



AP2620™

VoIP Gateway

High Performance VoIP Gateway Solution

VoIP Modules for AP2620



AddPac


AddPac Technology

Sales and Marketing

www.addpac.com




VoIP Modules



Target :
AP2610, AP2620, AP1700







New VoIP Modules

DSP

Target	Voice Modules	Module Features	Module Picture
AP2610	AP-E&M2	2-Port E&M Module	 A photograph of the AP-E&M2 2-Port E&M Module. The module is a dark grey or black rectangular device. On the front panel, there are two RJ45 ports labeled 'LINE 0' and 'LINE 1'. To the left of these ports is a small green LED indicator labeled '-ACT-' with '1' and '0' below it. To the right of the ports is a small silver screw terminal. The model number 'AP-E&M2' is printed on the top right corner of the front panel.





VoIP Modules

DSP

Target	VoIP Modules	Module Features	Module Picture
AP1700,AP2610 AP2620,AP3100P	AP-FXS4	4-Port FXS Module	
AP1700,AP2610 AP2620,AP3100P	AP-FXO4	4-Port FXO Module	
AP1700,AP2610 AP2620,AP3100P	AP-FXS2O2	2-Port FXS&2-Port FXO Module	
AP1700,AP2610 AP2620,AP3100P	AP-E&M4	4-Port E&M Module	
AP1700,AP2610 AP2620,AP3100P	AP-FXS3O1	3-Port FXS&1-Port FXO Module	
AP1700, AP2620	AP-E1	1-Port Digital E1/T1 Module	

Additional VoIP Modules

DSP

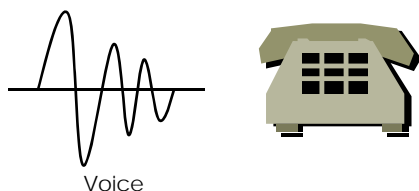
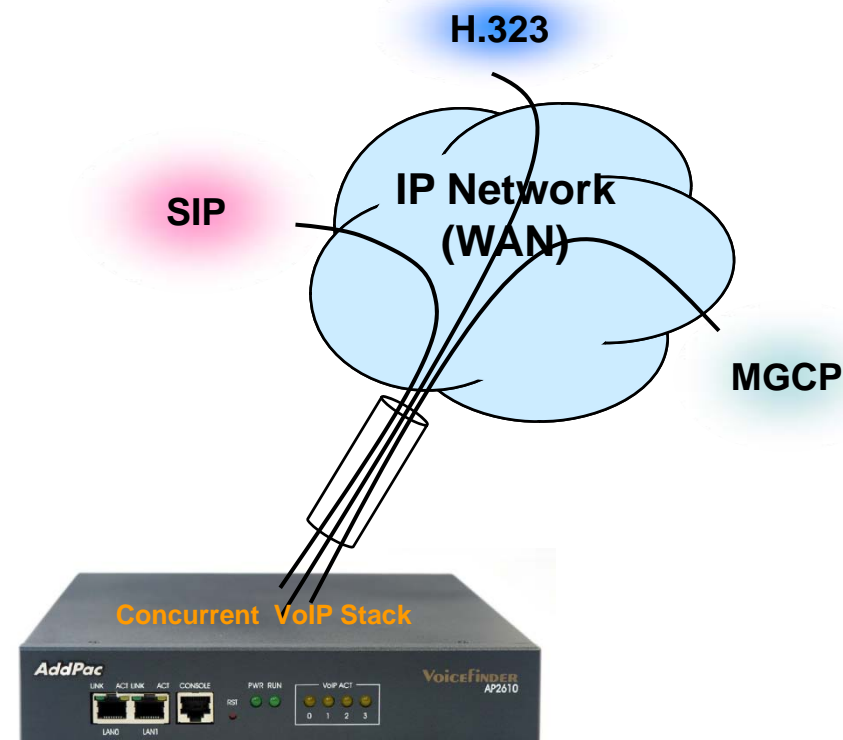
Target	Voice Modules	Module Features	Module Picture
AP2610, AP2620	AP-MP3	1-Pair Audio-In/Out Port, Direct MIC-In, Headphone High Quality Audio Band IP Broadcasting	
AP2610 AP2620	AP-AUDIO2	2-Pair Audio-In/Out Ports Voice Band IP Broadcasting	
AP2610, AP2620	AP-AUD1S3	1-Pair Audio-In/Out Ports, FXS Analog Interface Voice Band IP Broadcasting	
AP2610, AP2620	AP-AUD1S2O1	1-Pair Audio-In/Out Ports, FXS 2-Ports, FXO 1-Port Voice Band IP Broadcasting	



VoIP Gateway Service Features

VoIP (Voice over IP) Service

- **H.323, SIP Concurrent VoIP 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

- **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

- **Voice Codec**

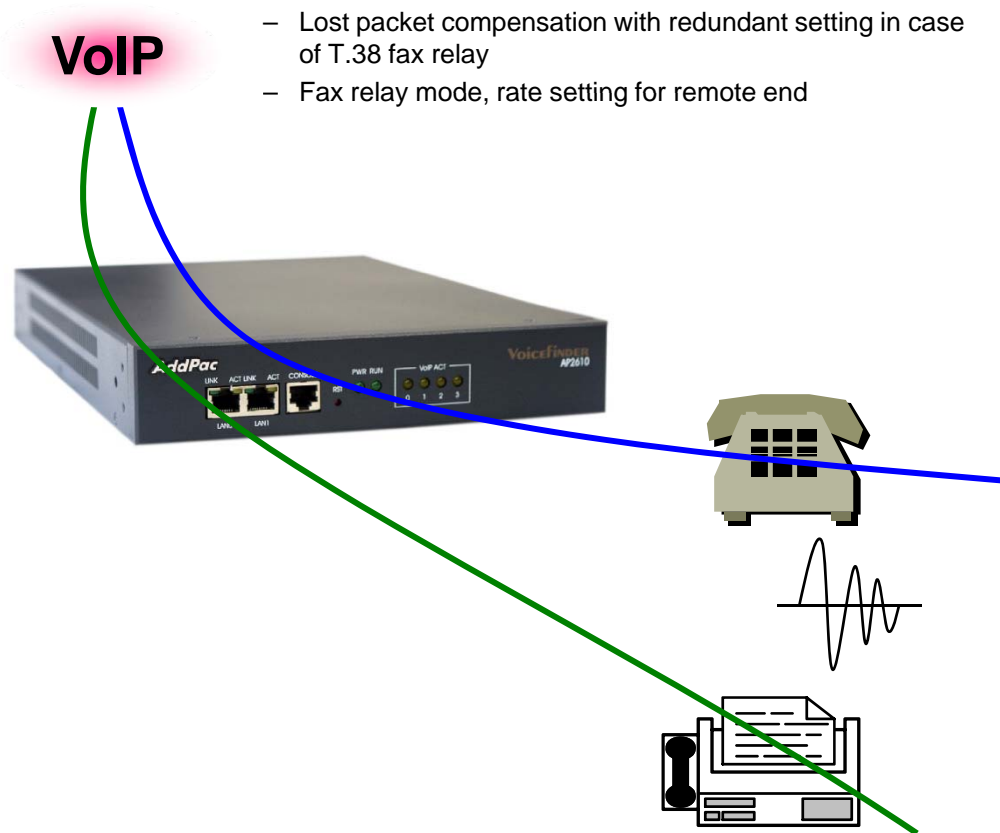
- G.711 A-Law, G.711 U-Law
- G.726 r16, G.726 r32
- G.729A
- G.723.1 r63, G.723.1 r53
- VAD (Voice Activity Detection) function support
- DTMF relay support (H.323, SIP, MGCP common) based on RFC2833

- **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



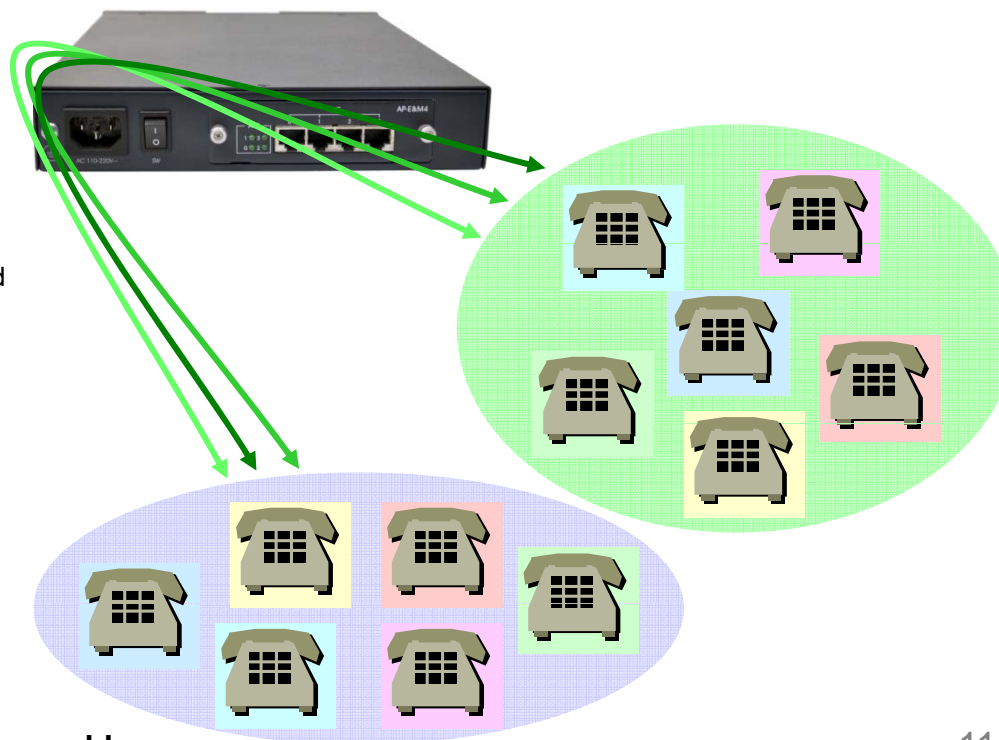
VoIP (Voice over IP) Service

• 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



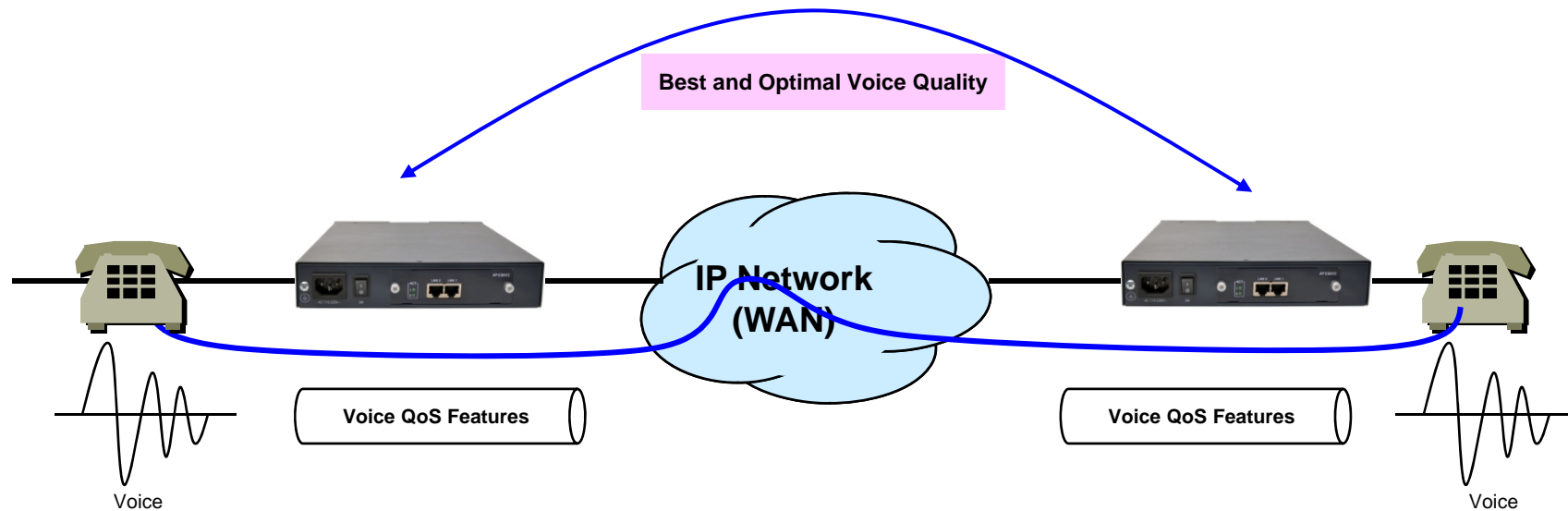
Advanced QoS Features

- **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

- **Basic Network Protocols**

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

- **Routing Protocol**

- IPv4 : Static

- **Service Protocol**

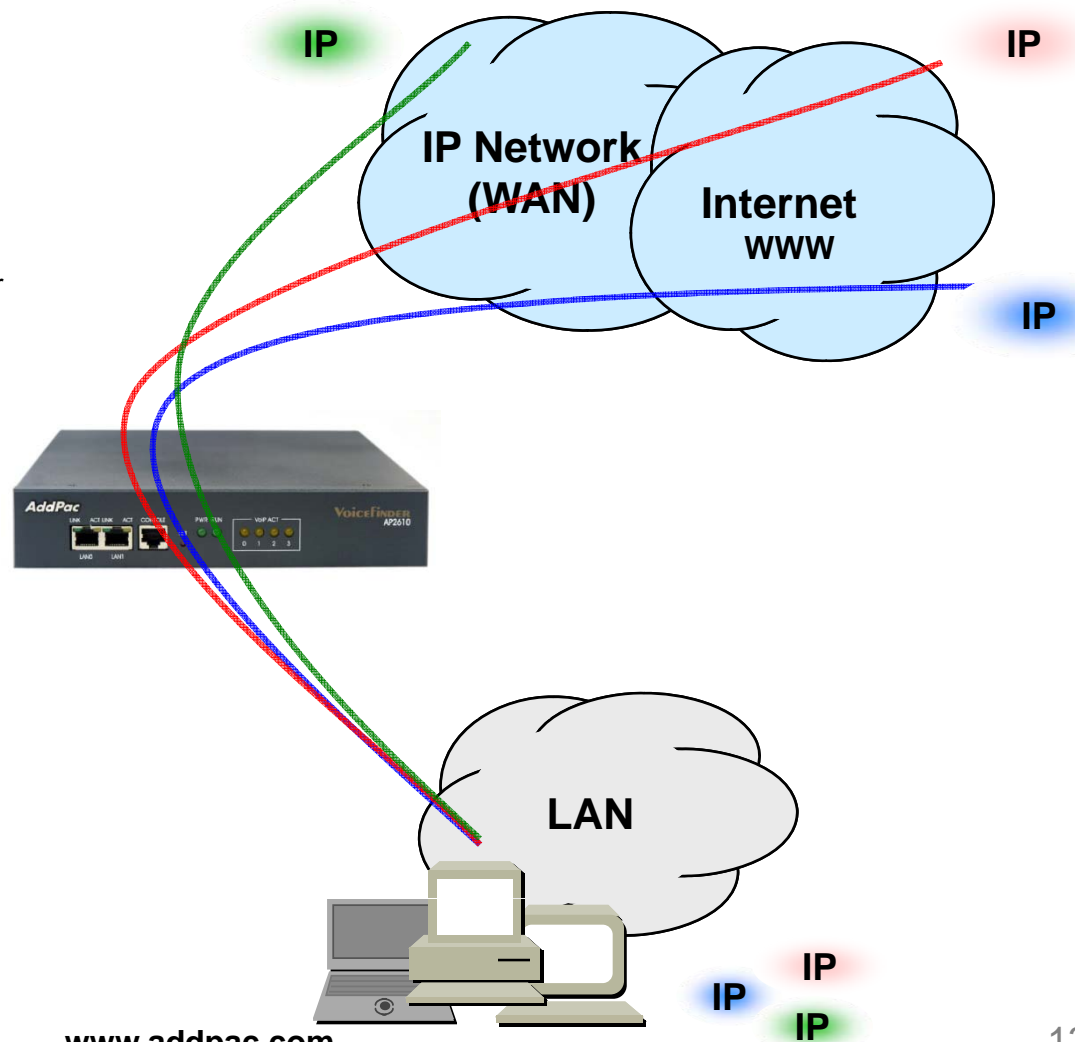
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

- **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

- **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**

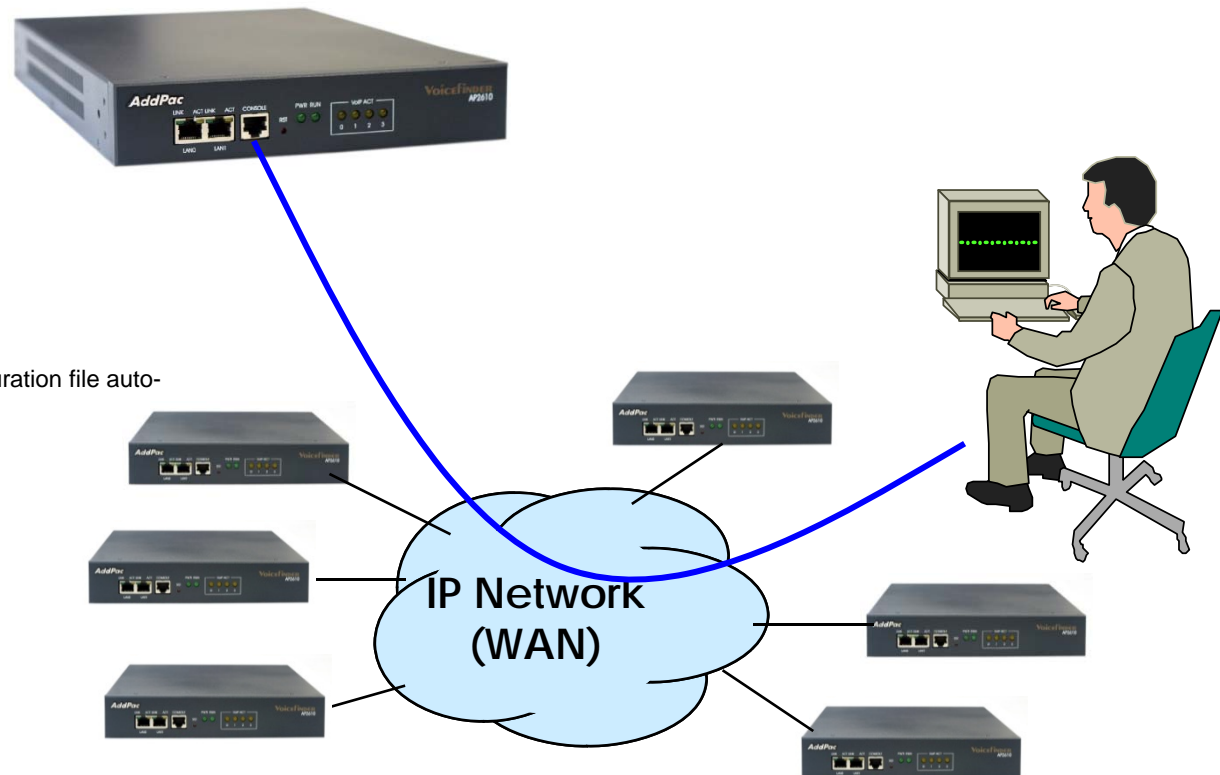
- HTTP server based APOS image and configuration file auto-upgrade support

- **Batch Job Function**

- Text based script downloading

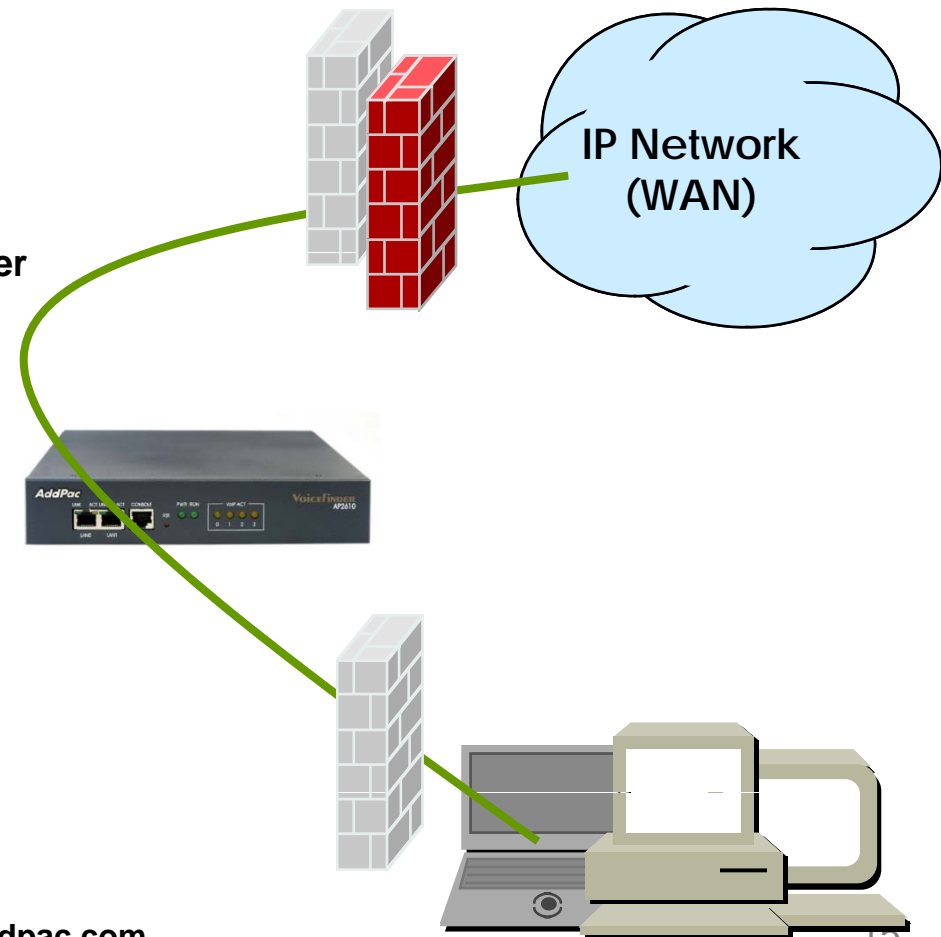
- **Interoperable with AP-VPMS Service**

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



Security Management

- 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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