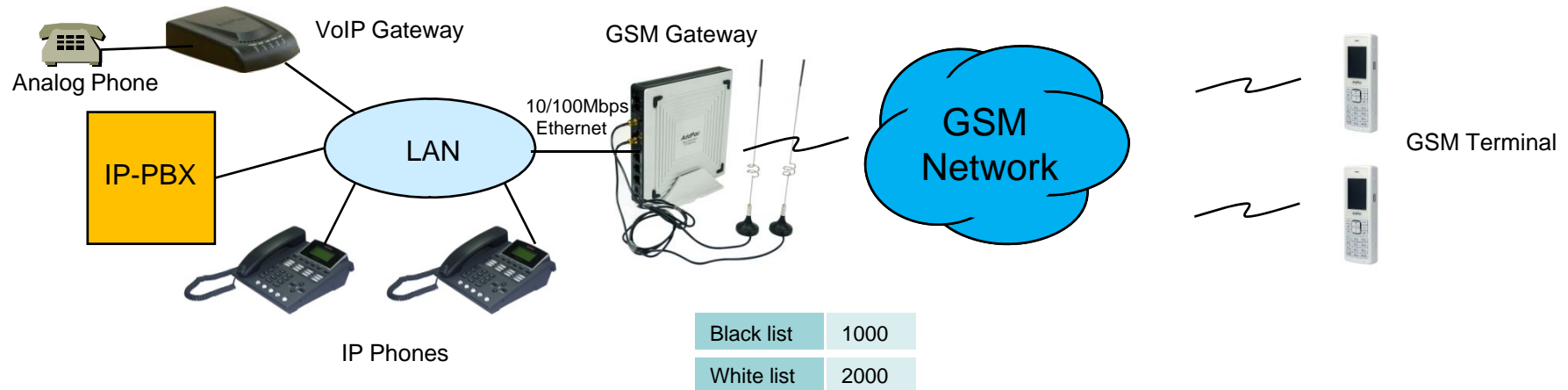
Black list	1000
White list	2000



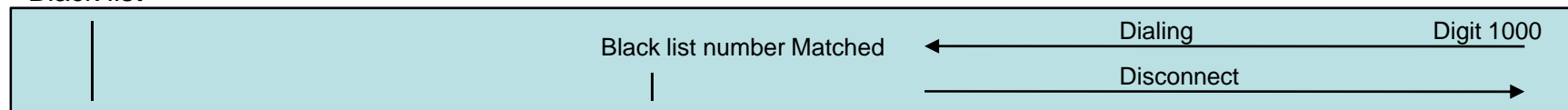
**Black list**  
: The number on black list is restricted to receive call.

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

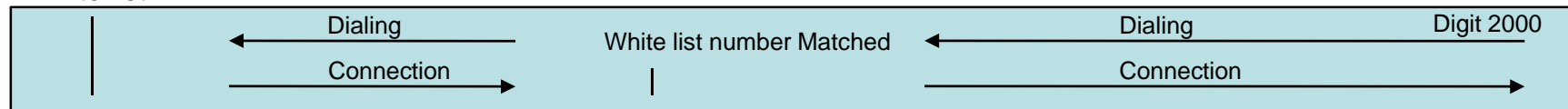
# VoIP to GSM Black / White list



Black list



White list



Black list

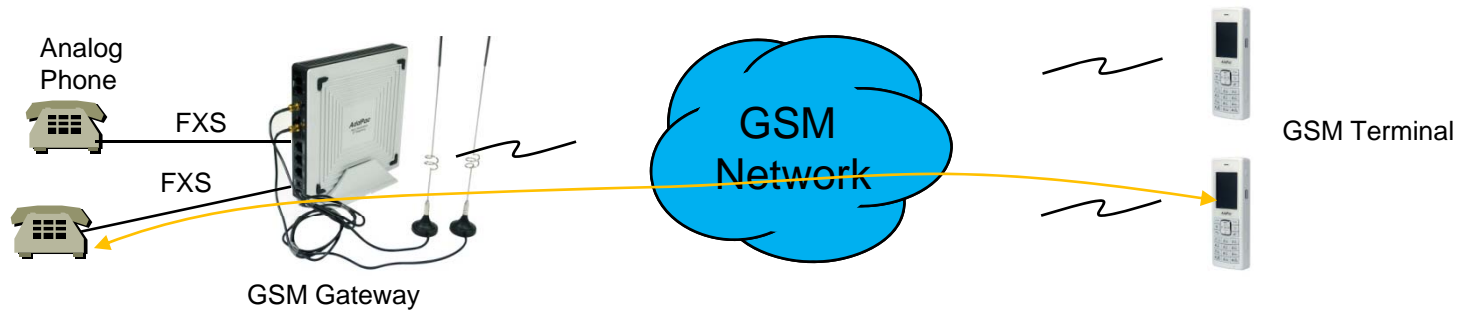
: The number on black list is restricted to receive call.

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)

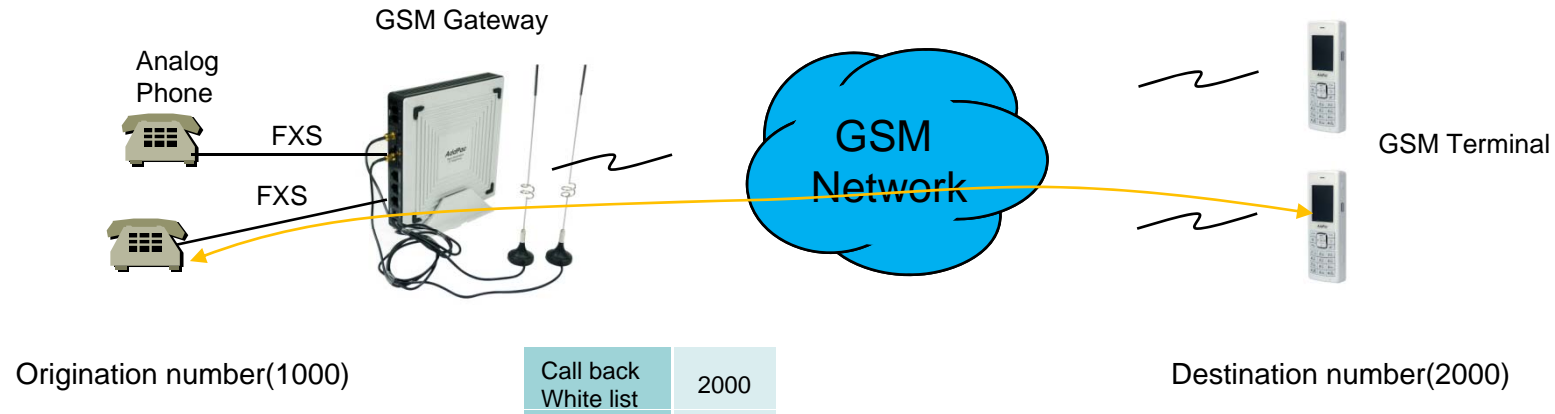
## WEB Callback Service



## 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



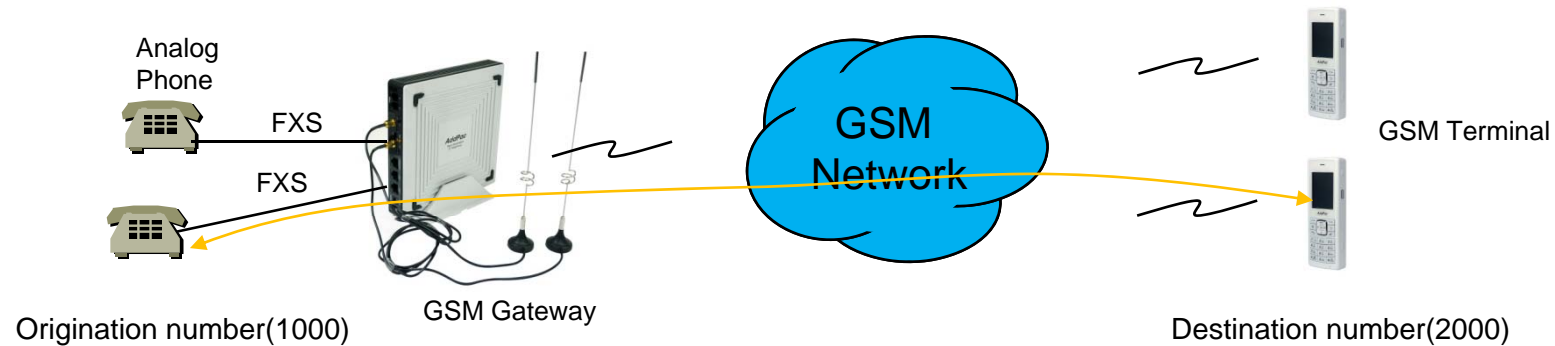
## GSM Callback Service



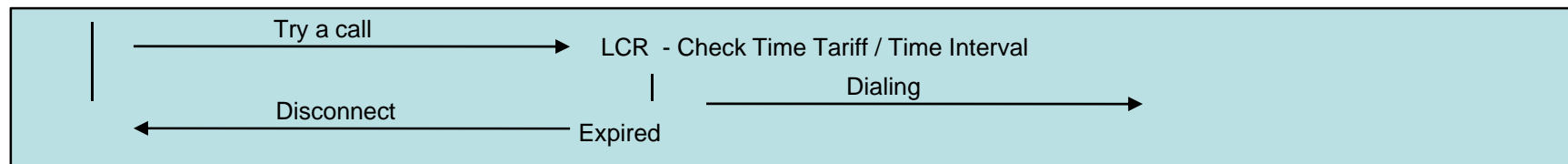
## GSM Callback Service

: 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

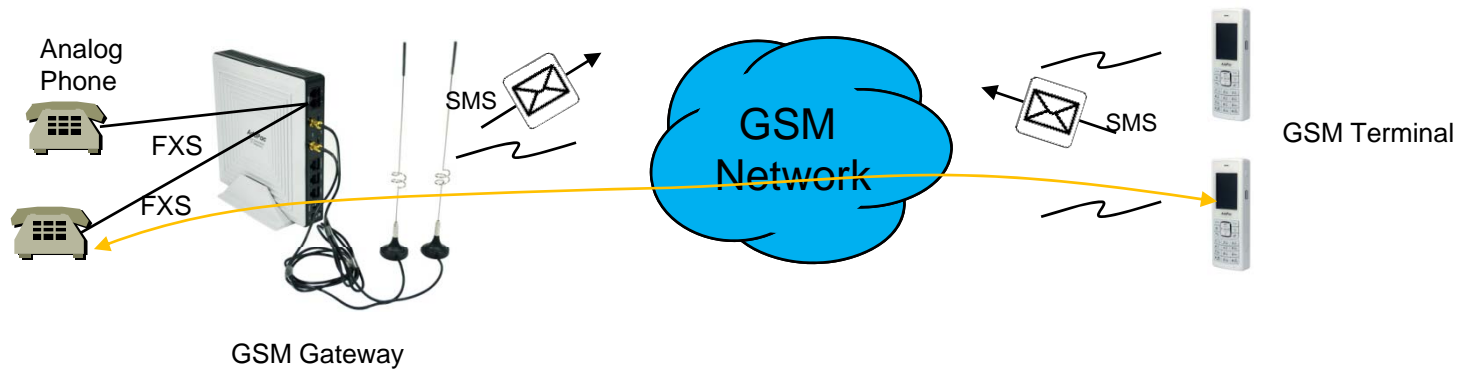
### GSM LCR Time Tariff

: 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



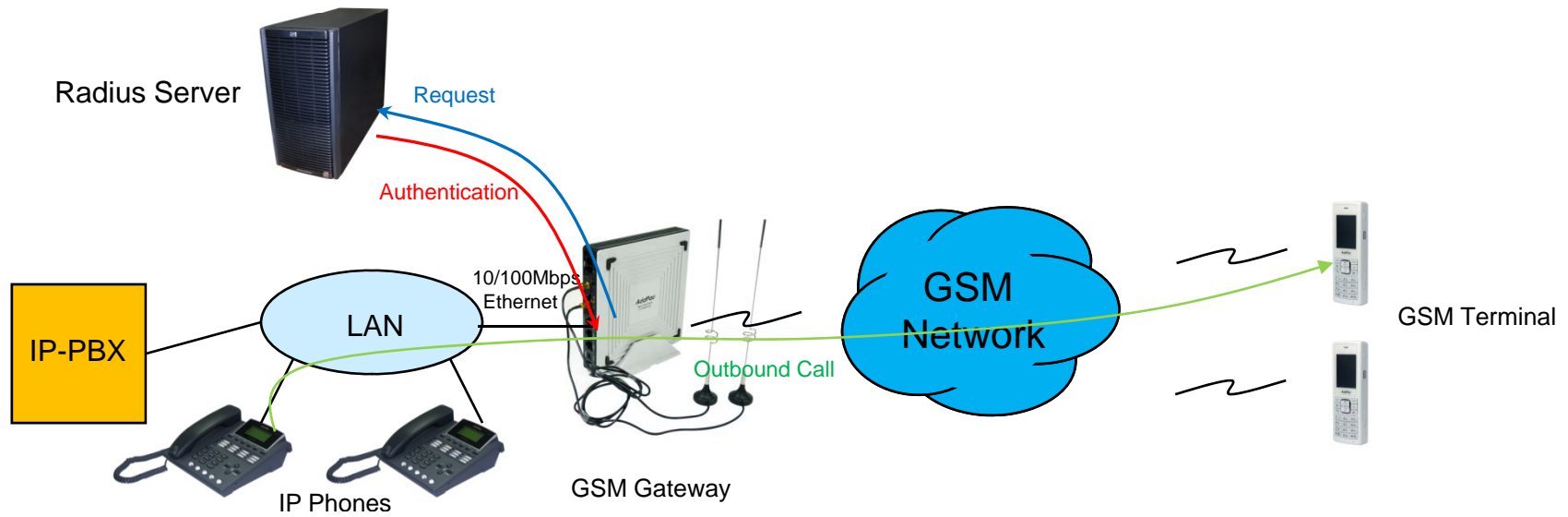
## GSM Messaging Service

- : 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)



# System – Network Setup

**Host Name**  
Create a representative name for the site to be installed

**Static IP**  
This is static IP mode. Specify the addressed IP from the service provider

**PPPoE**  
This is ADSL mode. This mode is used for addressing IP though authentication from the modem. At this time, the modem must be configured in a way that the device can be authenticated.

**DHCP**  
This is dynamic IP mode which is set at default. The IP can be addressed from the external DHCP server.

**VLAN**  
Configure VLAN mode and ID.

**WAN Link**  
Controls and recognizes WAN port. Specify the connection speed of WAN port. connection automatically.

**MAC**  
Change MAC address of WAN interface. Without address entry, use the basic MAC Address.

**Network Setup**

Hostname: GS1002

Static IP

IP Address: 172.16.9.16 (A.B.C.D)

Network Mask: 255.255.0.0 (A.B.C.D)

Default Router: 172.16.1.1 (A.B.C.D)

DNS Server: [ ] Primary DNS Server

[ ] Secondary DNS Server

PPPoE(ADSL)

Username: [ ]

Password: [ ]

DHCP

VLAN

ID: [0]

Auto

Speed:  100  10

Duplex:  full  half

Manual

MAC(Hardware) Address: [ ]-[ ]-[ ]-[ ]-[ ]-[ ]

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 : Registered  
CurrentCalls : 0 Call  
Network : Static 172.16.9.16  
Mac Address : 0002.a400.0000  
Unread Message:  
P0:0(0)  
P0:1(0)

**Description**

This command sets up WAN port. The static IP address is to be assigned to the WAN port. The static IP address can be changed. MAC Address change can be used only when necessary. It is recommended to use the address created by the user not the address of the device

**UnRead SMS Messages**

# System - Language

**Smart Web Manager**  
www.addpac.com

**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

**Smart Web Manager**  
www.addpac.com

**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", and "Service".

- Mode:** Includes radio buttons for "None(Disable Tunneling, default)", "PPTP(Point-to-Point Tunneling Protocol)", and "Authentication" options: "None (No Authentication)", "PAP (PPP Authentication Protocol)", and "CHAP(Challenge Handshake Authentication Protocol)". It also has fields for "Username", "Password", "Phone Number", and "Define".
- Source:** A dropdown menu currently set to "FastEthernet0/0".
- Destination:** A text input field followed by "A.B.C.D (Tunnel End Point Address)".
- Service:** Includes radio buttons for "Voice and Data Use Tunnel Interface (default)", "Voice Use Tunnel Interface, Data Use Ethernet Interface", and "Data Use Tunnel Interface, Voice Use Ethernet Interface".

Three yellow callout boxes on the left provide additional context:

- Tunneling Mode:** Configure Tunneling mode
- Tunneling source & destination:** Source LAN port & destination IP
- Tunneling Service:** Service mode

The right sidebar contains "Information" and "Description" sections. The "Information" section lists device details like "AddPac Technology", "Model: GS1002\_G2", and "Smart Web Version: 0.4". The "Description" section explains that these settings are for using PPTP and advises contacting the manufacturer if needed.

# System - NTP

**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**  
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**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

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

**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

# 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

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

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

**Smart Web Manager**  
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**System**

- Network Setup
- Language

**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

**DTMF/CODEC**

Voice CODEC

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

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 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

**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

**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

**Prefix Table**  
Configure GSM Peer with translation rule.

# Basic – Static Route

**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 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="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**

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

# Basic – Hot Line

**Smart Web Manager**  
www.addpac.com

**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**  
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 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**  
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**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)

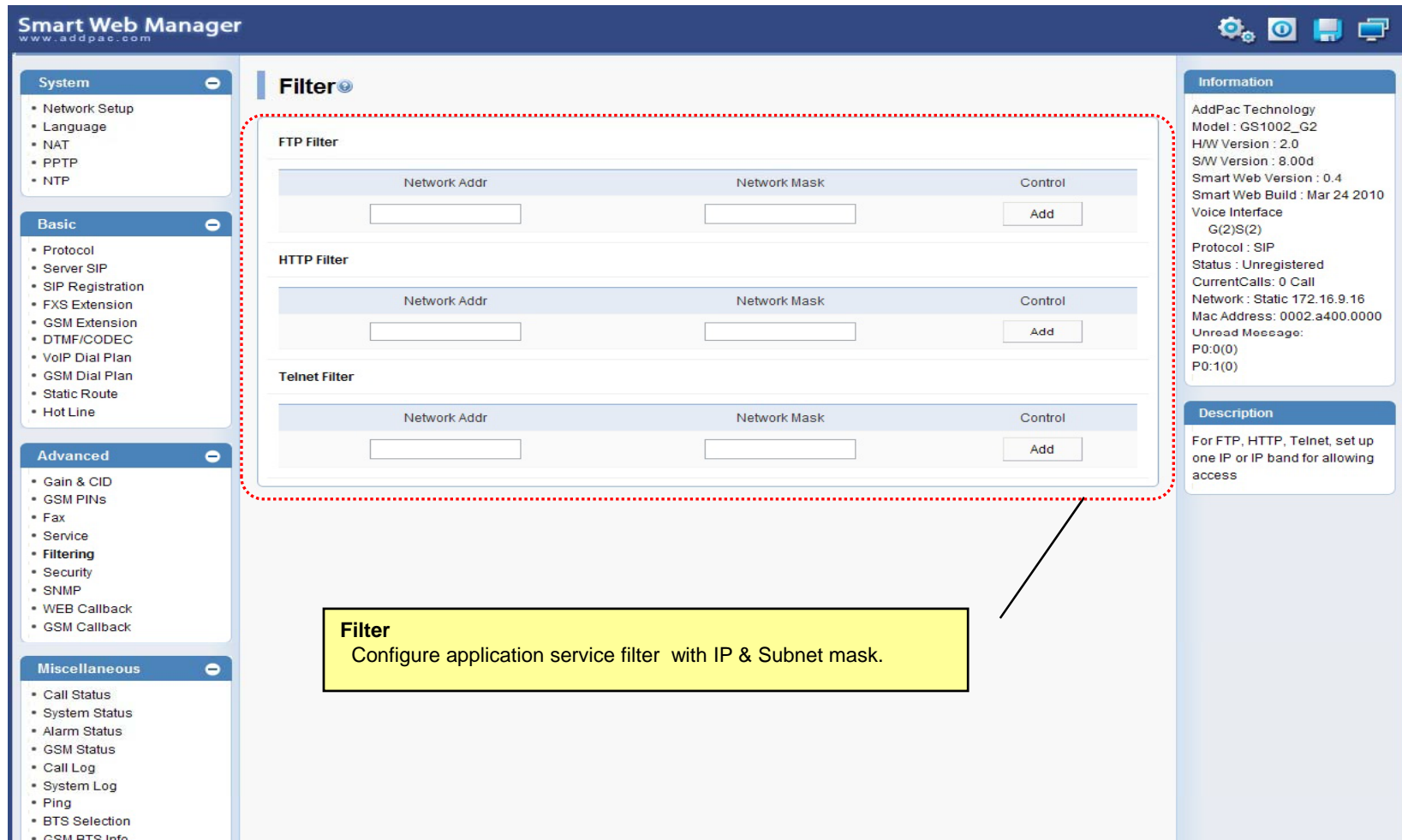
**Description**

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

**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

**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 Web Callback

**System**

- Network Setup
- Language
- NAT
- PPTP
- NTP

**Basic**

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

**Advanced**

- Gain & CID
- GSM PINs
- Fax
- Service

**WEB Callback**

- WEB Callback
- GSM Callback

### GSM WEB Callback

**Calling Number Whitelist**

index	DialPattern	Control
0	01023234444	<input type="checkbox"/>

0

Delete
Add

**WEB Callback**

Destination Numbers	Source Numbers	Control
8888	9999	<input type="button" value="Apply"/>

Call Fail

**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**

Execute GSM call service function and Configure call service whitelist

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

**WEB Callback**  
The employee working at the out of office can use web call agent.

**Status Viewer**  
Display real-time call status.

# 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.

**GSM Callback**

Calling Number Whitelist

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

Callback

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

**Callback**  
 The white list group is adapted to specific GSM port

**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

# 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	Port Group			
		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

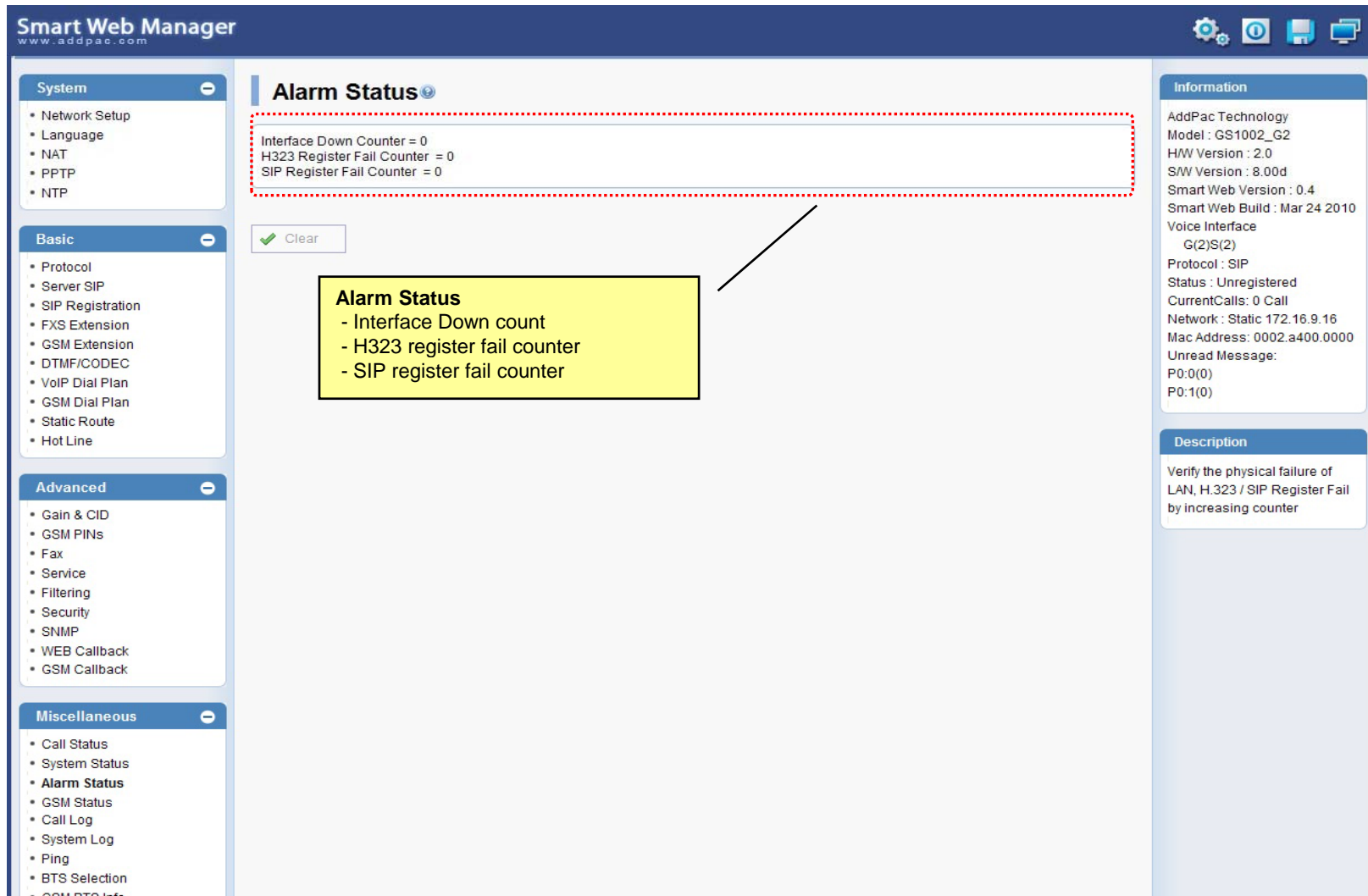
Voice Interface  
 G(2)S(2)  
 Protocol : SIP  
 Status : Unregistered  
 CurrentCalls: 0 Call  
 Network: Static 172.16.9.16  
 Mac Address: 0002.a400.0000  
 Unread Message:  
 P0:0(0)  
 P0:1(0)

**Description**

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



# Miscellaneous – Alarm Status



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

- Alarm Status** (indicated by a red dashed box in the original image):
  - Interface Down Counter = 0
  - H323 Register Fail Counter = 0
  - SIP Register Fail Counter = 0
- A 'Clear' button with a green checkmark icon.
- A yellow callout box with the text: **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 (GS1002\_G2), and various version numbers. 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)

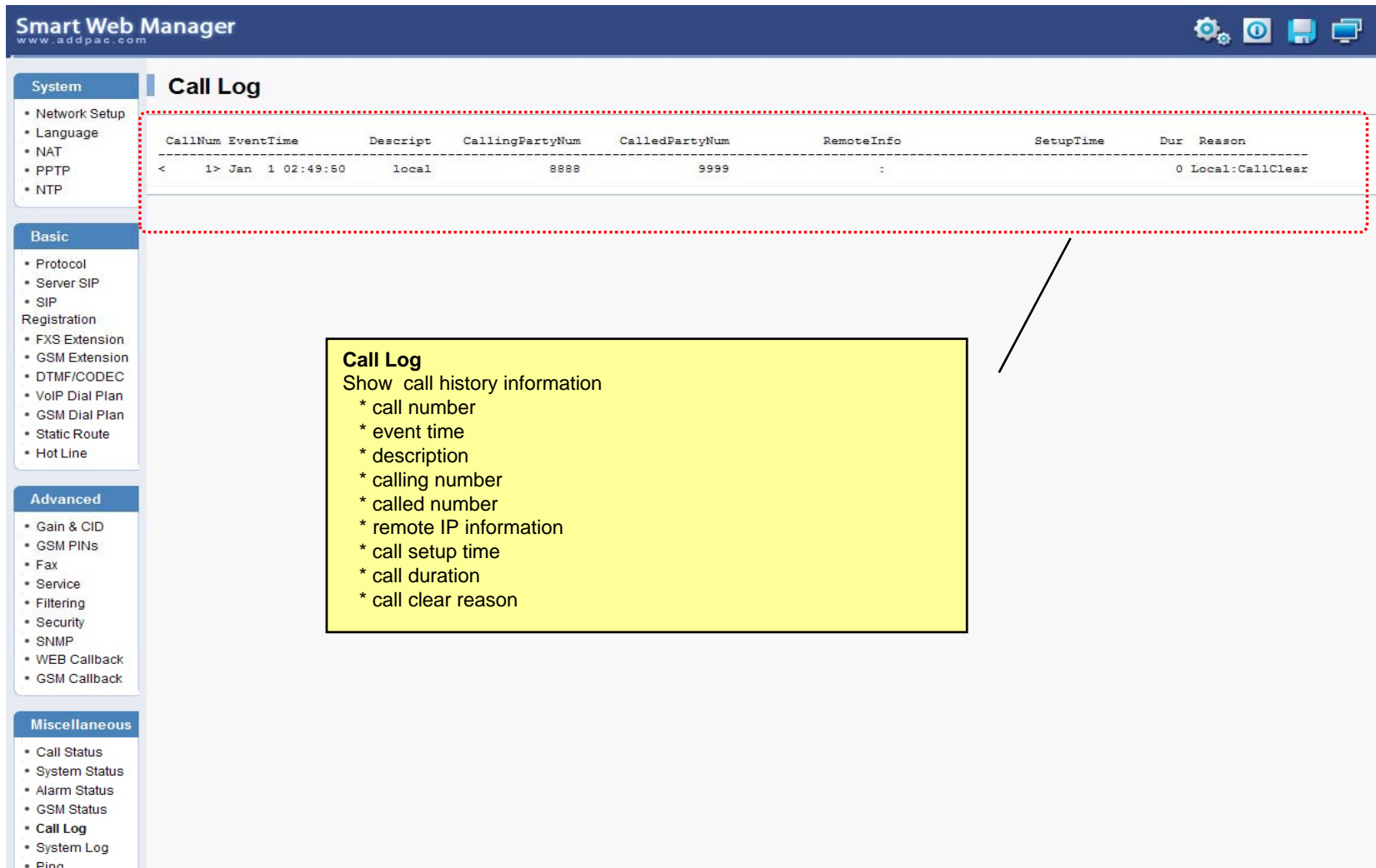
**Description**

Diaplay 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 left sidebar contains 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 a single data row. A yellow callout box provides a list of fields shown in the table.

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
0	<input type="text"/>	Add

**WhiteList**

Index	DialPattern	Control
0	2...	<input type="checkbox"/>
		Delete
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**  
www.addpac.com

**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

**Smart Web Manager**  
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**System**

- Network Setup
- Language
- NAT
- PPTP
- NTP

**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

**LCR Test**  
LCR simulator

**LCR Test**  
Show real time simulation status.

**Advanced**

- Gain & CID
- GSM PIN

# SMS – Inbox

**Smart Web Manager**  
www.addpac.com

**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 for sending an SMS message. The main content area is titled 'GSM SMS / New Message'. It contains a form with the following fields:

- Phone Number:
- Message:
- Port:

Below the form is a 'Send' button. A red dashed box highlights the form fields. A yellow callout box with a black border contains the text: 'New Message send a new message to the other GSM mobile phone.' An arrow points from this callout box to the form area. Above the form, it says 'Max size is 80 characters'.

The left sidebar contains a navigation menu with the following categories:

- 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

The right sidebar contains an 'Information' section with the following details:

- 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)

Below the information section is a 'Description' section with a text area.



# Thank you!

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