

# GSM based Voice Recording Solution



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**AddPac Technology**

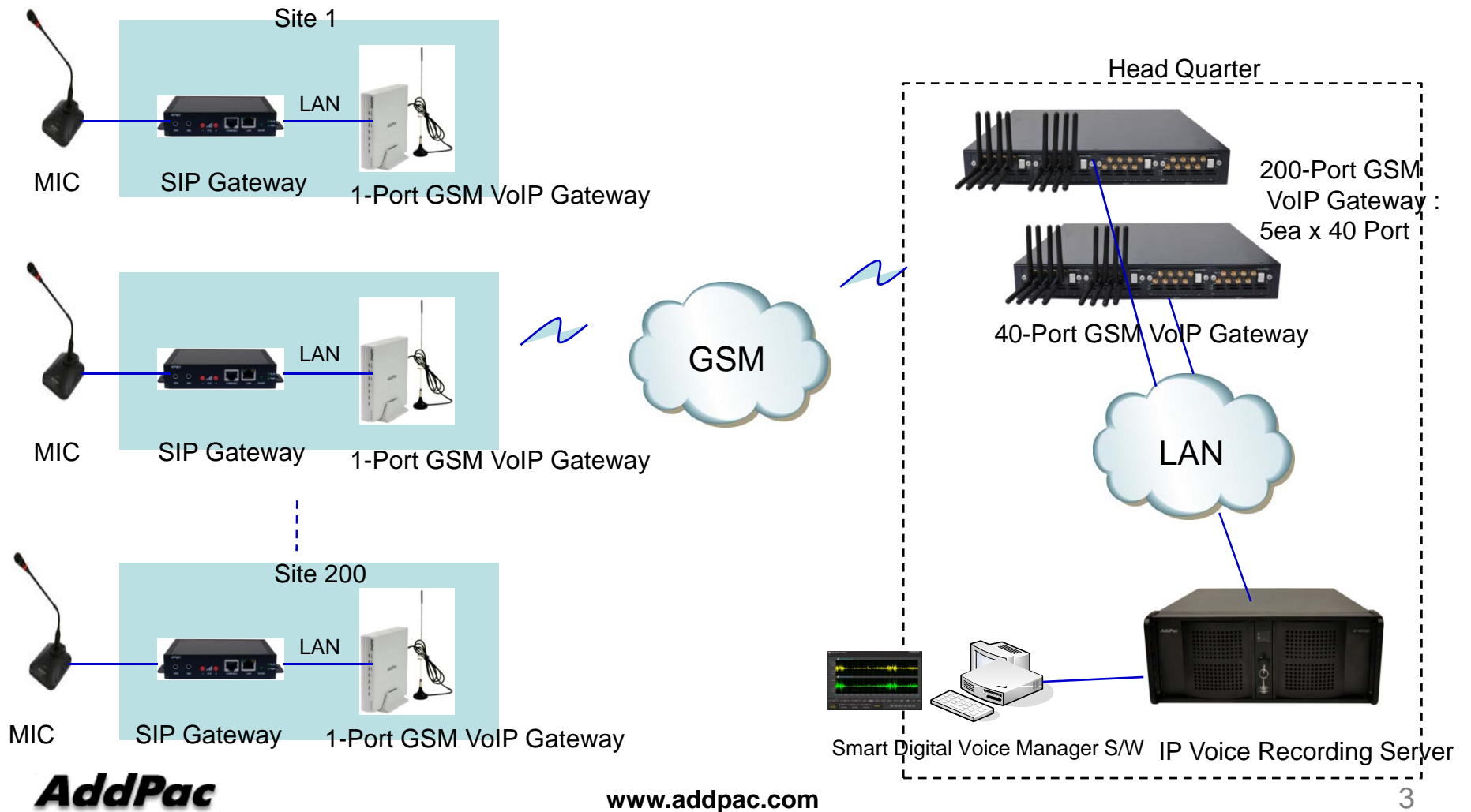
Sales and Marketing

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- Solution Table
- Solution Product Overview
  - AP-NR3000 IP Voice Recording Server
  - AP601 SIP Gateway with MIC Interface
  - AP-GV3200 40 Port GSM VoIP Gateway
  - AP-GS1001 GSM VoIP Gateway

# Network Diagram



# GSM based Voice Recording Solution

<p>IP Voice Recording Server AP-NR3000</p>	<p>SIP Gateway With MIC interface AP601</p>	<p>GSM VoIP Gateway For Center AP-GV3200</p>	<p>GSM VoIP Gateway For Site AP-GS1001</p>
			
<p>Linux based Voice Recording Server and Management Software.</p>	<p>Embedded Hardware based SIP Gateway with MIC Interface Voice Codec,G.711, G.726</p>	<p>40-Port GSM VoIP Gateway 8-Port GSM Module CPU Module Type H/W.</p>	<p>1-Port GSM VoIP Gateway SIP Protocol Field Proven</p>



# AP-NR3000 IP Voice Recording Server

# Product Overview

## AP-NR3000 Network Voice Recoding Server

- IP based Network Voice Recording Server
- Linux Operating System
- Powerful Management and User Friendly Features
- High-performance Voice Recoding Service
- External Media Gateway(ex:AP-MG3000) Interworking Support
- Firmware Upgradeable Architecture
- One(1) 10/100/1000Mbps Gigabit Ethernet Interface
- One(1) DVD Writer for Audio File Backup
- Up to Four(4) 3.5Inch SATA Hard Disk Interface Support
- USB Interface Support (Front Side, Back Side)

# Hardware Specification

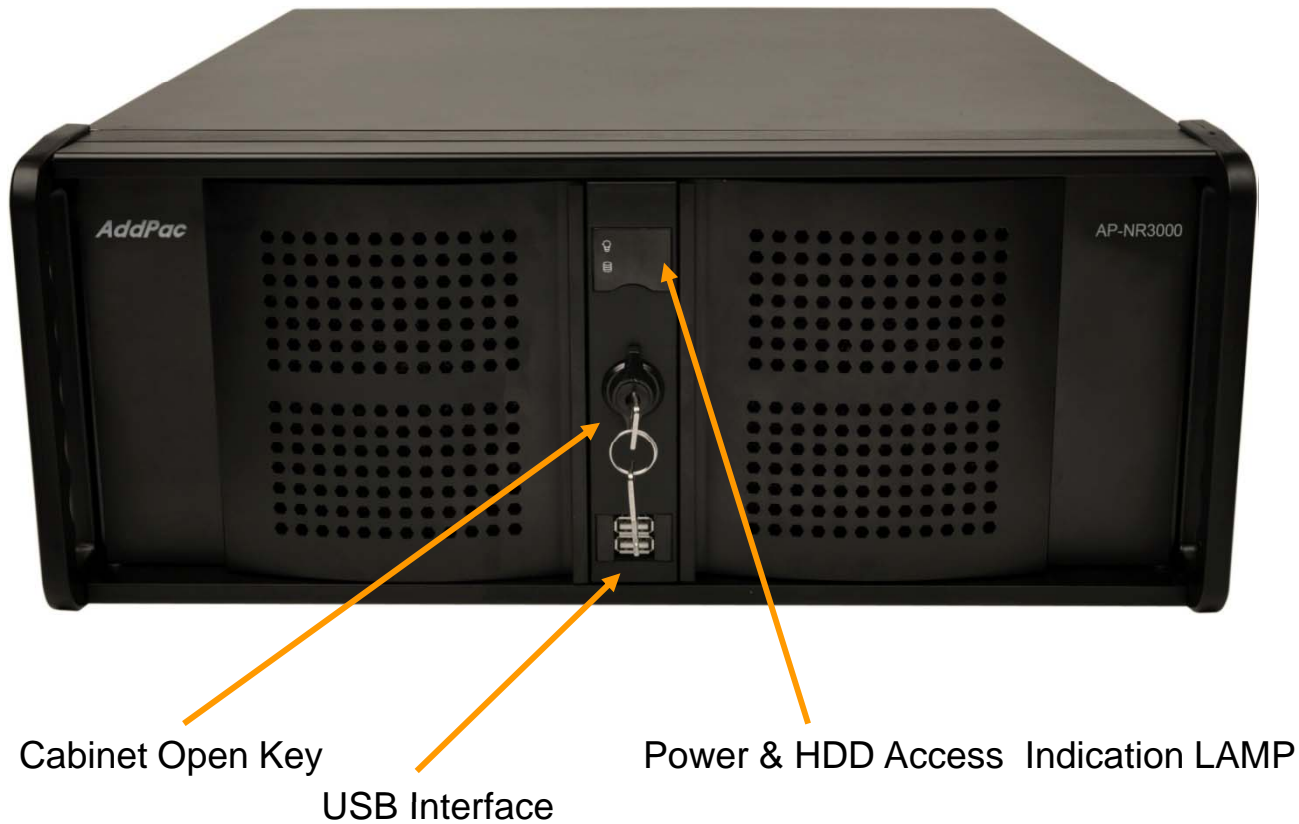
## AP-N3000 Network Voice Recoding Server

- High Performance Computing Power
- Main Chassis
  - Network Interface
    - One(1) 10/100/1000Mbps Fast Ethernet
  - Video Output Interface
    - HDMI, DVI,RGB Video Output
  - USB 2.0 Interfaces for Mouse, Keyboard, etc
  - One(1) DVD Writer for Video File Backup
  - Up to Four(4) Hard Disk

# Hardware Specification

AP-NR3000 Network Voice Recording Server

AP-NR3000 Front Side

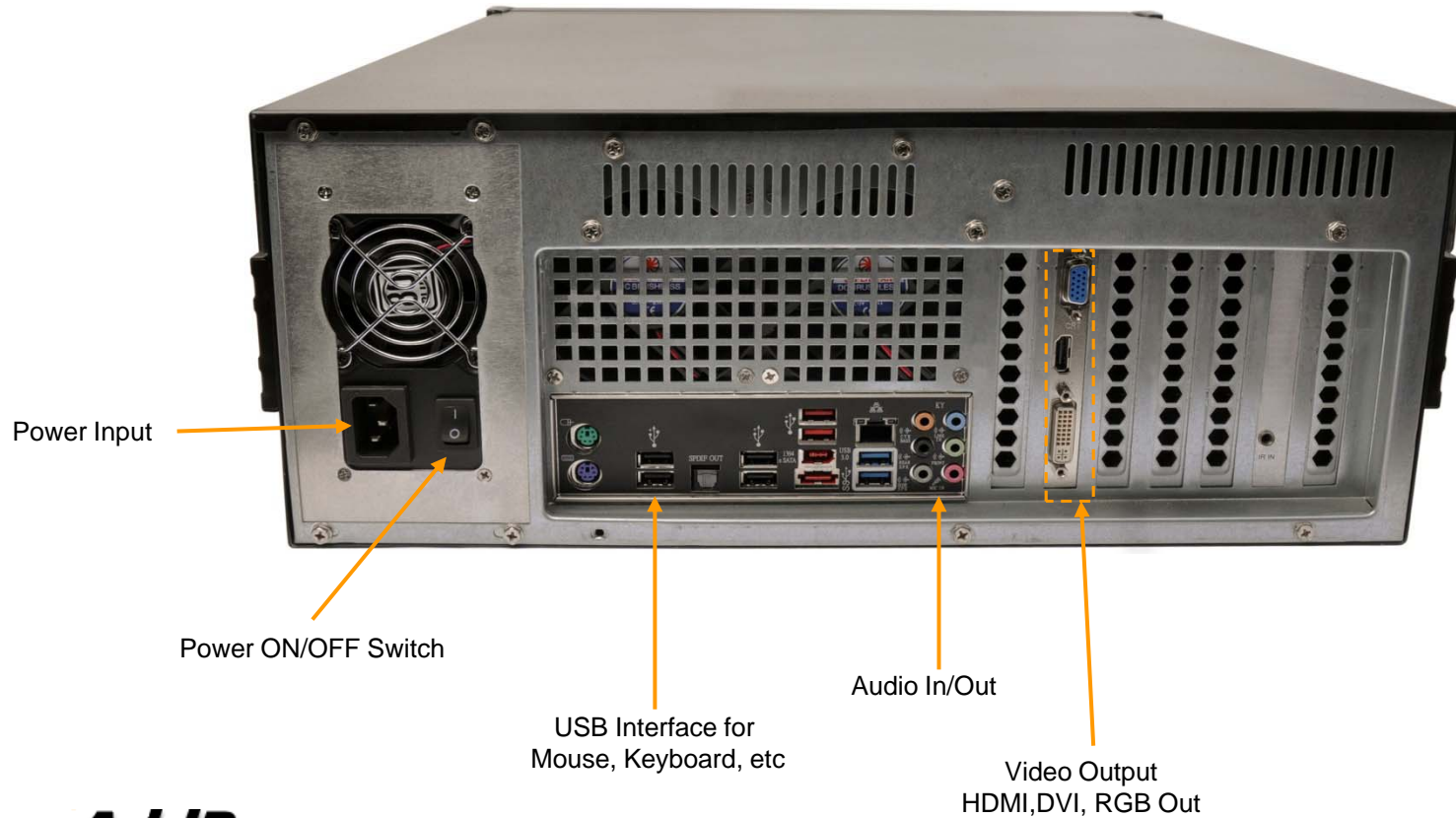




# Hardware Specification

## AP-NR3000 Network Voice Recoding Server

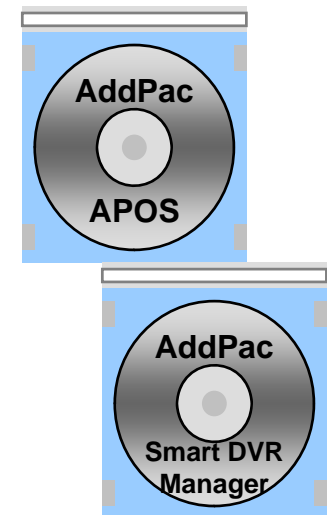
AP-NR3000 Back Side



# Software Service

## AP-NR3000 Network Voice Recoding Server

- **Built-in AddPac Internetworking Software**
  - Scalability, Functionality, and Stability Features
  - Advanced Network DV<sub>oice</sub>R Recording & Live Streaming Features
  - QoS Control Features
- **Firmware Upgradeable Architecture**
- **Industry Standard Network Protocol Features**
- **Highly User Friendly Management Features**
  - PC based Window Program
  - Smart DV<sub>oice</sub>R Manager



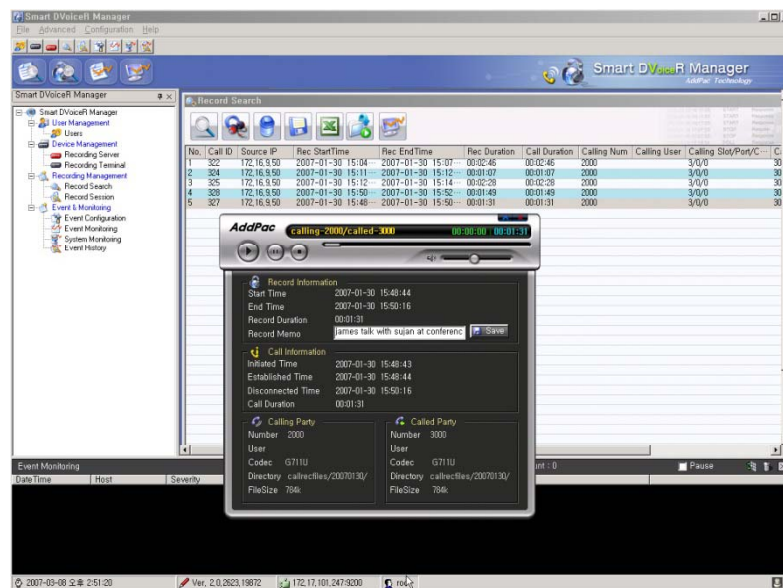
# Network Voice Recoding Components

## AP-NR3000 Network Voice Recoding Server

- Voice Recording & Manager Server
  - AP-NR500
  - AP-NR700
  - AP-NR3000
- Voice Recording Endpoints
  - Analog/Digital VoIP Gateways : AP2650, AP2640, AP2620, AP-MG3000etc
  - IP Phones : AP-IP300, AP-IP200, etc
  - Video Phones (Voice Capturing) : AP-VP300, AP-VP200, AP-VP150,etc
- Voice Transcoder (G.723,G.729-> G.711,etc)
  - AP-VTC1000
- Smart D<sub>voice</sub>R Management Program

# Smart DVoiceR Management Program

## AP-NR3000 Network Voice Recoding Server



- Call History Management (search/modify/delete/save)
- Media Play Management (Play/Stop/Seek/Pause)
- Live Call List Management, Live Call Monitoring
- Local Backup (File Manager Support, PC HDD, DVD) and Local Play
- User Management (registration/modify/delete/search)
- Server Status (CPU/Memory/HDD) & Event Monitoring
- Waveform Analyzing Function
- Recording Source Management (VoIP Gateway, IP Phone, etc)
- Live Recording Board

# AP601 SIP Gateway with MIC Interface



# Product Overview

## AP601 SIP Gateway with MIC Interface


- IP based Voice Broadcasting/Recording Terminal Solution
- Hardware Architecture for Voice Broadcasting/Recording Terminal Service
- Remote Broadcasting Service at terminal side
- High Quality Voice Codec Support (High Quality Codec, G.711, etc)
- SIP VoIP Signaling Protocol Support
- RTP/UDP Protocol Support
- Unicast and Multicast Broadcasting Scheme
- WBS (Window based Broadcasting Management System) Support
- One(1) channel SPK/MIC Port
- High-Quality Audio/Voice Service
- Firmware Upgradeable Architecture
- Broadcasting Solution with Outstanding Network Service Capability
- External Power Supply

# Hardware Specification

## AP601 IP Paging Terminal

A dark blue square with a light gray border containing the text "RISC CPU" in white.

RISC  
CPU

A dark blue square with a light gray border containing the text "High-end DSP" in white.

High-end  
DSP

- RISC Microprocessor Computing Power
- High-end Programmable DSP Hardware Architecture
- High Quality Audio Encoding/Decoding Service
- One(1) 10/100Mbps Fast Ethernet (RJ45)
- One(1) RS-232C Interface (RJ45)
- SPK/MIC Audio Interface Support
- Volume Control Button Support (Up, Down)
- External Power Supply Support

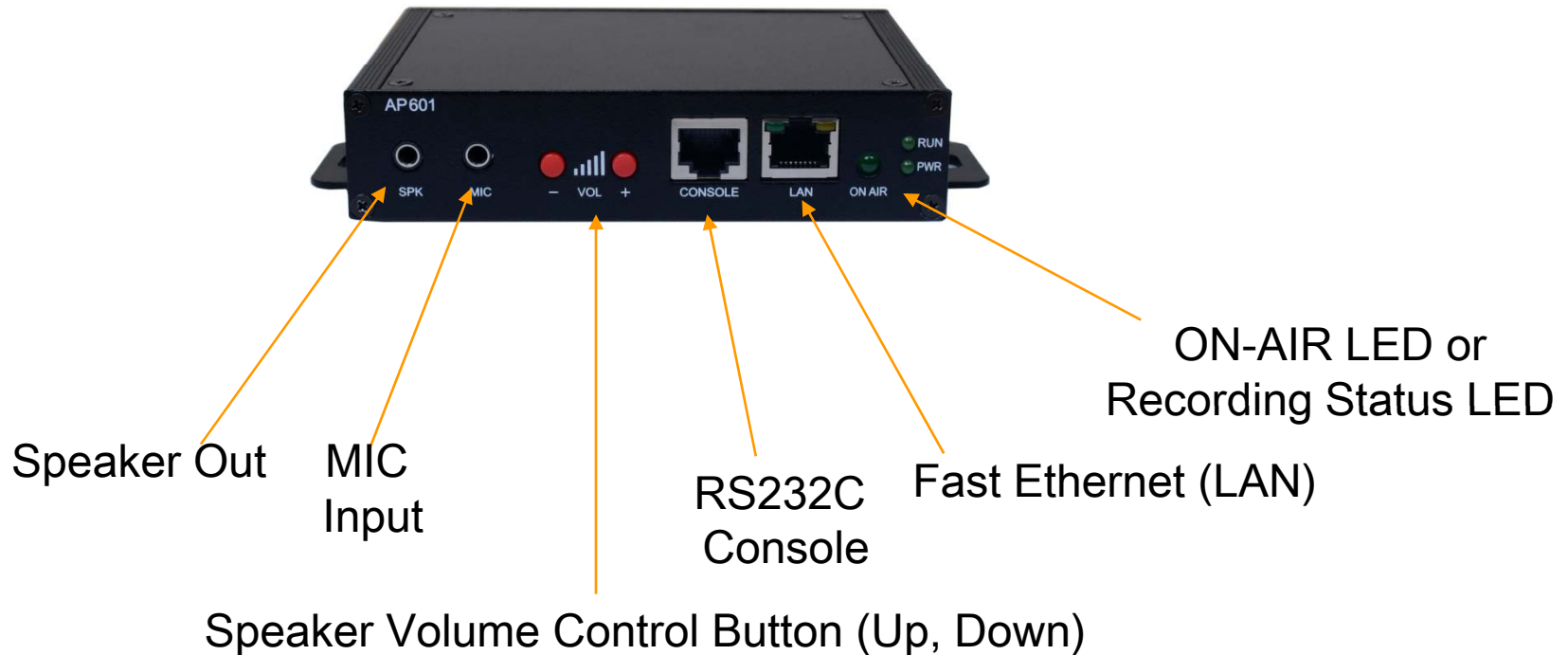
# Hardware Specification

AP601 IP Paging Terminal

RISC  
CPU

High-end  
DSP

## Front Side





# Hardware Specification

AP601 IP Paging Terminal

RISC  
CPU

High-end  
DSP

## Back Side



External Power  
Supply Adaptor

Power Switch

# AP-GV3200

## 40-Port GSM VoIP Gateway



# Contents

- Product Overview
- Hardware Specification
- VoIP (Voice over IP) service
- Advanced QoS Features
- Network Protocols
- Network Management
- Smart Web Management†
- Security Management



# Product Overview

## AP-GV3200 Multi-Port GSM VoIP Gateway

- Five(5) Module Slots for 8-Port GSM Module, Digital Interface (Up to 40-Port GSM, 1-2 Port Digital E1/T1 Module)
- Hot-Swap Function 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
- One(1) RS-232C Port for Command Line Interface
- Firmware Upgradeable Architecture
- Smart Web Manager Support
- Smart NMS(Network Management System) Support
- Advanced Voice QoS Mechanism
- Rack Mountable Chassis with Internal Power Supply

# Hardware Specification

## AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

- RISC Microprocessor Computing Power
- Five(5) Module Slot for GSM, Digital VoIP Interface
- 8-Port GSM Module(AP-N5-GSM8)
  - 8-Port SIM Card Slots
  - 8-Port Antenna Interface
  - Option : Two(2) GSM Antenna Interface (Internal 4 Channel Combiner x 2)
  - Hot-Swap
- VoIP Interface Module (Hot-Swap)
  - 1-Port Digital E1/T1 VoIP Module (AP-N1-1E1)
  - 2-Port Digital E1/T1 VoIP Module(AP-N1-2E1)
- Network Interface
  - Two(2) 10/100Mbps Fast Ethernet (RJ45)
- RS232C Console Interface for CLI
- Run LED, LAN LED, Port LEDs
- Internal Power Supply

# Hardware Specification

AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

<b>AP-GV3200 GSM VoIP Gateway</b>	<b>Basic Specifications</b>
<b>Voice Interface</b>	Five(5) Module Slots for GSM, Digital VoIP Interface
	AP-N5-GSM8x, AP-N1-E1, AP-N1-2E1
<b>Ethernet Interface</b>	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
<b>Flash Memory</b>	512Mbyte NAND Flash Memory
<b>Base Memory</b>	128Mbyte High-speed DDR2
<b>Power Requirement</b>	Power Supply / VAC 110~220V, 50/60Hz, 5V 30A
<b>Operating Temperature</b>	0°C ~ 45°C (32 °F ~ 122°F)
<b>Storage Temperature</b>	-40°C ~ 85°C (-40°C ~ 185°F)
<b>Relative Humidity</b>	5% ~ 95% (Non-condensing)
<b>Dimension (H x W x D)</b>	60mm x 440mm x 407mm
<b>Weight(kg)</b>	6.6Kg

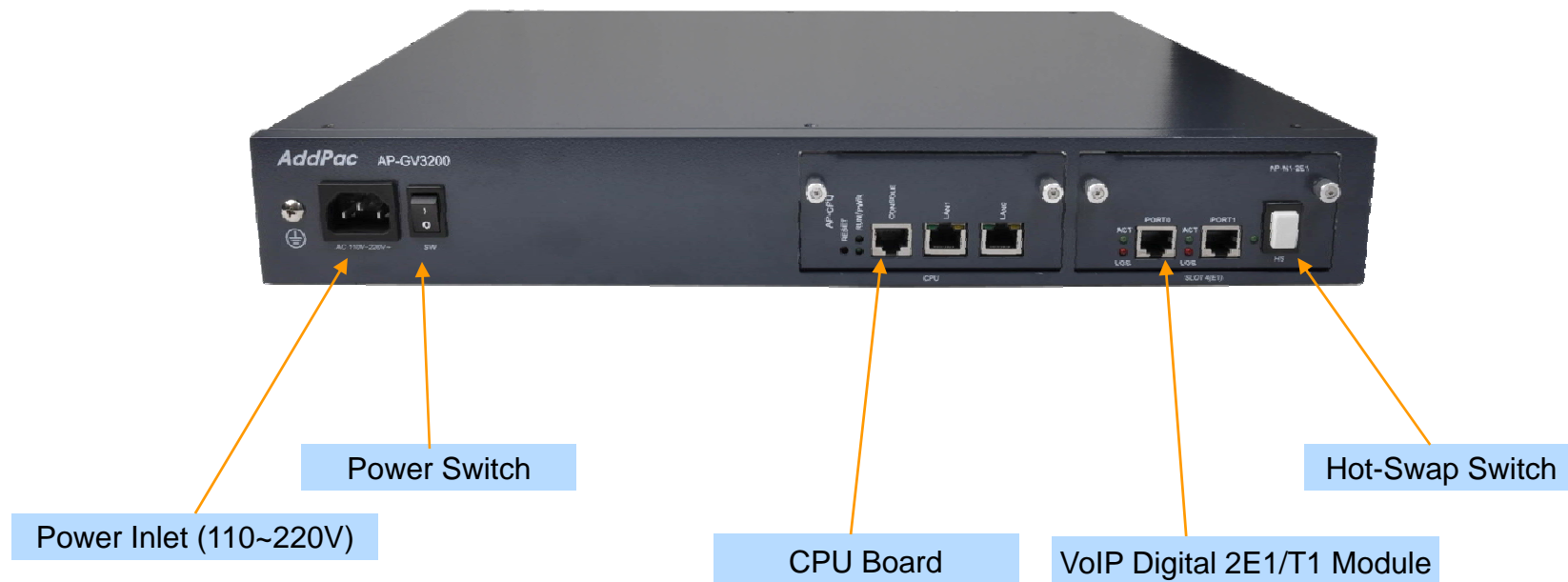
# Hardware Specification

AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## Front Side View



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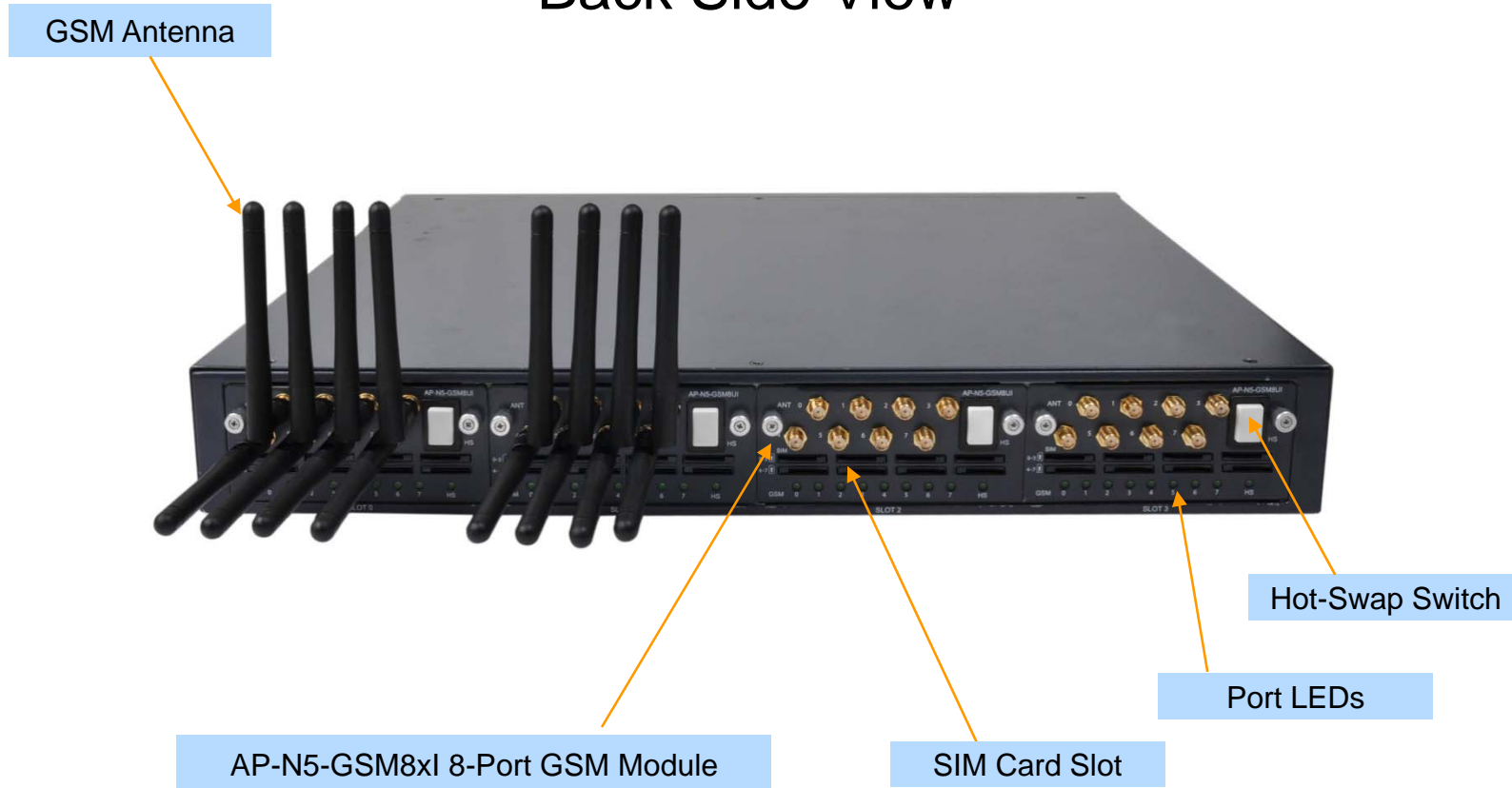
# Hardware Specification

AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## Back Side View





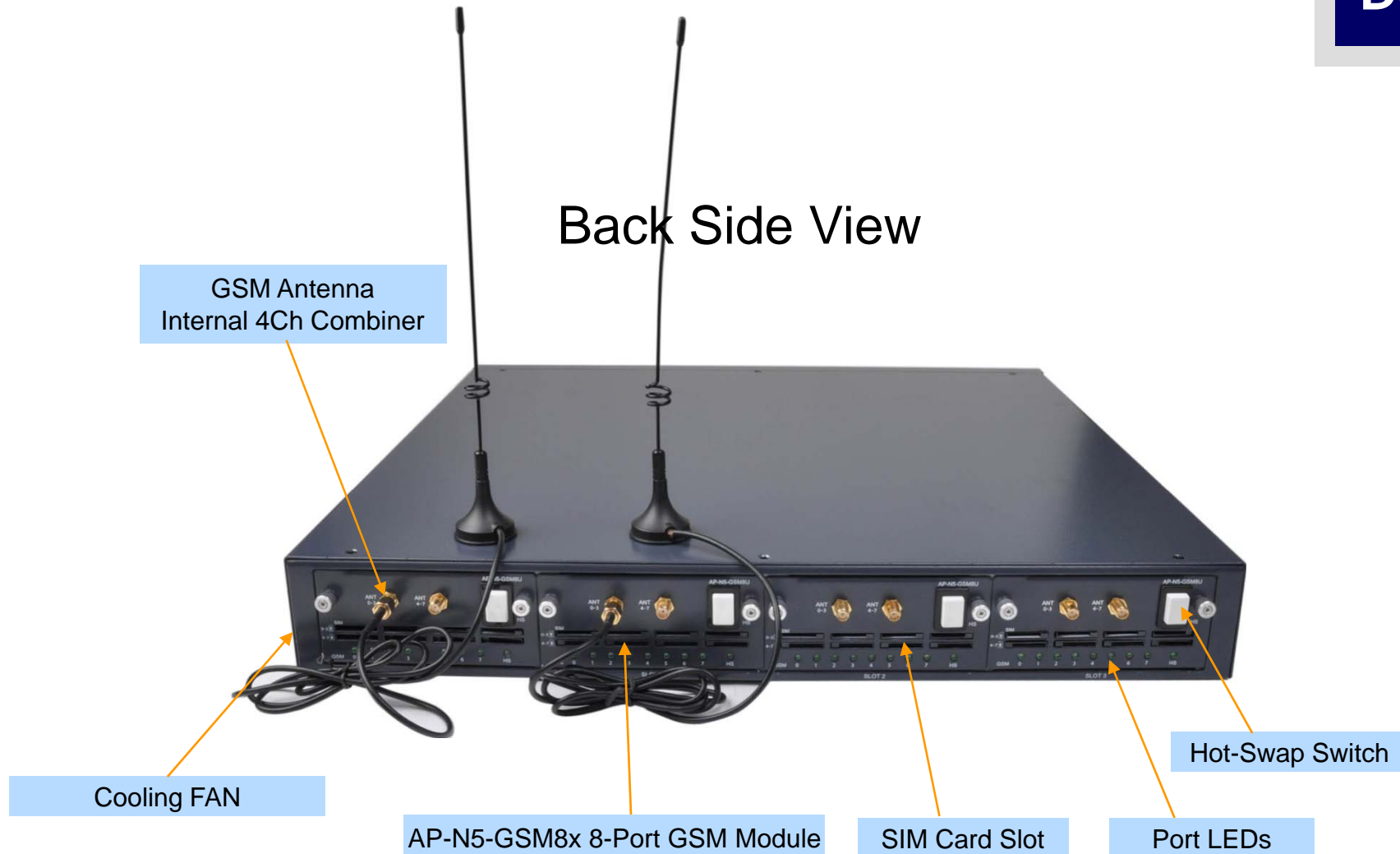
# Hardware Specification

AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## Back Side View



# GSM & Digital VoIP Modules

AP-GV3200 Multi-Port GSM VoIP Gateway

**AP-N5-GSM8x** 8-Port GSM Module  
4-Channel Antenna  
Combiner Model



**AP-N5-GSM8xI** 8-Port GSM Module  
Individual Antenna



**AP-N1-E1** 1-Port Digital E1/T1 Module



**AP-N1-2E1** 2-Port Digital E1/T1 Module



# Hardware Specification

AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

SIM Card Slot



AP-N5-GSM8xI  
(8-Port GSM Module)

GSM Antenna  
Internal 4Ch Combiner

Hot-Swap Switch

Port LEDs

# Hardware Specification

AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP



AP-N5-GSM8x  
(8-Port GSM Module)

GSM Antenna  
4channel Combiner

Port LEDs

SIM Card Slot

Hot-Swap Switch

# Hardware Specification

AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP



AP-N1-E1  
(1-Port Digital E1/T1 Module)



AP-N1-2E1  
(2-Port Digital E1/T1 Module)

# Hardware Specification

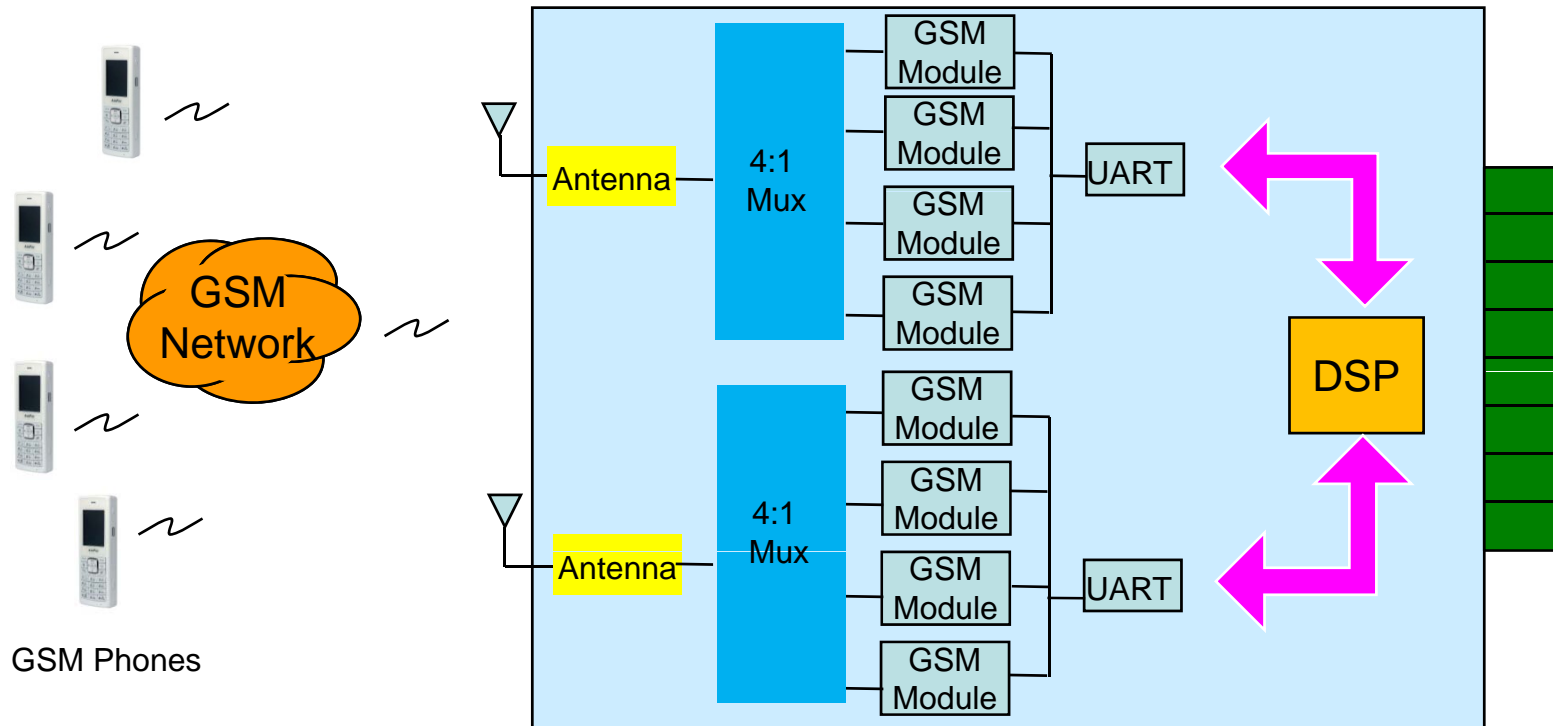
AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## GSM Gateway Service Diagram

8-Port GSM Module Internal H/W Block Diagram



# Hardware Specification

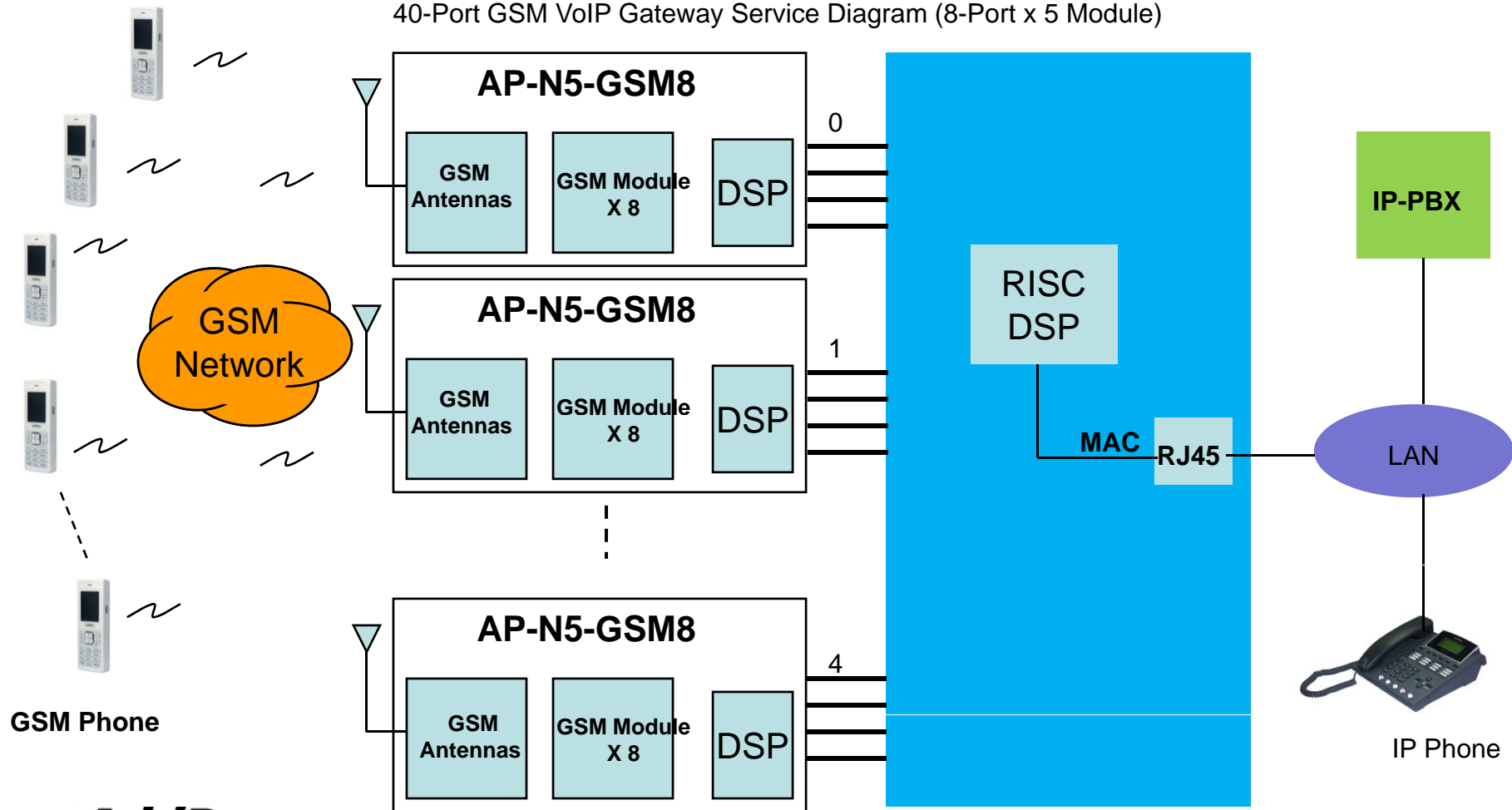
AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## GSM VoIP Gateway Service Diagram

40-Port GSM VoIP Gateway Service Diagram (8-Port x 5 Module)



GSM Phone

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# Hardware Specification

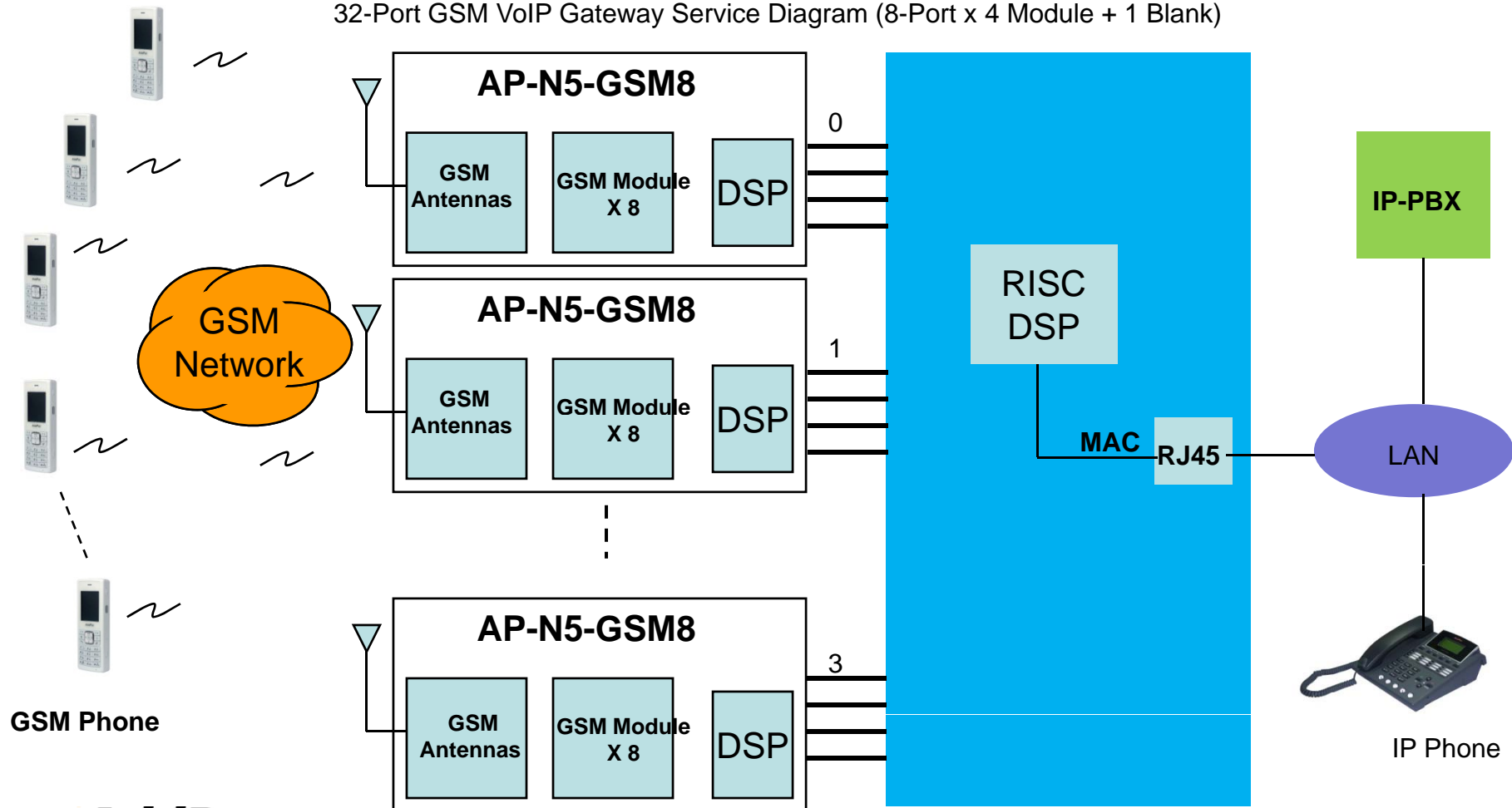
AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## GSM VoIP Gateway Service Diagram

32-Port GSM VoIP Gateway Service Diagram (8-Port x 4 Module + 1 Blank)



GSM Phone

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IP Phone



# Hardware Specification

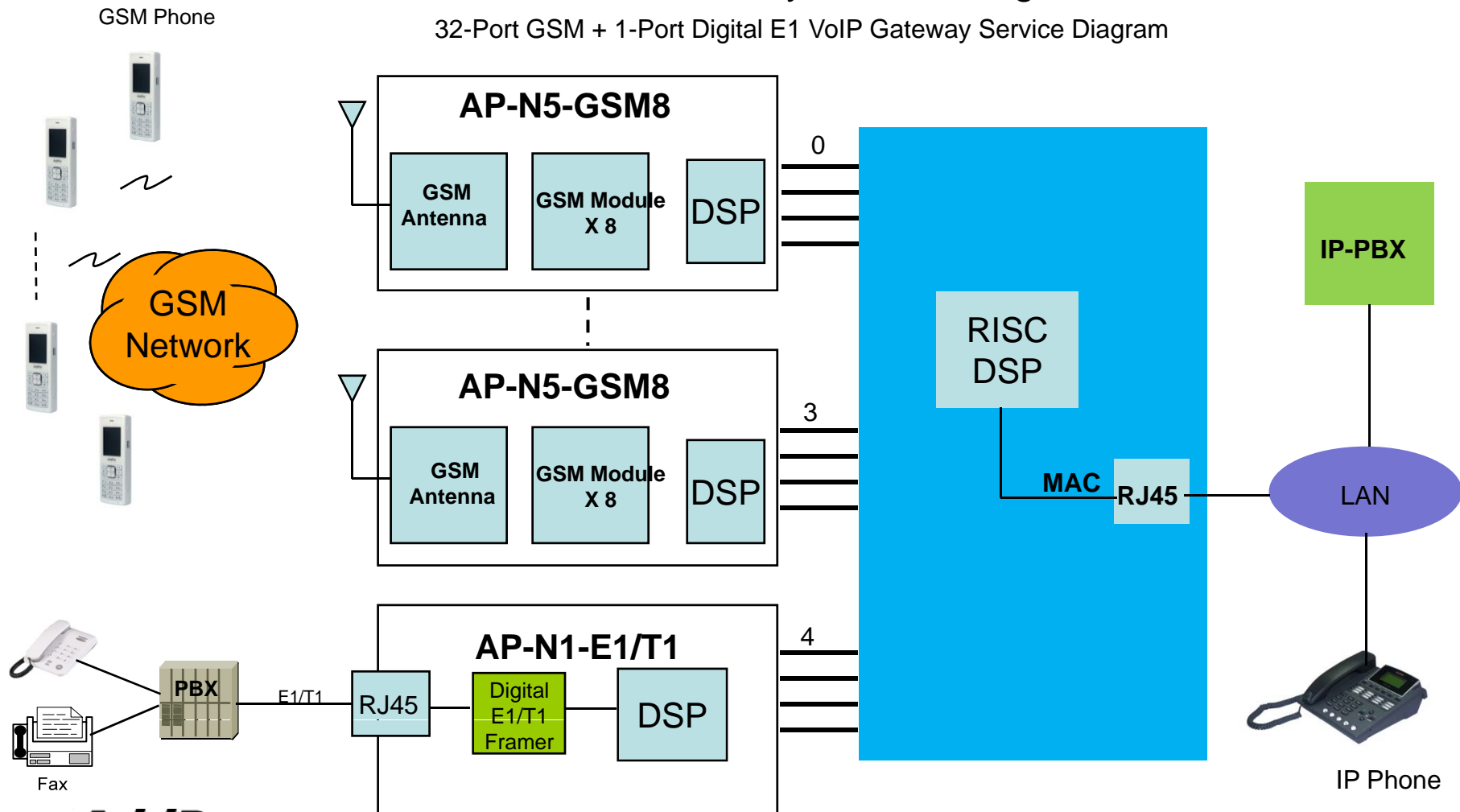
AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## GSM VoIP Gateway Service Diagram

32-Port GSM + 1-Port Digital E1 VoIP Gateway Service Diagram



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# Hardware Specification

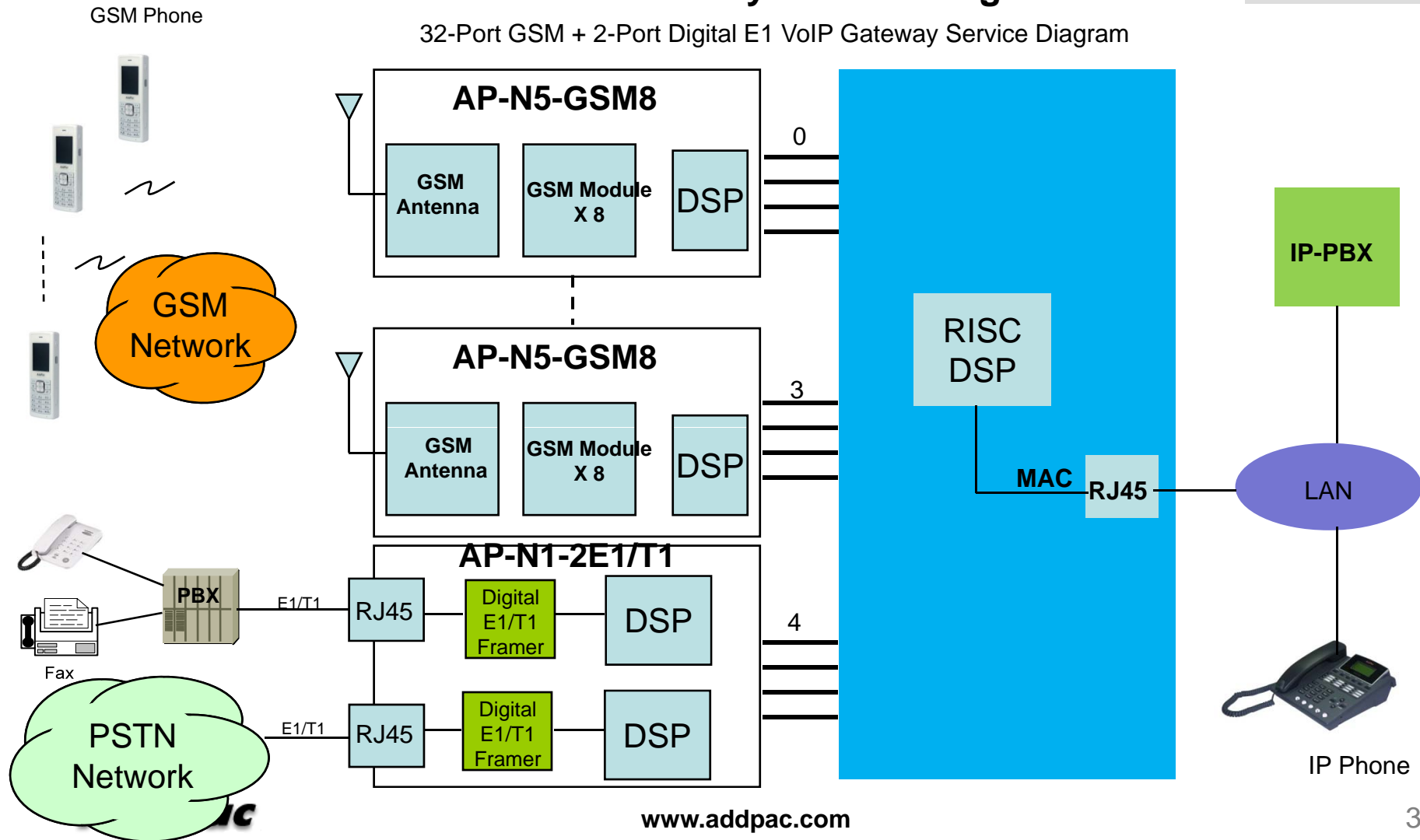
AP-GV3200 Multi-Port GSM VoIP Gateway

RISC  
CPU

High-end  
DSP

## GSM VoIP Gateway Service Diagram

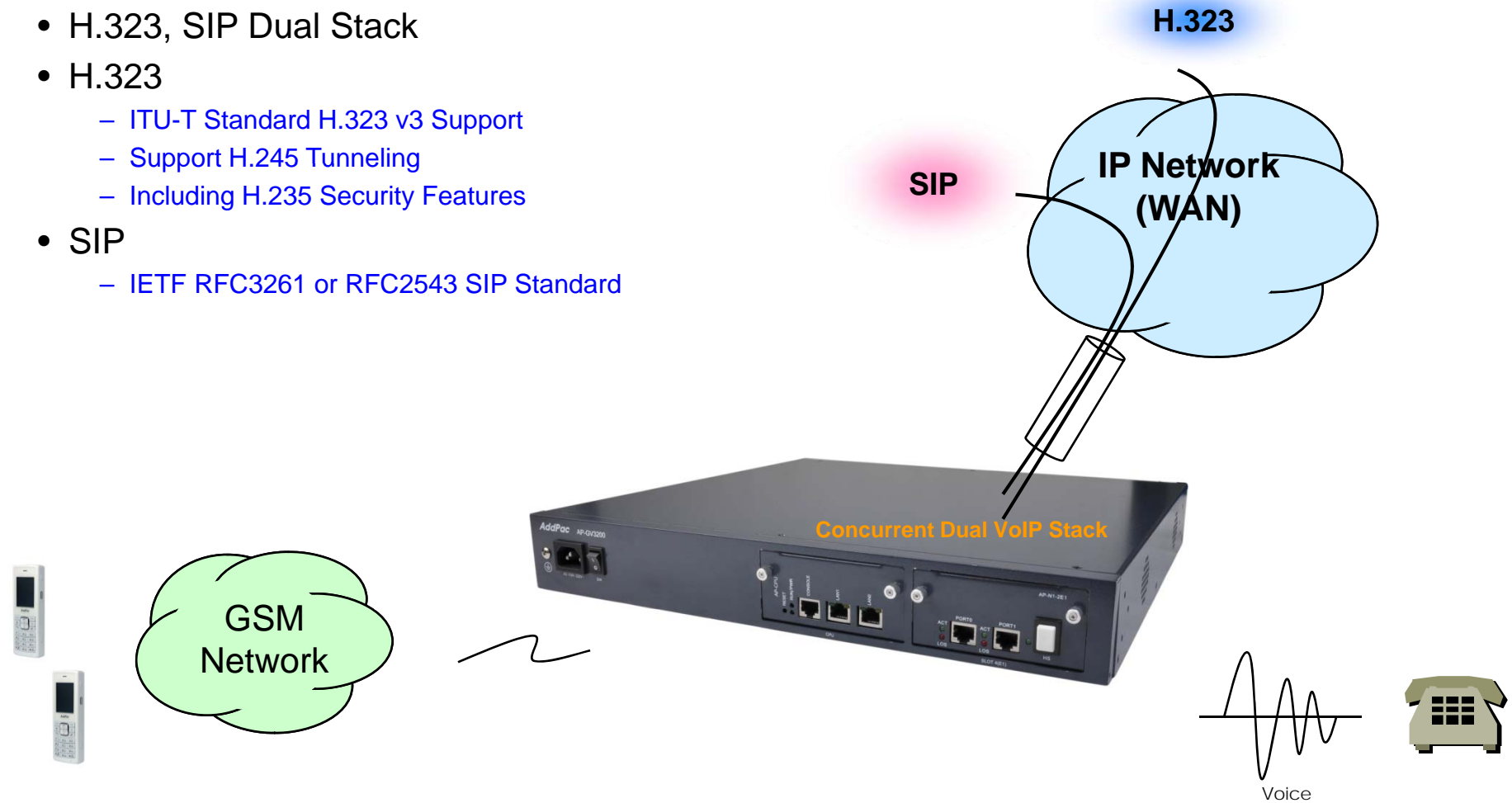
32-Port GSM + 2-Port Digital E1 VoIP Gateway Service Diagram



# VoIP (Voice over IP) Service

## AP-GV3200 Multi-Port GSM VoIP Gateway

- H.323, SIP Dual Stack
- H.323
  - ITU-T Standard H.323 v3 Support
  - Support H.245 Tunneling
  - Including H.235 Security Features
- SIP
  - IETF RFC3261 or RFC2543 SIP Standard



# VoIP (Voice over IP) Service

## AP-GV3200 Multi-Port GSM VoIP Gateway

- H.323

- Fast connect, normal connect support
- H.245 tunneling support
- Q.931 response message setting for inbound VoIP calls
- H.245 logical channel open timing selection function
- Start H.245 procedure support
- DTMF / Hook flash relay with H.245 alphanumeric / signal
- Secondary gatekeeper support
- Gatekeeper assignment according to the domain name
- Gatekeeper discovery with multicast
- Lightweight RRQ support
- Signaling TCP port assignment
- Resource threshold setting with RAI
- H.235 clear-token, crypto-token support
- canMapAlias support
- Technical prefix (supported prefix) support
- Public IP assignment in NAT environment

- SIP

- Gateway-based / Endpoint-based registration support
- Secondary proxy-server assignment function
- SIP signaling port change function
- SIP proxy server assignment according to the domain name
- T.38 real-time fax relay support
- DTMF relay support with RFC2833 / OPTION message
- Re-INVITE support

# VoIP (Voice over IP) Service

## AP-GV3200 Multi-Port GSM VoIP Gateway

- Voice Codec

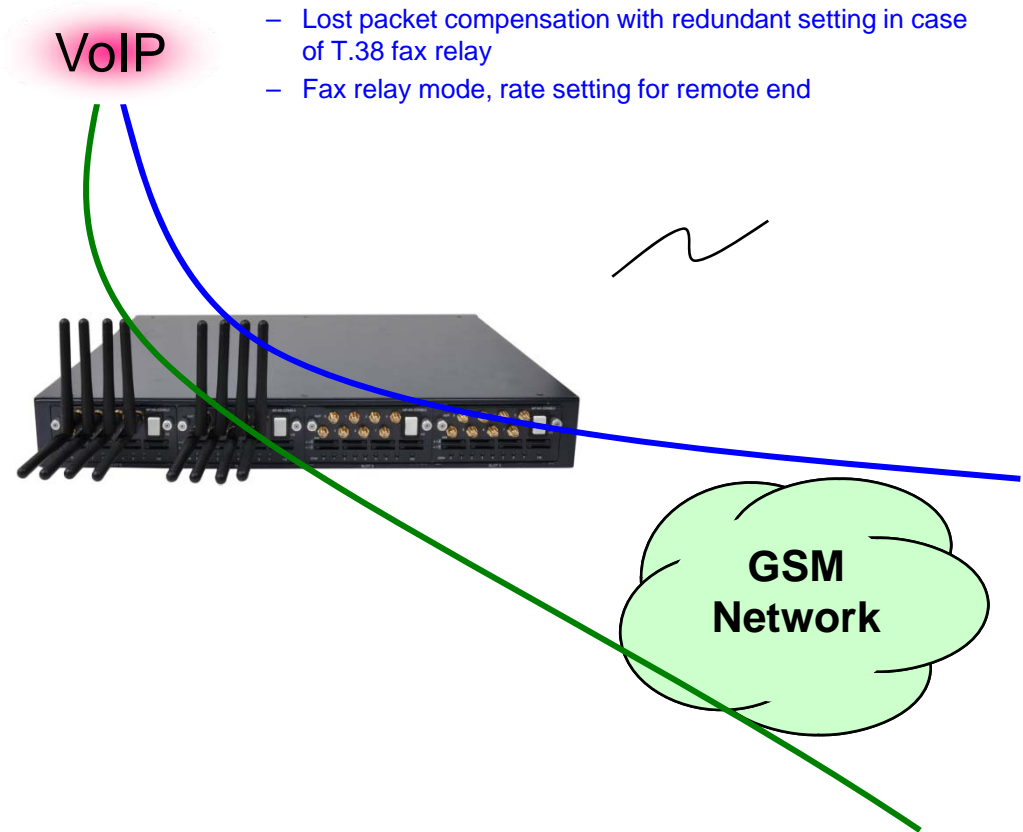
- G.711 A-Law, G.711 U-Law
- G.726 r16, G.726 r32
- G.729A
- G.723.1 r63, G.723.1 r53
- VAD (Voice Activity Detection) function support
- DTMF relay support (H.323, SIP, MGCP common) based on RFC2833

- RTP

- Redundant RTP packet transmission in case of severe packet loss
- Dynamic jitter buffer management and RTP packet jitter and loss compensation with heuristic & DSP error concealment
- Static jitter buffer setting support
- Voice frame per RTP packet number control for each codec
- In-band ring-back tone support
- Virtual ring-back tone support
- Tone parameter change support

- FAX

- Fax relay mode supporting T.38, inband-T.38, bypass mode
- Lost packet compensation with redundant setting in case of T.38 fax relay
- Fax relay mode, rate setting for remote end



# VoIP (Voice over IP) Service

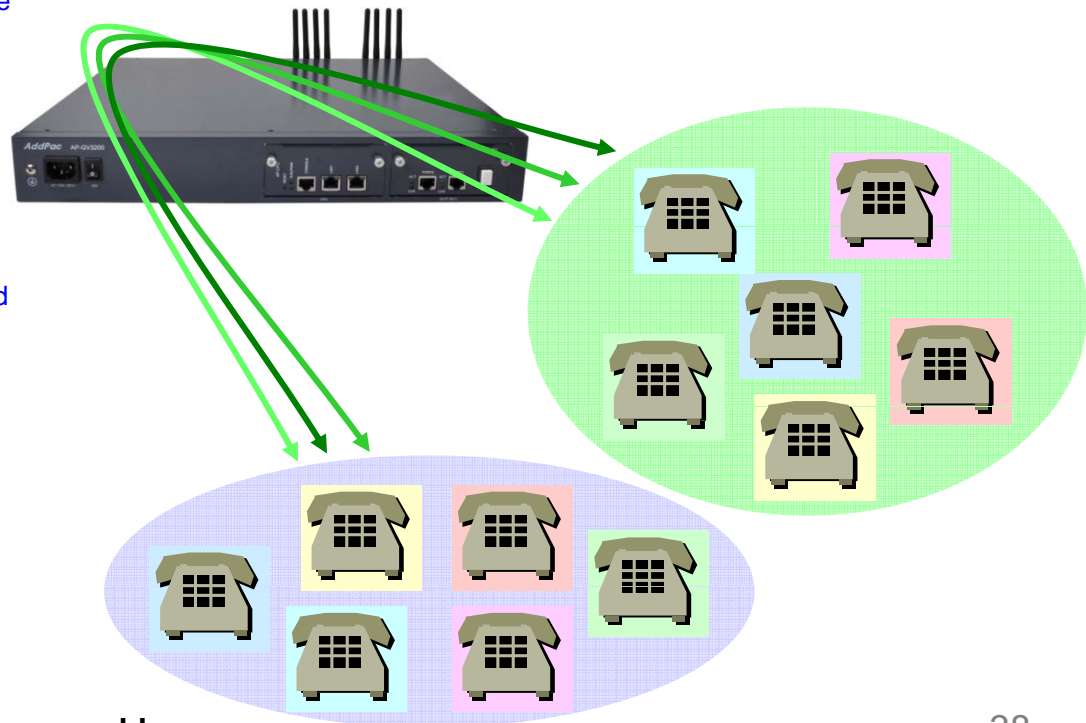
## AP-GV3200 Multi-Port GSM VoIP Gateway

- VoIP Call Controls

- Hot line connection function with PLAR (Private Line Auto Ring Down)
- Leased line emulation function
- Connection monitoring function
- Fault tolerant with Redundancy and Call Distribution among Gateways for load balancing
- Call attempt with IP address
- H.323, SIP, MGCP inbound call connection for each voice port
- Multiple E.164 setting for one voice port
- One E.164 or digit pattern can be assigned to more than one voice port
- Hunting with Longest match/ priority/ sequence/ random
- One stage call setup by Digit forwarding
- Call barring with specific digit patterns
- Calling and called number conversion for PSTN outbound calls
- PSTN rerouting in case of VoIP call attempt failure

- VoIP Call Controls (cont.)

- Call transfer for internal calls
- Call pickup for internal calls
- Calling and called number conversion for VoIP outbound calls
- Calling and called number conversion for VoIP inbound calls



# Advanced QoS Features

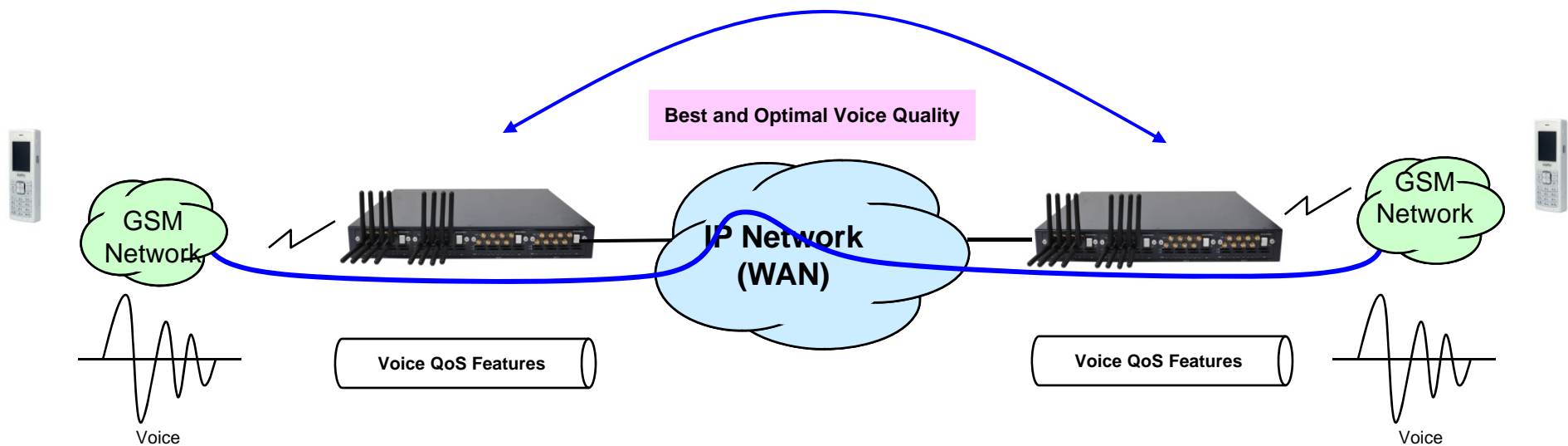
## AP-GV3200 Multi-Port GSM VoIP Gateway

- Enhances **Transmit** Voice QoS Features

- Voice Traffic Priority Queuing
- QoS Service Profiling
- Providing Virtual Network Transmit Algorithm
- Real-time Voice Traffic QoS Support
- RTP Packet Transmit Interval Control
- Supporting RTP Packet Redundancy Scheme
- IP Header Control such as ToS, Diffserv

- Enhances **Receive** Voice QoS Features

- Dynamic Jitter Buffer Management
- Error Concealment
- Support T.38 FAX Data Error Recovery Scheme



# Network Protocols

## AP-GV3200 Multi-Port GSM VoIP Gateway

### Basic Network Protocols

- ARP, IPv4, TCP, UDP, ICMP, SCTP, IGMP, MLD

### Routing Protocol

- IPv4 : Static

### Service Protocol

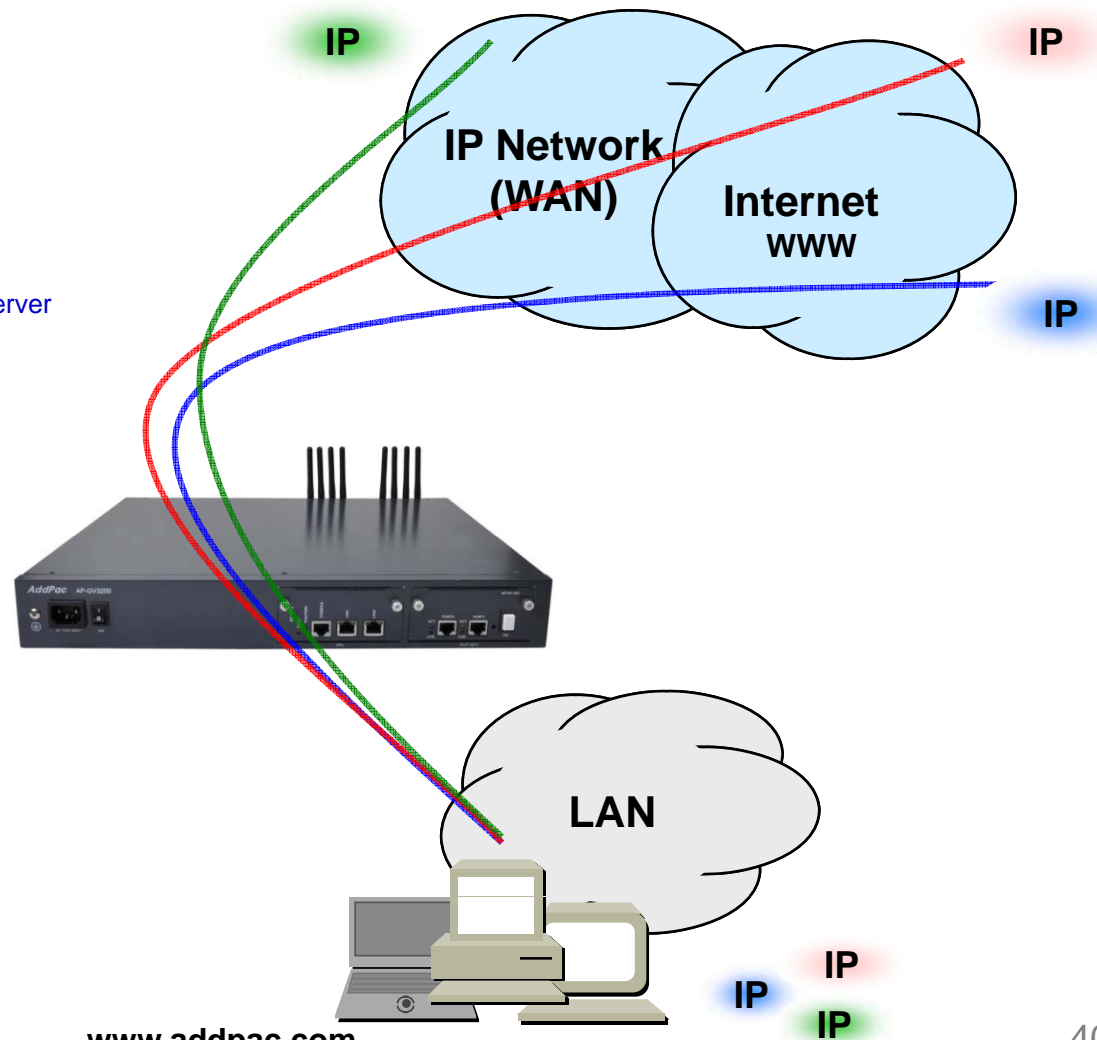
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

### IPv4 Address Configuration

- Fixed (Static)
- DHCP
- PPPoE

### Miscellaneous

- Cisco Style CLI
- Standard & Extended IPv4 Access List
- Multi-level User Account Management
- IP accounting
- STUN Client

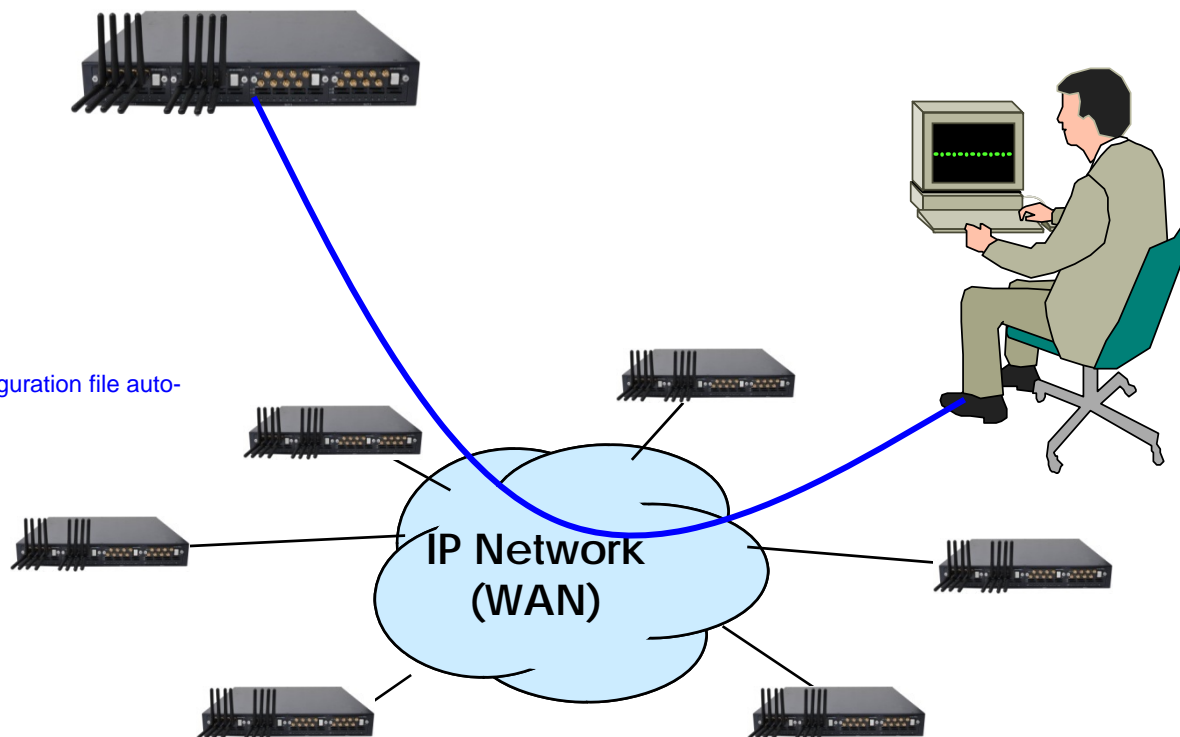




# Network Management

## AP-GV3200 Multi-Port GSM VoIP Gateway

- SNMP
  - Standard Simple Network Management Protocol( SNMP) Agent support
  - MIB v1 and v2 Support
- Web-based Management
  - Smart Easy Setup
  - Standard Voice Interface
  - Standard PSTN Back-up Interface
- Watch-dog Function
  - Hardware, Software watch-dog services
- Remote Management
  - Telnet
  - Rlogin
- Auto Upgrade Service
  - HTTP server based APOS image and configuration file auto-upgrade support
- Batch Job Function
  - Text based script downloading
- Interoperable with Smart NMS Service
  - AddPac Smart Network Management System



# Smart Web Manager

AP-GV3200 Multi-Port GSM VoIP Gateway

- **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
- **LCR(Least Cost Routing)**
  - Black & White List, Time Interval, Tariff Group, LCR Test
- **SMS**
  - Inbox, SMS New Message

# Smart Web Manager : Main Page Layout

## AP-GV3200 Multi-Port GSM VoIP Gateway

**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

System Information	
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**

# Smart Web Manager : System – Network Setup

## AP-GV3200 Multi-Port GSM VoIP Gateway

The screenshot shows the 'Network Setup' page in the Smart Web Manager interface. The page is divided into several sections, each with a radio button selection:

- Static IP:** Selected. Fields include Hostname (GS1002), IP Address (172.16.9.16), Network Mask (255.255.0.0), Default Router (172.16.1.1), and DNS Server (Primary and Secondary).
- PPPoE(ADSL):** Unselected. Fields include Username and Password.
- DHCP:** Unselected.
- VLAN:** Unselected. Field includes ID (0).
- WAN Link Control:** Selected. Options include Auto, Manual, Speed (100, 10), and Duplex (full, half).
- MAC(Hardware) Address:** Unselected. Field includes a 6-digit hexadecimal address.

On the right side, there is an 'Information' panel and a 'Description' panel. The 'Information' panel shows device details like Model (GS1002\_G2) and Network (Static 172.16.9.16). The 'Description' panel contains a note about WAN port setup and unreads SMS messages.

On the left side, there are several yellow callout boxes with arrows pointing to specific fields in the Network Setup page:

- 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 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.

At the bottom left of the page, there is an 'Apply' button with a green checkmark.

# Smart Web Manager : Basic – GSM Extension

## AP-GV3200 Multi-Port GSM VoIP Gateway

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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

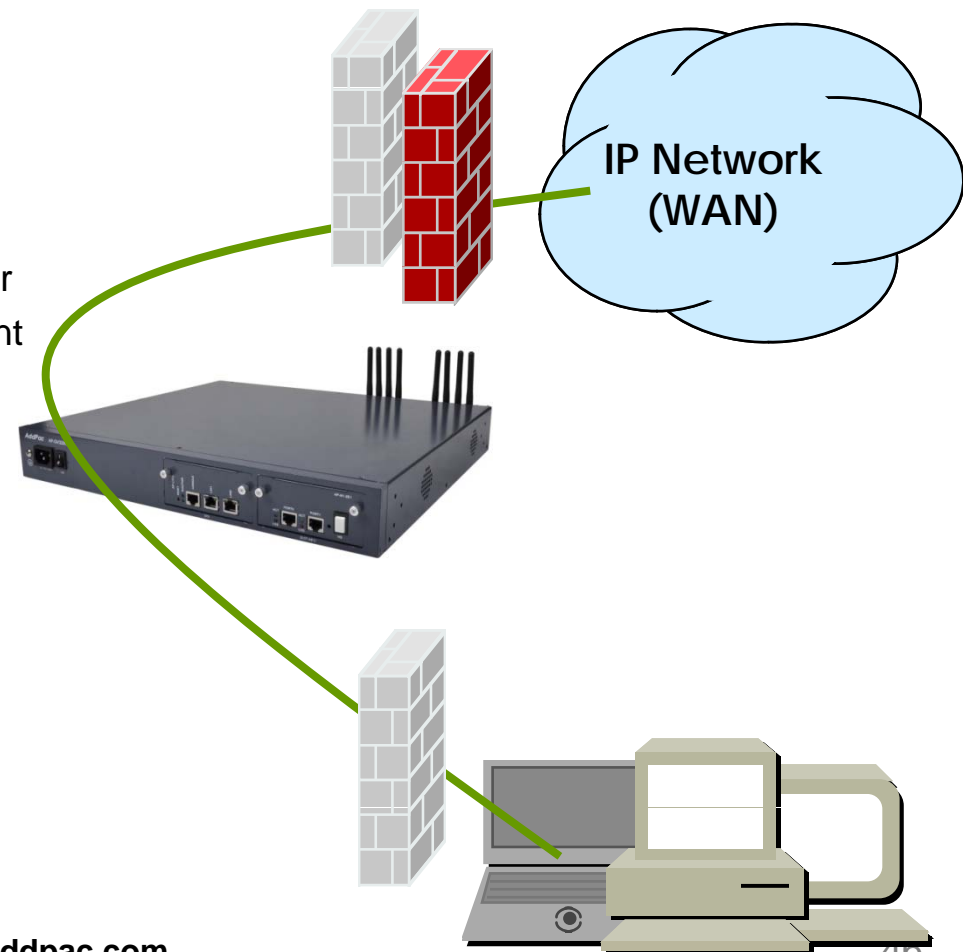
**GSM Extension**  
Configure GSM phone-number for receiving a call (usually 'T' is used for each port)

**GSM Extension with Translation**  
Used to GSM callback  
- The Received CID is not real serving number.  
- The specified translation rule is applied.

# Security Management

## AP-GV3200 Multi-Port GSM VoIP Gateway

- IP packet filtering
- IP access list
- User authentication function
  - Password Authentication Protocol (PAP)
  - Challenge Handshake Authentication Protocol (CHAP)
- Enable/Disable specific protocols
- Auto-square connect of Telnet session
- Account Management function for multi-level user
- SNMP/TELNET/FTP/HTTP/TFTP port assignment function
- SNMP/TELNET/FTP access list management
- Boot mode security checking function



# AP-GS1001

## 1-Port GSM VoIP Gateway



**AddPac**

[www.addpac.com](http://www.addpac.com)

# Contents

- Product Overview
- Hardware Specification
- GSM Module Specification
- APOS Technology
- VoIP (Voice over IP) service
- Advanced QoS Features
- Network Protocols
- Network Management
- Smart Web Management
- Security Management
- Application Service
- Ordering Information



# Product Overview

## AP-GS1001 1-Port FXS GSM Gateway

- Analog Interface (FXS)/VoIP Interface(LAN) Both Support
- H.323/SIP/MGCP Triple 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-GS1001 1-Port FXS GSM Gateway

RISC  
CPU

High-end  
DSP

- RISC Microprocessor Computing Power
- 1-Port GSM Gateway
- 1-Port SIM Card Slot
- 1-Port GSM Antenna Interface
- VoIP Gateway Interface
  - AP-GS1001 Model A: Basic Configuration
  - AP-GS1001 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

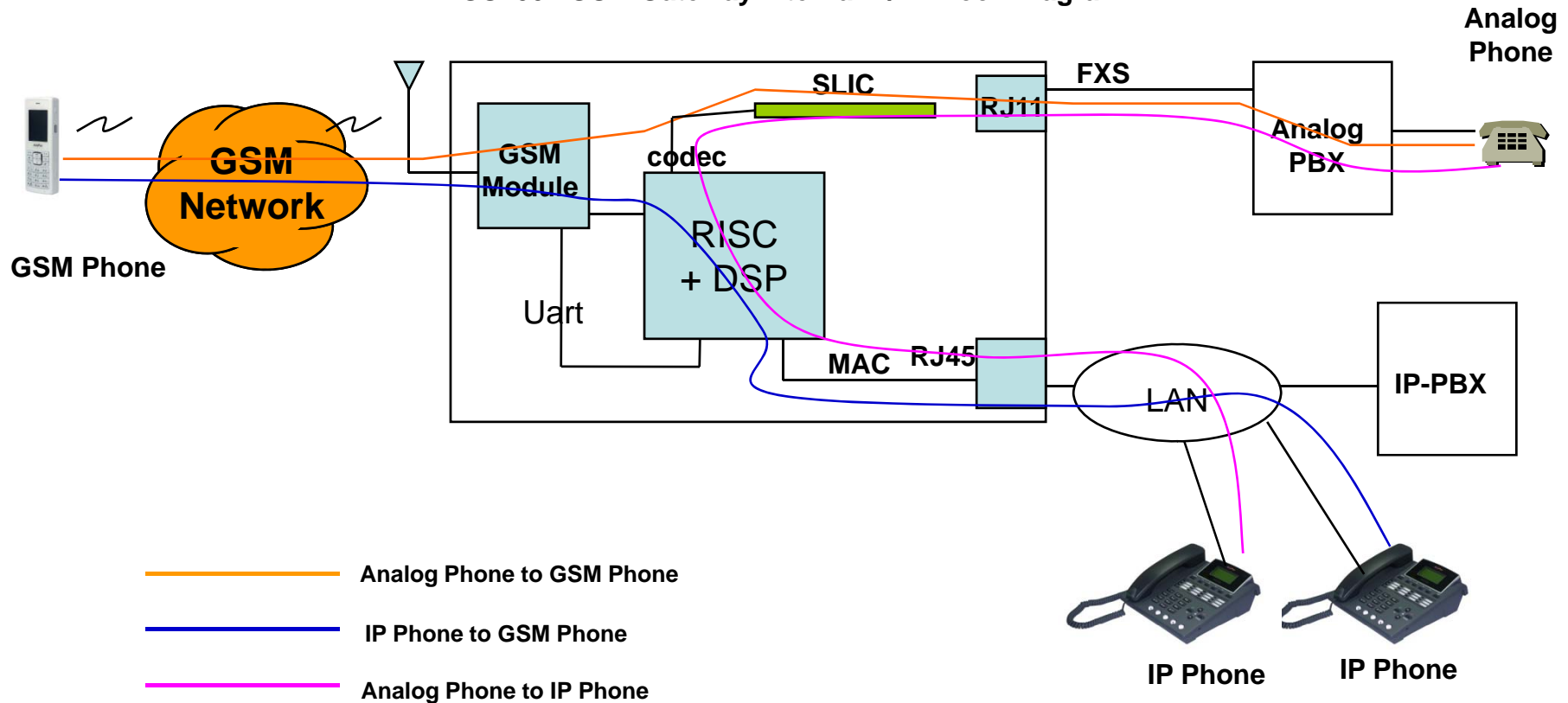
AP-GS1001 1-Port FXS GSM Gateway

RISC  
CPU

High-end  
DSP

## GSM Gateway Service Diagram

AP-GS1001 GSM Gateway Internal H/W Block Diagram



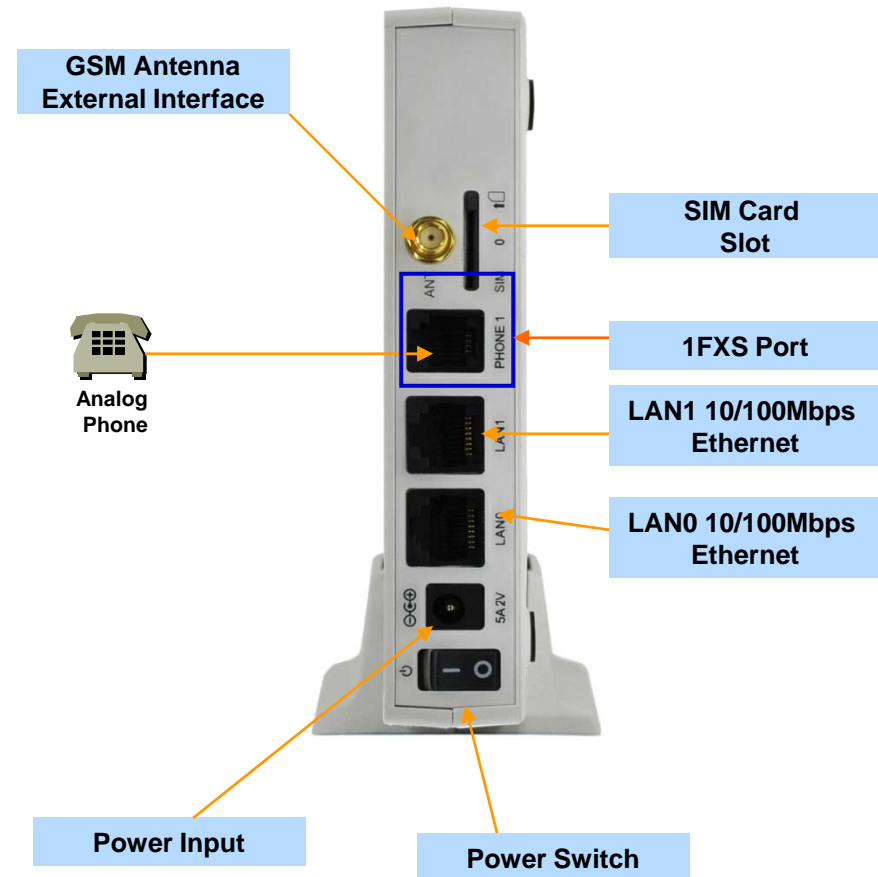
# Hardware Specification

## AP-GS1001 1-Port FXS GSM Gateway

### Hardware Specifications

AP-GS1001 GSM Gateway	Basic Specifications
Voice Interface	No FXS Port : Model A One(1)-Port FXS Analog Interface : Model B
GSM Antenna Interface	External GSM Antenna
SIM Card Slot Interface	GSM SIM Card Slot
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	32 Mbyte High-speed SDRAM
Power Requirement	Power Supply Adaptor / VAC 110~220V, 50/60Hz, 5V, 2A
Operating Temperature	0°C ~ 45°C (32 °F ~ 122°F)
Storage Temperature	-40°C ~ 85°C (-40°C ~ 185°F)
Relative Humidity	5% ~ 95% (Non-condensing)
Dimension	142mm x 142mm x 28mm (W x D x H)
Weight (Kg)	0.42Kg

### Network interface Configurations



# GSM Module Specification

## AP-GS1001 1-Port FXS GSM Gateway

- Bearers : GSM + GPRS class 10
- Quad-Band EGSM 850/900/1800/1900MHz
- Normal Sensitivity
  - 850MHz Rx            -104 dBm
  - 900MHz Rx           -104 dBm
  - 1800MHz Rx         -102 dBm
  - 1900MHz Rx         -102 dBm
- Tx Performances
  - 850MHz Tx            +33 dBm
  - 900MHz Tx            +33 dBm
  - 1800MHz Tx          +30 dBm
  - 1900MHz Tx          +30 dBm
- Power Consumption
  - 17 uA Sleep Mode, 1.7mA Idle Mode, 400mA GPRS class10 (33 dBm)
- Codec
  - FR-EFR-HR-AMR

# VoIP (Voice over IP) Service

AP-GS1001 1-Port FXS GSM Gateway

- **H.323, SIP, and MGCP Triple Stack**
- **H.323**
  - ITU-T Standard H.323 v3 Support
  - Support H.245 Tunneling
  - Including H.235 Security Features
- **SIP**
  - IETF RFC3261 or RFC2543 SIP Standard
- **MGCP**
  - IETF RFC2705bis-02 Standard MGCP 1.0



# VoIP (Voice over IP) Service

## AP-GS1001 1-Port FXS GSM Gateway

- **H.323**

- Fast connect, normal connect support
- H.245 tunneling support
- Q.931 response message setting for inbound VoIP calls
- H.245 logical channel open timing selection function
- Start H.245 procedure support
- DTMF / Hook flash relay with H.245 alphanumeric / signal
- Secondary gatekeeper support
- Gatekeeper assignment according to the domain name
- Gatekeeper discovery with multicast
- Lightweight RRQ support
- Signaling TCP port assignment
- Resource threshold setting with RAI
- H.235 clear-token, crypto-token support
- canMapAlias support
- Technical prefix (supported prefix) support
- Public IP assignment in NAT environment

- **SIP**

- Gateway-based / Endpoint-based registration support
- Secondary proxy-server assignment function
- SIP signaling port change function
- SIP proxy server assignment according to the domain name
- T.38 real-time fax relay support
- DTMF relay support with RFC2833 / OPTION message
- Re-INVITE support

- **MGCP**

- Secondary call agent assignment function
- Default package assignment
- Announcement Server Package, Generic Media Package, Handset Package, Line Package, Trunk Package support
- MGCP call agent assignment according to the domain name
- T.38 real-time fax relay support
- DTMF relay support based on RFC2833

# VoIP (Voice over IP) Service

## AP-GS1001 1-Port FXS GSM Gateway

- **Voice Codec**

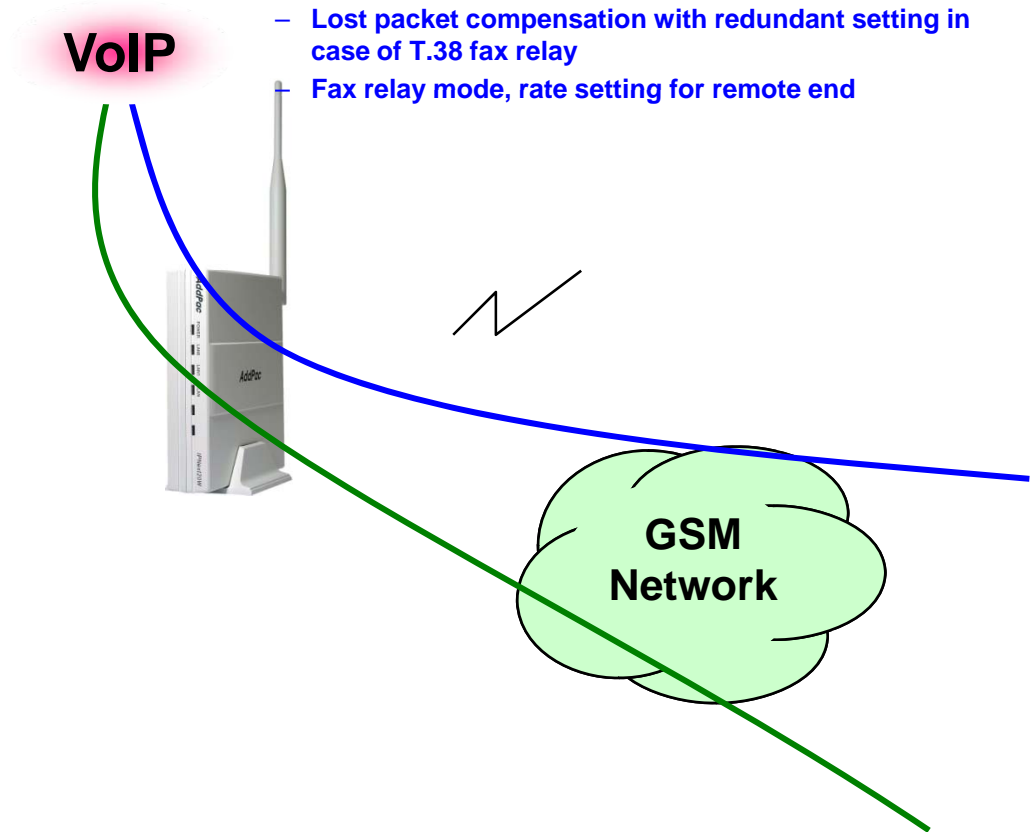
- G.711 A-Law, G.711 U-Law
- G.726 r16, G.726 r32
- G.729A
- G.723.1 r63, G.723.1 r53
- VAD (Voice Activity Detection) function support
- DTMF relay support (H.323, SIP, MGCP common) based on RFC2833

- **RTP**

- Redundant RTP packet transmission in case of severe packet loss
- Dynamic jitter buffer management and RTP packet jitter and loss compensation with heuristic & DSP error concealment
- Static jitter buffer setting support
- Voice frame per RTP packet number control for each codec
- In-band ring-back tone support
- Virtual ring-back tone support
- Tone parameter change support

- **FAX**

- Fax relay mode supporting T.38, inband-T.38, bypass mode
- Lost packet compensation with redundant setting in case of T.38 fax relay
- Fax relay mode, rate setting for remote end





# VoIP (Voice over IP) Service

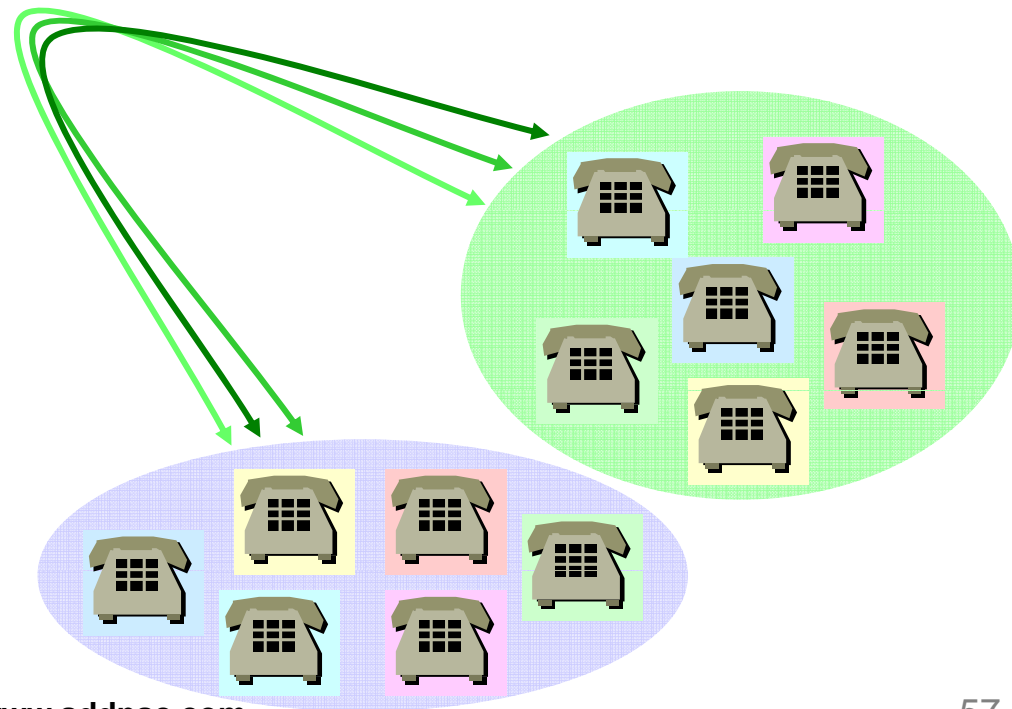
## AP-GS1001 1-Port FXS GSM Gateway

### • VoIP Call Controls

- Hot line connection function with PLAR (Private Line Auto Ring Down)
- Leased line emulation function
- Connection monitoring function
- Fault tolerant with Redundancy and Call Distribution among Gateways for load balancing
- Call attempt with IP address
- H.323, SIP, MGCP inbound call connection for each voice port
- Multiple E.164 setting for one voice port
- One E.164 or digit pattern can be assigned to more than one voice port
- Hunting with Longest match/ priority/ sequence/ random
- One stage call setup by Digit forwarding
- Call barring with specific digit patterns
- Calling and called number conversion for PSTN outbound calls
- PSTN rerouting in case of VoIP call attempt failure

### • VoIP Call Controls (cont.)

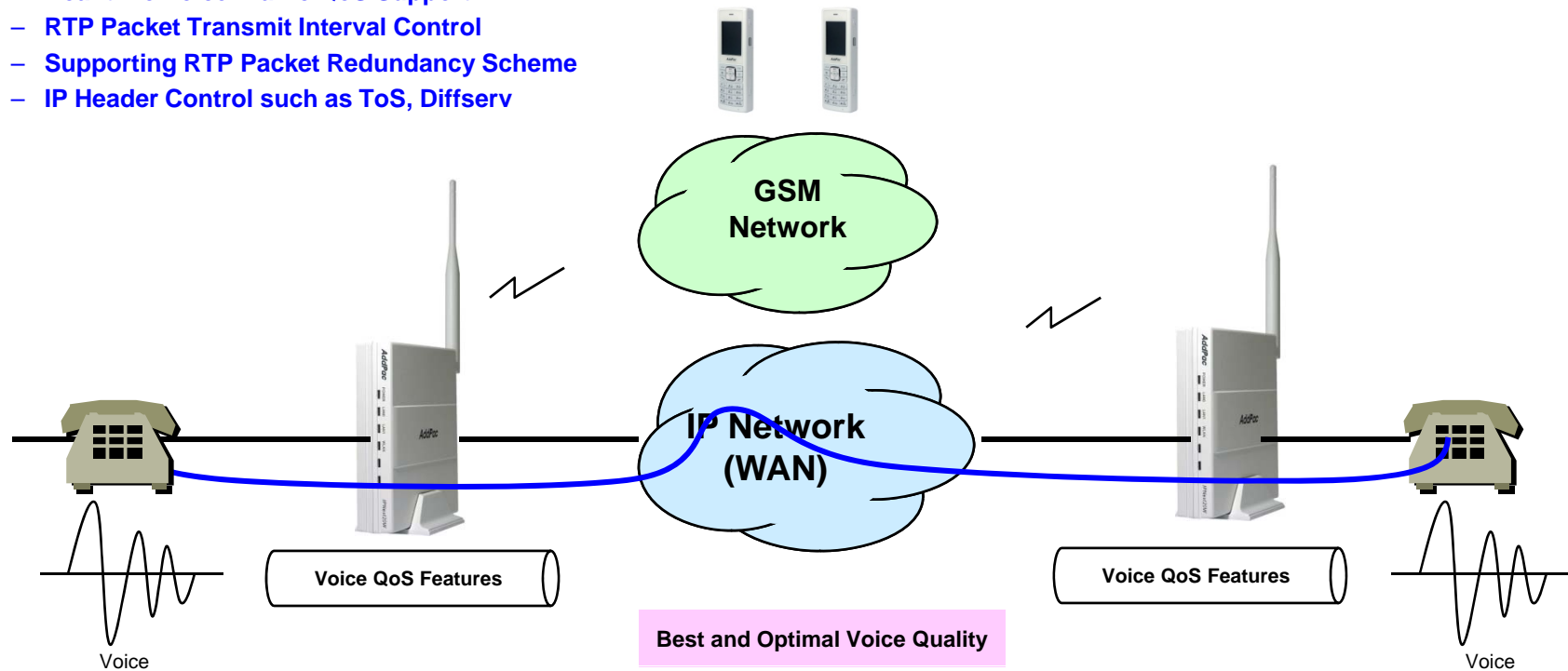
- Call transfer for internal calls
- Call pickup for internal calls
- Calling and called number conversion for VoIP outbound calls
- Calling and called number conversion for VoIP inbound calls
- Fax broadcasting call control



# Advanced QoS Features

## AP-GS1001 1-Port FXS GSM Gateway

- Enhances **Transmit** Voice QoS Features
  - Voice Traffic Priority Queuing
  - QoS Service Profiling
  - Providing Virtual Network Transmit Algorithm
  - Real-time Voice Traffic QoS Support
  - RTP Packet Transmit Interval Control
  - Supporting RTP Packet Redundancy Scheme
  - IP Header Control such as ToS, Diffserv
- Enhances **Receive** Voice QoS Features
  - Dynamic Jitter Buffer Management
  - Error Concealment
  - Support T.38 FAX Data Error Recovery Scheme



# Network Protocols

AP-GS1001 1-Port FXS GSM Gateway

## Basic Network Protocols

- ARP, IPv4, TCP, UDP, ICMP, SCTP, IGMP, MLD

## Routing Protocol

- IPv4 : Static

## Service Protocol

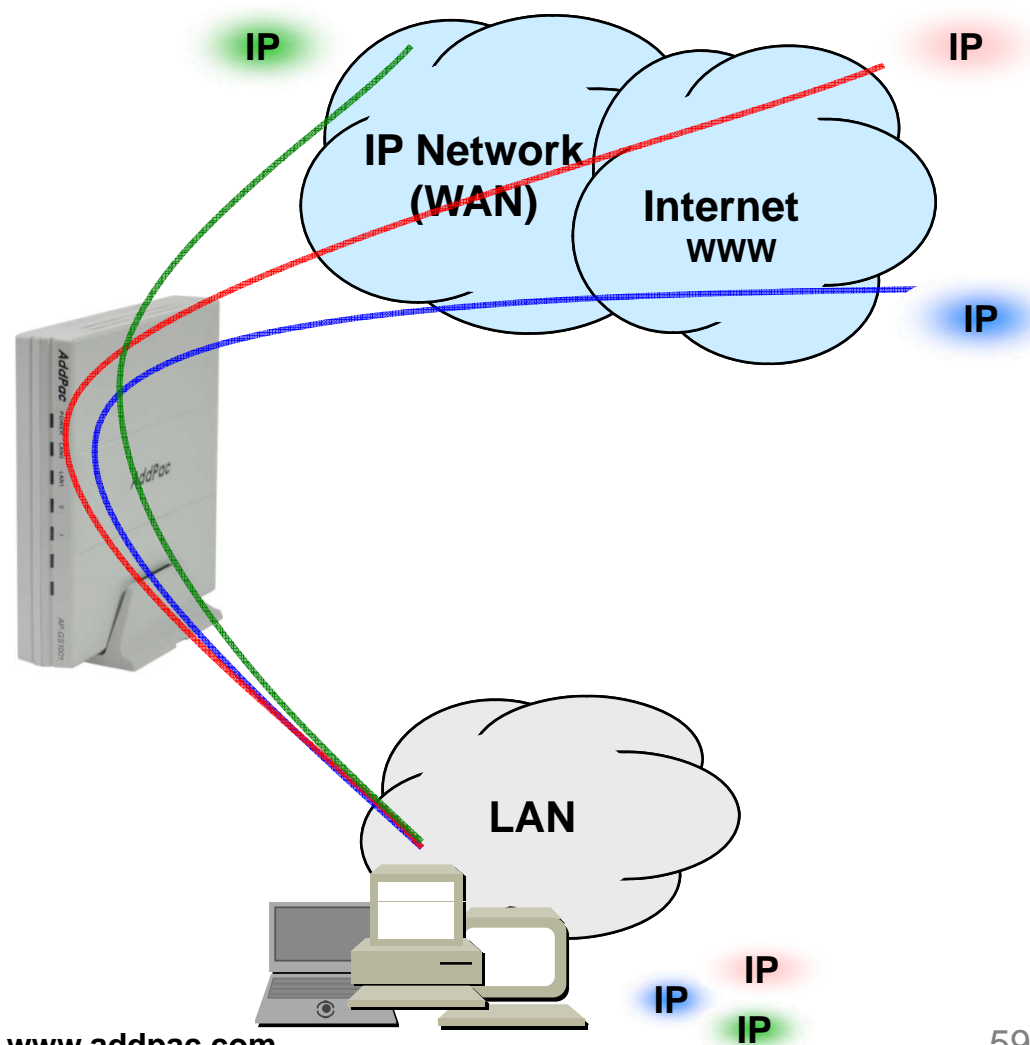
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

## IPv4 Address Configuration

- Fixed (Static)
- DHCP
- PPPoE

## Miscellaneous

- Cisco Style CLI
- Standard & Extended IPv4 Access List
- Multi-level User Account Management
- IP accounting
- STUN Client



# Network Management

## AP-GS1001 1-Port FXS GSM Gateway

- **SNMP**

- Standard Simple Network Management Protocol( **SNMP**) Agent support
- **MIB v1 and v2 Support**

- **Web-based Management**

- **Smart Easy Setup**
- **Standard Voice Interface**
- **Standard PSTN Back-up Interface**

- **Watch-dog Function**

- **Hardware, Software watch-dog services**

- **Remote Management**

- **Telnet**
- **Rlogin**

- **Auto Upgrade Service**

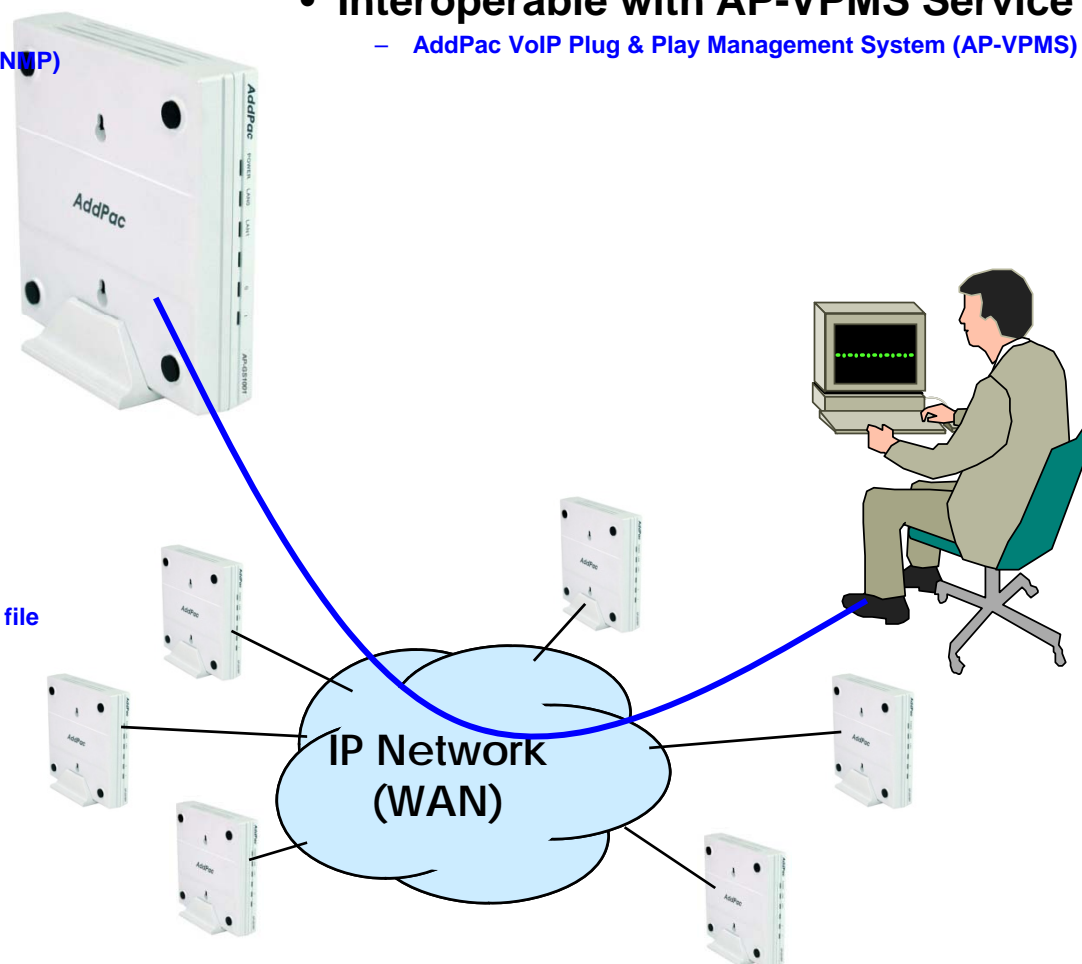
- **HTTP server based APOS image and configuration file auto-upgrade support**

- **Batch Job Function**

- **Text based script downloading**

- **Interoperable with AP-VPMS Service**

- **AddPac VoIP Plug & Play Management System (AP-VPMS)**



# Smart Web Manager : Main Page Layout

## AP-GS1001 1-Port FXS GSM Gateway

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

- System
  - Network Setup
  - Language
  - NTP
  - Backup/Restore
- Basic
  - Protocol
  - Server SIP
  - Server H.323
  - Tel Number
  - FXS/FXO/E1 Group
  - E1 Trunk
  - DTMF/CODEC
  - Dial Plan/Prefix
  - Static Route
  - Hot Line
- Advanced
  - Gain
  - Fax
  - Service
  - Filtering
  - Security
  - SNMP
- Miscellaneous
  - Call Status
  - System Status
  - Alarm Status
  - Call Log
  - System Log
  - Test Call
  - Ping

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

**System Information**

HW Version	2.0
S/W Version	ap1800k_web_g2_v8_47T.bin 8.47
MAC Address	0002.a511.2245
VoIP Protocol	SIP
Voice Interface Module	S(4)O(4) : E1(2)
Registration Status	Unregistered
Supported Codec List	g711alaw g711ulaw g7231r53 g7231r63 g726r32 g729
Network Information	Static 172.16.50.114
WAN LINK Status	100Mbps FULL Duplex Link UP
LAN LINK Status	Link Down
Current Time	Thu Oct 1 13:06:23 2009
System Startup Time	Thu Oct 1 12:56:46 2009
System Running Time	0 days 00:09:37
Total Calls	0

**Information**  
Display the current system version and status summary

AddPac Tehonology  
Model : AP1800K\_G2  
H/W Version : 2.0  
S/W Version : 8.47  
Smart Web Version : 0.3  
Smart Web Build : Oct 1 2009  
Voice Interface  
S(4)O(4) : E1(2)  
Protocol : SIP  
Status : Unregistered  
CurrentCalls : 0 Call  
Network : Static 172.16.50.114  
Mac Address: 0002.a511.2245

**Description**  
WAN 포트에 대한 설정입니다. Static IP의 경우 고정 IP 주소로 사업자로부터 할당 받은 주소 정보를 입력합니다. DHCP와 PPPoE의 경우 유동 IP로 장비의 주소가 변경될 수 있습니다. DHCP 및 PPPoE는 사용자 환경에 맞도록 설정하십시오. MAC 주소 변경은 필요시 장비에 설정된 주소를 사용하지 않고사용자가 설정한 주소를 사용하는 방안으로 반드시 필요한 경우에 한하여 사용하여야 합니다.

**Workspace**  
Workspace for detailed action

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

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# Smart Web Manager : E1 Trunk Configuration (Example)

## AP-GS1001 1-Port FXS GSM Gateway

**E1 Port**  
E1 Port Configuration  
Example : Slot is selected  
by tap button. Group in  
each E1 port can be  
added or deleted.

**E1 Port Configuration**  
Provide the detailed  
configuration menu about  
selected E1 port

**Tool Tip**  
Display the help  
message according to  
mouse move

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

**E1 Trunk**

Slot 1/0	Slot 1/1	Slot/Port	group Num (0)	Time slot Range(1-31,16-31,1,2,3)	Control
1/0		1/0	0	1-31	Delete
1/0		1/0	<input type="text" value="0"/>	<input type="text"/>	Add

**E1 Port (Slot 1/0)**

**Clock-Source** Slave

**Signaling-type** ISDN-PRI

**ISDN-PRI**

Protocol-emulate	<input type="radio"/> Network	<input checked="" type="radio"/> User
Virtual-Connect	<input type="radio"/> Enable	<input checked="" type="radio"/> Disable
Immediate-disc	<input type="radio"/> Enable	<input checked="" type="radio"/> Disable
Dial-Tone-Generate	<input type="radio"/> Enable	<input checked="" type="radio"/> Disable
Comband-Type	<input checked="" type="radio"/> a-law	<input type="radio"/> u-law
	N303	<input type="text" value="2"/> (1~10sec)
Q931 Timer	T303	<input type="text" value="4"/> (1~400sec)
	T310	<input type="text" value="10"/> (5~400sec)

**R2-MFC** Get-Calling-number  Enable  Disable

**Busyout** Action

**Get-Calling-number**  
enable로 체크해 주어야 할 내용으로 사설 교환기로 부터 발신번호 요청을 합니다.  
R2 방식일 경우에는 발신자 번호를 대국측에서 요청을 하여야만 받을 수 있습니다.  
기본적으로 R2 pdd 값을 줄이기 위해서 초기 값은 0으로 설정되어 있습니다.

Apply

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**Information**  
AddPac Tehonology  
Model : AP1800K\_G2  
H/W Version : 2.0  
S/W Version : 8.47  
Smart Web Version : 0.3  
Smart Web Build : Oct 1 2009  
Voice Interface  
S(4)O(4) : E1(2)  
Protocol : SIP  
Status : Unregistered  
CurrentCalls : 0 Call  
Network : Static 172.16.50.114  
Mac Address: 0002.a511.2245

**Description**  
E1 포트별 동작 방법을 설정합니다.

# Smart Web Manager : Call Status (Example)

## AP-GS1001 1-Port FXS GSM Gateway

**E1 Port**  
Real-time display about digital E1 time slot status (idle, dialing, incoming call, outgoing call, call blocking, etc)  
Provide a specific port blocking function

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

**Port Status Display**

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

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

**System**

- Network Setup
- Language
- NTP
- Backup/Restore

**Basic**

- Protocol
- Server SIP
- Server H.323
- Tel Number
- FXS/FXO/E1 Group
- E1 Trunk
- DTMF/CODEC
- Dial Plan Prefix
- Static Route
- Hot Line

**Advanced**

- Gain
- Fax
- Service
- Filtering
- Security
- SNMP

**Miscellaneous**

- Call Status
- System Status
- Alarm Status
- Call Log
- System Log
- Test Call
- Ping

**Call Status**

**Port Status (E1)**

Port#	Channel Group	Control
SLOT 1/0	Channel 01 02 03 04 05 06 07 08 09 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31	Unblock
	Status	Block
SLOT 1/1	Channel 01 02 03 04 05 06 07 08 09 10 11 12 13 14 15 16 17 18 19 20 21 22 23 24 25 26 27 28 29 30 31	Unblock
	Status	Block

**Port Status (Analog)**

Slot	Port	0(FXS)	1(FXS)	2(FXS)	3(FXS)	4(FXO)	5(FXO)	6(FXO)	7(FXO)
SLOT 0	Status	C	I	I	I	I	I	I	I
	Select	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
	Control	Unblock	Block						

포트 연결 상태 : (Connected) (Disconnected || Blocked)  
 포트 통화 상태 : (Idle) (Ring || Dial) (Called) (Calling) (Blocked)

**Call Status**

Port	Direction	Established Time	Calling Number	Called Number	CODEC	Src/Dest. IP
1/0:30	E1Local	01/01 09:10:38	6404	9000	g711u	N.A

**Information**

AddPac Tehonology  
 Model : AP1800K\_G2  
 HW Version : 2.0  
 SW Version : 8.47  
 Smart Web Version : 0.3  
 Smart Web Build : Oct 1 2009  
 Voice Interface  
 S(4)O(4) : E1(2)  
 Protocol : SIP  
 Status : Unregistered  
 CurrentCalls : 1 Call  
 Network : Static 172.16.50.114  
 Mac Address : 0002.a511.2245

**Description**

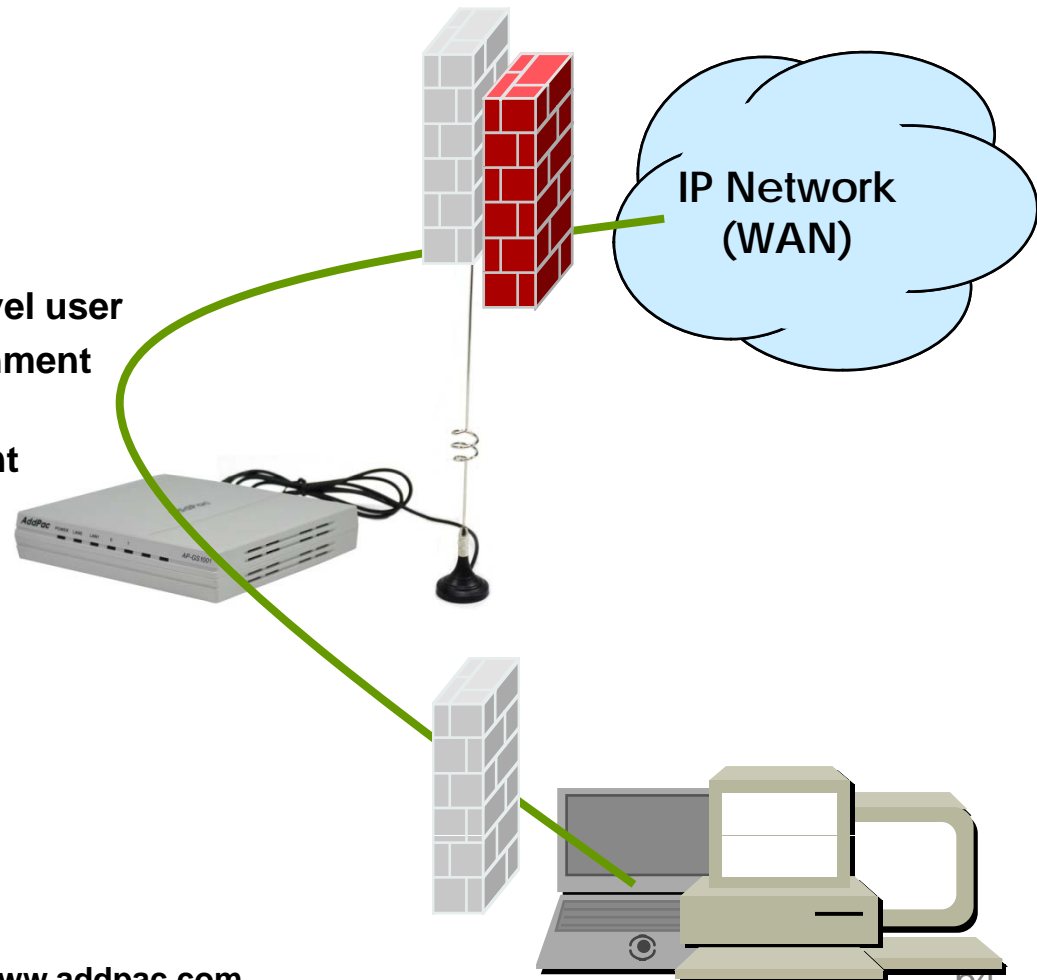
포트의 상태 정보를 확인하고 현재 통화 정보를 조회합니다.

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# Security Management

## AP-GS1001 1-Port FXS GSM Gateway

- IP packet filtering
- IP access list
- User authentication function
  - Password Authentication Protocol (PAP)
  - Challenge Handshake Authentication Protocol (CHAP)
- Enable/Disable specific protocols
- Auto-square connect of Telnet session
- Account Management function for multi-level user
- SNMP/TELNET/FTP/HTTP/TFTP port assignment function
- SNMP/TELNET/FTP access list management
- Boot mode security checking function







# Thank you!

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