

AP602

SIP VoIP Paging Terminal

High Performance SIP based Paging Terminal



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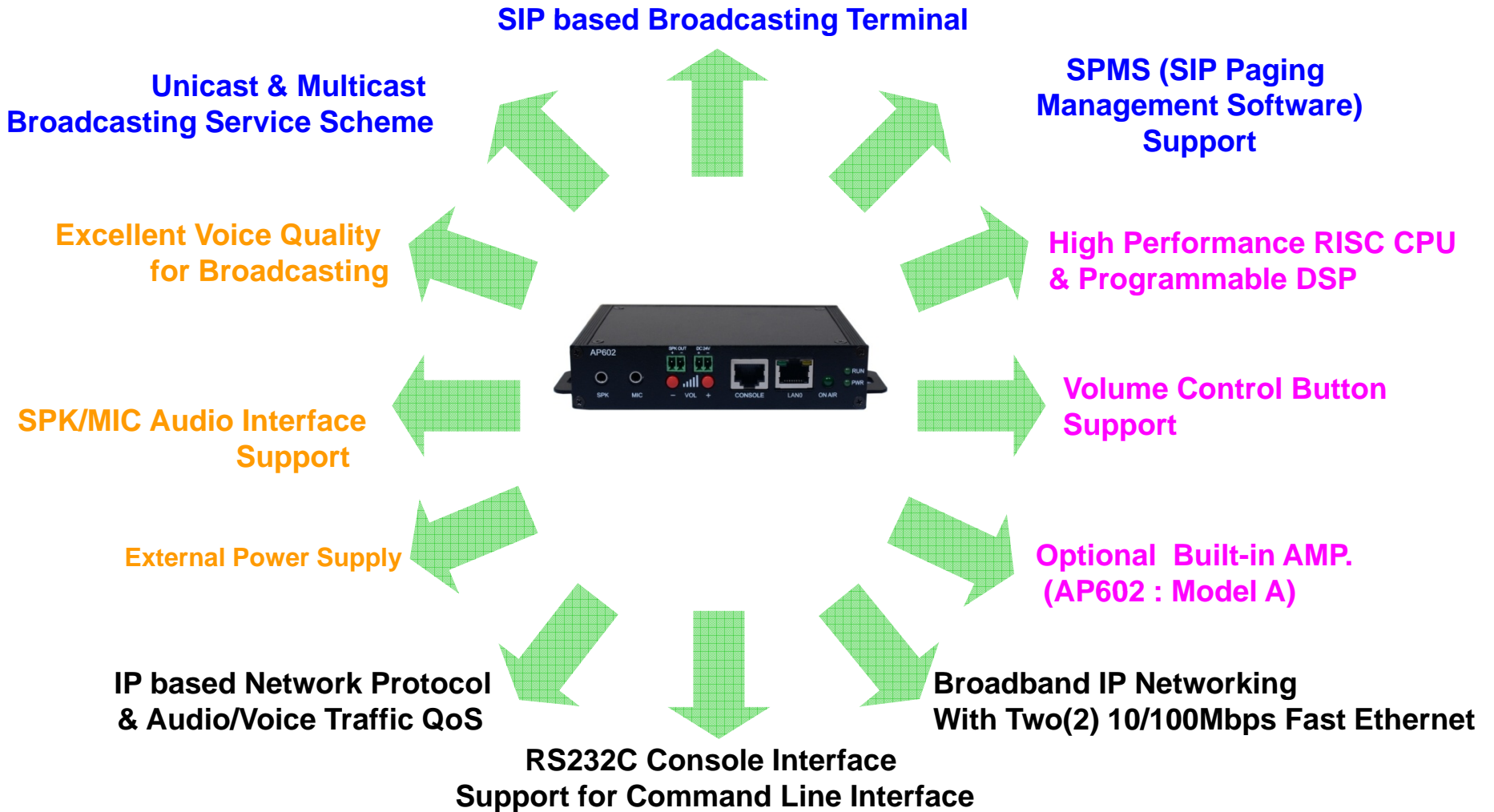
Product Overview

AP602 SIP VoIP Paging Terminal

- SIP based Voice Broadcasting Terminal Solution
- Hardware Architecture for Voice Broadcasting Terminal Service
- Remote Broadcasting Service at terminal side
- High Quality Voice Codec Support (High Quality Codec, G.711, G.726, etc)
- RTP/UDP Protocol Support
- Unicast and Multicast Broadcasting Scheme
- SPMS (SIP Paging Management Software) Support
- One(1) MIC Port 3.5mm Connector, One(1) SPK Port : 3.5mm Connector
- Volume Control Button Support (Up, Down)
- Option (AP602 : Model A) : AMP. Built-in, Terminal SPK Port
- Two(2) 10/100Mbps Fast Ethernet LAN Port
- High-Quality Audio/Voice Service
- Firmware Upgradeable Architecture
- Broadcasting Solution with Outstanding Network Service Capability
- External Power Supply

Product Highlights

AP602 SIP VoIP Paging Terminal



Hardware Specification

AP602 SIP VoIP Paging Terminal

RISC
CPU

High-end
DSP

- RISC Microprocessor Computing Power
- High-end Programmable DSP Hardware Architecture
- High Quality Audio Encoding/Decoding Service
- Two(2) 10/100Mbps Fast Ethernet (RJ45)
- One(1) RS-232C Interface (RJ45)
- SPK/MIC Audio Interface Support
- Volume Control Button Support (Up, Down)
- External Power Supply Support
- Option : 40Watt Digital AMP. Built-in (AP602 : Model A)
- Option : I/O Interface (Alarm Input 1Port, Relay Output 1Port, I2C Interface)

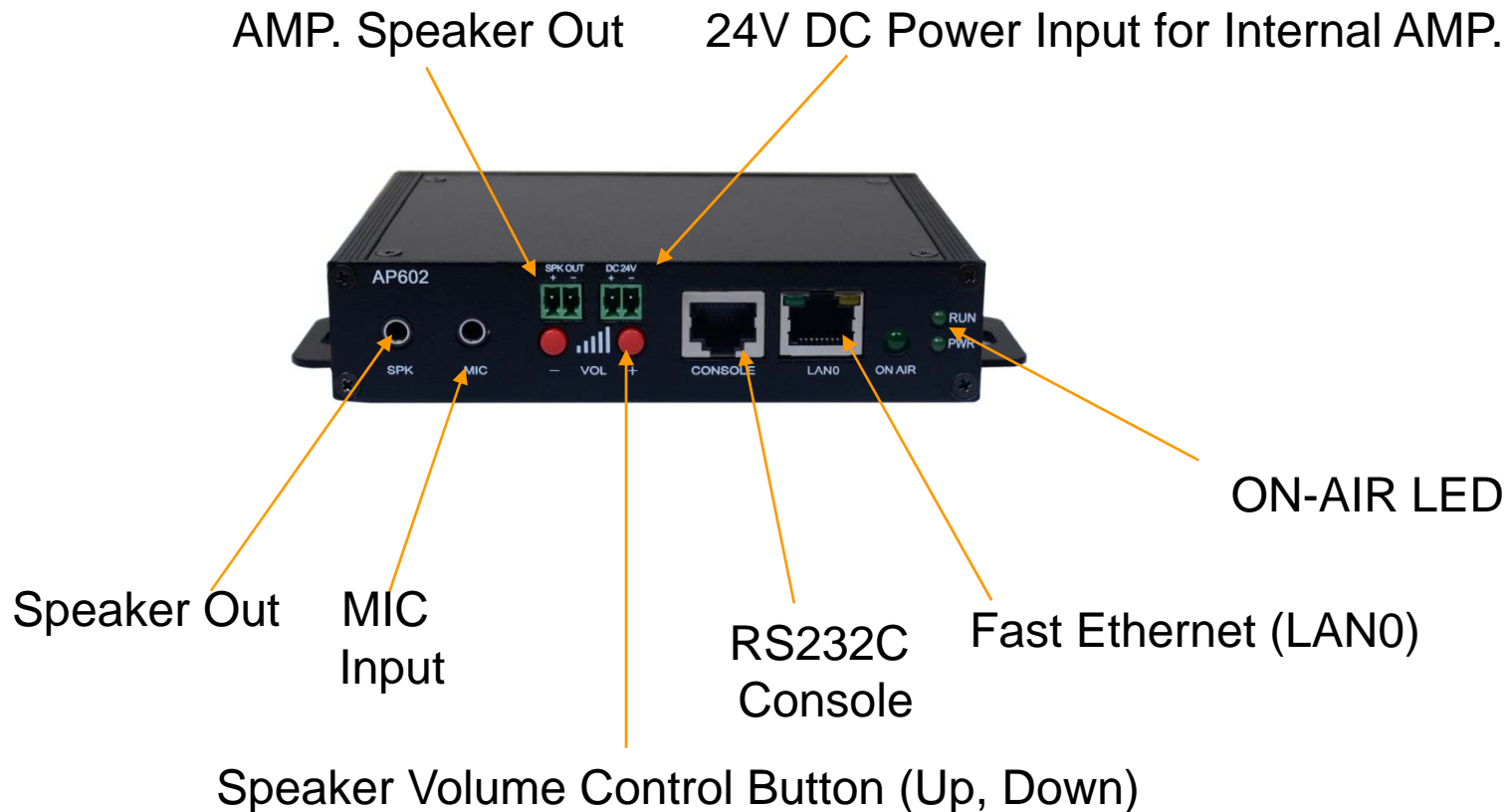
Hardware Specification

AP602 SIP VoIP Paging Terminal

RISC
CPU

High-end
DSP

Front Side (AMP. Built-In Model : AP602 : Model A)



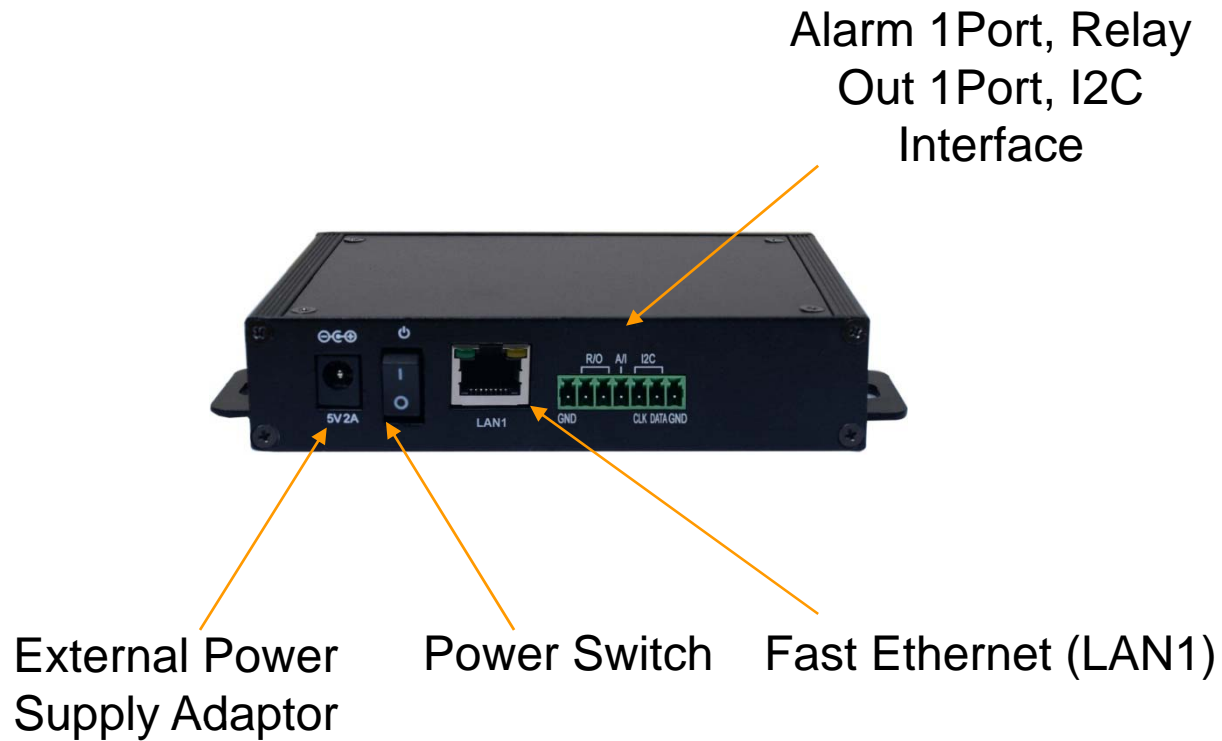
Hardware Specification

AP602 SIP VoIP Paging Terminal

RISC
CPU

High-end
DSP

Back Side



APOS™ Service Features

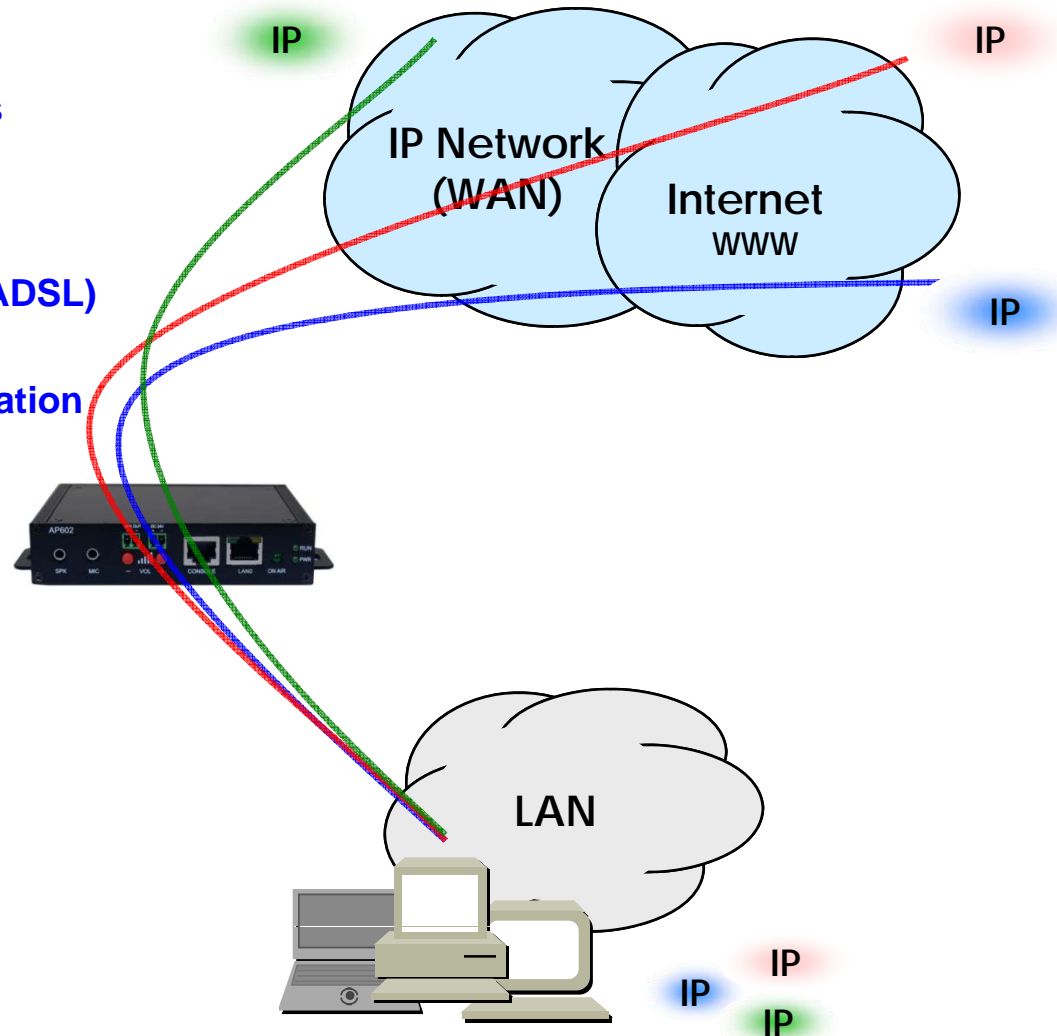
AP602 SIP VoIP Paging Terminal

- **IP Routing Protocols**

- Multi-protocol Internetworking Services
- Static & Default IP routing

- **WAN Protocols**

- Point-to-Point Protocol (PPPoE for ADSL)
- IEEE 802.3 Ethernet
- PPTP support for secure communication



APOS™ Service Features

AP602 SIP VoIP Paging Terminal

- **Network Managements**
 - Standard SNMP Agent (MIB v2) Support
 - Remote Management using Console, Telnet
 - Web based Management using HTTP Server Interface
- **Security Functions**
 - Standard & Extended IP Access List
 - Enable/Disable for Specific Network Protocols
 - Multi-level User Account Management
 - Auto-disconnect for Telnet/Console Sessions
 - PPP User Authentication Supports (PAP & CHAP)
- **Operation & Managements**
 - System Performance Analysis for Process, CPU, Connection Interface
 - Debugging, System Auditing, and Diagnostics Support
 - System Booting and Auto-rebooting with Watchdog Feature
 - System Managements with Data Logging
 - IP Traffic Statistics with Accounting

APOS™ Service Features

AP602 SIP VoIP Paging Terminal

- **Network Protocols**
 - DHCP Server & Relay Functions
 - Network Address Translation (NAT) Function
 - Port Address Translation (PAT) Function
 - Transparent Bridging (IEEE Standard) Function
 - Spanning Tree Bridging Protocol Support
 - Remote Bridging Support
 - Concurrent Routing and Bridging Support
 - Cisco Style Command Line Interface (CLI)
 - Network time Protocol (NTP) Support

VoIP Service Features

AP602 SIP VoIP Paging Terminal

- **SIP Protocol Service**

- 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 Service Features

AP602 SIP VoIP Paging Terminal

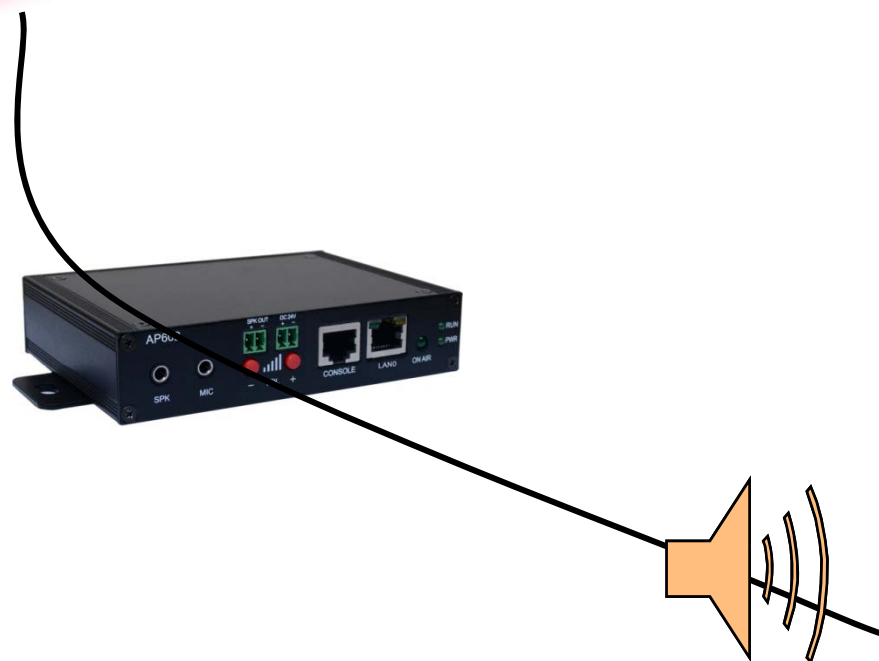
- **Voice Codec**

- 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 Protocols**

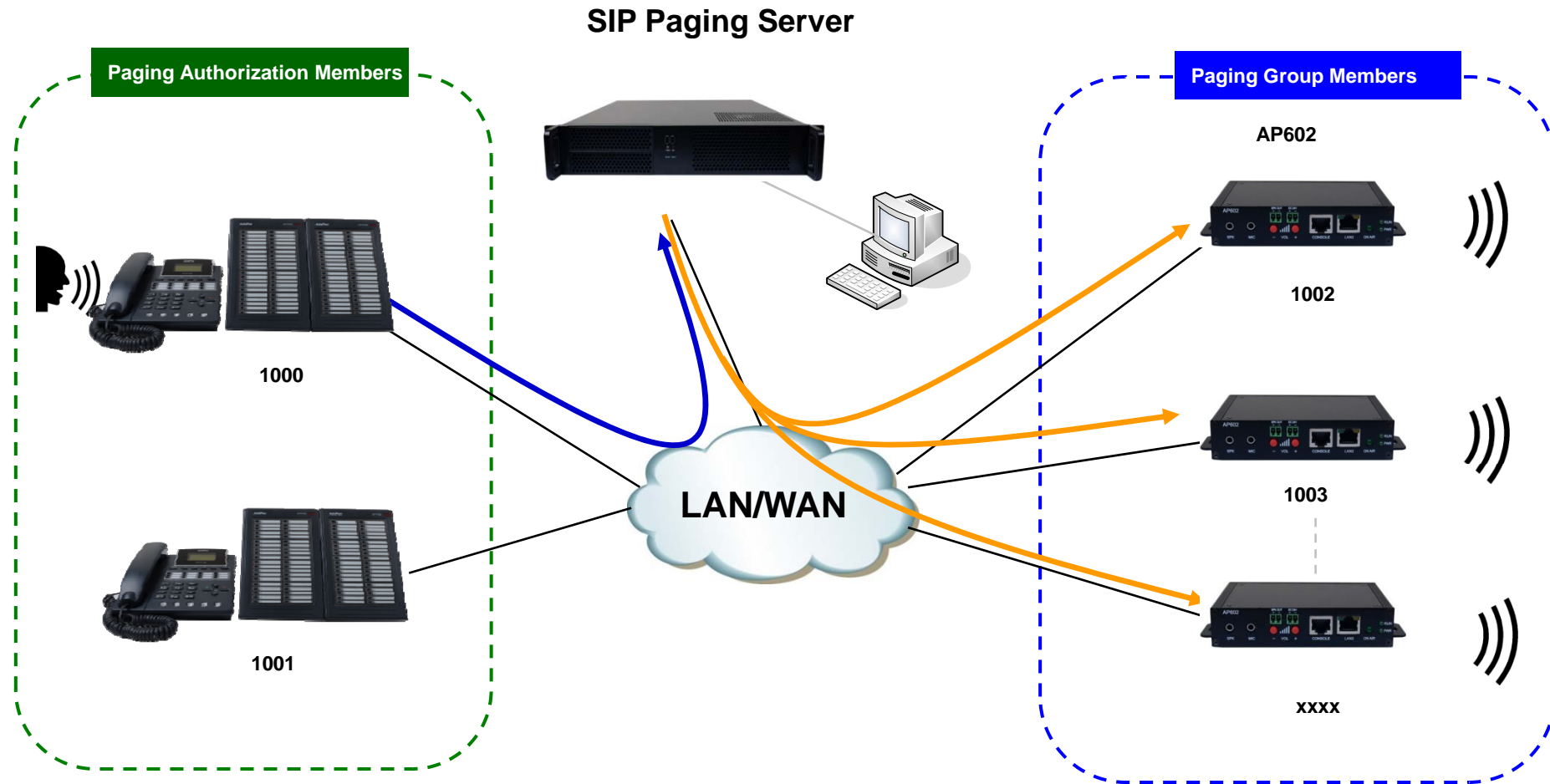
- Redundant RTP packet transmission in case of severe packet loss
- Dynamic jitter buffer management and RTP packet jitter and loss compensation with heuristic & DSP error concealment
- Static jitter buffer setting support
- Voice frame per RTP packet number control for each codec

IP Broadcasting



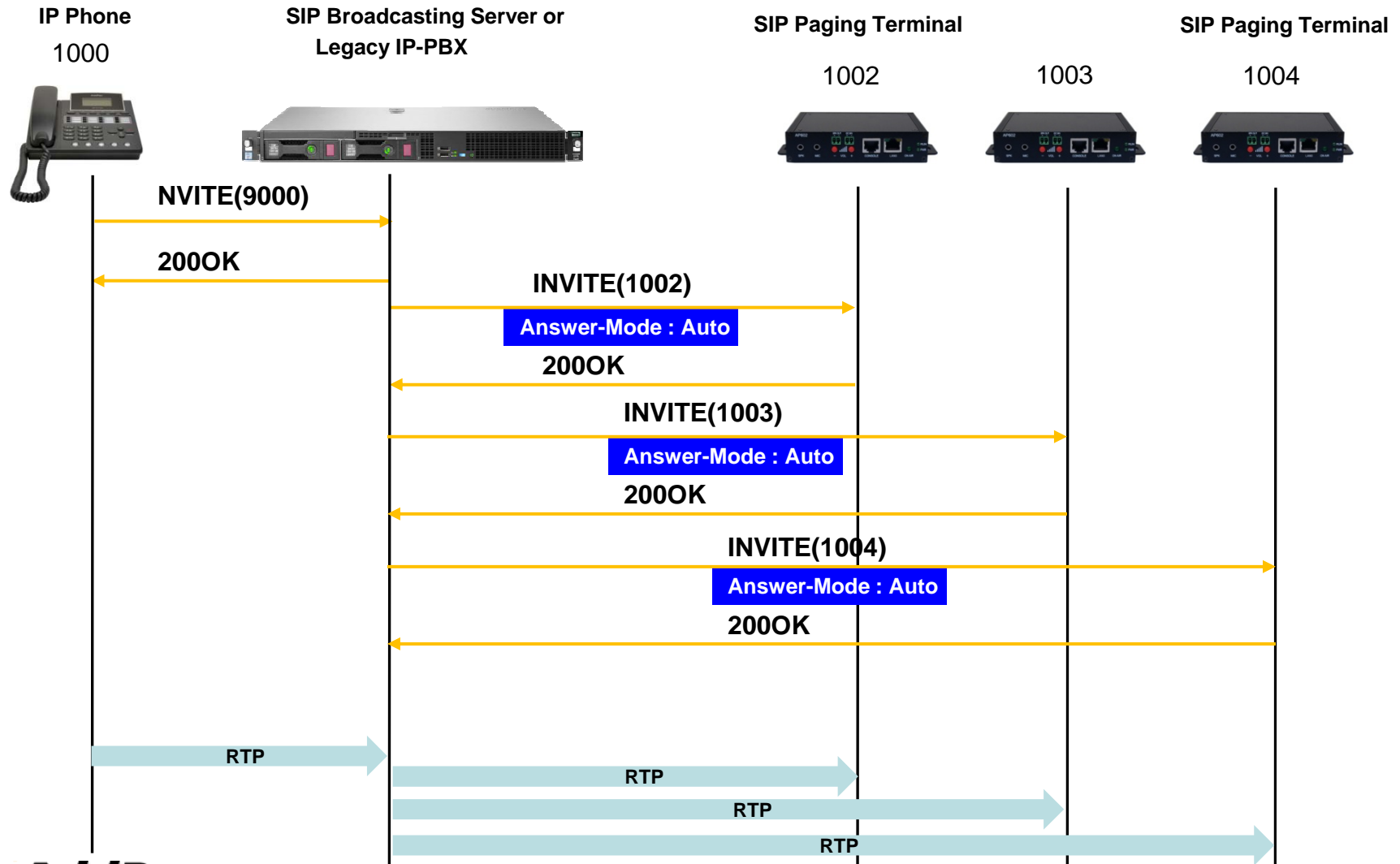
SIP Broadcasting Network Diagram

AP602 SIP VoIP Paging Terminal



Paging Group Signaling Flow

AP602 SIP VoIP Paging Terminal



Ordering Information

- **AP602 SIP VoIP Paging Terminal Hardware**
 - AP602 Main Body
 - Two(2) Fast Ethernet Port
 - SPK, MIC Port (3.5mm x 2)
 - SPK AMP. Out (Option : Model A : 40Watt)
 - Volume Up Down Button
 - I/O Interface : Alarm InPut 1 Port, Relay Output 1Port, I2C Interface
 - Voice Codec : G.722(16KHz), G.711,G.726,G.729A,G,723.1, etc
 - RISC Microprocessor with High-end Programmable DSP Architecture
 - Including Network Cable Set & Power Supply, etc.
- **Built-in APOS Internetworking Software for AP602**
- **Including 1 Year Hardware Warranty**
- **Product Documents**
 - Install and Operation Guide (PDF)
- **Pricing**
 - AddPac Technology Regional Sales Manager
 - Authorized Sales and Marketing Representatives
 - Please Contact www.addpac.com



Thank you!

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