

AP-SVG3000

STT (Speech to Text) VoIP Gateway

STT VoIP Gateway Service Mode
(Radio PTT Talk Listening Service)



AddPac

AddPac Technology

Sales and Marketing

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Contents

- Product Overview
- Hardware Specification
- VoIP (Voice over IP) Service
- STT VoIP Gateway Service Mode
 - STT VoIP Gateway to STT VoIP Gateway
 - STT VoIP Gateway to IP Phone for PTT Talk Listening
 - STT VoIP Gateway to IP Voice Recording Server
 - STT VoIP Gateway to STT Application
- IP Voice Recording Server Solution for PTT Talk Recording

Product Overview

AP-SVG3000 8 Port STT VoIP Gateway

- H.323/SIP Dual Concurrent Stack Embedded
- High Performance RISC & Programmable DSP Architecture
- Various STT Audio Interface Module(16KHz G.722, etc) Support
- G.722(16KHz) beside 8KHzG.711/G.726, etc VoIP Codec Support for Backward Compatibility
- Independent Octal (8) STT VoIP Gateway Module
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- SNMS (Smart Network Management System) for Large Scale Deployment
- Advanced Voice QoS Mechanism
- 19 Inch Rack Mountable Design with Internal Power Supply
- Four(4) VoIP Module Slot

Hardware Specification

AP-SVG3000 8 Port STT VoIP Gateway

RISC
CPU

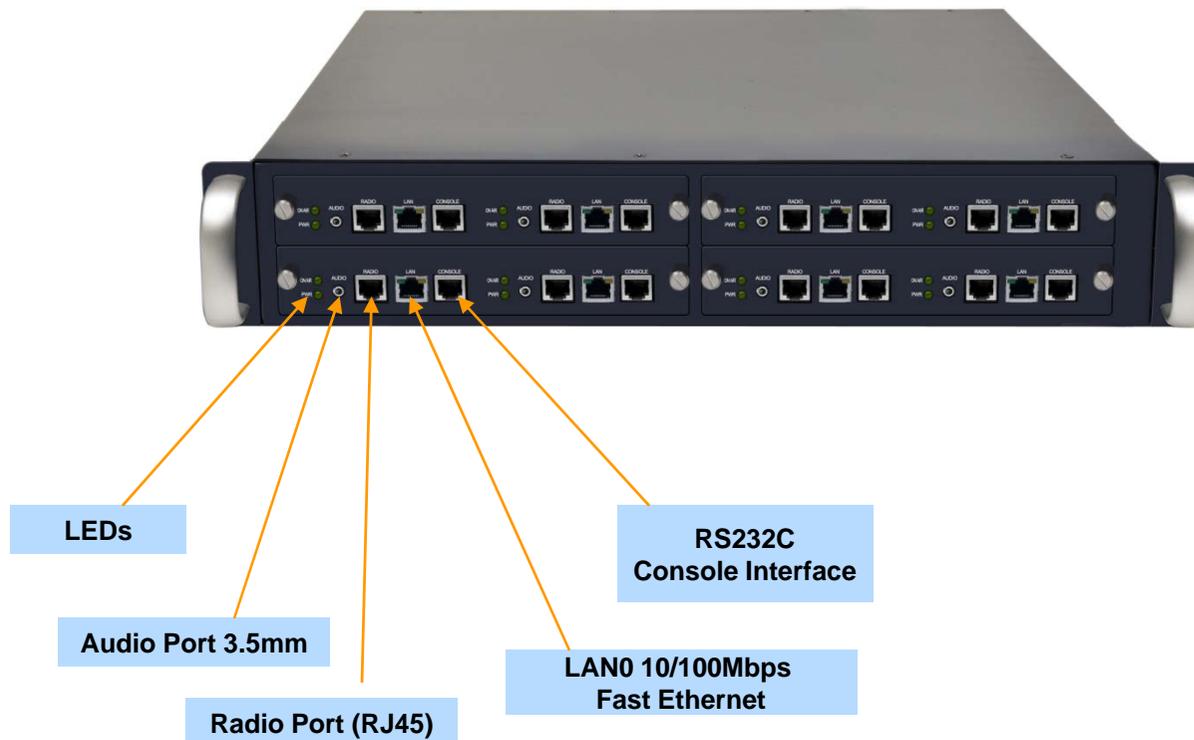
High-end
DSP

- RISC Microprocessor Computing Power
- Up to 8 Port STT VoIP Gateway
- Four VoIP Module Slots
 - 2-Port STT Audio Module Card (Audio (3.5mm), Radio(RJ45))
 - One(1) 10/100Mbps Fast Ethernet (RJ45)/ Port
 - One(1) RS232C Console Interface (RJ45)/Port
 - Status LEDs/Port
- Internal Power Supply

Hardware Specification

AP-SVG3000 8 Port STT VoIP Gateway

Front Side



Hardware Specification

AP-SVG3000 8 Port STT VoIP Gateway

Back Side



Hardware Specification

AP-SVG3000 8 Port STT VoIP Gateway

Example : RADIO Interface for Radio Interworking

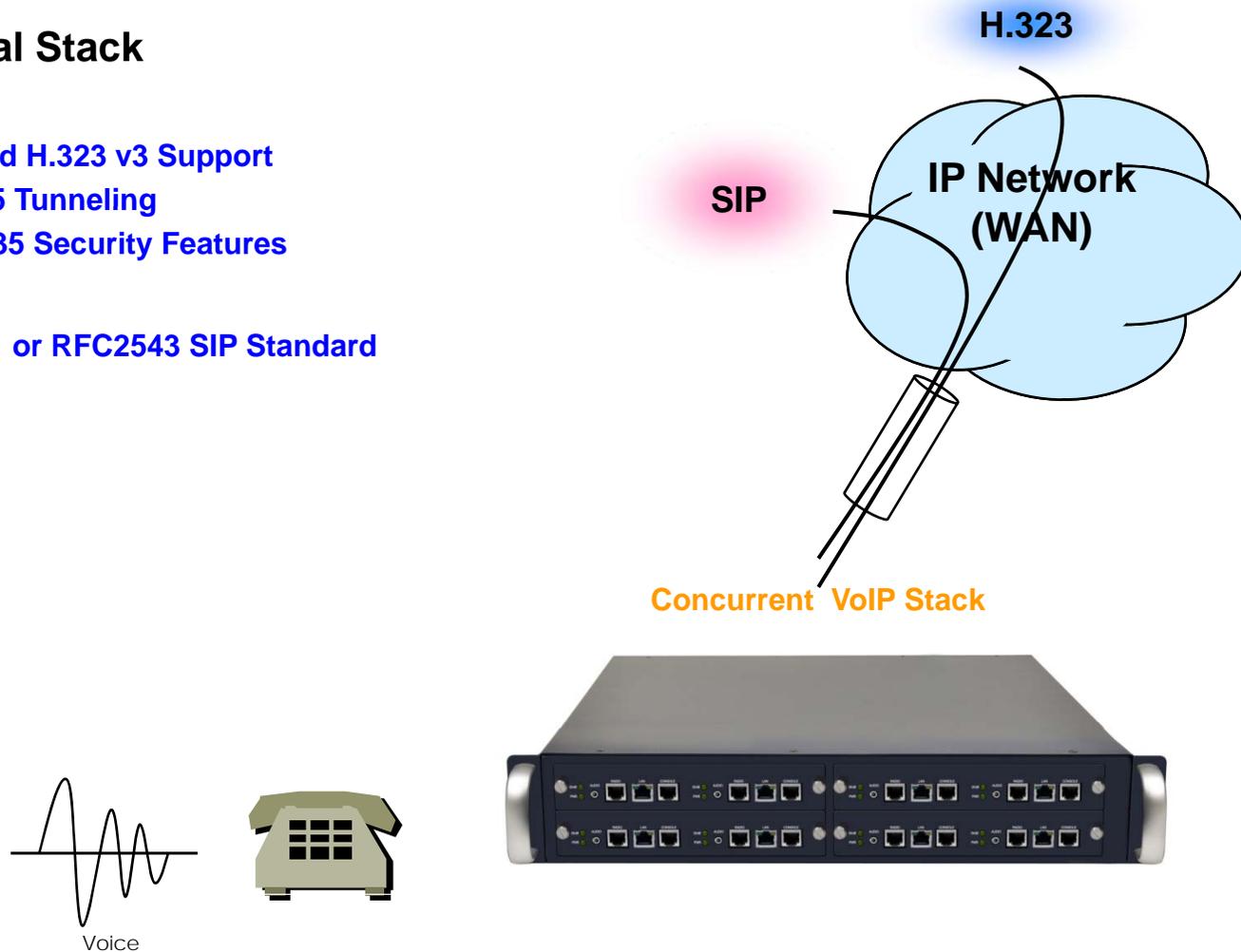
RADIO Port (Tx, Rx, PTT Tx, PTT Rx)



AP-STT2 Two(2) Port STT VoIP Module

VoIP (Voice over IP) Service

- 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

- **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



VoIP (Voice over IP) Service

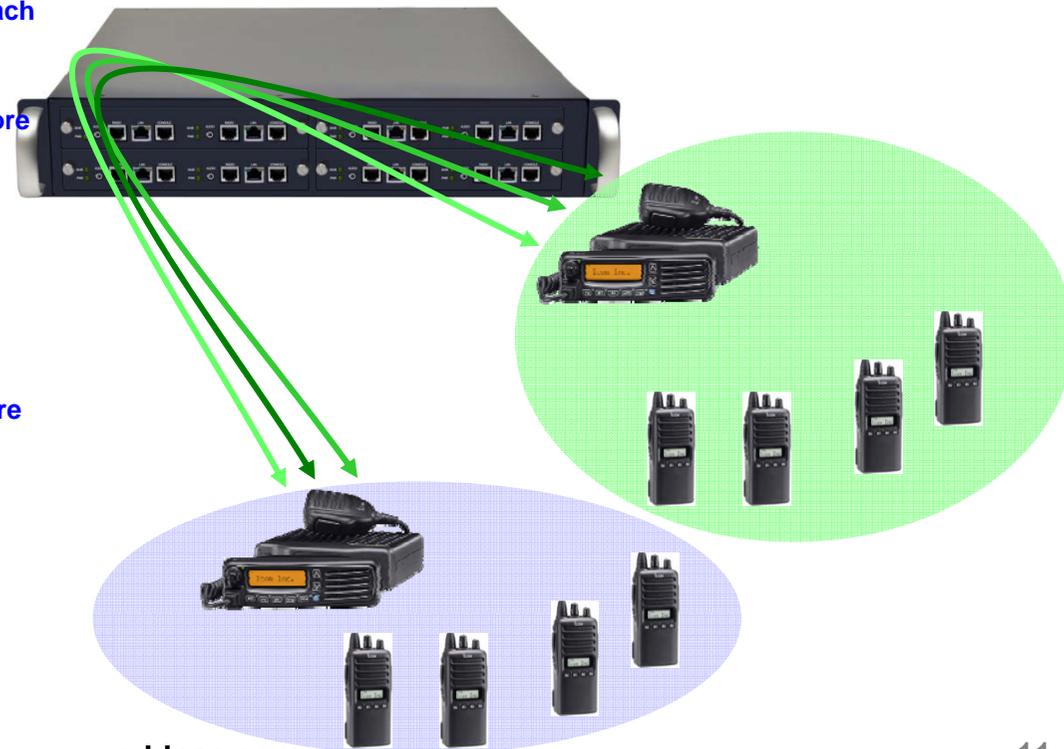
AP-LMR2000 LMR Gateway

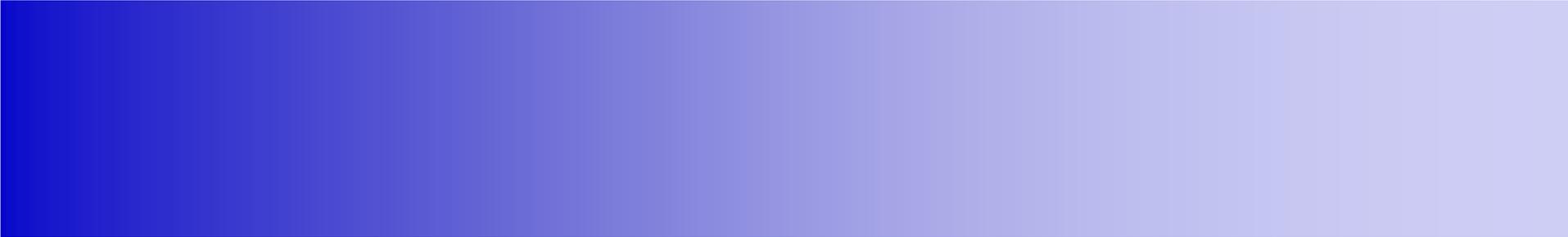
• VoIP Call Controls

- 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

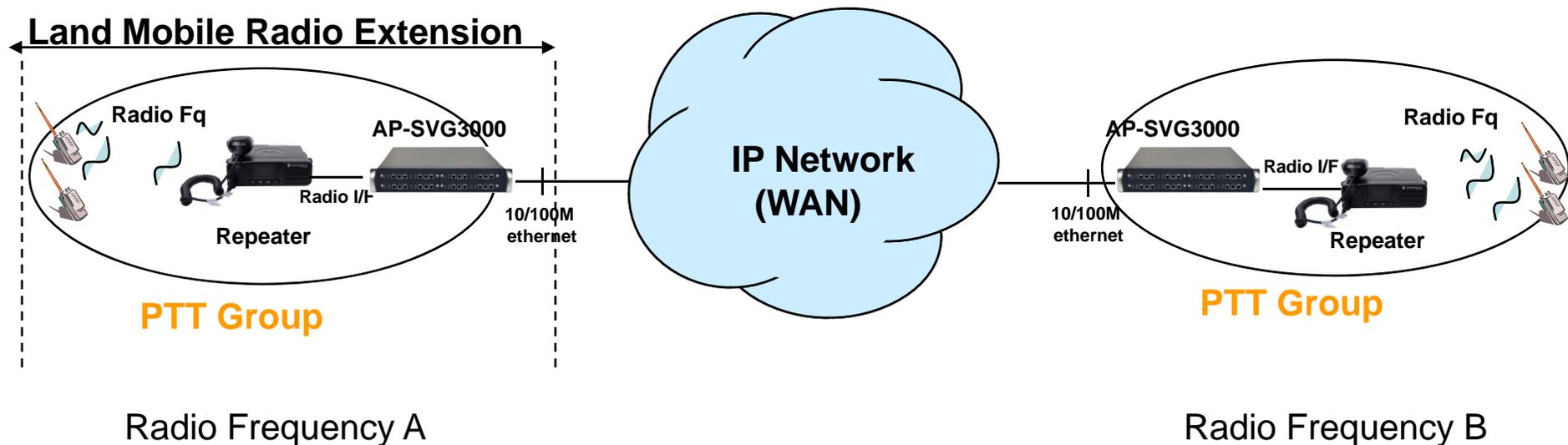




STT VoIP Gateway Service Mode

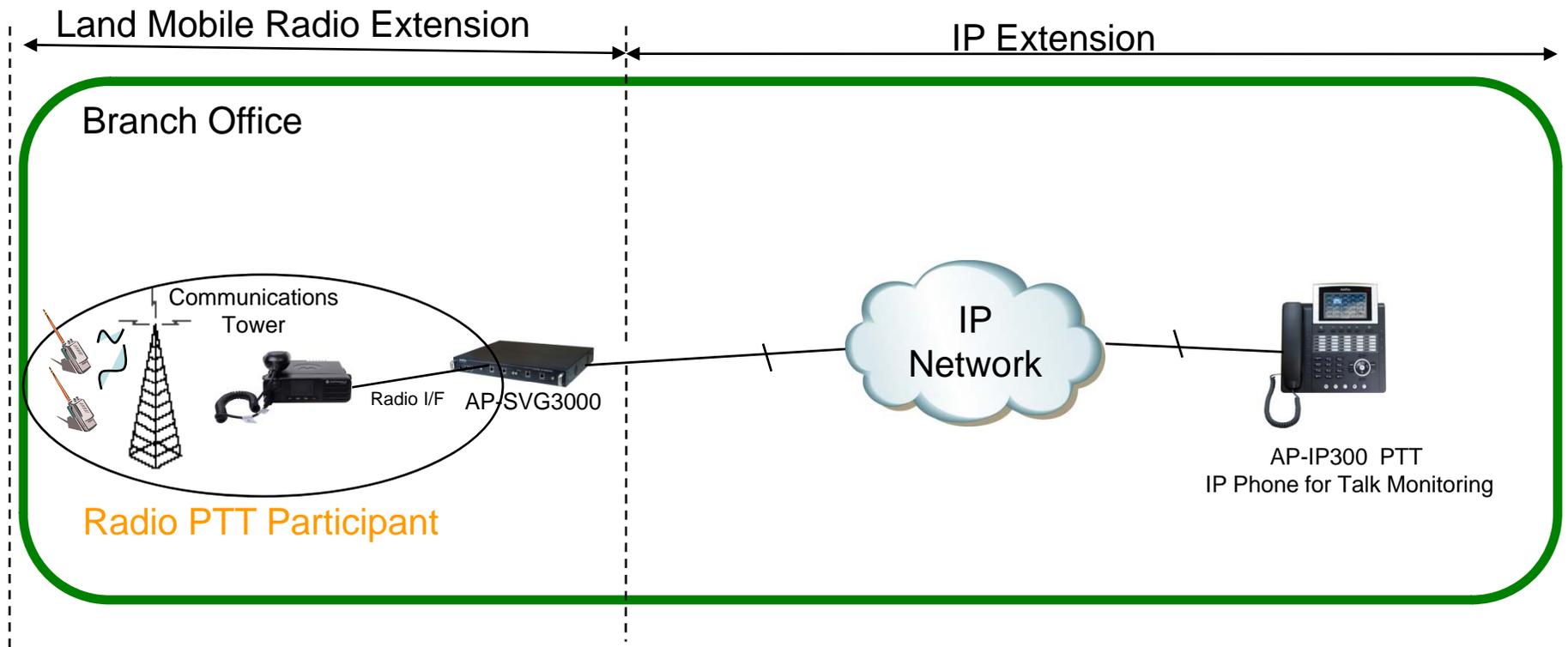
RoIP Peer to Peer Service Mode

STT VoIP Gateway to STT VoIP Gateway Service Mode



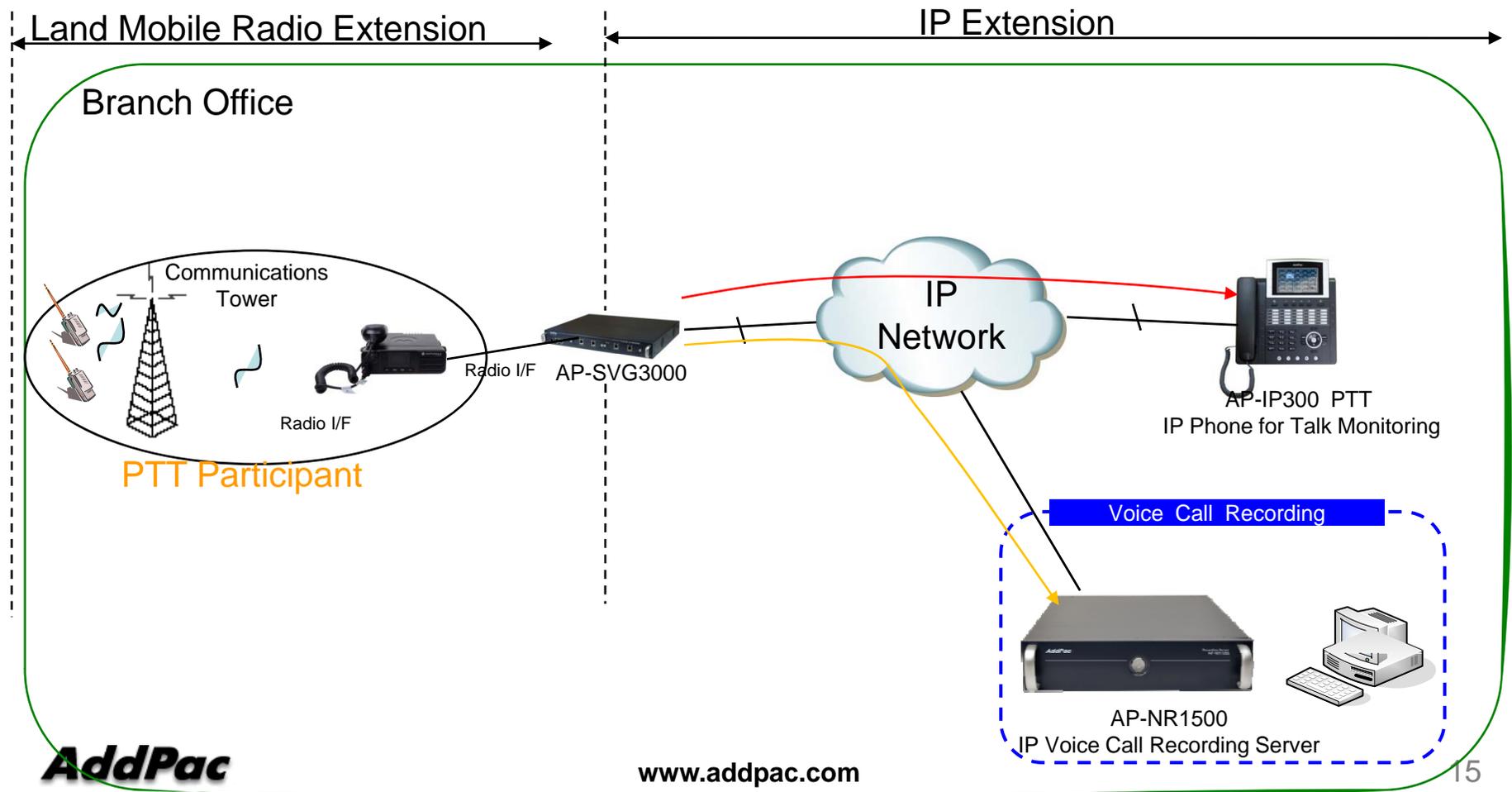
Radio Talk Monitoring in IP Network

- STT VoIP Gateway to IP Phone Service Diagram

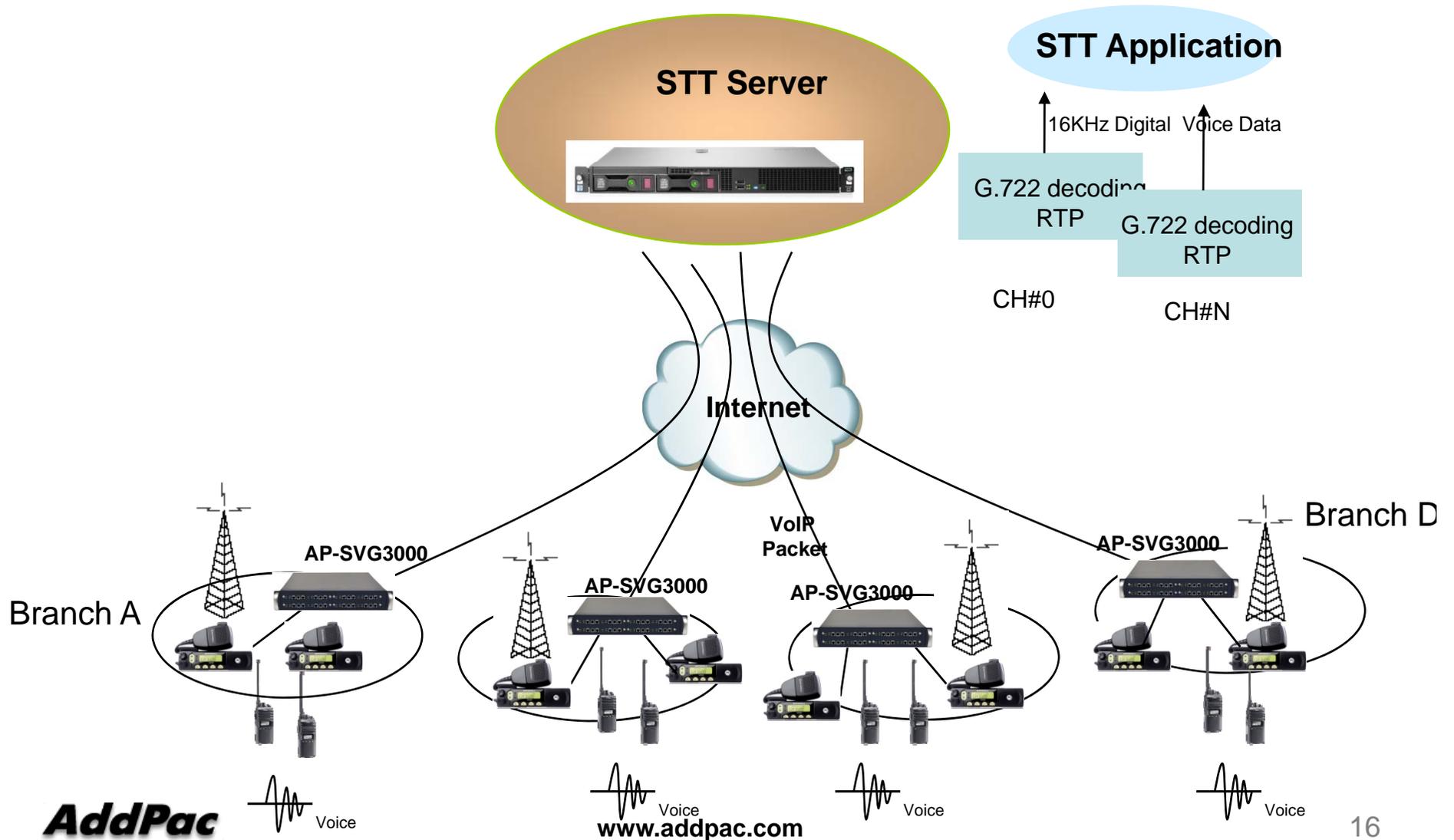


Radio Talk Recording in IP Network

- STT VoIP Gateway to Voice Recording Server Service Diagram



Radio Talk STT Application



AP-NR1500

IP Voice Call Recording Server



Product Overview

AP-NR1500 IP Voice Call Recording Server

- IP based Network Voice Call Recording Server
- Linux Operating System
- Powerful Management and User Friendly Features
- High-performance Voice Recording Service
- External AddPac IP Terminal (Ex: IP Phone, IP Intercom, IP Emergency Phone) Interworking Support
- Firmware Upgradeable Architecture
- One(1) 10/100/1000Mbps Gigabit Ethernet Interface
- Up to Two(2) 3.5Inch SATA Hard Disk Interface Support
- Two(2) USB Interface Support
- One(1) RS232C Console Interface

Hardware Specification

AP-N1500 IP Voice Call Recording Server

- High Performance Computing Power
- Network Interface
 - One(1) 10/100/1000Mbps Gigabit Ethernet Port
- Two(2) USB 2.0 Interfaces for Mouse, Secondary Storage, etc
- One(1) RS232C Console Interface (RJ45)
- Up Two(2) SATA type Hard Disk (4~8 Tera HDD Capacity)
- Power On/Off Soft Switch with LED Indication Lamp (Front Side)

Hardware Specification

AP-NR1500 IP Voice Call Recording Server

AP-NR1500 Front Side

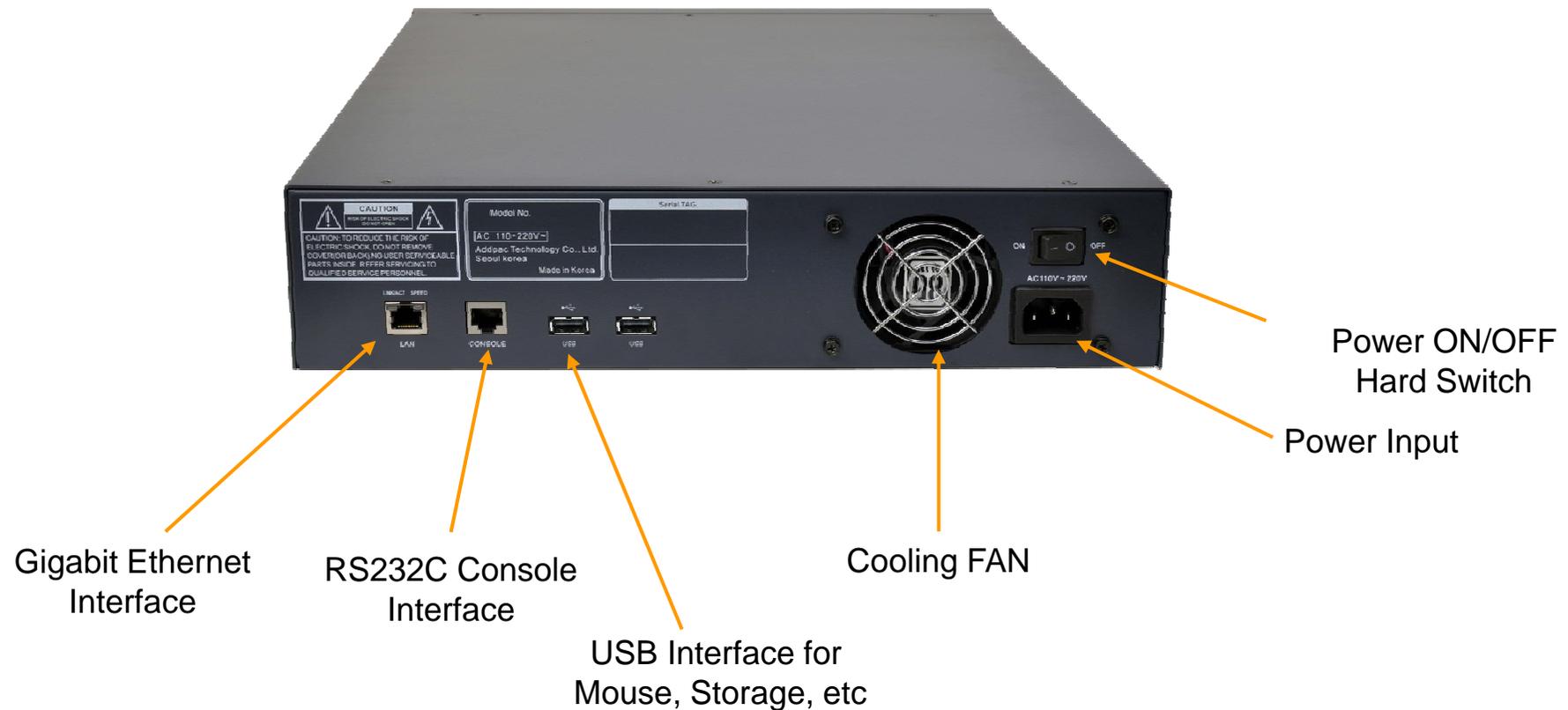


Power On/Off Switch with LED Indication LAMP

Hardware Specification

AP-NR1500 IP Voice Call Recording Server

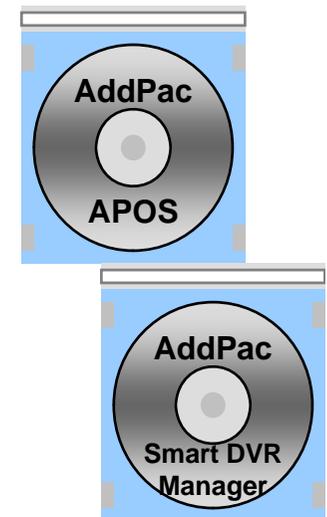
AP-NR1500 Back Side



Software Service

AP-NR1500 IP Voice Call Recording Server

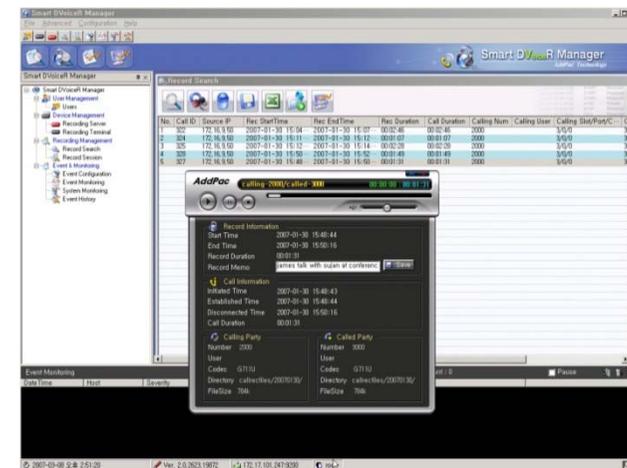
- **Built-in AddPac Internetworking Software**
 - Scalability, Functionality, and Stability Features
 - Advanced Network DV_{oice}R Recording & Live Streaming Features
 - QoS Control Features
- **Firmware Upgradeable Architecture**
- **Industry Standard Network Protocol Features**
- **Highly User Friendly Management Features**
 - PC based Window Program
 - Smart DV_{oice}R Manager



Smart DVoiceR Management Program

AP-NR1500 IP Voice Call Recording Server

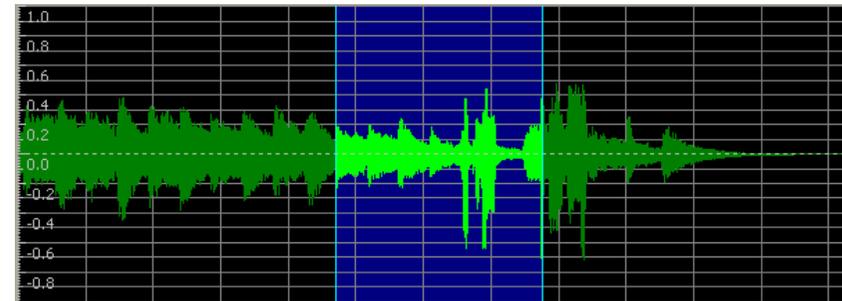
- Call History Management (search/modify/delete/save)
- Media Play Management (Play/Stop/Seek/Pause)
- Live Call List Management, Live Call Monitoring
- Local Backup (File Manager Support, PC HDD, DVD) and Local Play
- User Management (registration/modify/delete/search)
- Server Status (CPU/Memory/HDD) & Event Monitoring
- Waveform Analyzing Function
- Recording Source Management (VoIP Gateway, IP Phone, etc)
- Live Recording Board

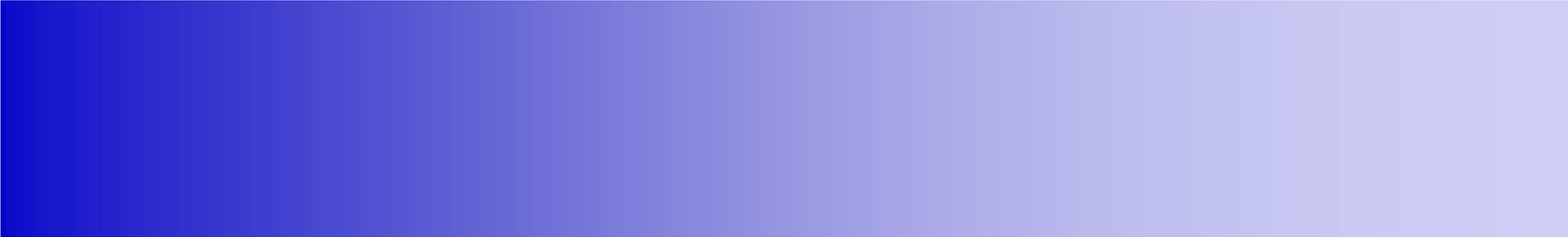


Application Service

AP-NR1500 IP Voice Call Recording Server

- Call Center Application
- Enterprise Application
- All IP Network Application





Thank you!

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