SS7 VoIP Gateway Solution

AddPac Technology

Sales and Marketing

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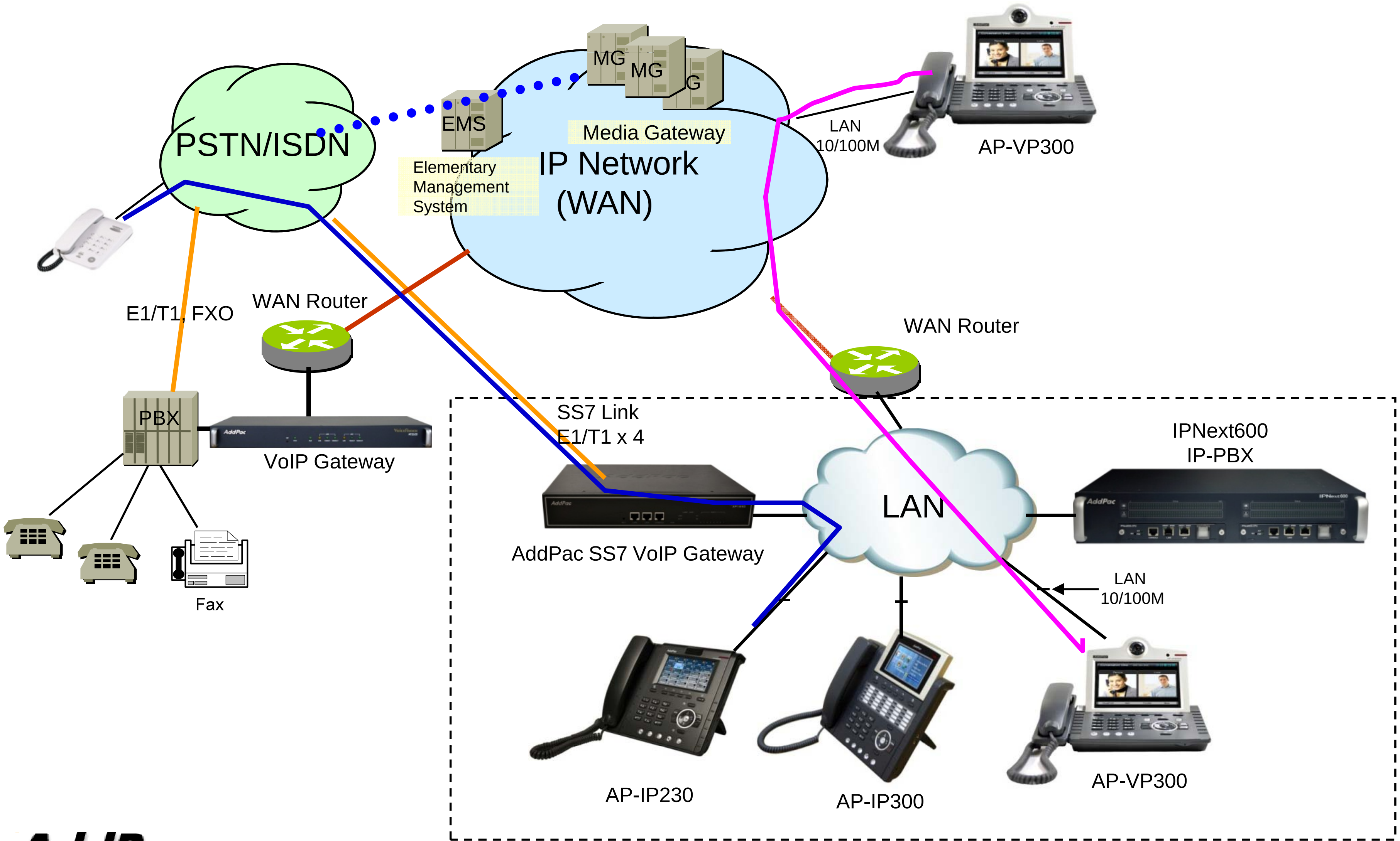
Contents

- SS7 VoIP Gateway Service Diagram
- SS7 VoIP Gateway Comparison Table
  - Digital VoIP Gateways(1~16 E1/T1)
- VoIP Modules for Rack Mountable Equipment
- SS7 Signaling Service Features
  - SS7 Signaling Features
  - ISUP (ISDN User Part) Features
- VoIP Gateway Service Features

SS7 VoIP Gateway Service Diagram



SS7 VoIP Gateways (1~16 E1/T1)

Product	AP1800	AP1850	AP-MG3000	AP-MG3800	AP-MG5000
					
Available Modules	AP-N1-FXS8 AP-N1-FXO8 AP-N1-FXS4O4 AP-N1-E1	AP-N1-E1 AP-N1-2E1	APv2-1E1 APv2-2E1 APv2-4E1	HIM-VoIP4E1 (4 E1/T1 Module)	HIM-4E1 (4 E1/T1 Module)
VoIP Signaling	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323	SIP, H.323
Digital E1/T1	Up to 2E1	Up to 4E1	Up to 4E1	Up to 8 E1	Up to 16E1
Digital Signaling	ISDN PRI, R2, SS7	ISDN PRI, R2, SS7	ISDN PRI, R2, SS7	ISDN PRI, R2, SS7	ISDN PRI, R2, SS7
Module Slot	Two(2)	Two(2)	Two(2)	Two(2)	Four(4)
LAN Port	2	2	2	2	2
Console	1	1	1	1	1
Power	Single PSU	Single PSU	Single PSU	Single PSU	Dual PSU

VoIP Modules

Target :
AP1800, AP1850

VoIP Modules

DSP

Target	VoIP Modules	Module Features	Module Picture
AP1800 AP1850	AP-N1-E1	1-Port Digital E1/T1 Module	
AP1800 AP1850	AP-N1-2E1	2-Port Digital E1/T1 Module	

Target : AP-MG3000

VoIP Modules

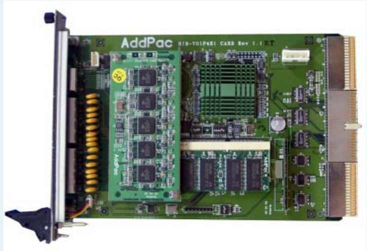
DSP

Target	VoIP Modules	Module Features	Module Picture
AP-MG3000	APv2-1E1	1-Port Digital E1/T1 Module	
AP-MG3000	APv2-2E1	2-Port Digital E1/T1 Module	
AP-MG3000	APv2-4E1	4-Port Digital E1/T1 Module	

Target :
AP-MG3800, AP-MG5000

VoIP Modules

DSP

Target	VoIP Modules	Module Features	Module Picture
AP-MG3800, AP-MG5000	HIM-VOIP4E1	4-Port Digital E1/T1 Module	 

SS7 Signaling Service Features

SS7 Gateway Feature

- Support Multiple E1 / T1 Interface
- Support Incoming and Outgoing Calls
- MTP2 (Q.703) / MTP3 (Q.704) / ISUP (Q.76x)
- SS7 A/F Link Support
- Support multiple linksets with different Point Code
- Support links with multiple links for load balance and failover
- Variant : ANSI, ITU-T, Japan(*), China(*)

ISUP (ISDN User Part) (1/2)

- Basic call messages (IAM, ACM, ANM, REL, RLC)
- CIC Reset (GRS, GRA, RSC)
- Connect message (CON)
- CIC Blocking/Unblocking (CGB, CGBA, CGU, CGUA, BLO, BLA)
- Continuity Check (COT, CCR)
- Call Progress message (CPG)
- Inbound and outbound calling works.

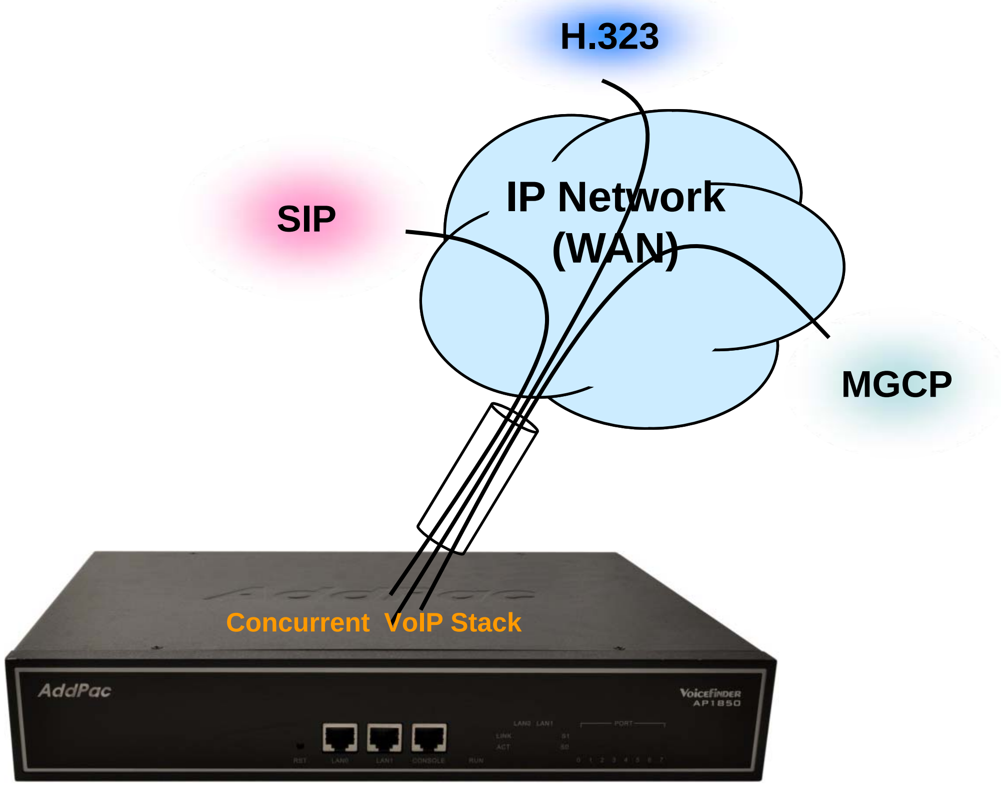
ISUP (ISDN User Part) (2/2)

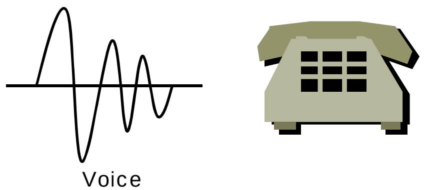
- ITU style SS7 support
- ANSI-style signalling support
- Called and Calling Nature of Address Indicator
- CallerID presentation and screening
- UCIC and LPA messages
- ANI2 - Originating line interface parameter (ANSI)
- Charge number parameter (ANSI)

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





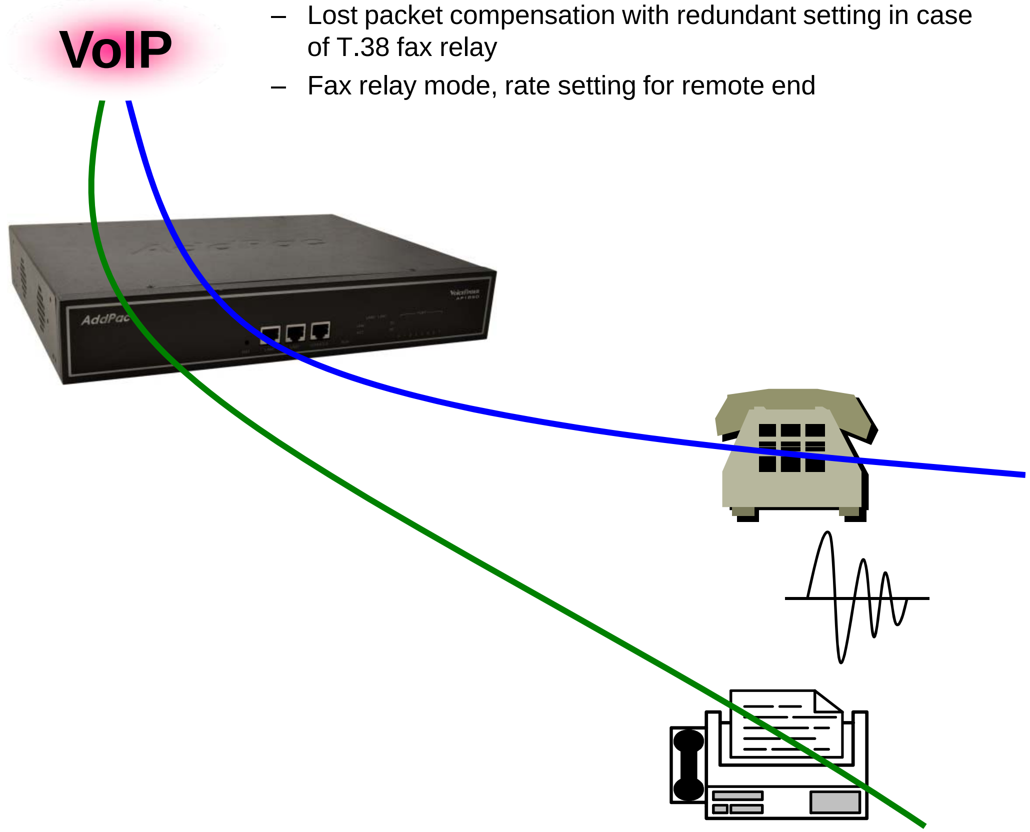
VoIP (Voice over IP) Service

- **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

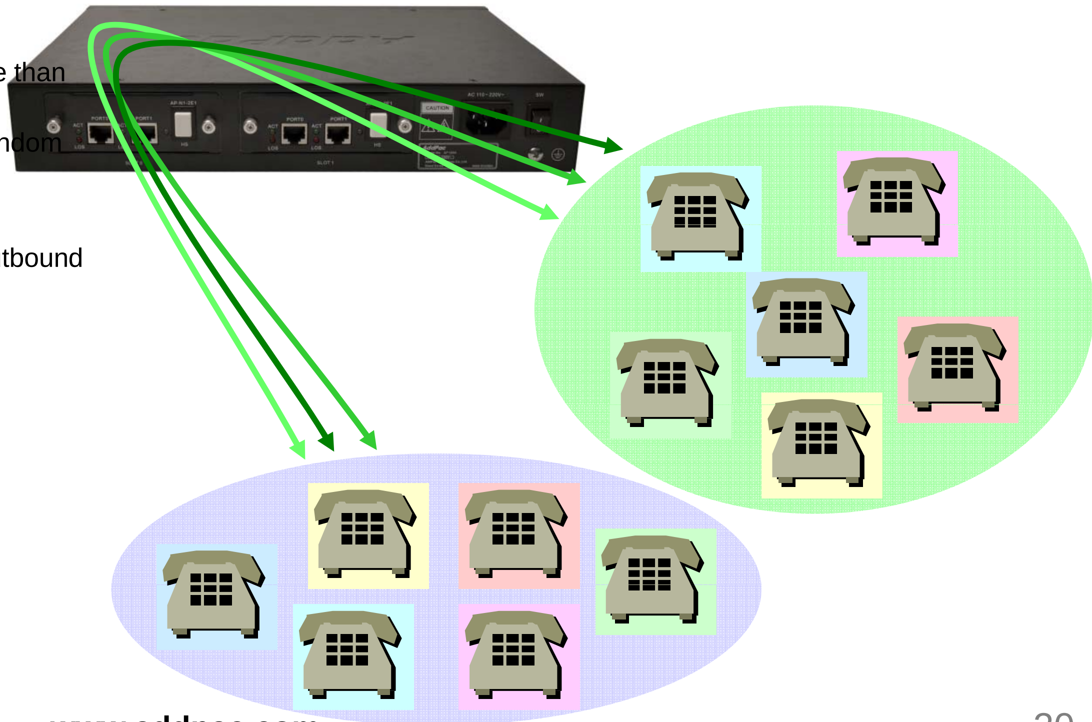
- **Voice Codec**
  - G.711 A-Law, G.711 U-Law
  - G.726 r16, G.726 r32
  - G.729A
  - G.723.1 r63, G.723.1 r53
  - VAD (Voice Activity Detection) function support
  - DTMF relay support (H.323, SIP, MGCP common) based on RFC2833
- **RTP**
  - Redundant RTP packet transmission in case of severe packet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

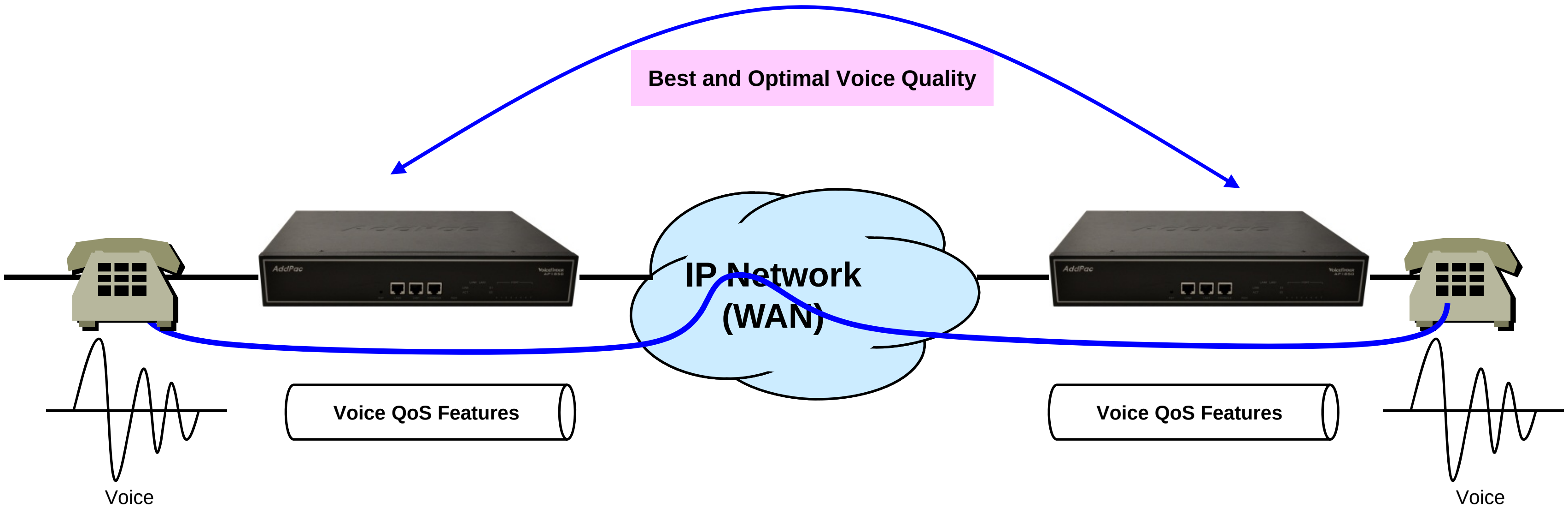
- **VoIP Call Controls**
  - Hot line connection function with PLAR (Private Line Auto Ring Down)
  - Leased line emulation function
  - Connection monitoring function
  - Fault tolerant with Redundancy and Call Distribution among Gateways for load balancing
  - Call attempt with IP address
  - H.323,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

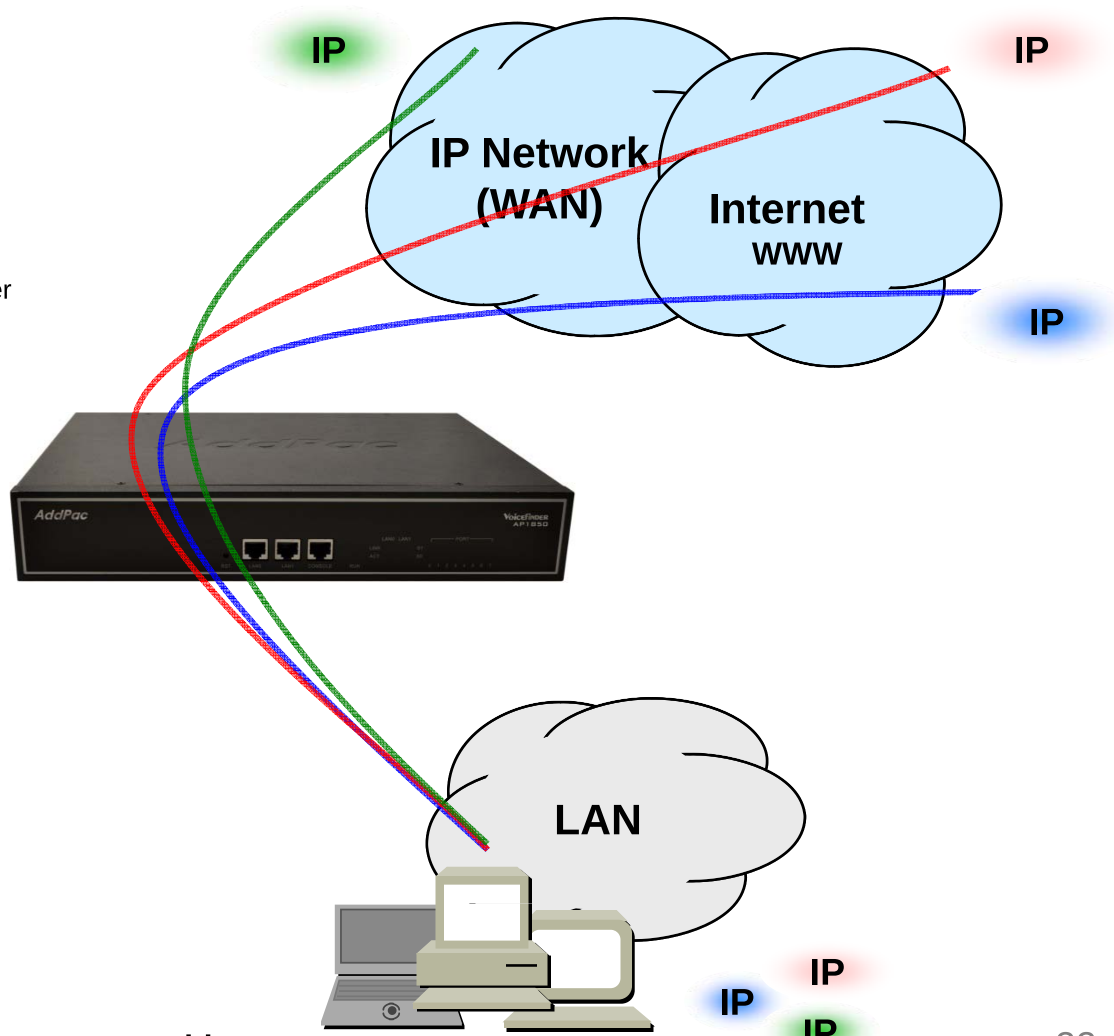
- Basic Network Protocols
 - ARP, IPv4, TCP, UDP, ICMP, SCTP, IGMP, MLD

- 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

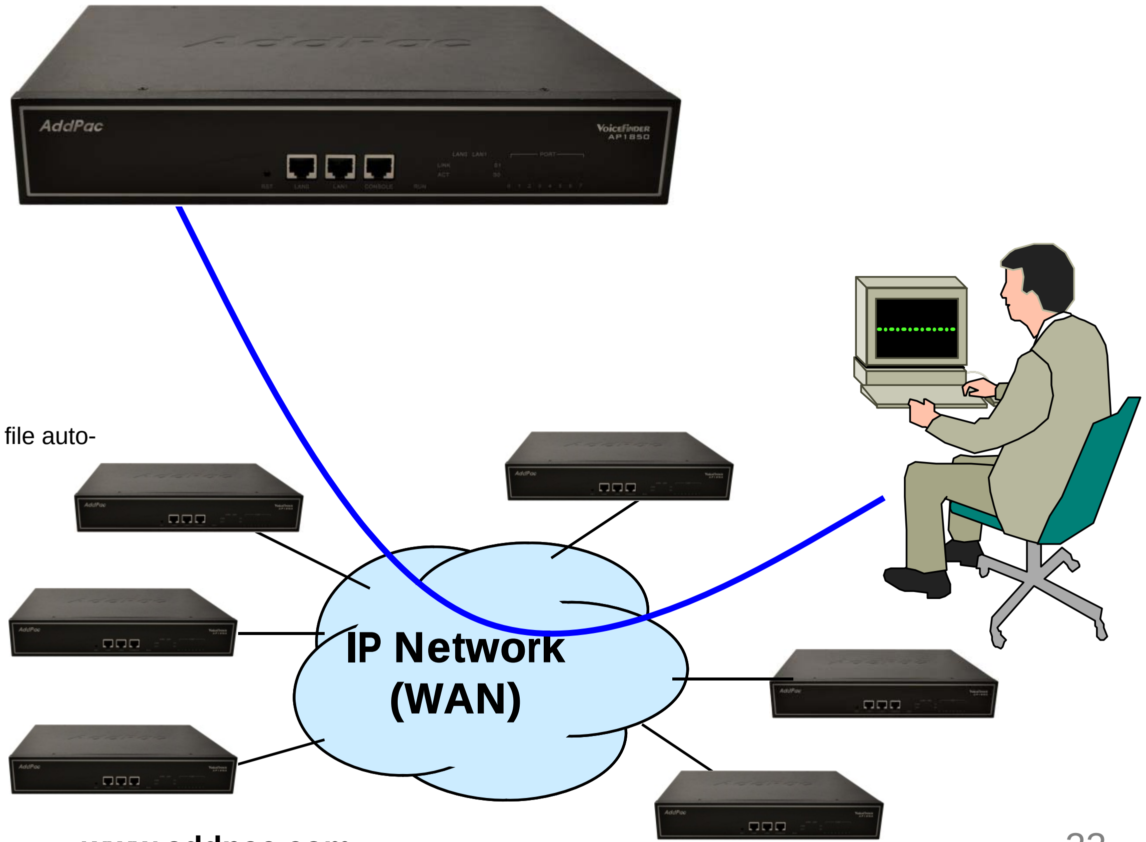
- Miscellaneous
 -Cisco Style CLI
 - Standard & Extended IPv4 Access List
 - Multi-level User Account Management
 - IP accounting
 - STUN Client



Network Management

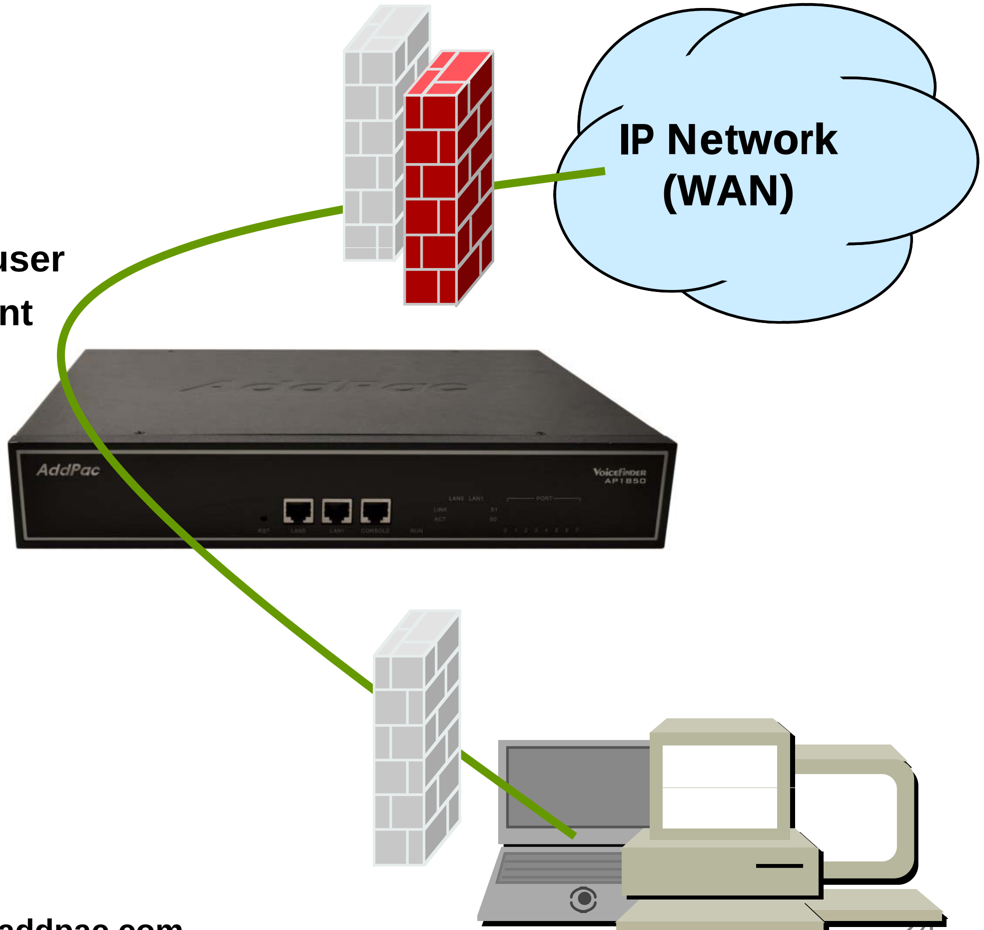
- SNMP
  - Standard Simple Network Management Protocol(SNMP) Agent support
  - MIB v1 and v2 Support
- Web-based Management
  - Smart Easy Setup
  - Standard Voice Interface
  - Standard PSTN Back-up Interface
- Watch-dog Function
  - Hardware, Software watch-dog services
- Remote Management
  - Telnet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