



AP-LMS1500

Multi-Port LMS VoIP Gateway

High Performance LMS(Land to Mobile Station) VoIP Gateway Solution

Bluetooth VoIP, 3G VoIP Service



AddPac

AddPac Technology

2013, Sales and Marketing

www.addpac.com

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Bluetooth VoIP Module

Product Overview

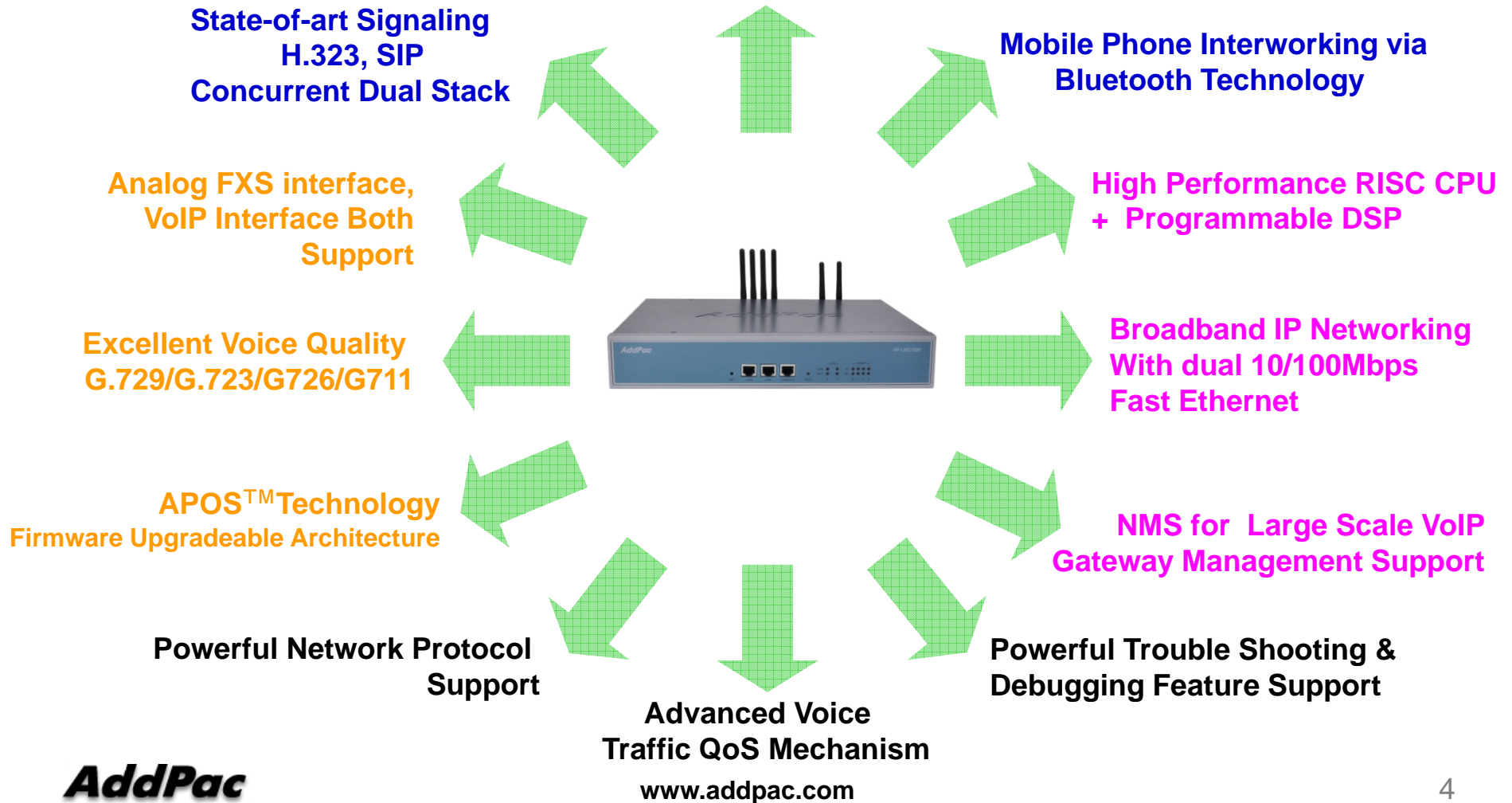
AP-LMS1500 Multi-Port LMS VoIP Gateway

- Two(2) Module Slots for 4-Port Bluetooth Module, 3G WCDMA, 8FXS/8FXO Analog Interface (Up to 8-Port Bluetooth Channel)
- Analog Interface (FXS)/VoIP Interface(LAN) Both Support
- Mobile Phone Interworking Solution via Bluetooth Technology
- 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
- Smart Web Manager for LMS VoIP Gateway
- Smart NMS for Large Scale Deployment
- Advanced Voice QoS Mechanism
- Light and Compact Design with Internal Power Supply

Product Highlights

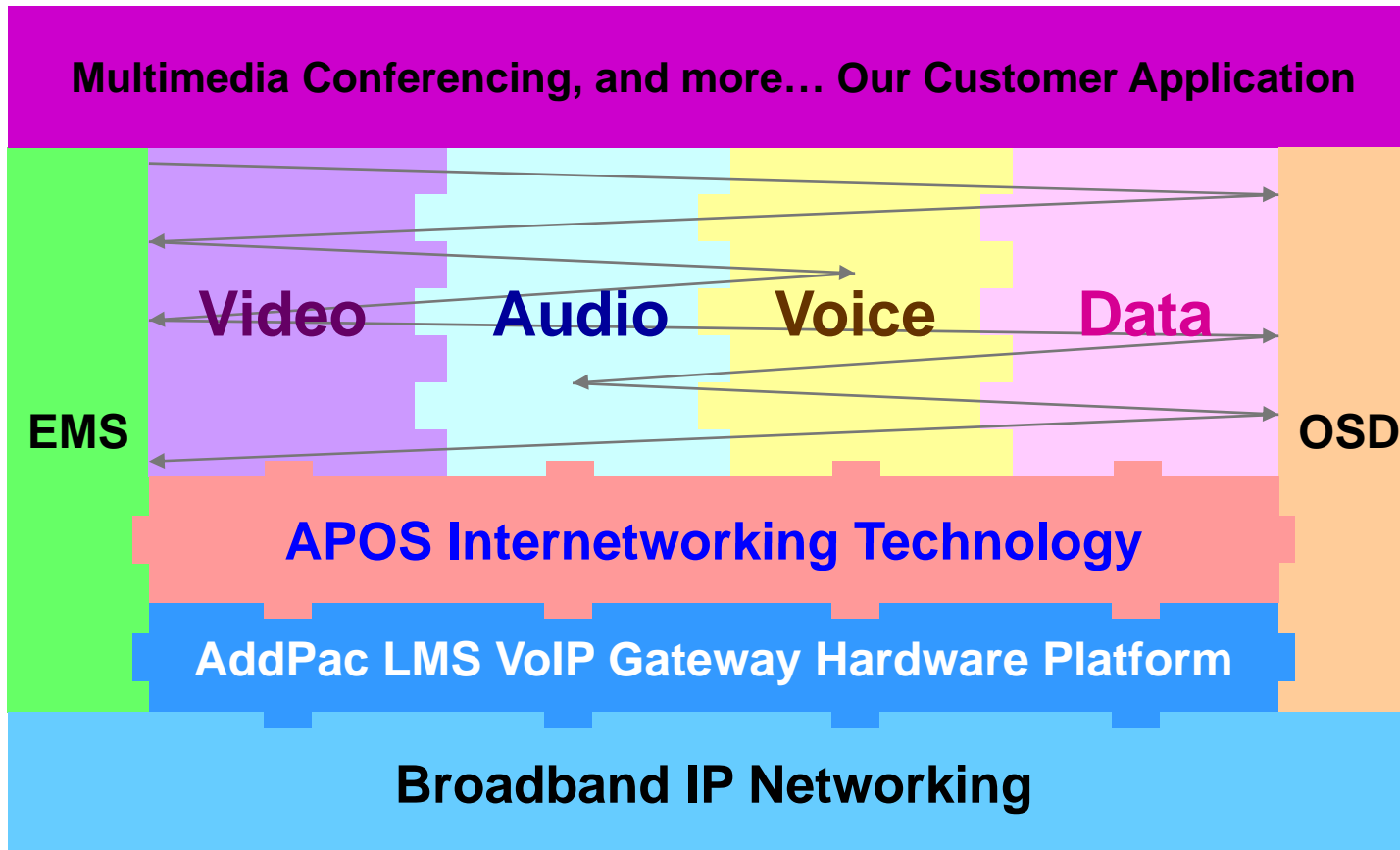
AP-LMS1500 Multi-Port LMS VoIP Gateway

Powerful LMS VoIP Gateway



APOS Technology

AP-LMS1500 Multi-Port LMS VoIP Gateway



- APOS : AddPac Internetworking Operating System
- OSD : On- Screen Display
- EMS : Element Management System

Hardware Specification

AP-LMS1500 Multi-Port LMS VoIP Gateway

RISC
CPU

High-end
DSP

- RISC Microprocessor Computing Power
- Two(2) Module Slot for Bluetooth, 3G, Analog Interface
- 4-Port Bluetooth Module(AP-N1-LMS4)
 - Four(4) Bluetooth Antenna Interface
 - Hot-Swap
- VoIP Interface Module
 - 8-Port FXS Module (AP-N1-FXS8)
 - 8-Port FXO Module (AP-N1-FXO8)
 - Hot-Swap
- 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

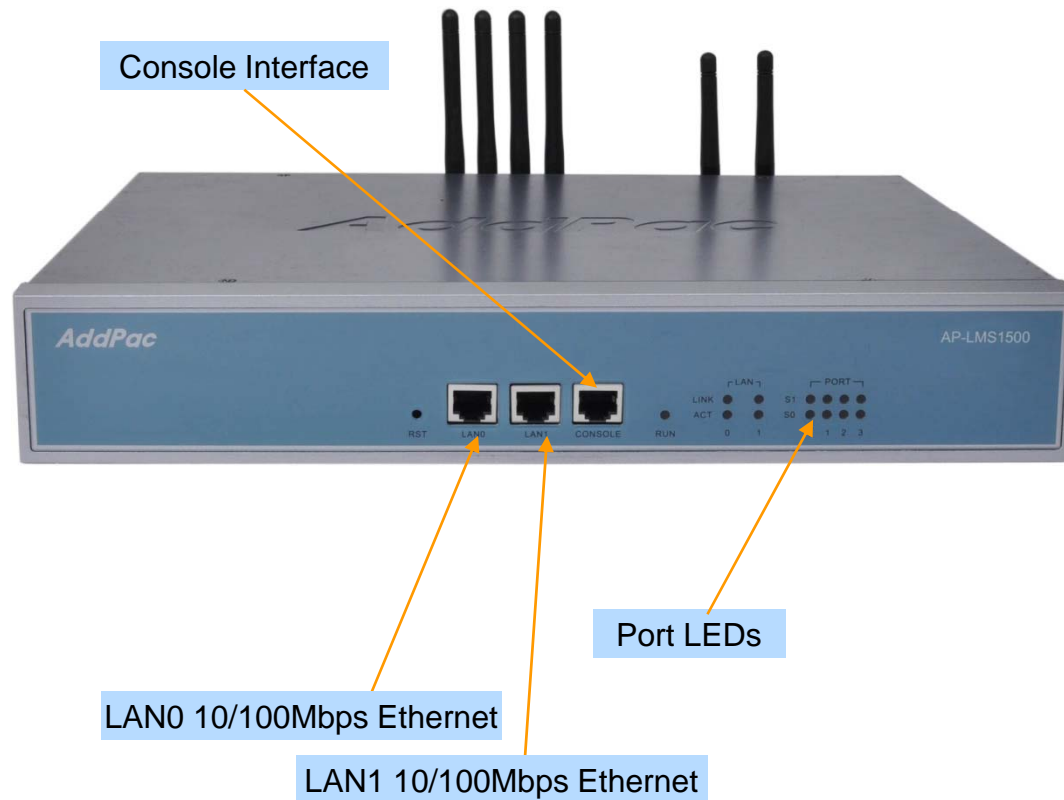
AP-LMS1500 Multi-Port LMS VoIP Gateway

AP-LMS1500 LMS VoIP Gateway	Basic Specifications
Mobile Interface	Two(2) Module Slots for LMS, 3G WCDMA, etc
	AP-N1-LMS4, AP-N1-3G4I, AP-N1-FXS8, AP-N1-FXO8, AP-N1-FXS4O4
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	32Mbyte High-speed SDRAM
Power Requirement	Power Supply / VAC 110~220V, 50/60Hz, 5V 15A, 12V 8A
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)
Dimensions	H x W x D (55mm x 340 x 267mm)
Weight(kg)	2.5Kg

Hardware Specification

AP-LMS1500 Multi-Port LMS VoIP Gateway

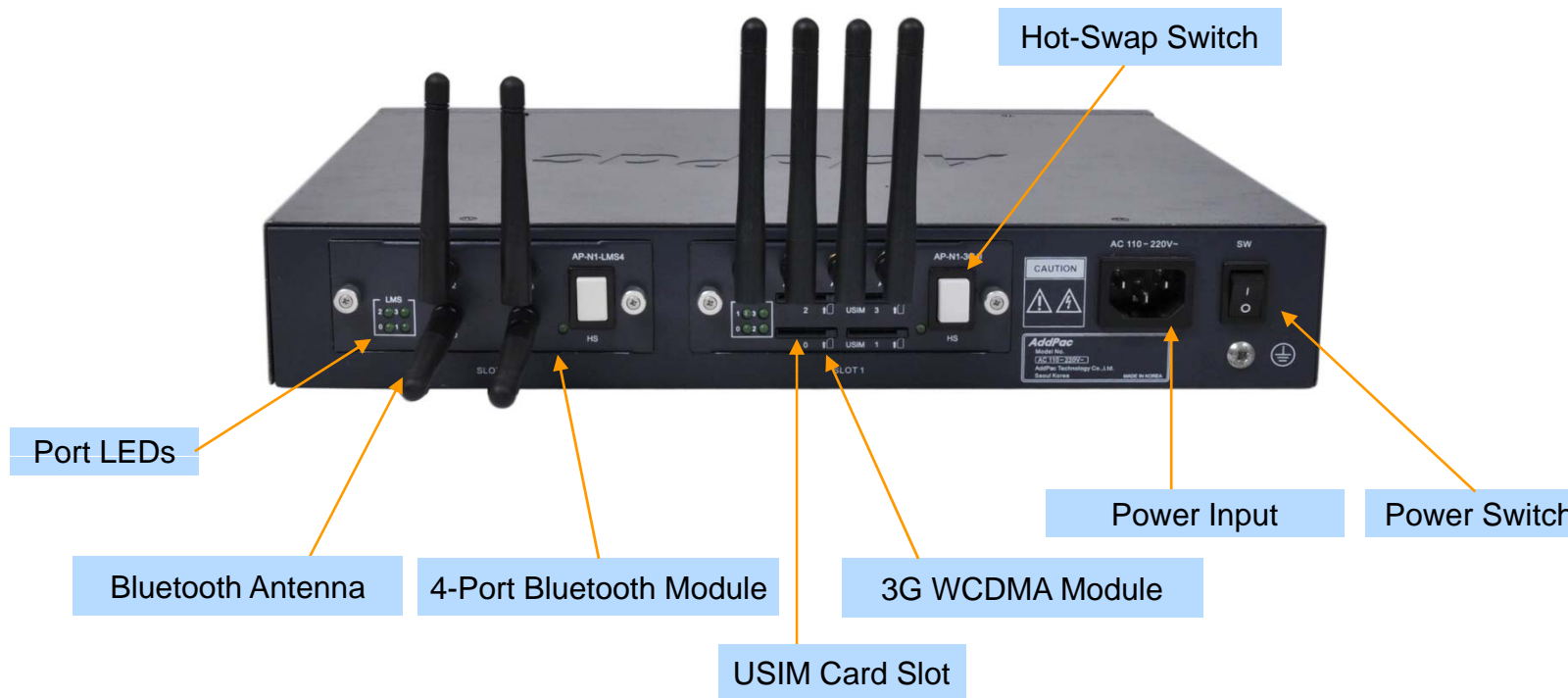
Front Side View



Hardware Specification

AP-LMS1500 Multi-Port LMS VoIP Gateway

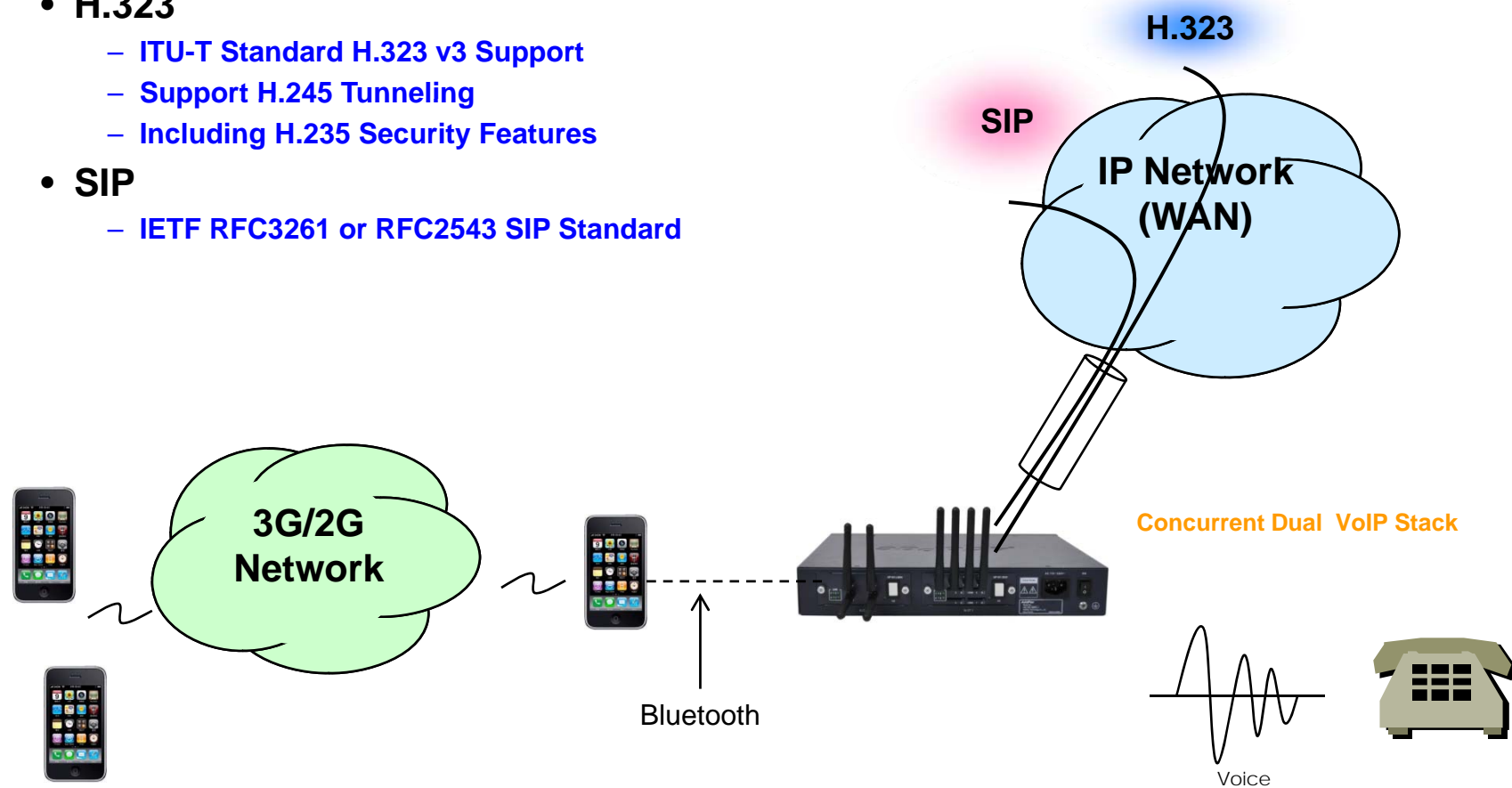
Back Side View (8-Port Bluetooth + 8-Port 3G WCDMA)



VoIP (Voice over IP) Service

AP-LMS1500 Multi-Port LMS 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-LMS1500 Multi-Port LMS 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

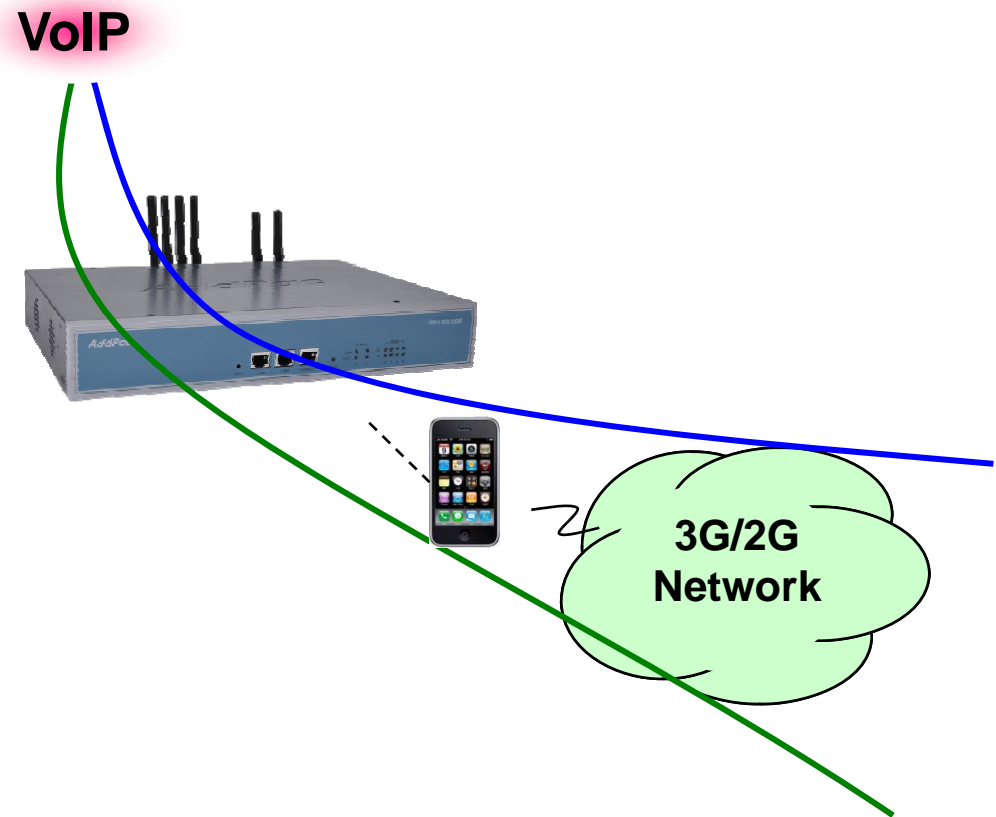
AP-LMS1500 Multi-Port LMS VoIP Gateway

- **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



VoIP (Voice over IP) Service

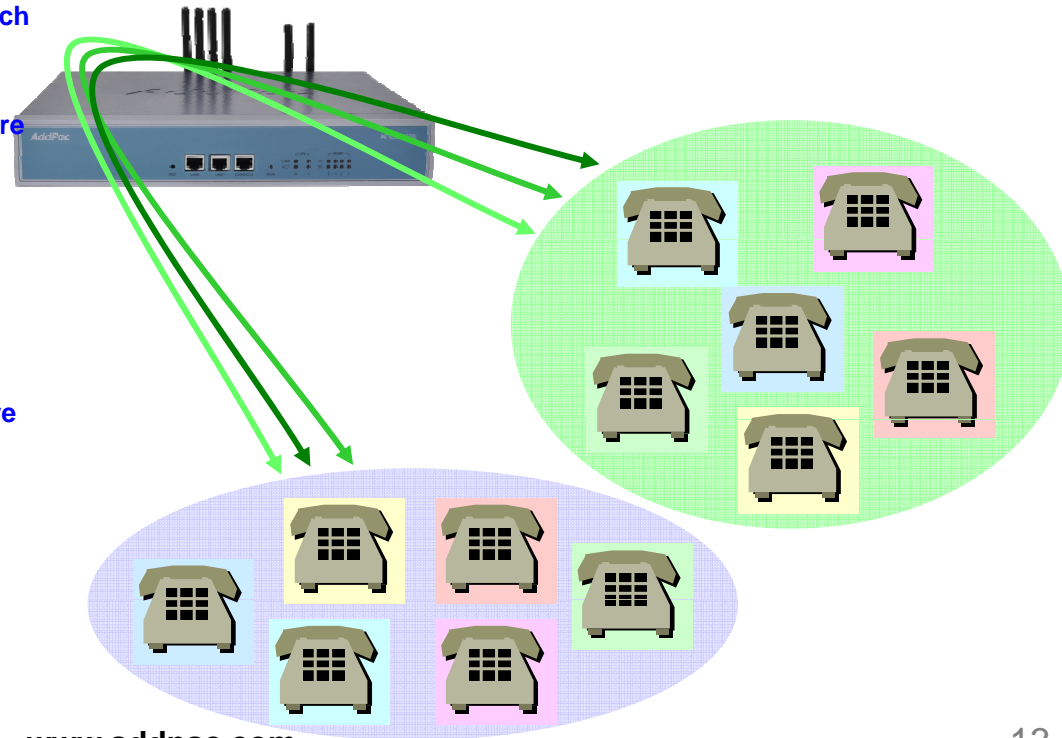
AP-LMS1500 Multi-Port LMS VoIP Gateway

• 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.)

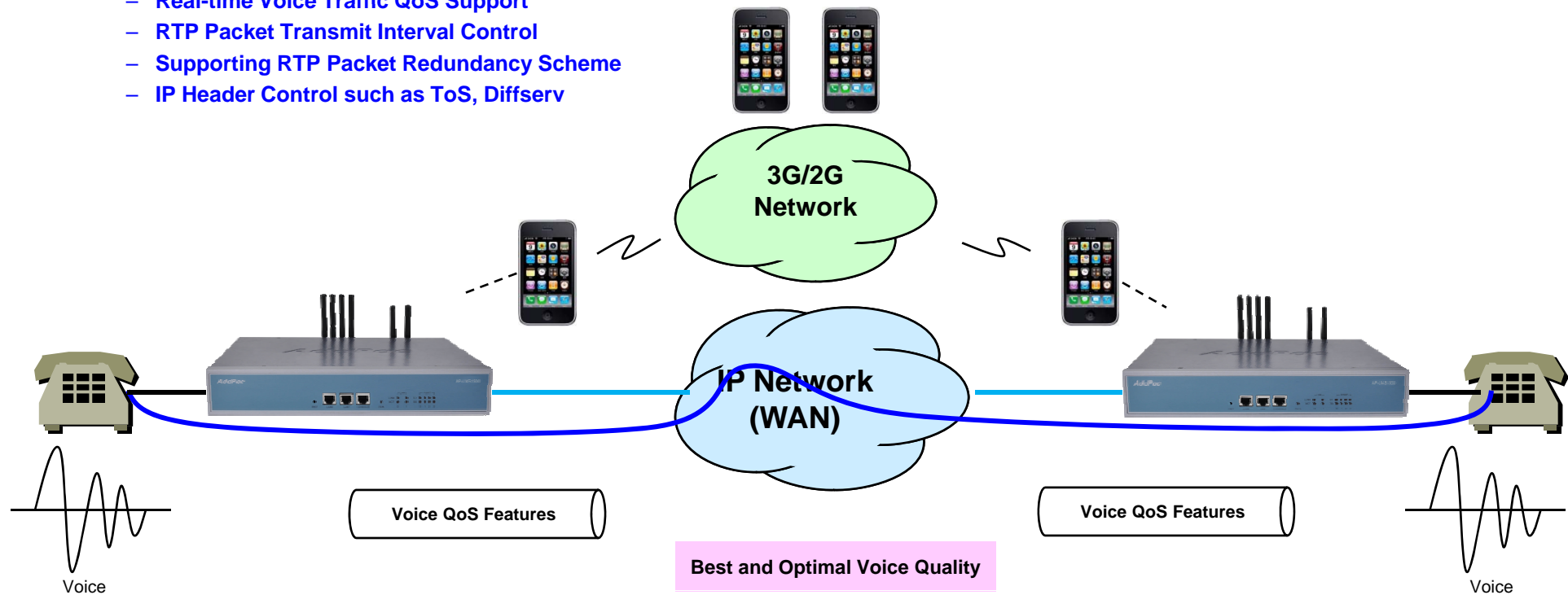
- Call transfer for internal calls
- Call pickup for internal calls
- Calling and called number conversion for VoIP outbound calls
- Calling and called number conversion for VoIP inbound calls
- Fax broadcasting call control



Advanced QoS Features

AP-LMS1500 Multi-Port LMS 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-LMS1500 Multi-Port LMS VoIP Gateway

Basic Network Protocols

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

Routing Protocol

- IPv4 : Static

Service Protocol

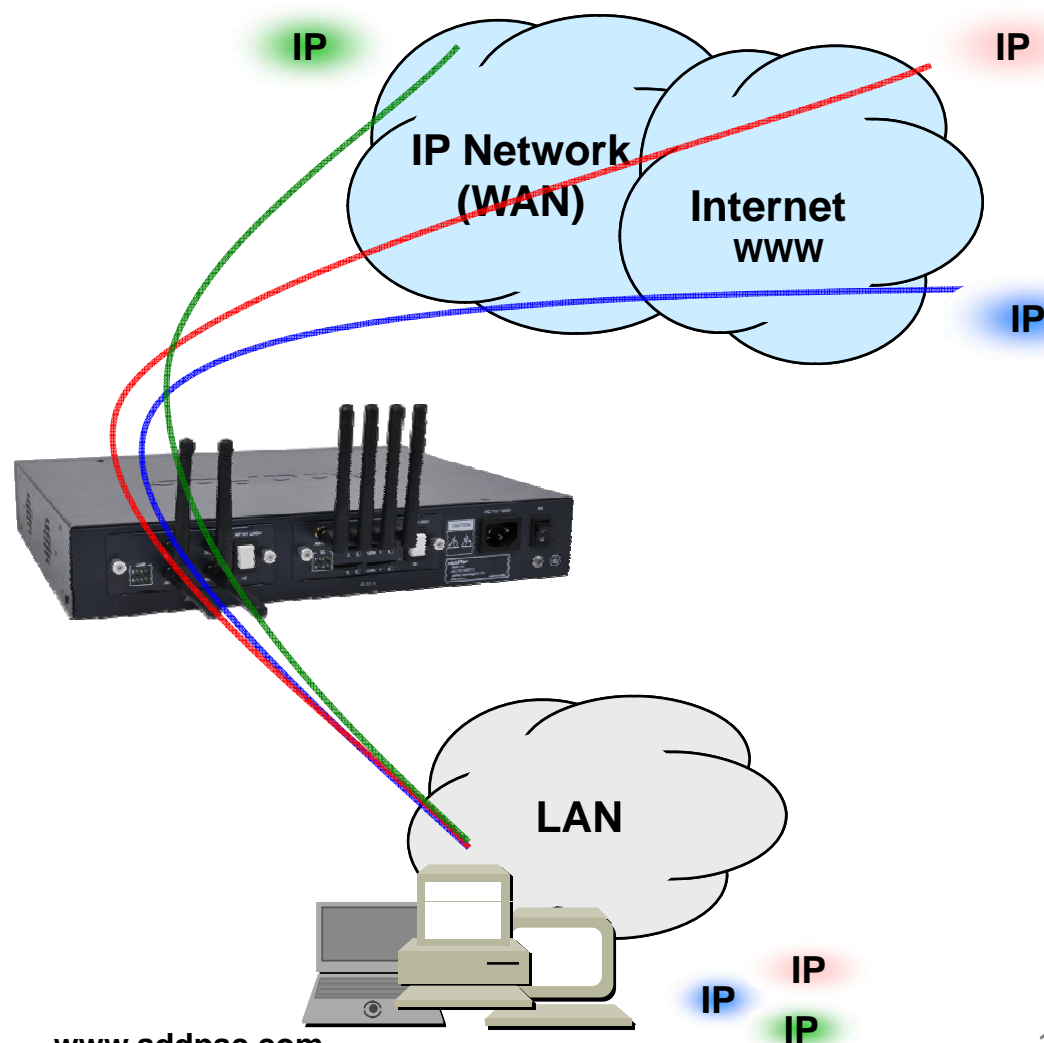
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

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-LMS1500 Multi-Port LMS 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**

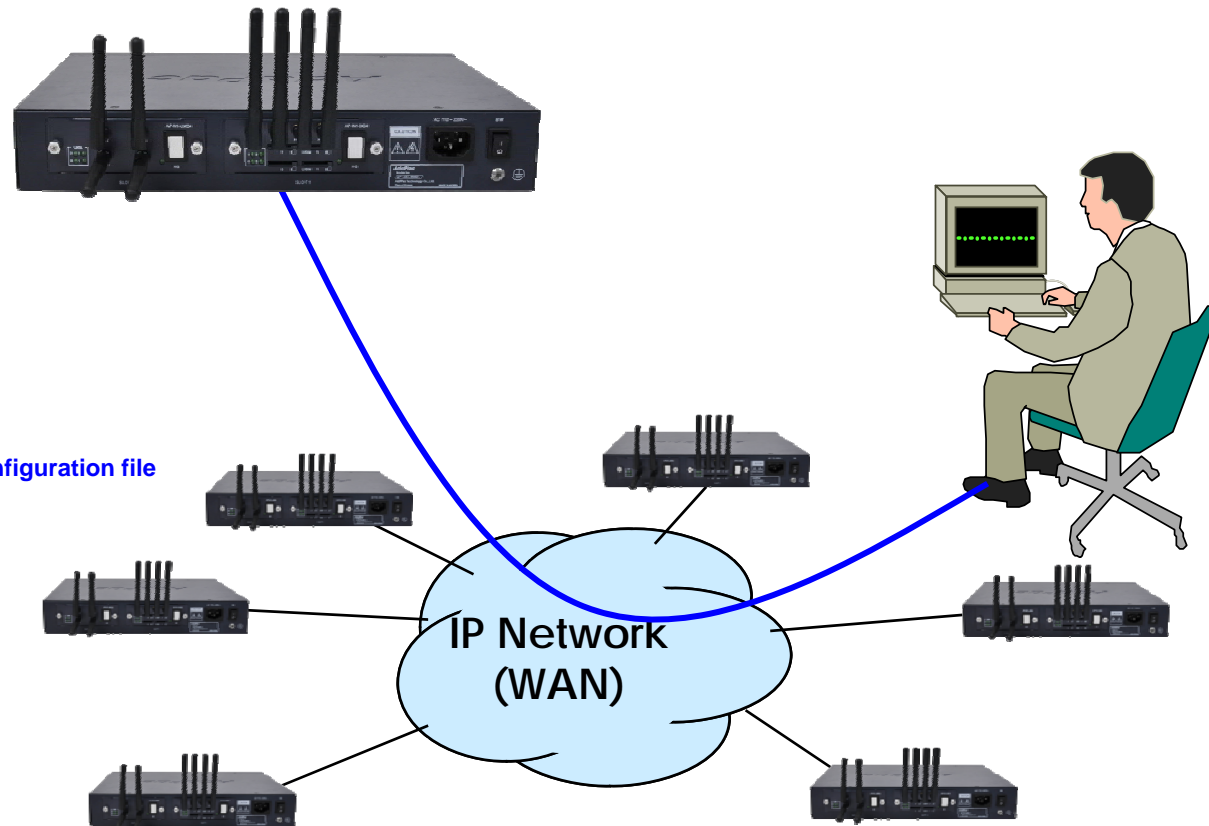
- HTTP server based APOS image and configuration file auto-upgrade support

- **Batch Job Function**

- Text based script downloading

- **Interoperable with AP-VPMS Service**

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



Smart Web Manager : Main Page Layout

AP-LMS1500 Multi-Port LMS VoIP 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 : SIP Server (Example)

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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

Smart Web Manager : FXS Extension (Example)

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Port Information
voice port type & physical port

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"/>

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

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)

Smart Web Manager : Call Status (Example)

AP-LMS1500 Multi-Port LMS VoIP Gateway

The screenshot shows the 'Call Status' page in the Smart Web Manager interface. The page is divided into several sections:

- System:** A sidebar menu with options like Network Setup, Language, NAT, DTMF, DTMF/CODEC, VoIP Dial Plan, GSM Dial Plan, Static Route, Hot Line, Security, SNMP, WEB Callback, GSM Callback, and Miscellaneous.
- Call Status:** The main content area, which includes:
 - Port Status (Analog):** A table showing the status of four ports (0, 1, 2, 3) in Slot 0. All ports are currently 'Idle'. Below the table are 'Unblock' and 'Block' buttons.
 - Call Status:** A table with columns for Port, Direction, Established Time, Calling Number, Called Number, CODEC, and Src/Dest. IP.
 - Legend:** A color-coded legend for Connection State (Connected, Disconnected, Blocked) and Call State (Idle, Ring, Called, Calling, Blocked).
- Information:** A sidebar on the right containing system details such as Model (GS1002_G2), H/W Version (2.0), S/W Version (8.00d), Smart Web Version (0.4), and Voice Interface details.
- Description:** A sidebar on the right with the text: 'Verify port status and retrieve the present call information'.

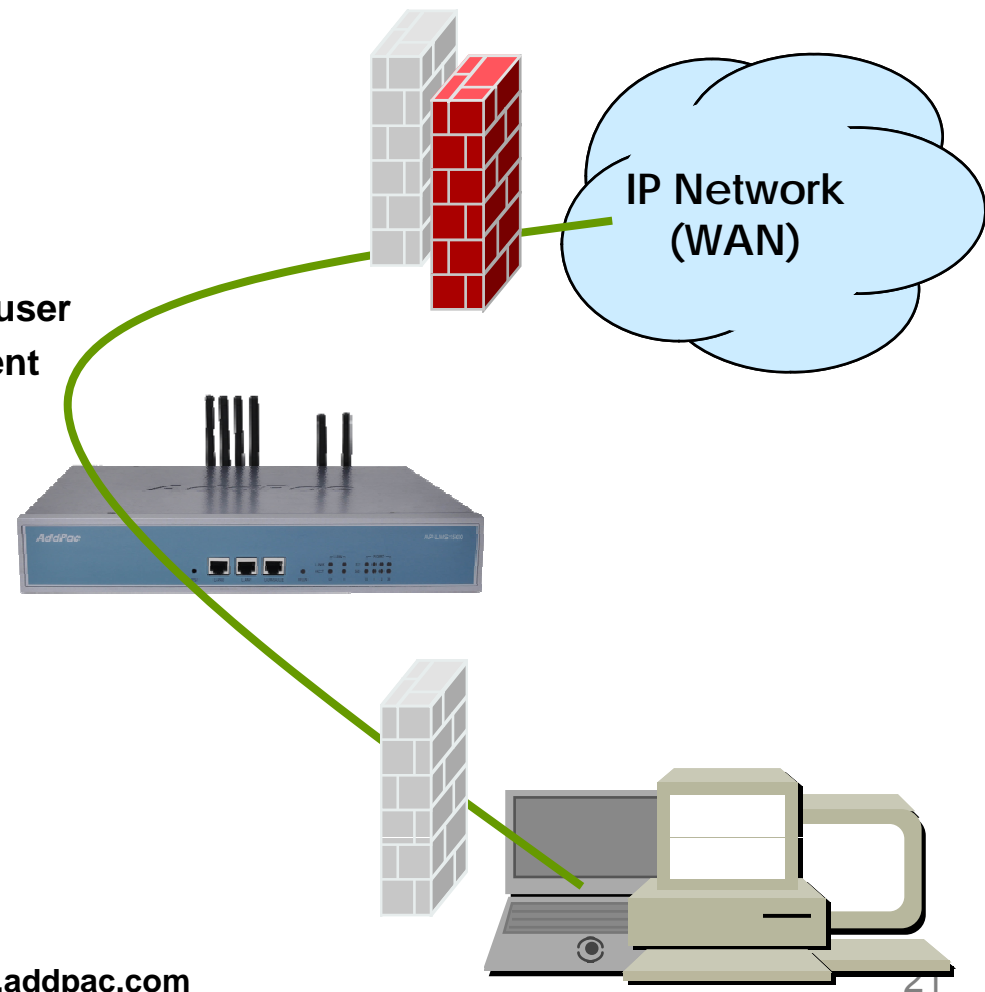
Two callout boxes provide additional context:

- 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).

Security Management

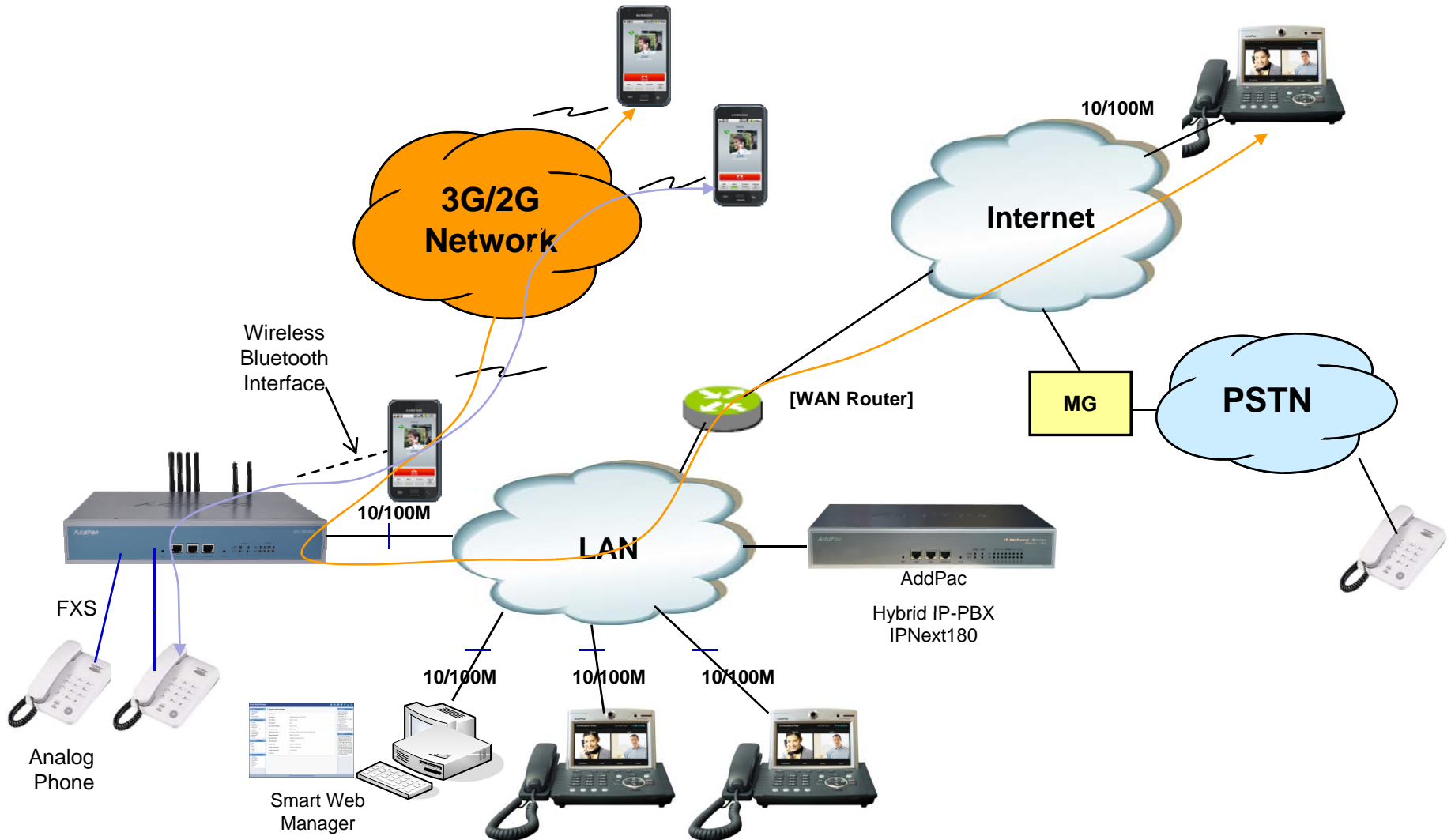
AP-LMS1500 Multi-Port LMS VoIP Gateway

- 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



Standard Application

AP-LMS1500 Multi-Port LMS VoIP Gateway



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Ordering Information

- AP-LMS1500 Multi-Port LMS VoIP Gateway Hardware
 - AP-LMS1500 LMS VoIP Gateway Main Body
 - RISC Microprocessor with High-end Programmable DSP Architecture
 - Two(2) Module Slots
 - 2-ports 10/100Mbps Fast Ethernet(RJ45)
 - Including Network Cable, Antenna & Internal Power Supply, etc.
- Built-in APOS Internetworking Software for AP-LMS1500
- Including 1 Year Hardware Warranty
- Product Documents
 - Install and Operation Guide (PDF)
- Pricing
 - AddPac Technology Regional Sales Manager
 - Authorized Sales and Marketing Representatives
 - Please Contact www.addpac.com

LMS VoIP Gateway Series

Thank you!

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Sales and Marketing

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