

# 2 Port STT (Speech to Text) VoIP Module for AP-SVG3000



**AddPac**

**AddPac Technology**

Sales and Marketing

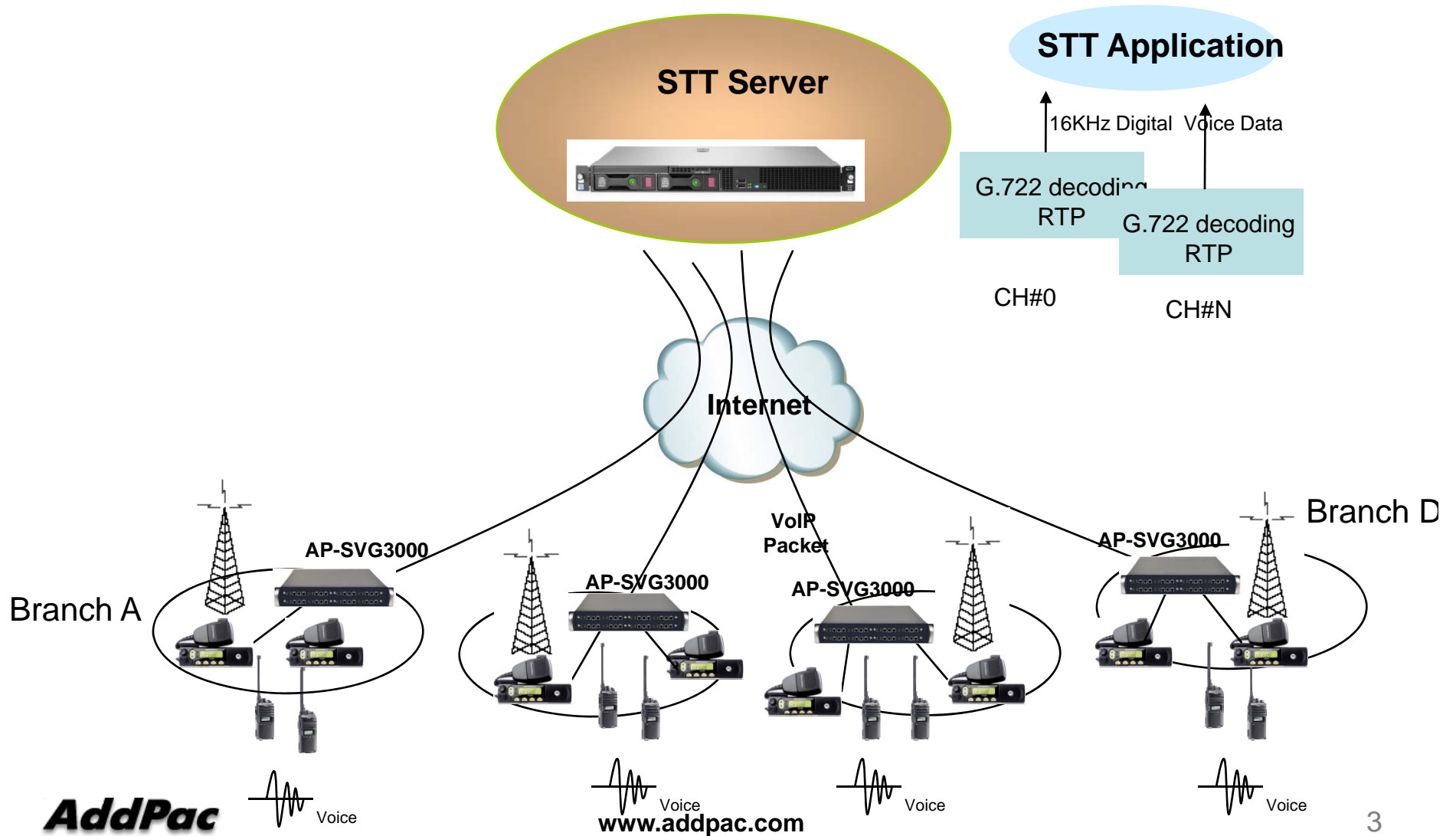
[www.addpac.com](http://www.addpac.com)

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- STT VoIP Service Network Diagram
- AP-SVG3000 H/W Platform
- AP-STT2 Module VoIP Service Features



# STT (Speech to Text) VoIP Service



# AP-SVG3000

## 8-Port STT VoIP Gateway



# 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

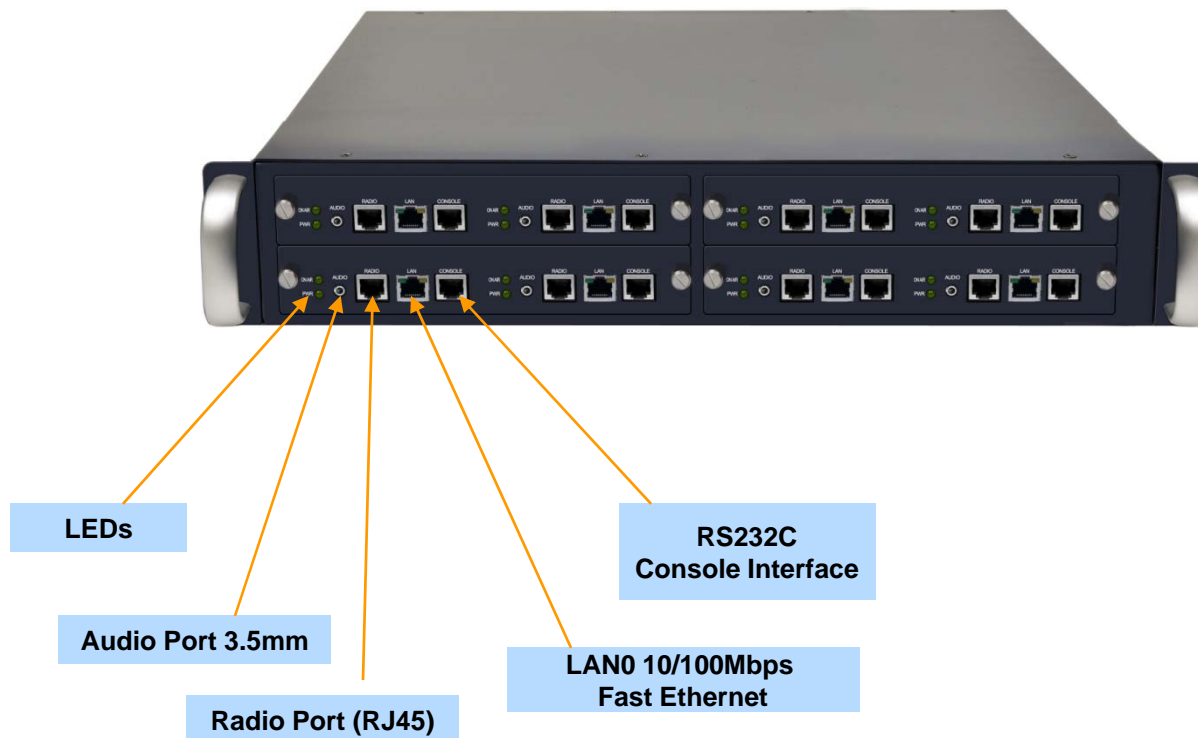
High-end  
DSP

- 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







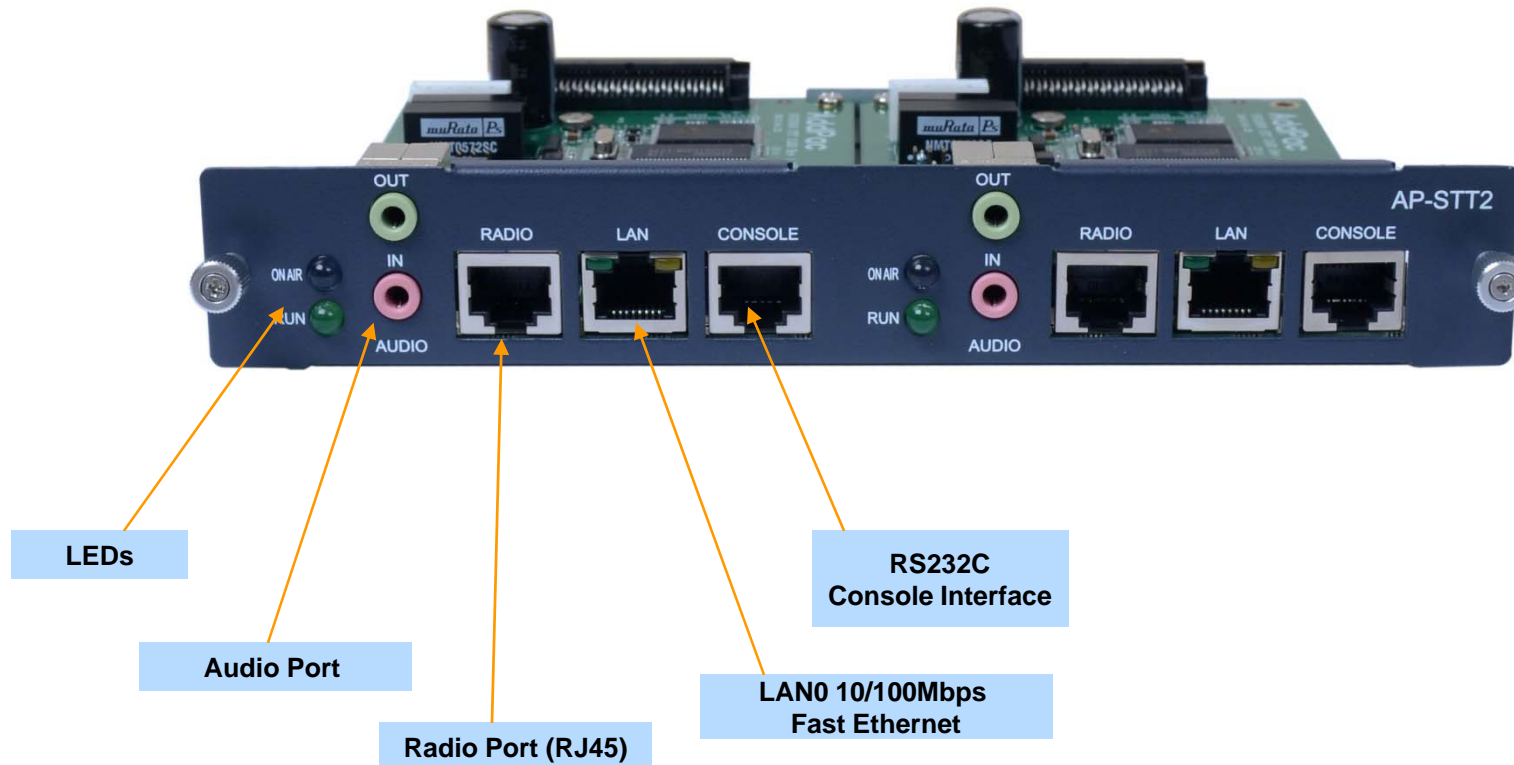
# STT2 VoIP Gateway Module

# Contents

- Hardware Specification
- Software Specification

# Hardware Specification

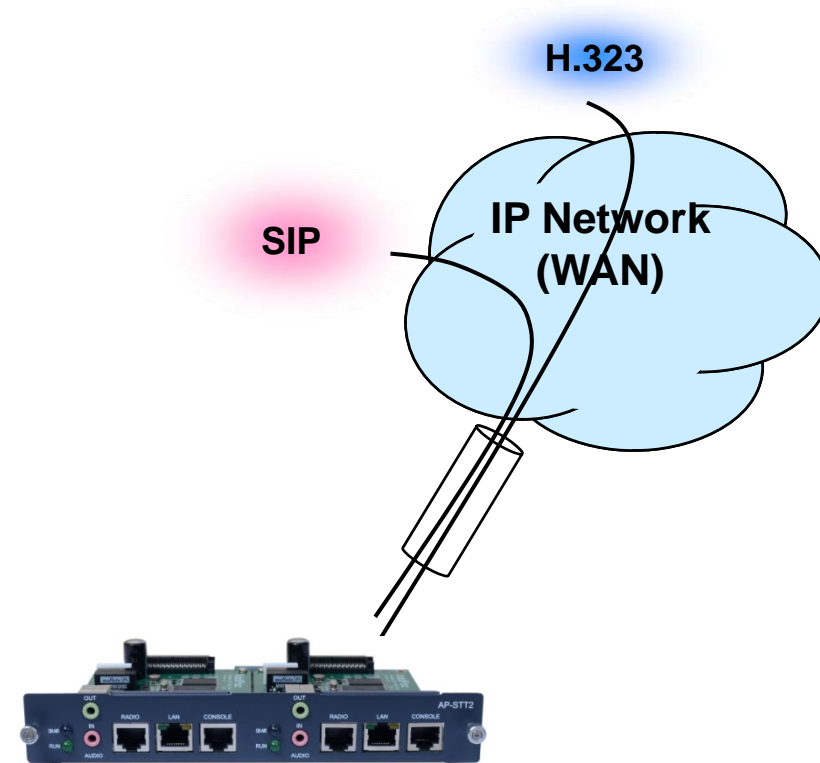
## STT2 VoIP Gateway Module



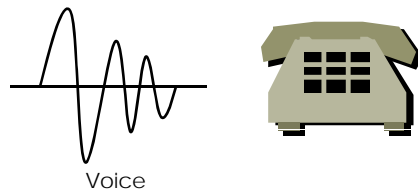
# VoIP (Voice over IP) Service

## STT2 VoIP Gateway Module

- **H.323, SIP Dual VoIP Signaling 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



Concurrent Dual VoIP Stack



# VoIP (Voice over IP) Service

## STT2 VoIP Gateway Module

- **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

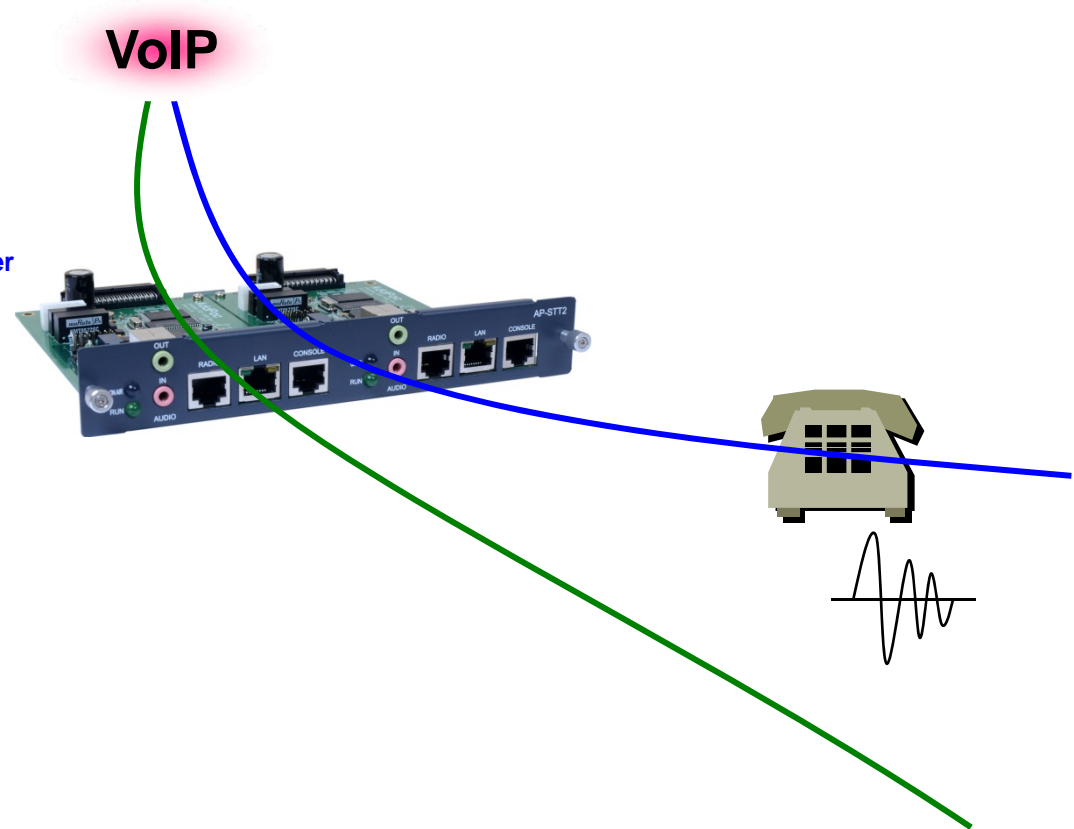
## STT2 VoIP Gateway Module

- **Voice Codec**

- G.722 16KHz
- G.711 A-Law, G.711 U-Law
- G.726 r16, G.726 r32
- VAD (Voice Activity Detection) function support
- DTMF relay support (H.323, SIP) 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

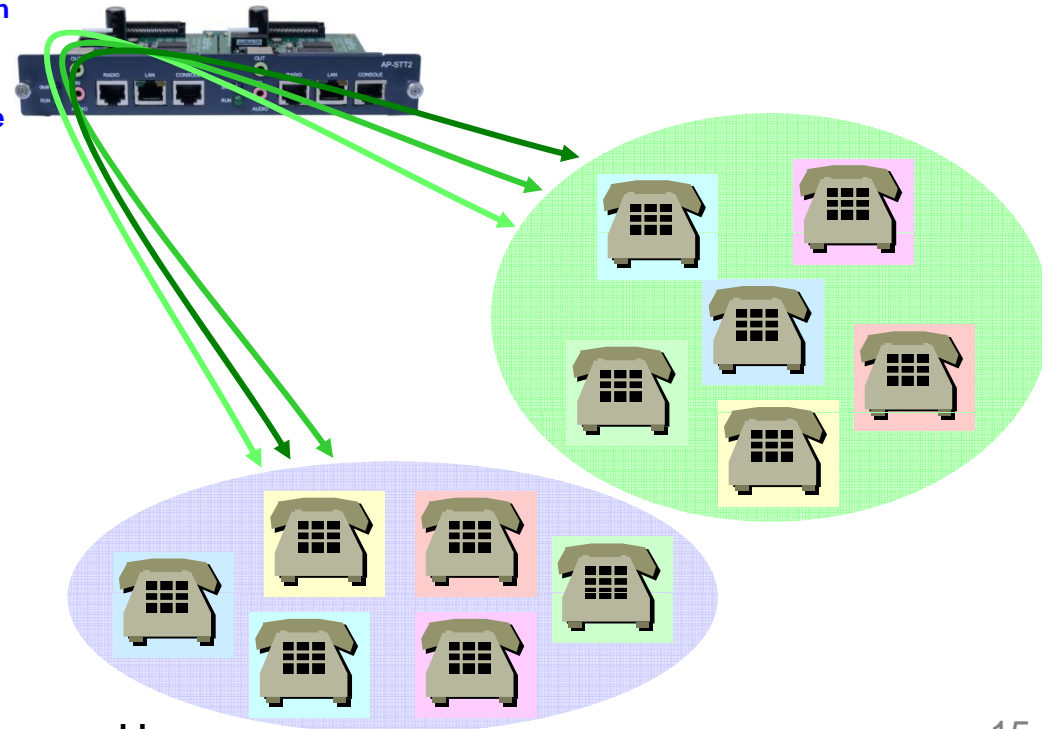
## STT2 VoIP Gateway Module

### • 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.)

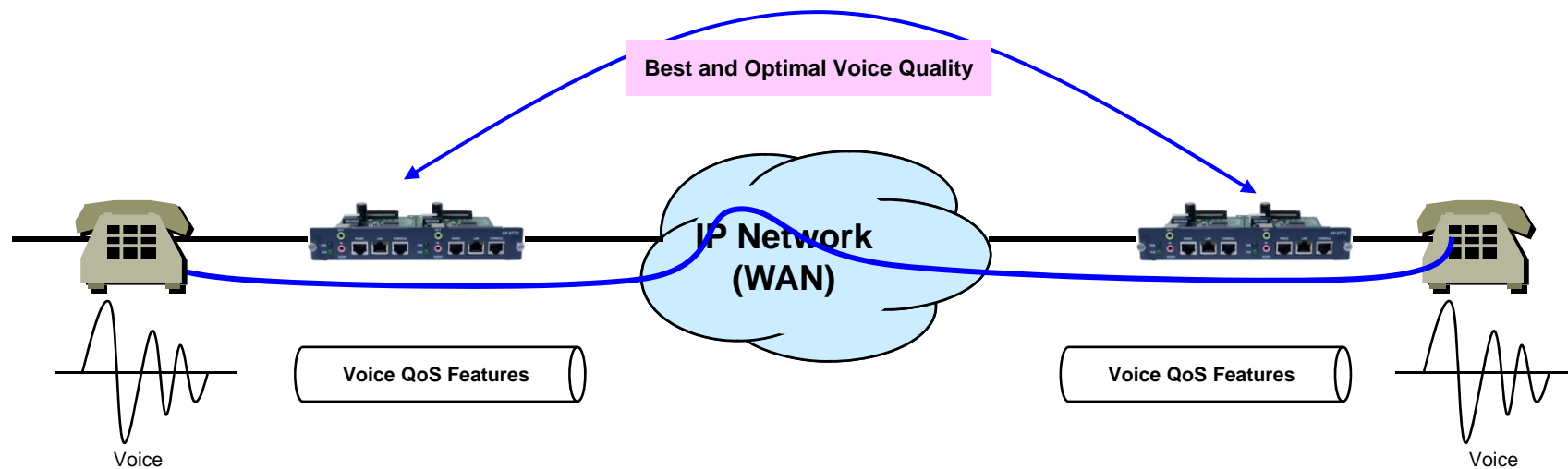
- Call transfer for internal calls
- Call pickup for internal calls
- Calling and called number conversion for VoIP outbound calls
- Calling and called number conversion for VoIP inbound calls
- Fax broadcasting call control



# Advanced QoS Features

## STT2 VoIP Gateway Module

- 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

## STT2 VoIP Gateway Module

### Basic Network Protocols

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

### Routing Protocol

- IPv4 : Static

### Service Protocol

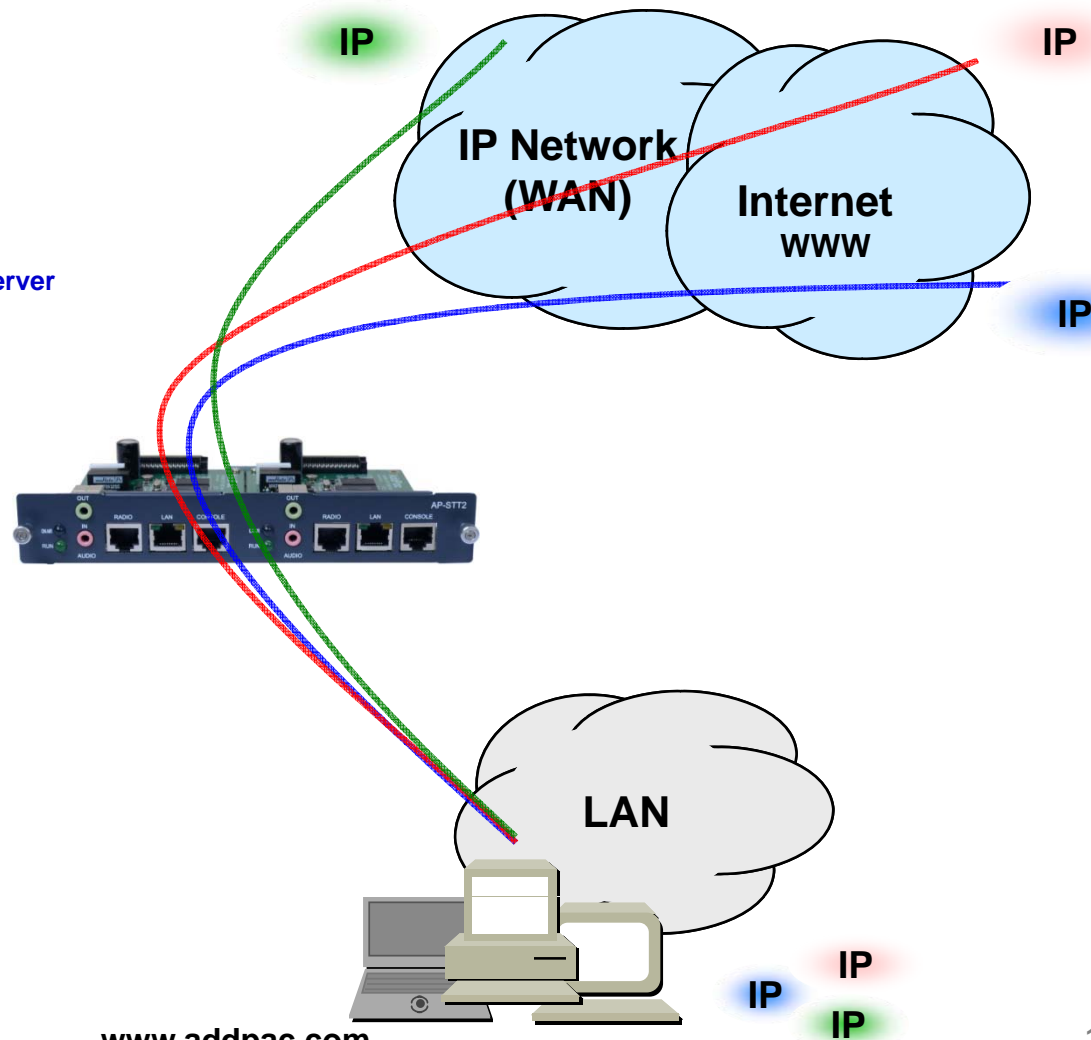
- FTP, Telnet, TFTP, DHCP Server/Relay, SNMP Server
- CDP (Cisco Discovery Protocol)
- DNS Resolver , DDNS(nsupdate)
- Bridge
- Syslog

### 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

## STT2 VoIP Gateway Module

- **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**

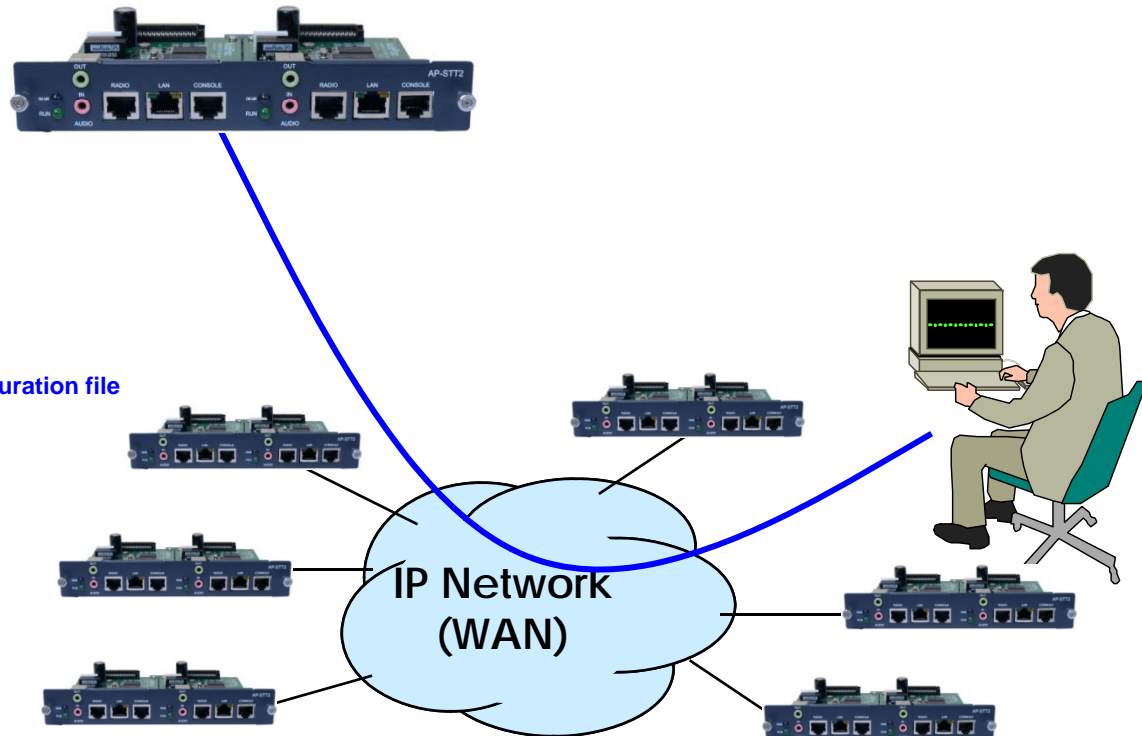
- HTTP server based APOS image and configuration file auto-upgrade support

- **Batch Job Function**

- Text based script downloading

- **Interoperable with AP-VPMS Service**

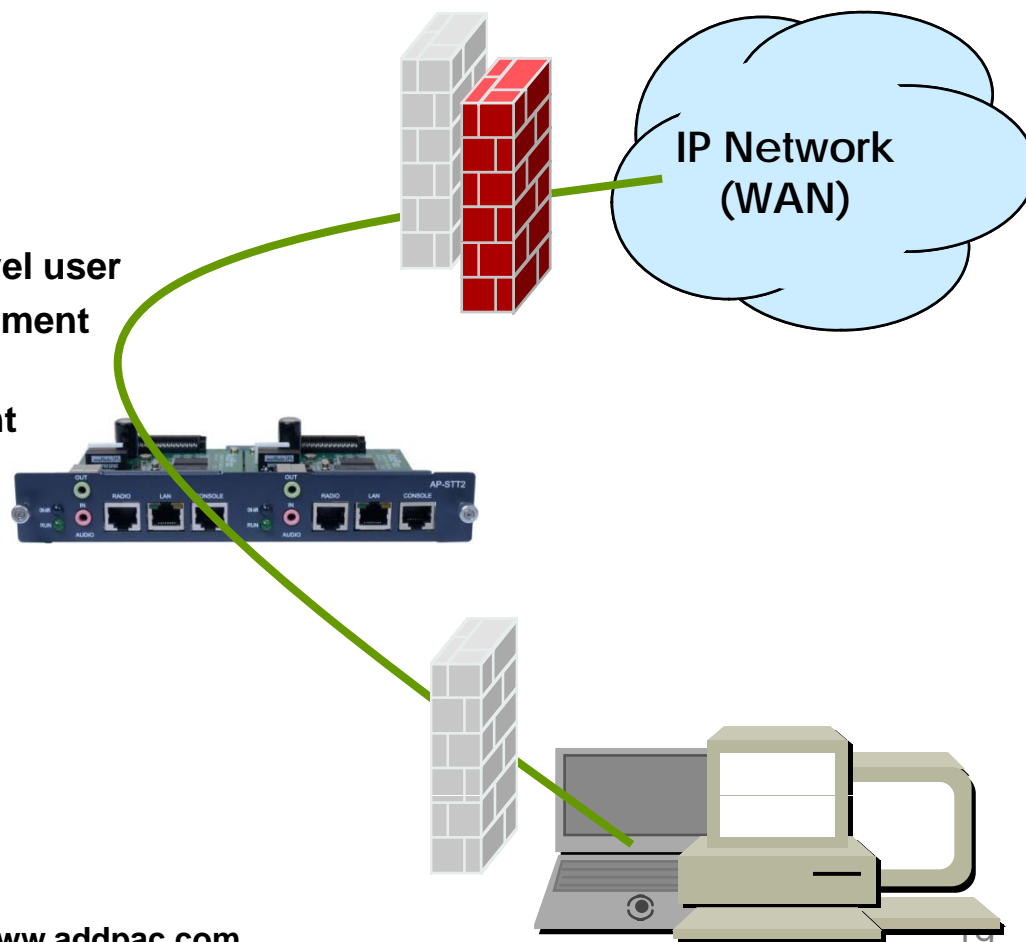
- AddPac VoIP Plug & Play Management System (AP-VPMS)



# Security Management

## STT2 VoIP Gateway Module

- 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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