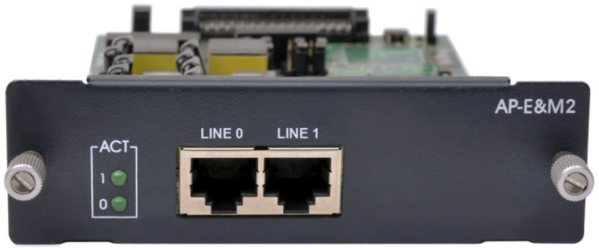
AP2610^{TM} VoIP Gateway

High Performance VoIP Gateway Solution

2-Port E&M VoIP Module



AddPac

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Sales and Marketing

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VoIP Modules

Target :
AP2610, AP2620, AP1700

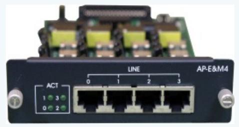
New VoIP Modules

DSP

Target	Voice Modules	Module Features	Module Picture
AP2610	AP-E&M2	2-Port E&M Module	

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

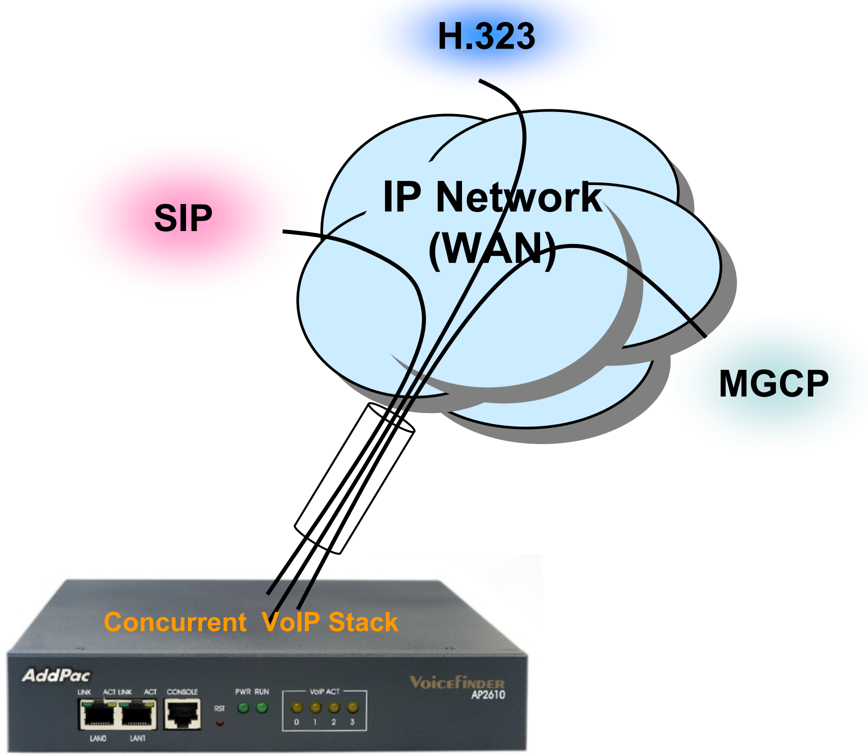
DSP

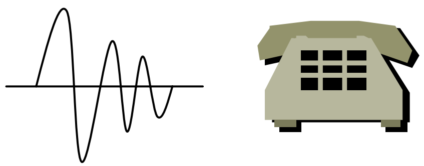
Target	Voice Modules	Module Features	Module Picture
AP2610, AP2620	AP-MP3	1-Pair Audio-In/Out Port, Direct MIC-In, Headphone High Quality Audio Band IP Broadcasting	
AP2610 AP2620	AP-AUDIO2	2-Pair Audio-In/Out Ports Voice Band IP Broadcasting	
AP2610, AP2620	AP-AUD1S3	1-Pair Audio-In/Out Ports, FXS Analog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





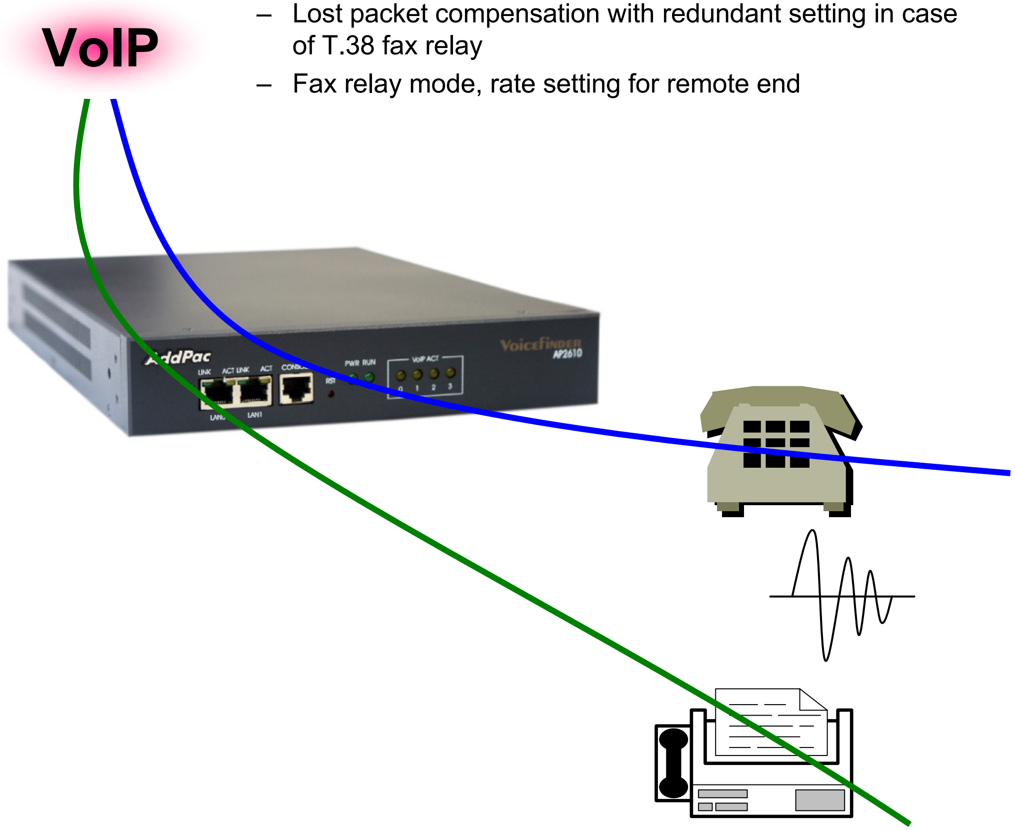
VoIP (Voice over IP) Service

- **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

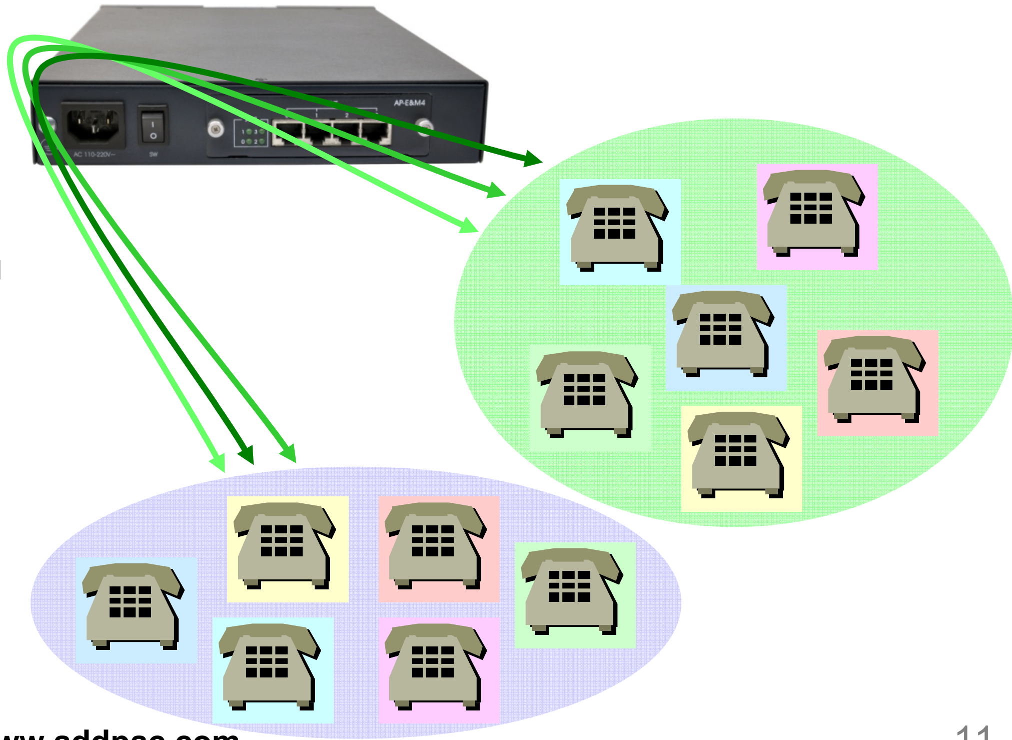
- **Voice Codec**
  - G.711 A-Law, G.711 U-Law
  - G.726 r16, G.726 r32
  - G.729A
  - G.723.1 r63, G.723.1 r53
  - VAD (Voice Activity Detection) function support
  - DTMF relay support (H.323, SIP, MGCP common) based on RFC2833
- **RTP**
  - Redundant RTP packet transmission in case of severe packet loss
  - Dynamic jitter buffer management and RPT 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

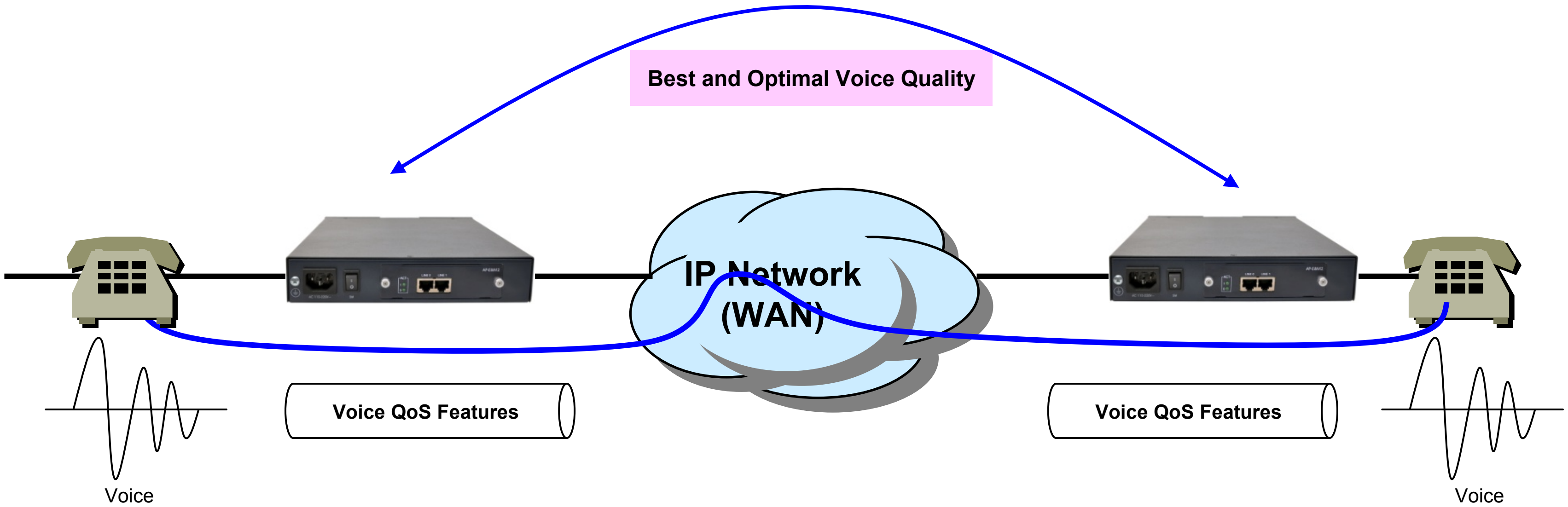
- **VoIP Call Controls**
  - Hot line connection function with PLAR (Private Line Auto Ring Down)
  - Leased line emulation function
  - Connection monitoring function
  - Fault tolerant with Redundancy and Call Distribution among Gateways for load balancing
  - Call attempt with IP address
  - H.323, 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



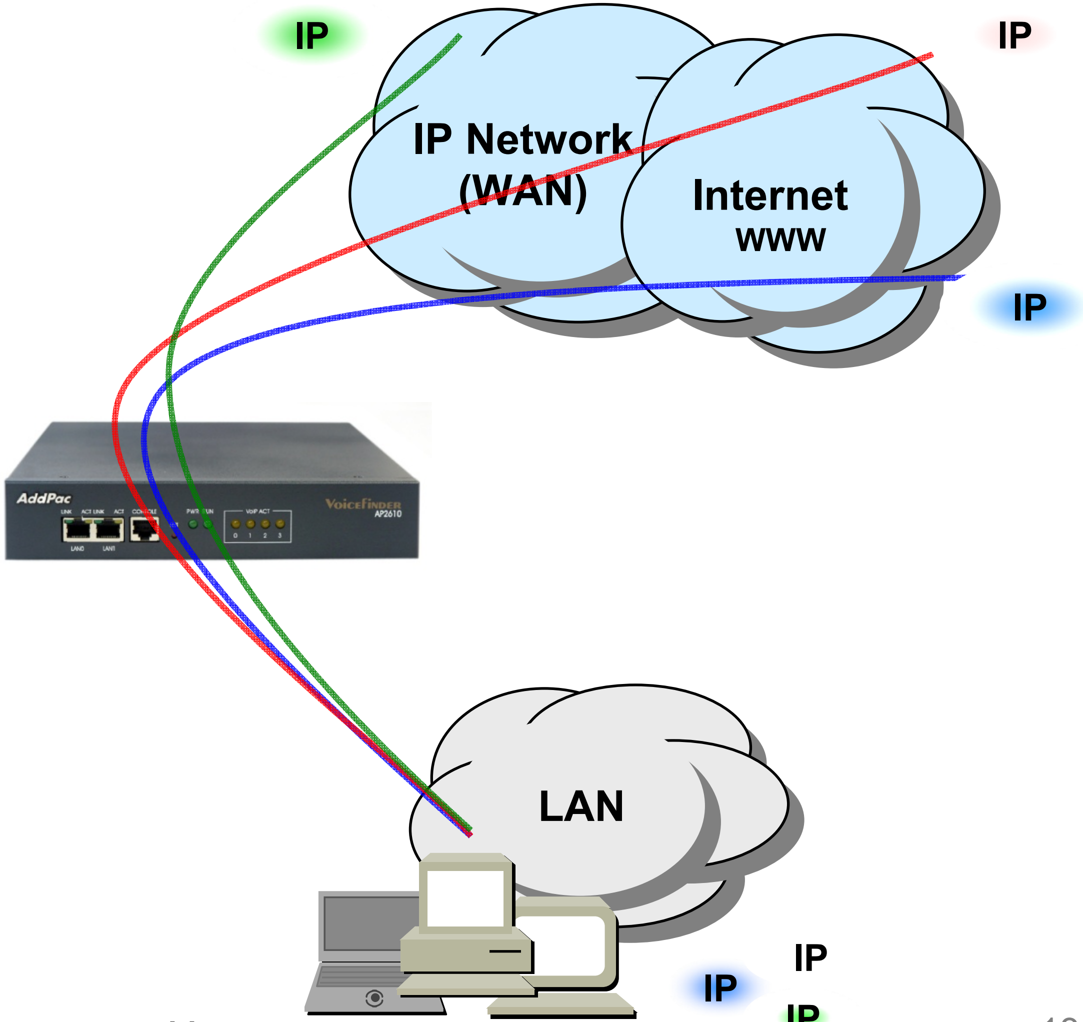
Advanced QoS Features

- **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

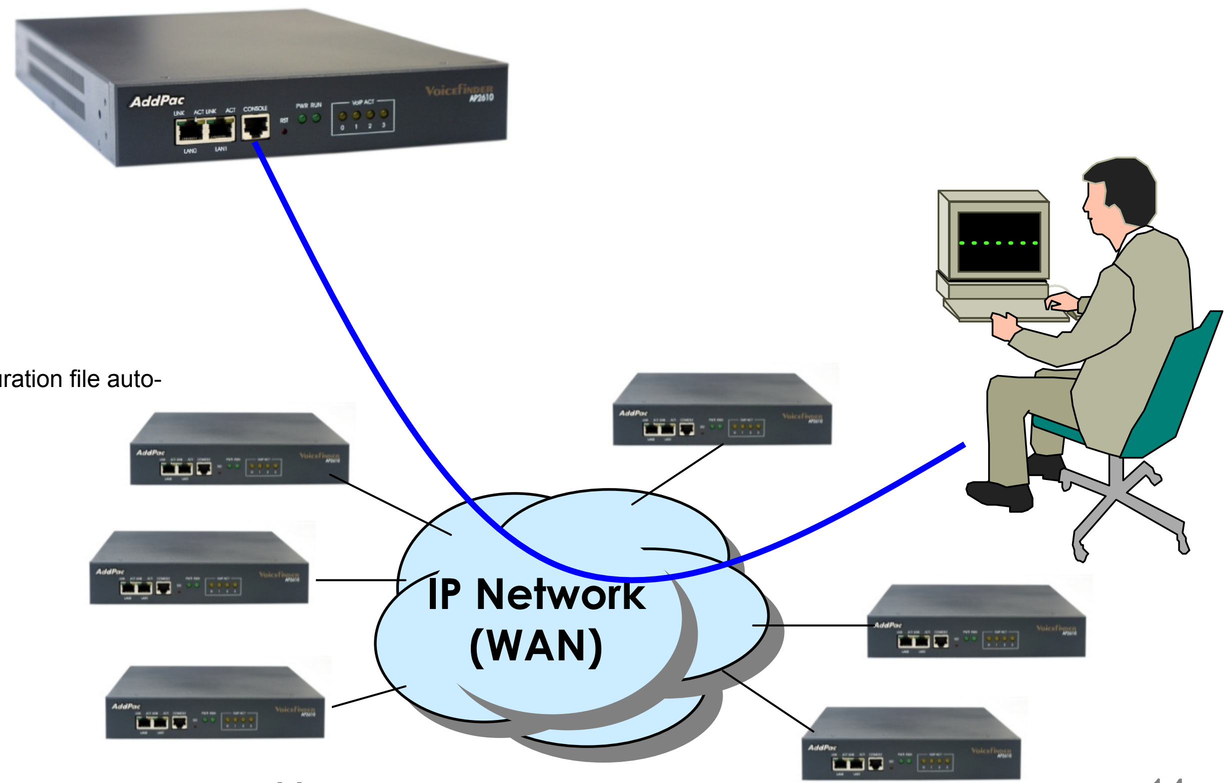
- **Basic Network Protocols**
  - ARP, IPv4, TCP, UDP, ICMP, SCTP, IGMP, MLD
- **Routing Protocol**
  - IPv4 : Static
- **Service Protocol**
  - FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
  - CDP (Cisco Discovery Protocol)
  - DNS Resolver , DDNS(nsupdate)
  - Bridge
  - Syslog
- **IPv4 Address 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

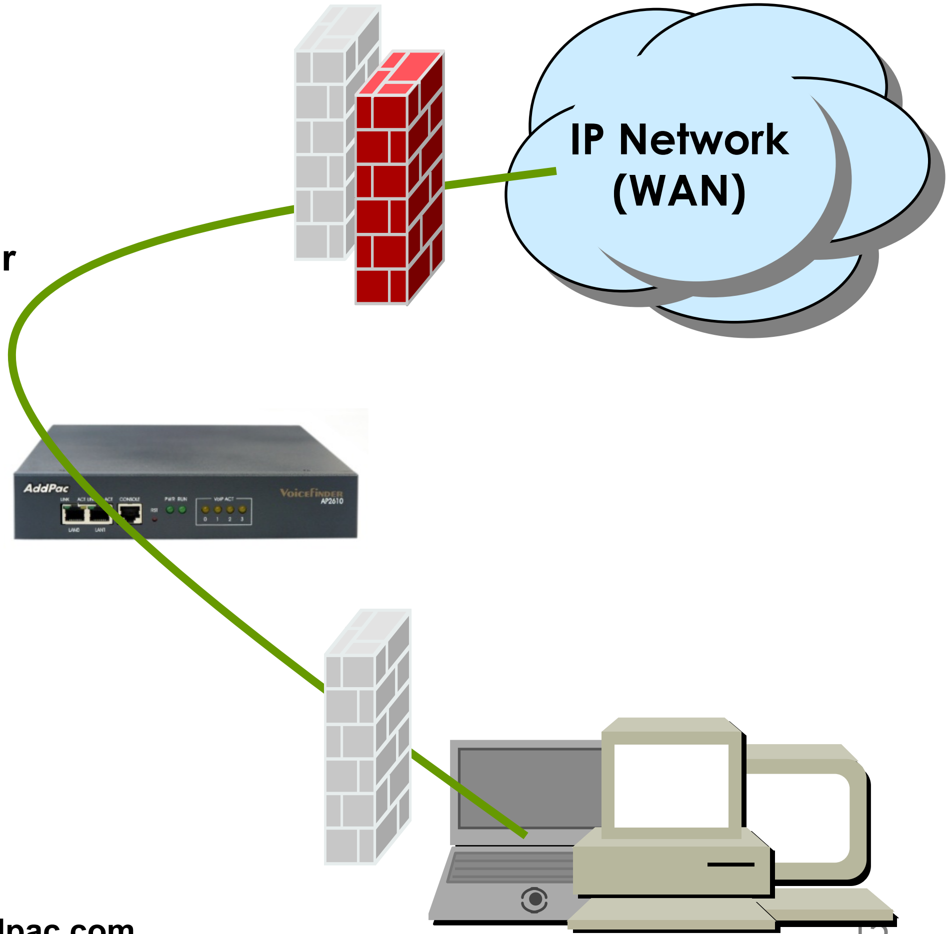
- SNMP
  - Standard Simple Network Management Protocol(SNMP) Agent support
  - MIB v1 and v2 Support
- Web-based Management
  - Smart Easy Setup
  - Standard Voice Interface
  - Standard PSTN Back-up Interface
- Watch-dog Function
  - Hardware, Software watch-dog services
- Remote Management
  - Telnet
  - 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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