

AP-GDP120™ GSM Desk Phone

High Performance GSM Multiservice Phone Solution



Preliminary Product Overview

(Without notice, following described technical spec. can be changed)



AddPac

AddPac Technology

2011, Sales and Marketing

www.addpac.com

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- Hardware Specification
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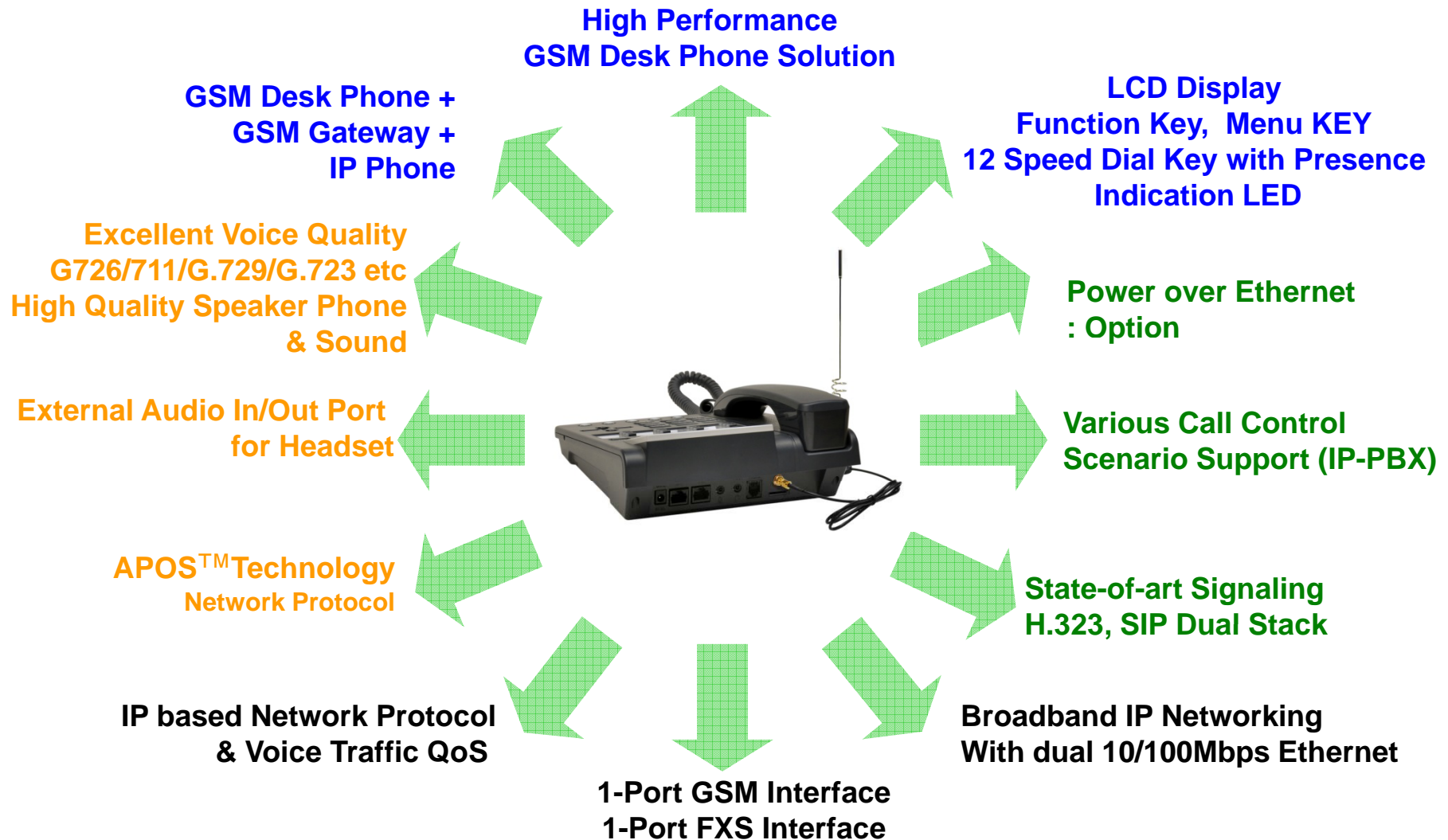
Product Overview

AP-GDP120 GSM Desk Phone

- GSM Desk Phone Solution
- GSM Gateway Solution
- IP Phone Solution
- VoIP Gateway Solution
- IP Phone + VoIP Gateway Solution (concurrent)
- 12 Speed-Dial Key with Presence Indication Lamp
- High-performance Audio, and Voice Service
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection

Product Highlights

AP-GDP120 GSM Desk Phone



Multimedia Service

AP-GDP120 GSM Desk Phone



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

Hardware Specification

AP-GDP120 GSM Desk Phone

- RISC+DSP Microprocessor Computing Power (Dual Processor Architecture)
- 1-Port GSM Interface : Antenna, SIM Card Slot
- VoIP Gateway Interface
 - [AP-GDP120 Model A: Basic Configuration](#)
 - [AP-GDP120 Model B: Additonal One\(1\) FXS Port](#)
- Optional PoE (Power over Ethernet)
- High quality Audio and Voice Interface
 - Stereo Audio Input Connector
 - Stereo Audio Output Connector
- Network Interface
 - Two(2) 10/100Mbps Fast Ethernet
- LCD Window : Graphic LCD (4 Line Text)
- 12 Speed-Dial Key with Presence Indication LAMP
- Power Supply
 - External Power Adaptor (5V, 2A)

Hardware Specification

AP-GDP120 GSM Desk Phone

Hardware Specifications

AP-GDP120 GSM Desk Phone	Basic Specifications
CPU	RISC Microprocessor
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
GSM Interface	1-Port Antenna Interface, SIM Card Slot
GSM Gateway Port (Optional)	1-Port FXS Port(RJ-11)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	32Mbyte High-speed SDRAM
Power Requirement	External Power Supply Adaptor / VAC 110~220V, 50/60Hz, 10Watt(5V,2A)
	Power over Ethernet (option)
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 (70mm x 200mm x 210mm)
Weight (g)	1.2Kg

Hardware Specification

AP-GDP120 GSM Desk Phone

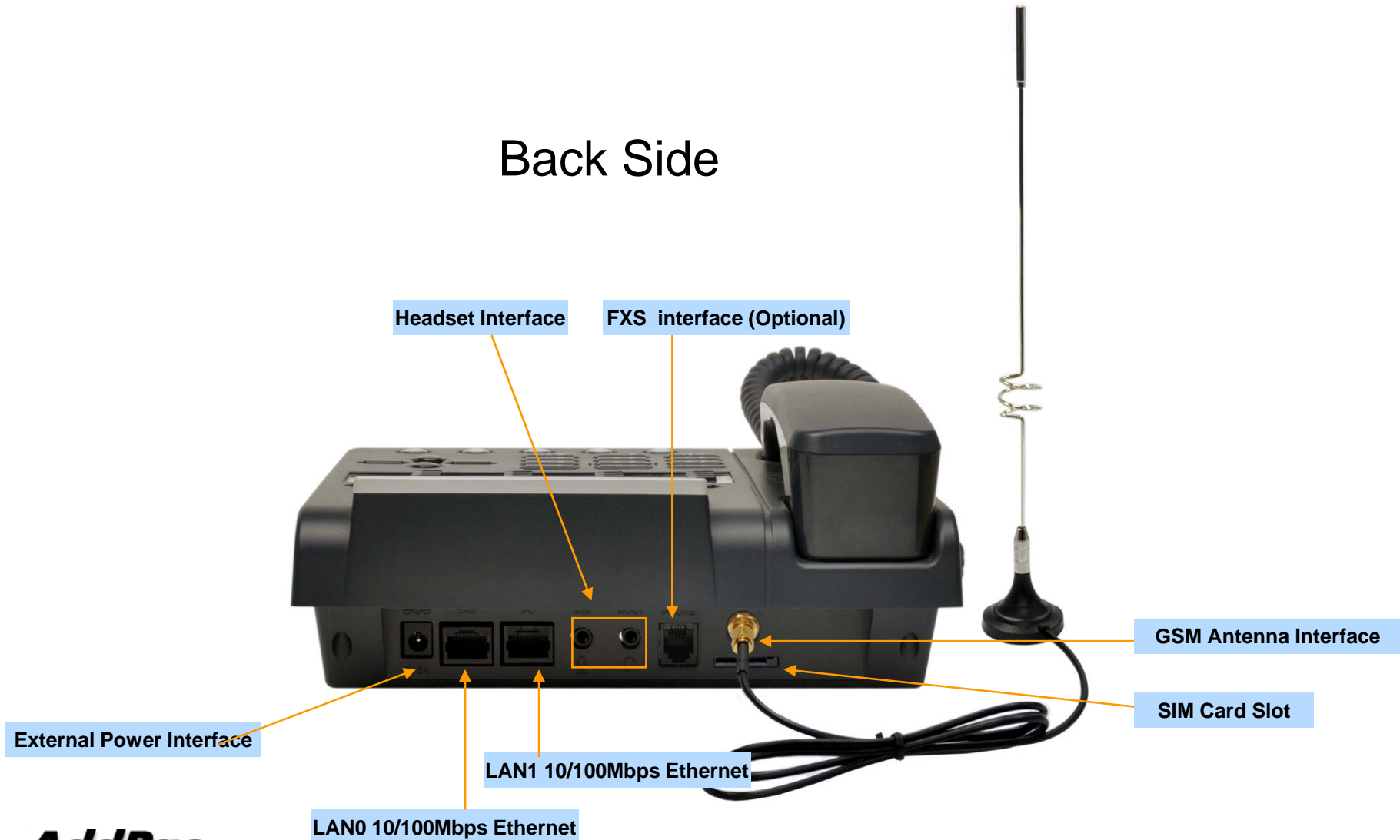
Front Side



Hardware Specification

AP-GDP120 GSM Desk Phone

Back Side



GSM Module Specification

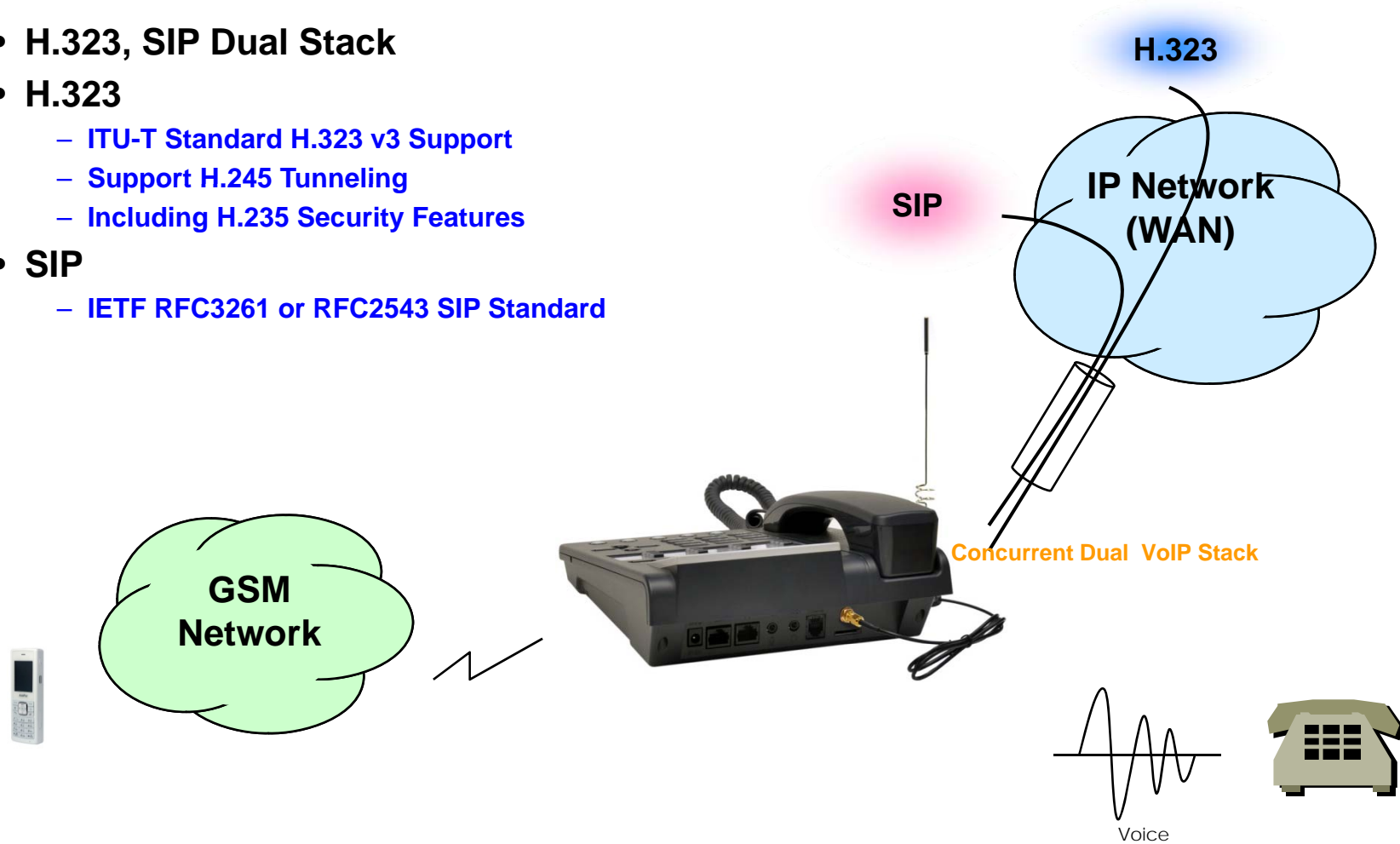
AP-GDP120 GSM Desk Phone

- Bearers : GSM + GPRS multislots class 10
- Quad-Band EGSM 850/900/1800/1900MHz
- Normal Sensitivity
 - 850MHz Rx -102 dBm (min.) -110 dBm (typ)
 - 900MHz Rx -102 dBm (min.) -110 dBm (typ)
 - 1800MHz Rx -102 dBm (min.) -109 dBm (typ)
 - 1900MHz Rx -102 dBm (min.) -109 dBm (typ)
- Tx Performances
 - 850MHz Tx +33 dBm (max.)
 - 900MHz Tx +33 dBm (max.)
 - 1800MHz Tx +30 dBm (max.)
 - 1900MHz Tx +30 dBm (max.)
- Power Class 4 (33dBm nominal maximum output power)
- Codec
 - FR-EFR-HR-AMR

VoIP (Voice over IP) Service

AP-GDP120 GSM Desk Phone

- 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-GDP120 GSM Desk Phone

- **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-GDP120 GSM Desk Phone

- **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

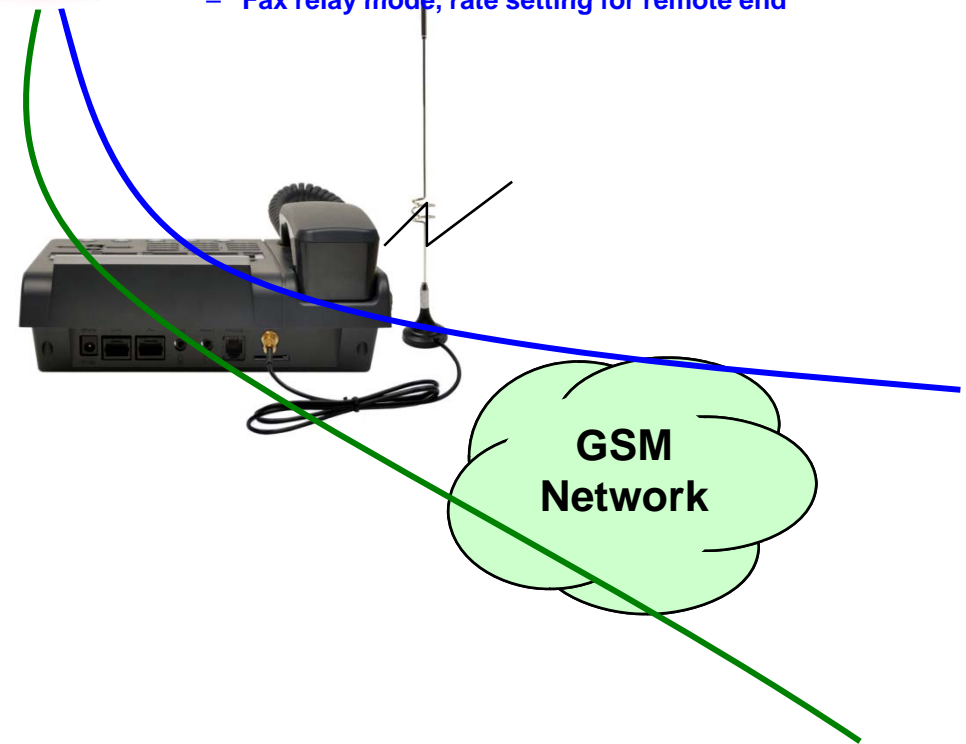
- **RTP**

- 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



VoIP (Voice over IP) Service

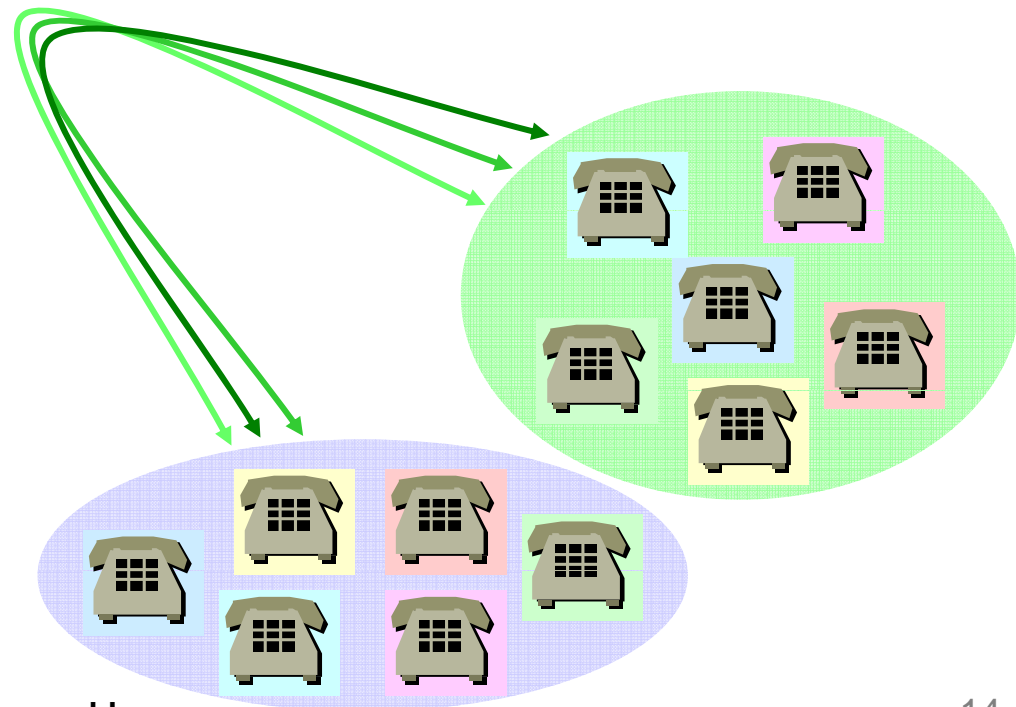
AP-GDP120 GSM Desk Phone

• 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

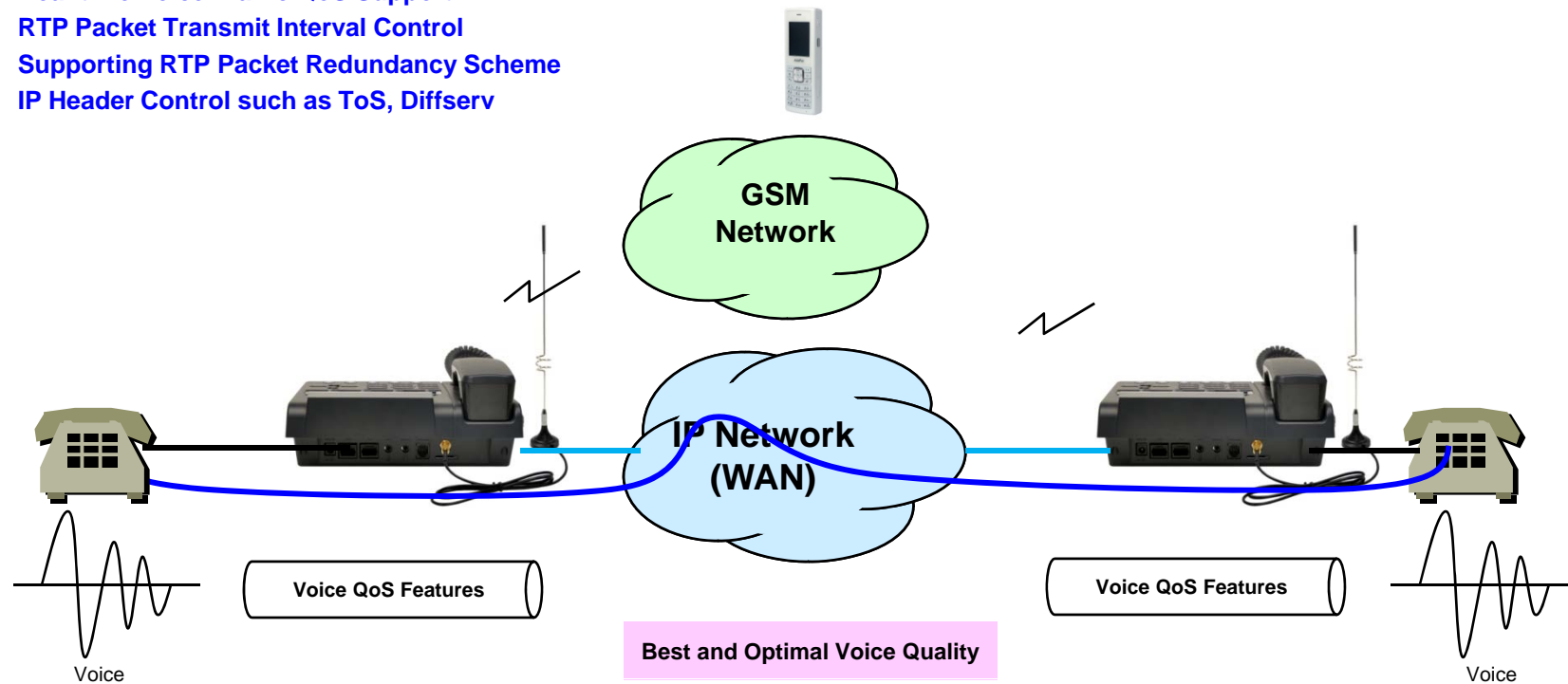
AP-GDP120 GSM Desk Phone

- 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-GDP120 GSM Desk Phone

Basic Network Protocols

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

Routing Protocol

- IPv4 : Static

Service Protocol

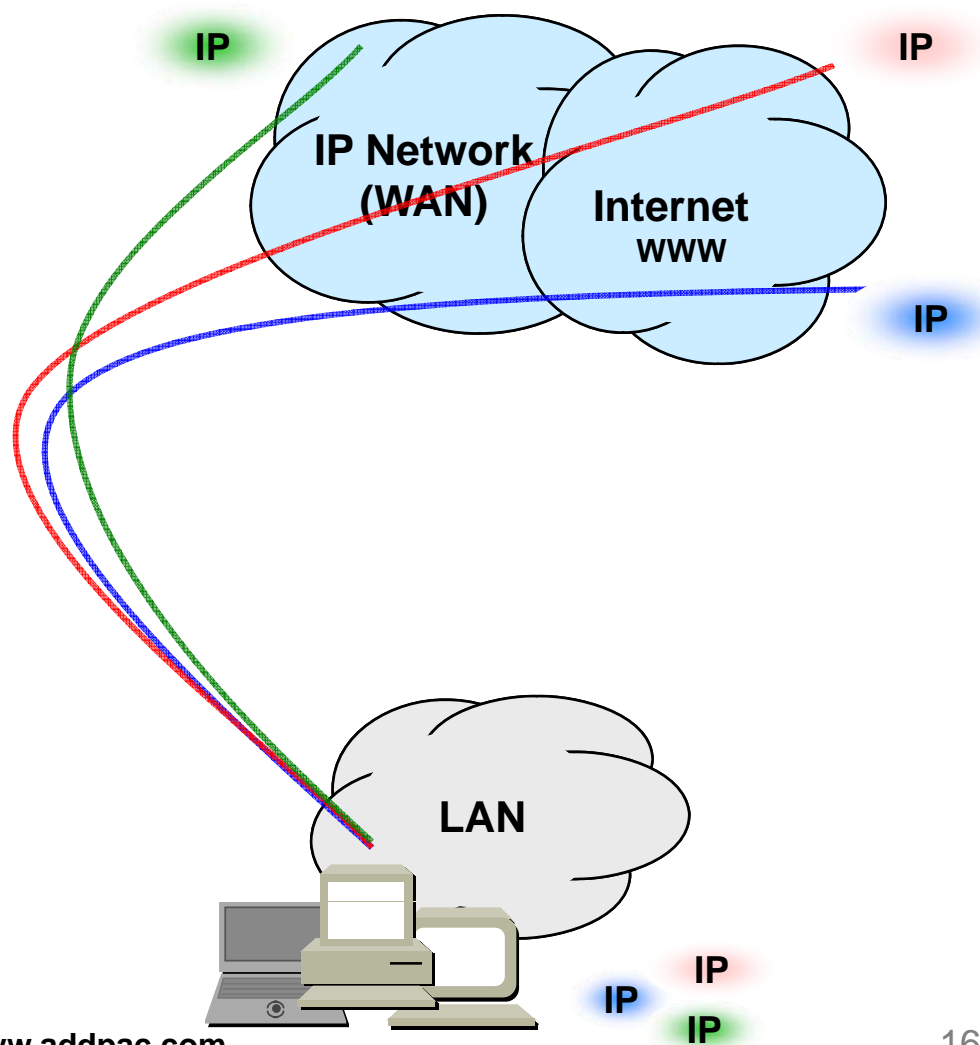
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

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-GDP120 GSM Desk Phone

- **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**

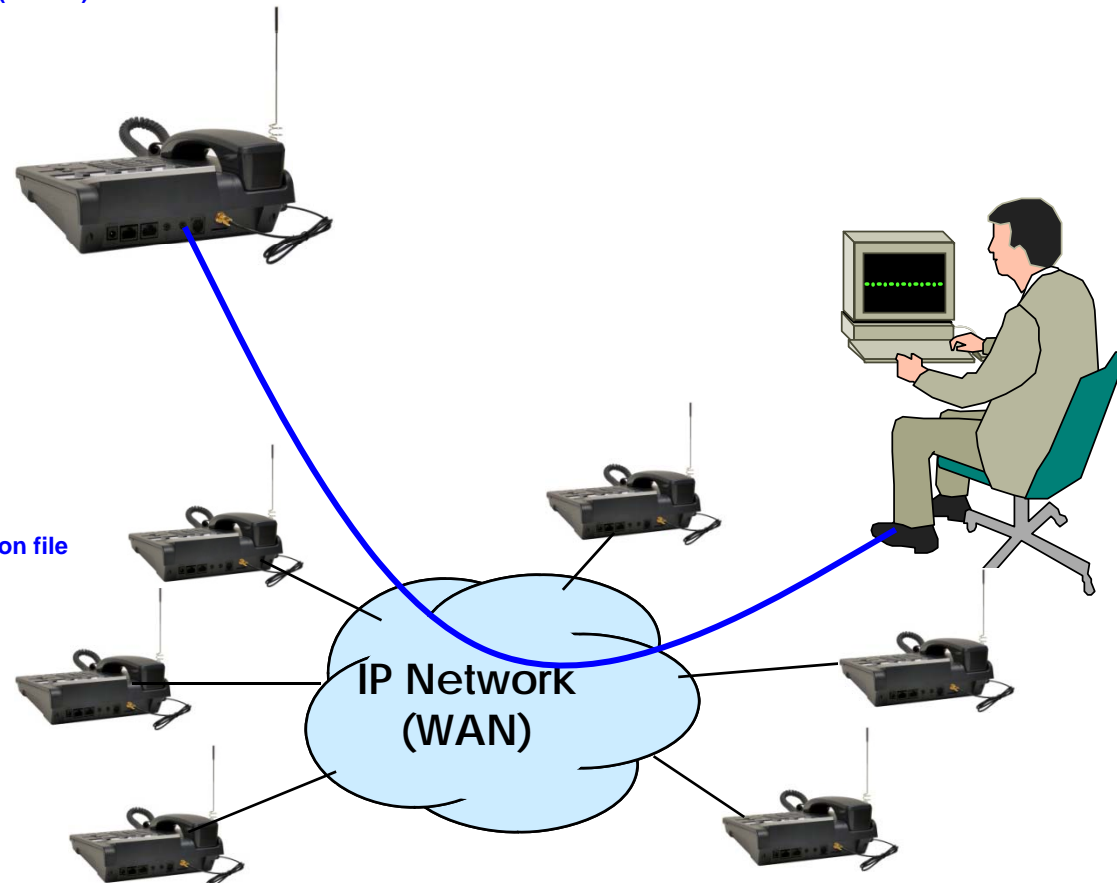
- HTTP server based APOS image and configuration file auto-upgrade support

- **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

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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
www.addpac.com

SIP (Session Initiation Protocol)

Use SIP Server Yes No

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

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

Local Domain name (SIP userpart of authentication)

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

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

Session Re-Fresh INVITE UPDATE

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

Apply

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 : GSM Extension (Example)

AP-GDP120 GSM Desk Phone

The screenshot shows the Smart Web Manager interface for configuring GSM extensions. The main content area is titled "GSM Extension" and is divided into three sections: "Port Information", "GSM Extension Configuration", and "GSM Extension with Translation".

Port Information: A table showing port configurations for SLOT0, P0, P1, P2, and P3. P0 and P1 are set to GSM, while P2 and P3 are set to FXS.

Port	P0	P1	P2	P3
SLOT0	GSM	GSM	FXS	FXS

GSM Extension Configuration: A table for configuring individual extensions. The first row shows index 0, port 0/0, numbers 'T', preference 0, and huntstop 'X'. There are input fields for port, numbers, and preference, and a "Delete" button.

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

GSM Extension with Translation: A table for defining translation rules. The first row shows port P0:0, destination pattern '33', digits to insert '8', and number of digits to delete '1'. The second row shows port P0:1 with empty fields for the other columns. There is an "Apply" button.

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

Callouts:

- 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.

Information Panel (Right):

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 Panel (Right):

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

Smart Web Manager : Call Status (Example)

AP-GDP120 GSM Desk Phone

System

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

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
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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

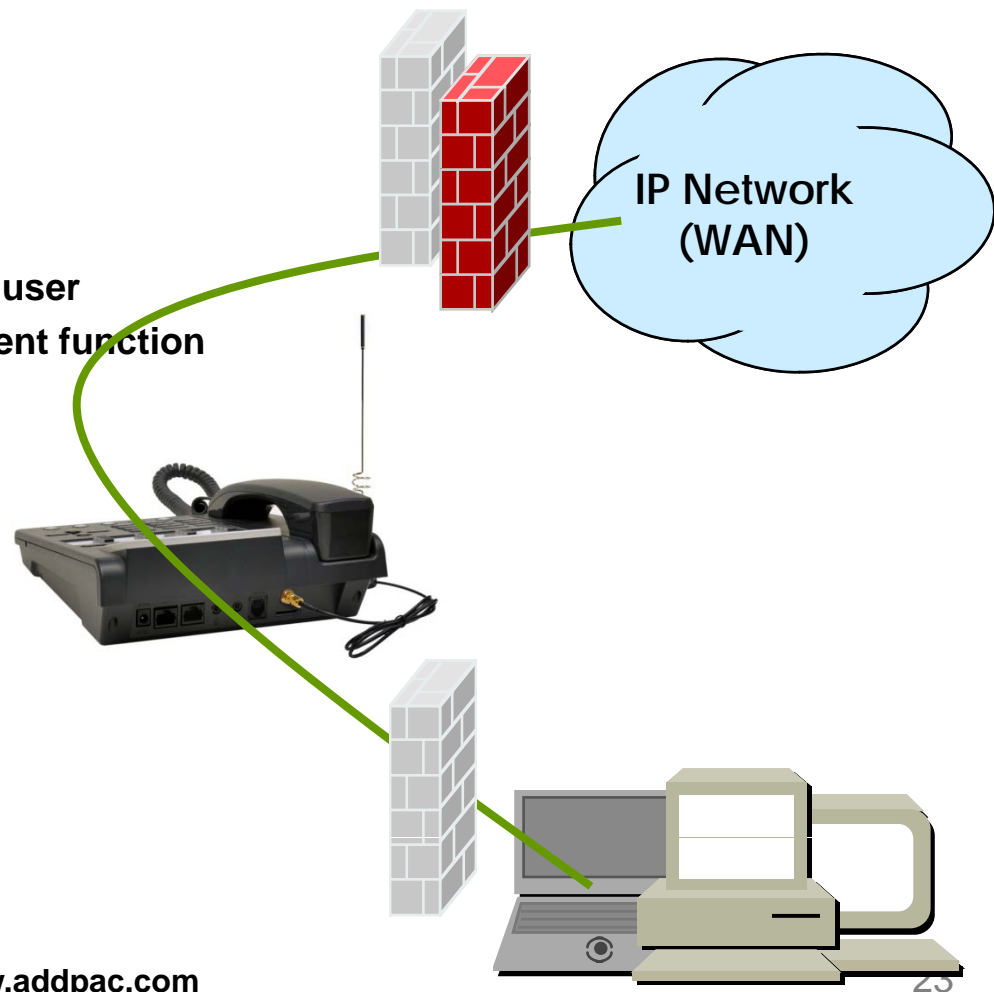
Analog Port
Real-time display about analog port status (occupation, call status). Provide a specific port blocking function

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

Security Management

AP-GDP120 GSM Desk Phone

- 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



Application Area

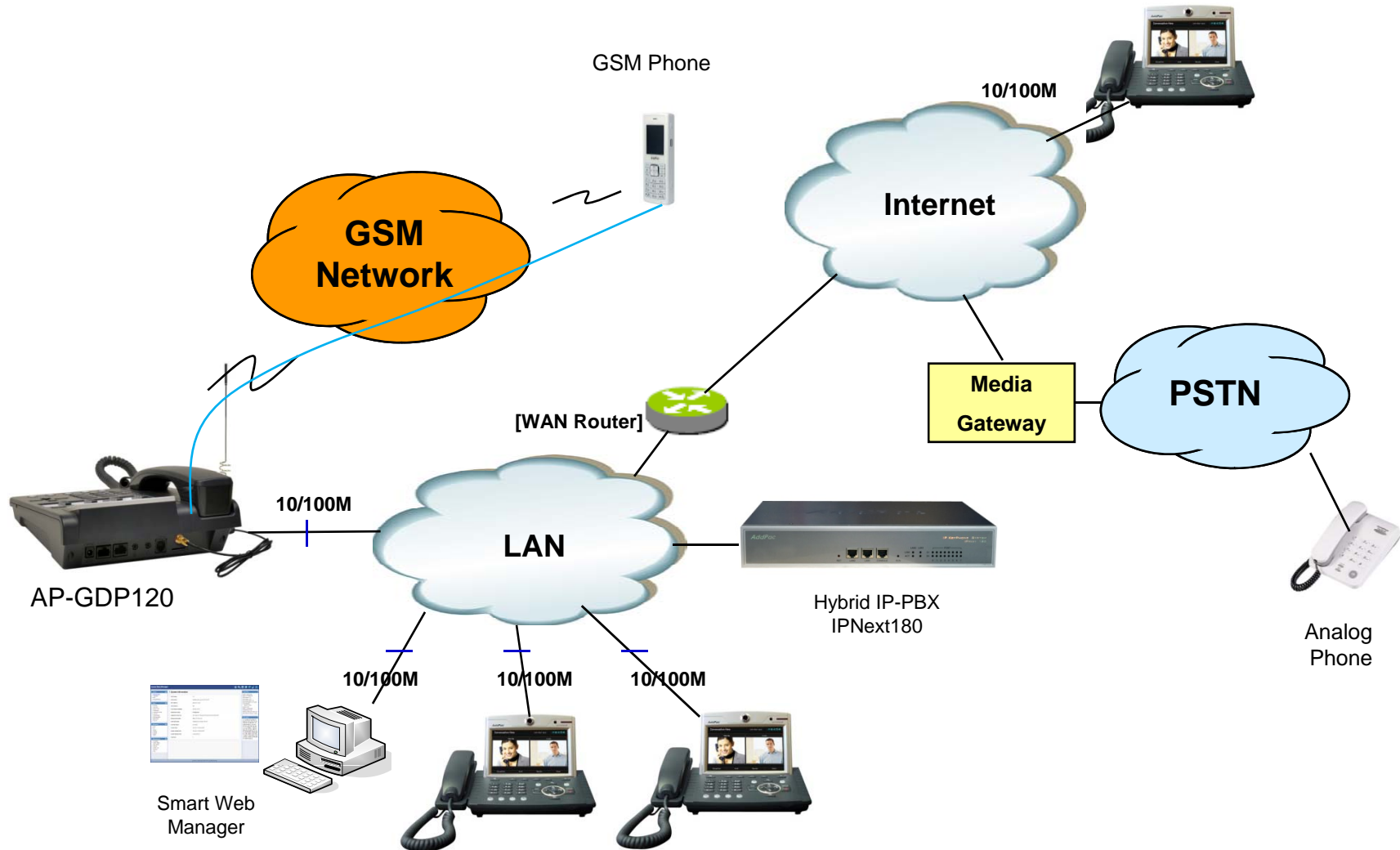
AP-GDP120 GSM Desk Phone

- GSM Desk Phone Application
- GSM Gateway Application (FXS to GSM)
- GSM Gateway Application (IP to GSM)
- IP Phone Application (IP-PBX Application)
- VoIP Gateway Application
- GSM Desk Phone + VoIP Gateway



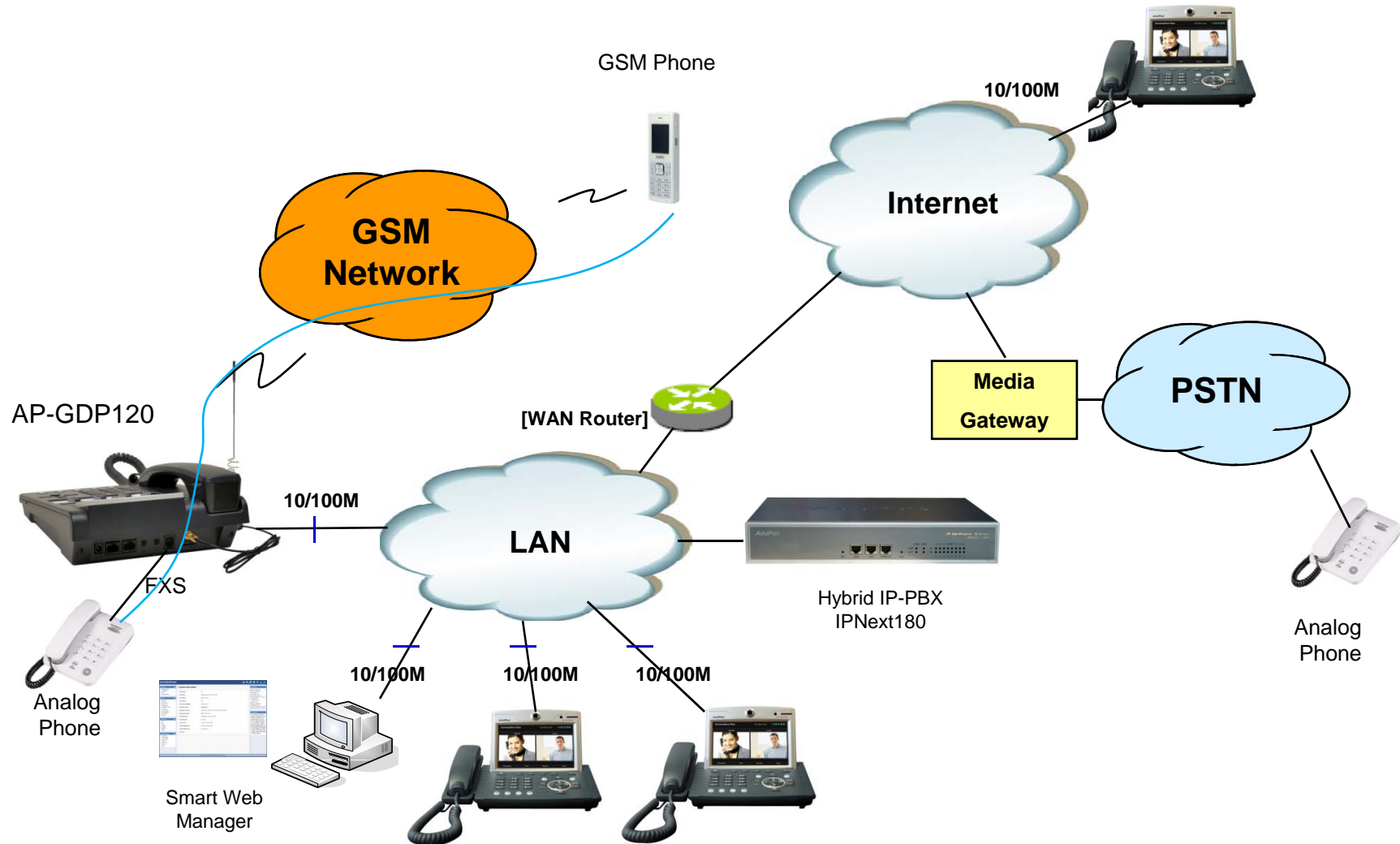
GSM Desk Phone Application

AP-GDP120 GSM Desk Phone



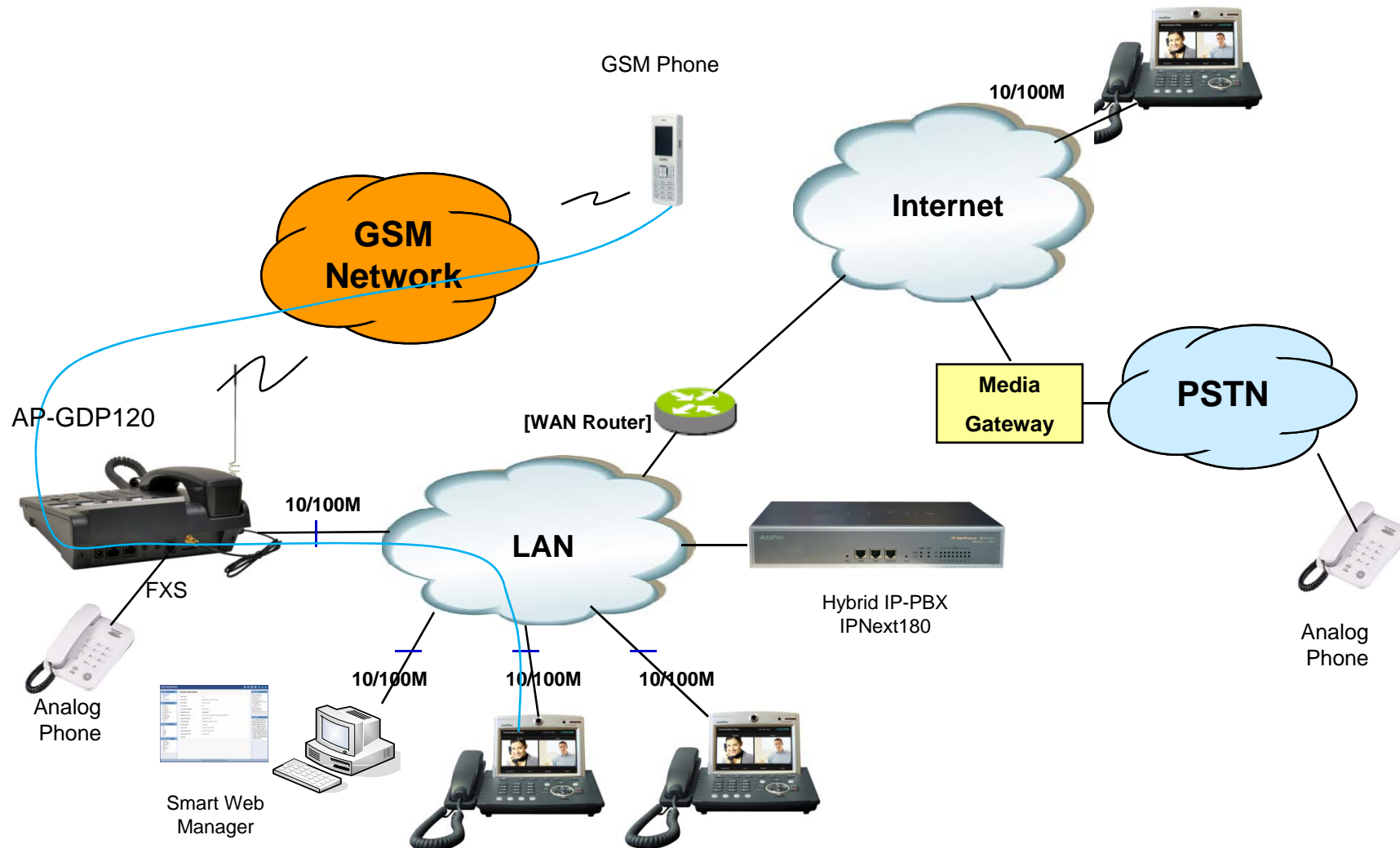
GSM Gateway Application (FXS to GSM)

AP-GDP120 GSM Desk Phone



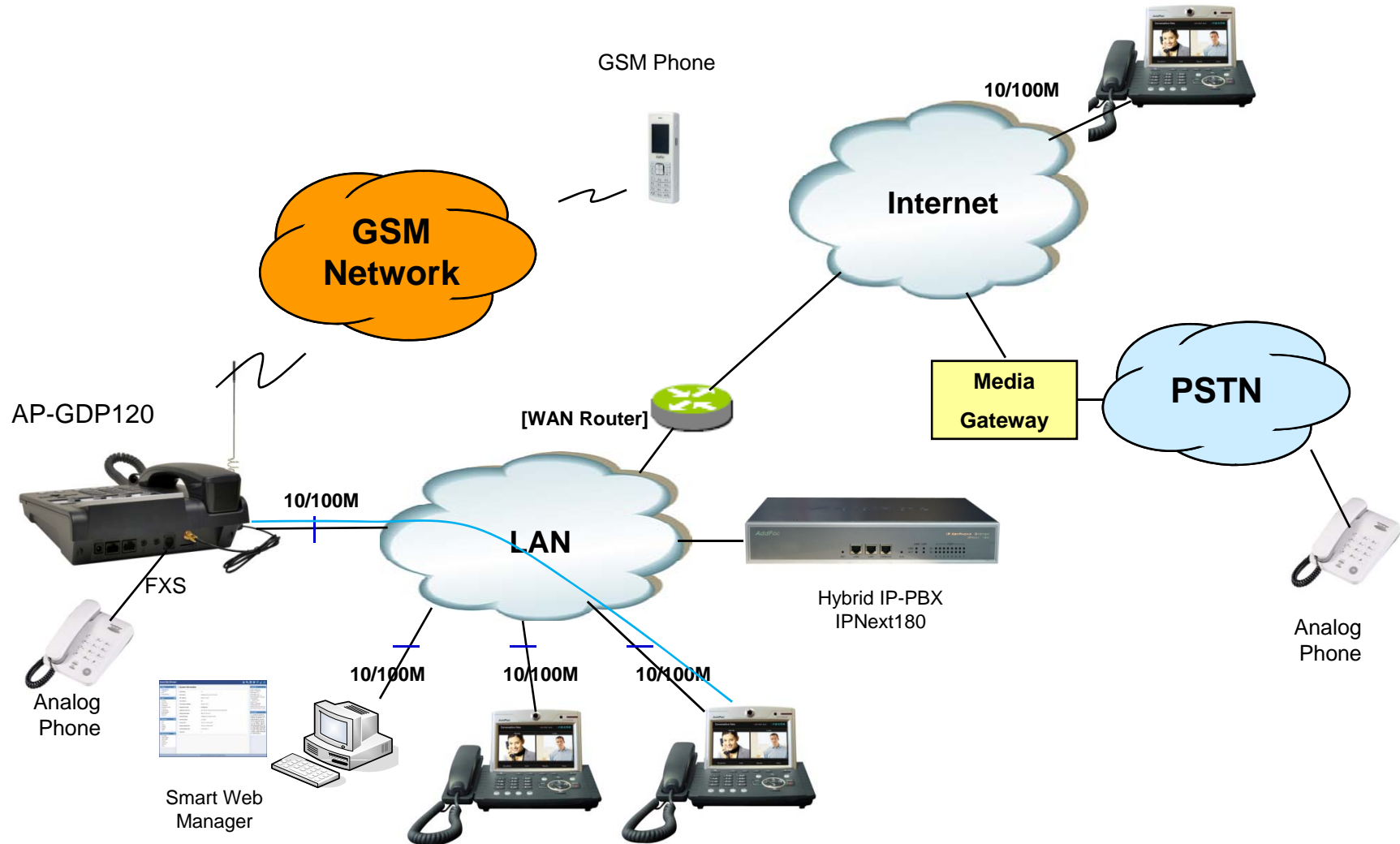
GSM Gateway Application (IP to GSM)

AP-GDP120 GSM Desk Phone



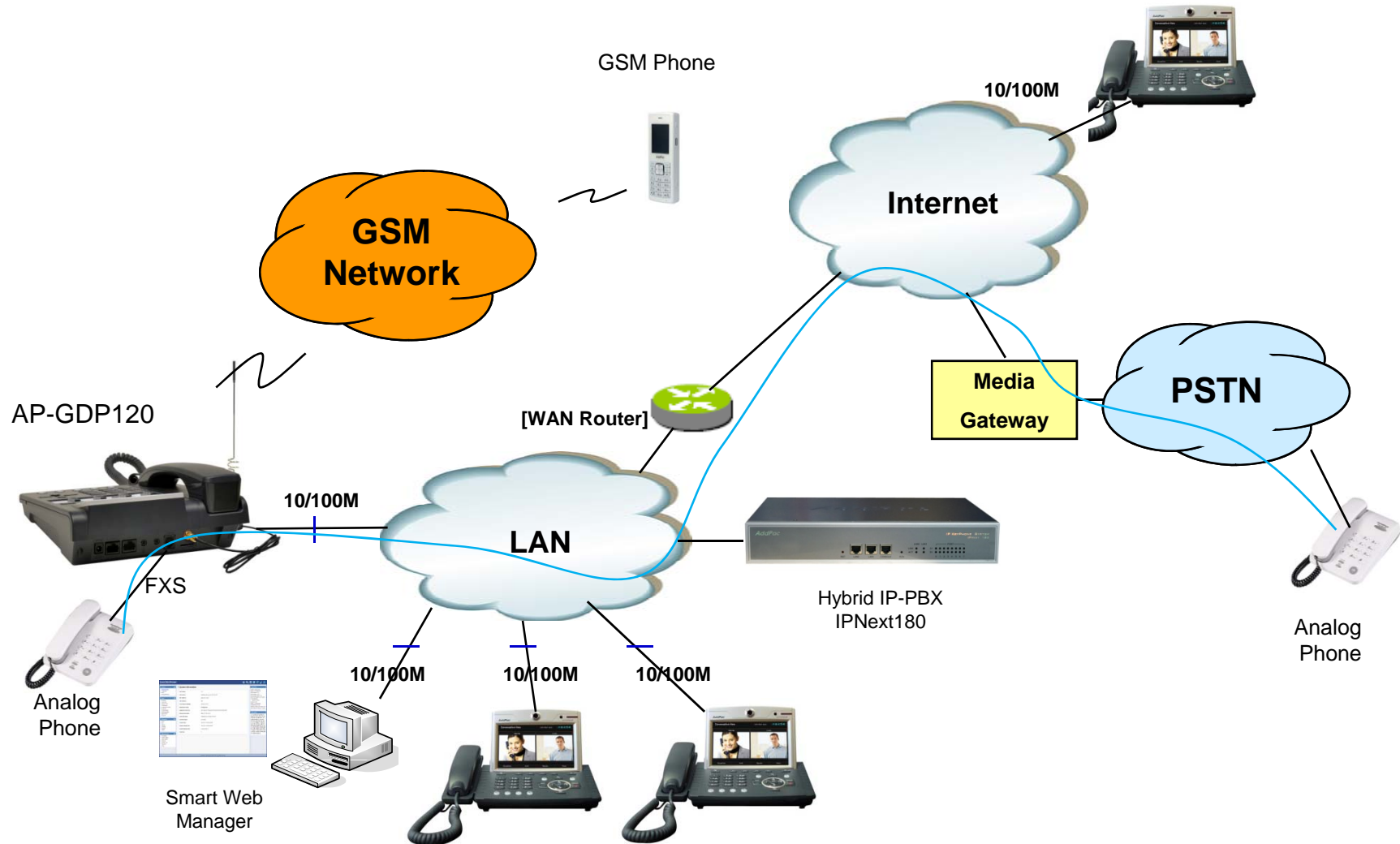
IP Phone Application

AP-GDP120 GSM Desk Phone



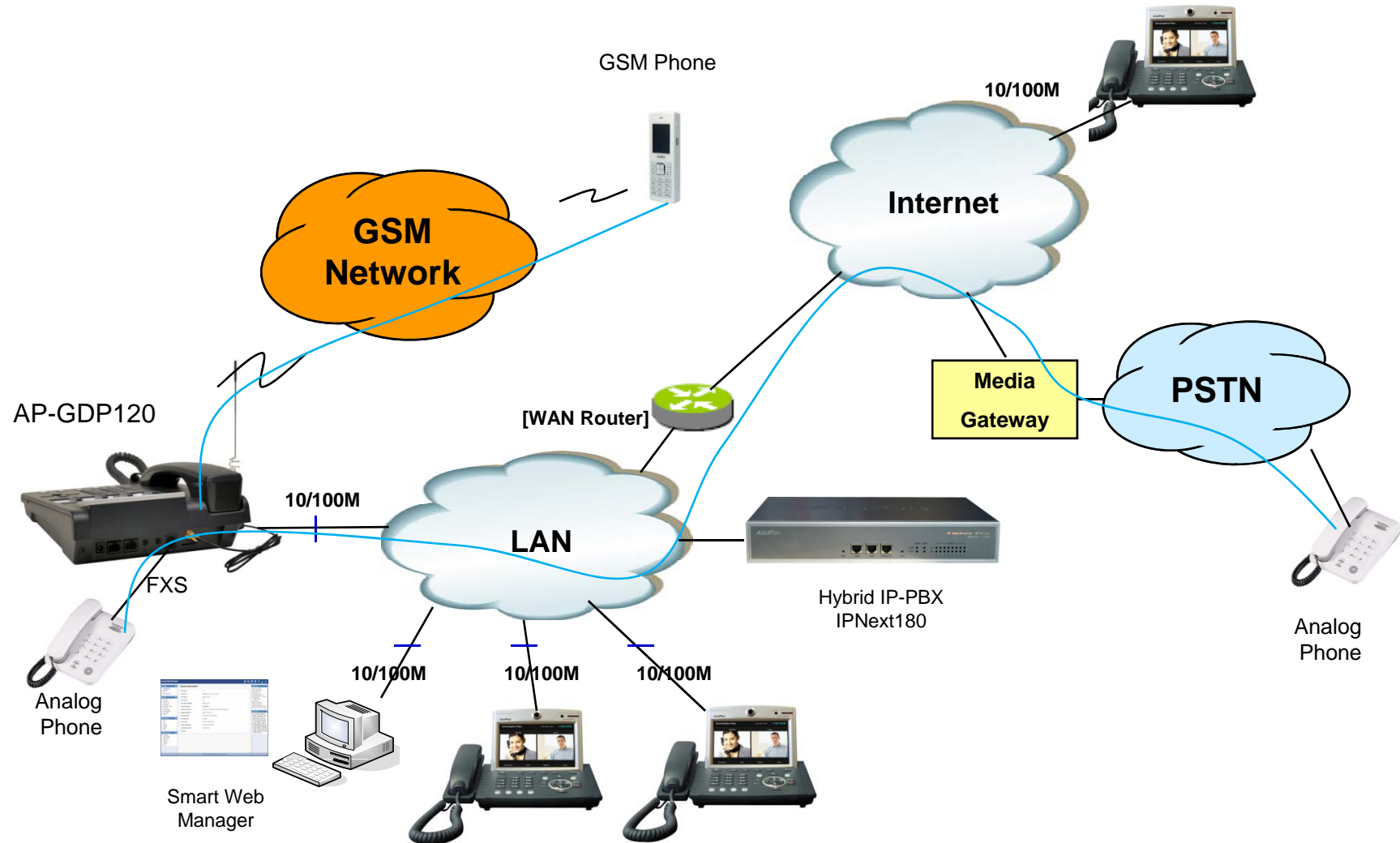
VoIP Gateway Application

AP-GDP120 GSM Desk Phone



GSM Desk Phone + VoIP Gateway Application

AP-GDP120 GSM Desk Phone



Ordering Information

- **AP-GDP120 IP Phone Hardware**
 - AP-GDP120 IP Phone Main Body
 - 12 Speed-Dial Key with Presence Indication LAMP
 - External 3.5mm Audio Input/Output
 - 2-ports 10/100Mbps Fast Ethernet
 - 1-Port GSM Antenna Interface, SIM Card Slot
 - Optional FXS Interface Port for GSM Gateway
 - Optional PoE (Power over Ethernet)
 - Including Network Cable Set & Ext. Power Supply, etc.
- **Built-in APOS Internetworking Software for AP-GDP120**
- **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



Thank you!

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