

AP1602

SIP Audio Broadcasting Terminal

High Performance SIP Audio Broadcasting Terminal



AddPac

AddPac Technology

Sales and Marketing

www.addpac.com

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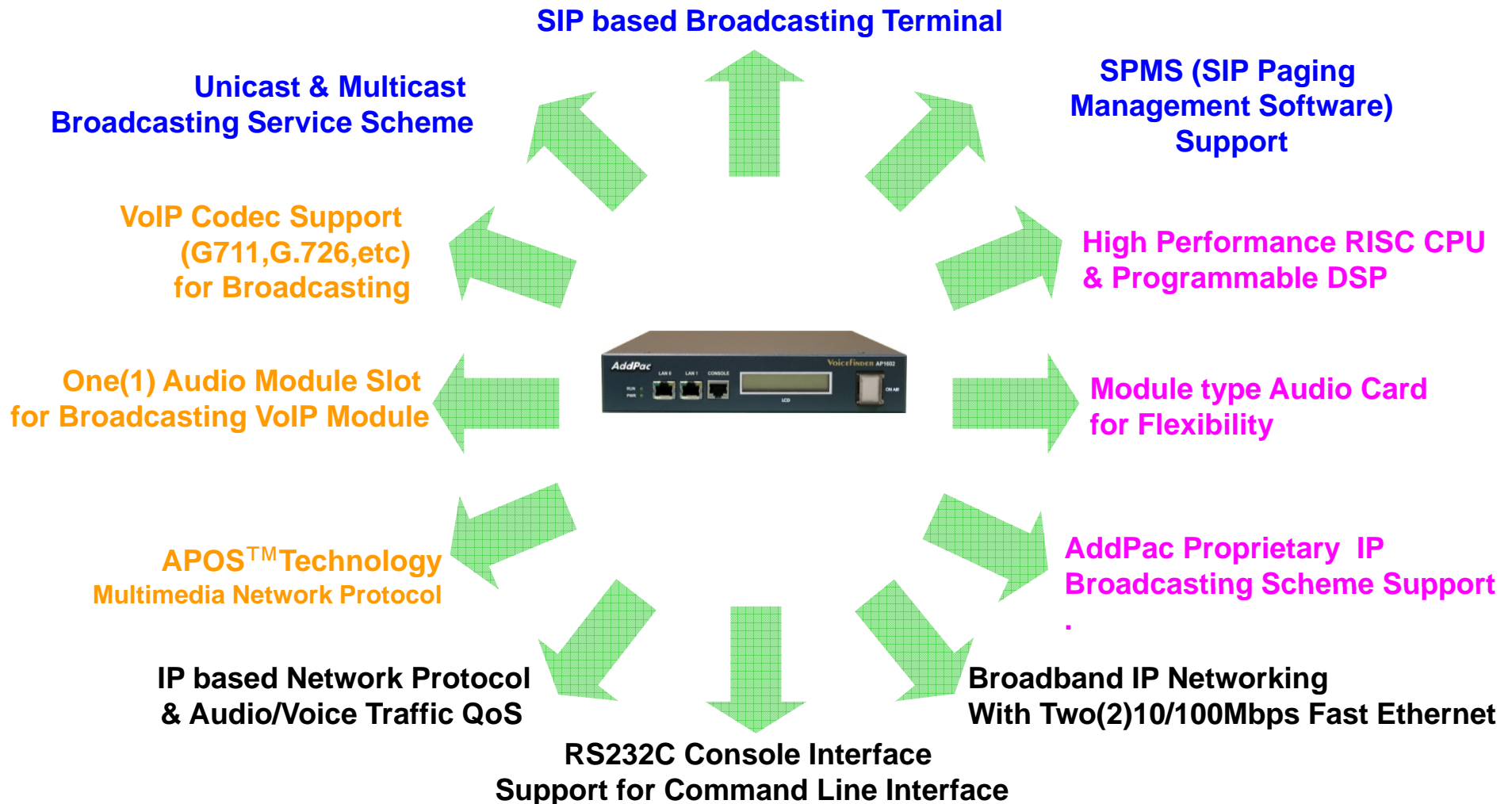
Product Overview

AP1602 SIP Audio Broadcasting Terminal

- SIP Protocol based Audio Broadcasting Terminal Solution
- RTP (Real-time Transport Protocol) Support for Media Transmission
- IP based Audio Broadcasting Terminal Solution (AddPac Proprietary Protocol)
- Hardware Architecture for Audio Broadcasting Terminal Service
- One(1) Module Slot for Audio Encoding & Decoding Service
- Remote Broadcasting Service at terminal side
- VoIP Codec Support (G.711, G.726, etc)
- Unicast and Multicast Broadcasting Scheme
- SPMS (SIP Paging Management Software) Support
- Various Audio Broadcasting Module Support
- On-AIR Blue LAMP
- Firmware Upgradeable Architecture
- Broadcasting Solution with Outstanding Network Service Capability

Product Highlights

AP1602 SIP Audio Broadcasting Terminal



Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

- RISC Microprocessor Computing Power
- High-end Programmable DSP Hardware Architecture
- One(1) Module Slot for Audio Broadcasting Codec Module
- VoIP Audio Encoding/Decoding Service
- ON-AIR Blue LAMP
- Two(2) 10/100Mbps Fast Ethernet Interface
- Option Module : AP-AUDIO2
 - Two(2) 3.5mm Audio Input/Output Interface
- Option Module : AP-AUD1S3
 - One(1) 3.5mm Audio Input/Output Interface
 - Three(3) FXS VoIP Interface

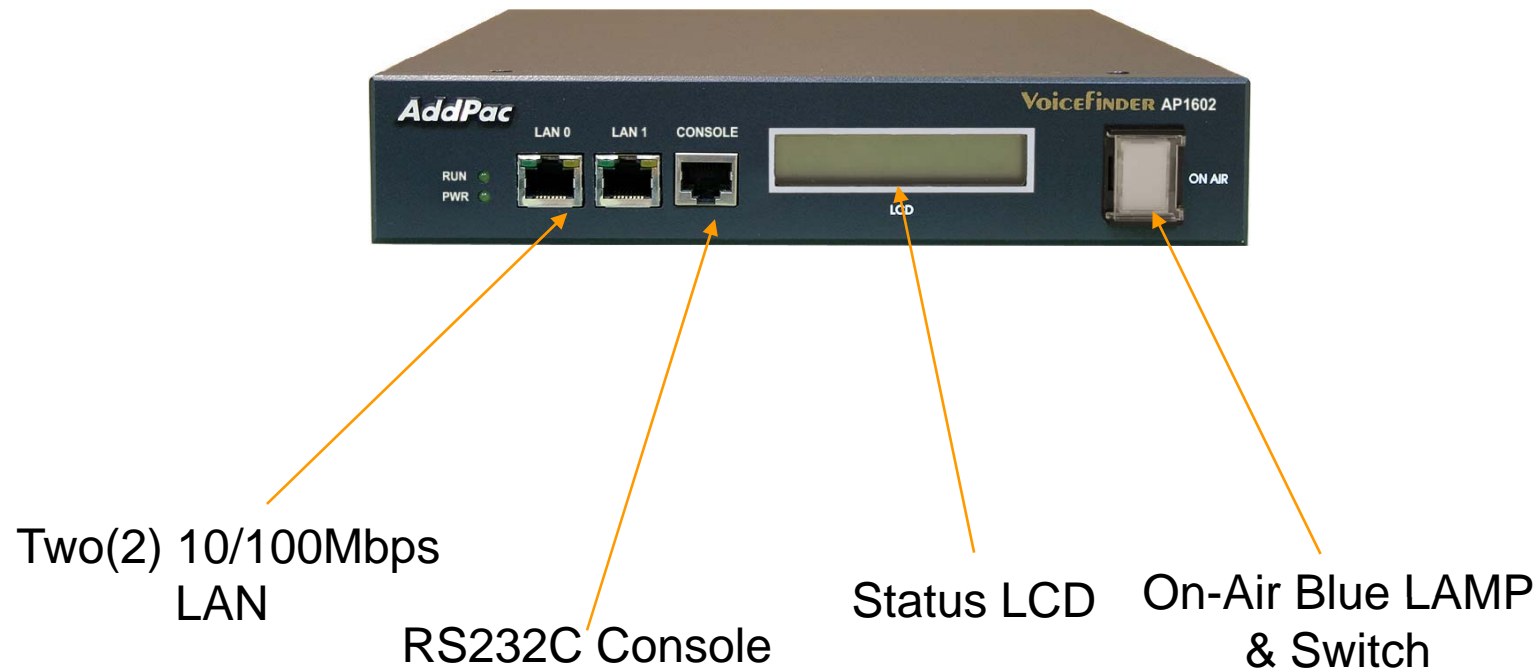
Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP1602 Front Side



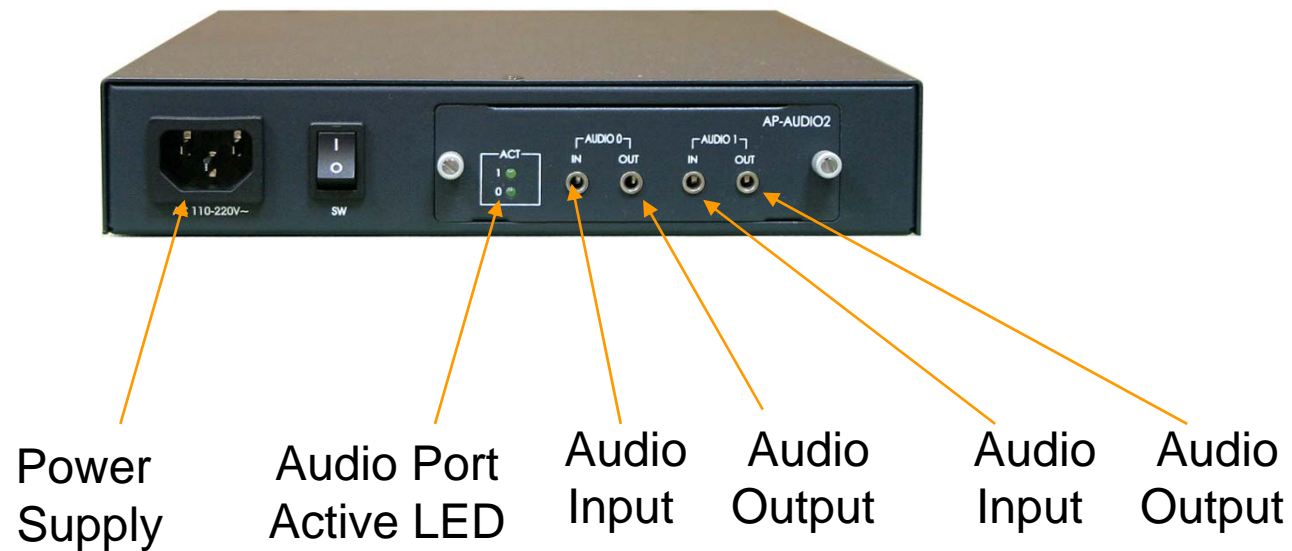
Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP1602 Back Side



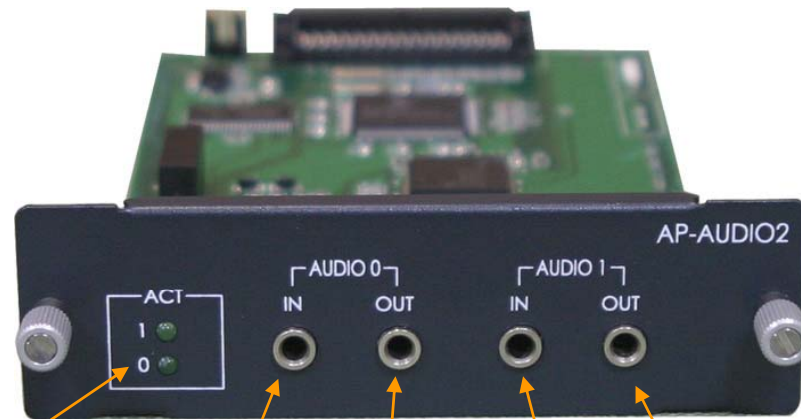
Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP-AUDIO2 Board



Audio Port
Active LED

Audio
Input

Audio
Output

Audio
Input

Audio
Output

Audio 0 Channel

Audio 1 Channel

AddPac

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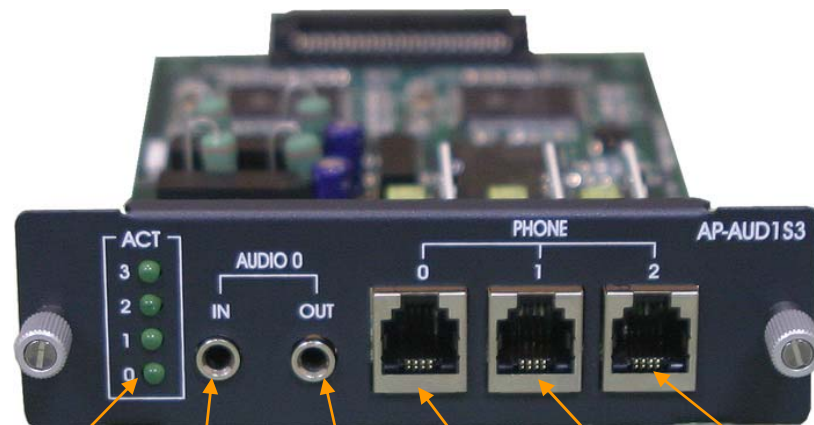
Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP-AUD1S3 Board



Audio Port
Active LED

Audio
Input

Audio
Output

FXS
Port 0

FXS
Port 1

FXS
Port 2

Audio 0 Channel

Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP1602 Audio Module

| Audio Module Type | Audio Module Features |
|--|---|
| AP-AUDIO2 | Two(2)-Channel Audio In/Out Port |
|  | Audio Encoding/Decoding Service |
| | Audio IN : MIC IN Audio OUT :Line OUT 3.5mm Stereo JACK |
| | G.711, G.726, G.729A, G.723.1 Audio Codec |


Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP1602 Audio Module

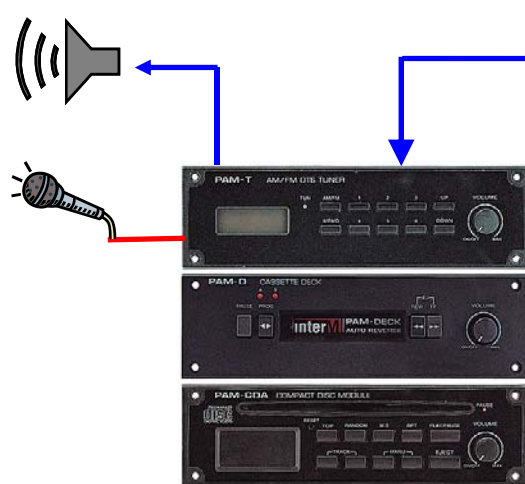
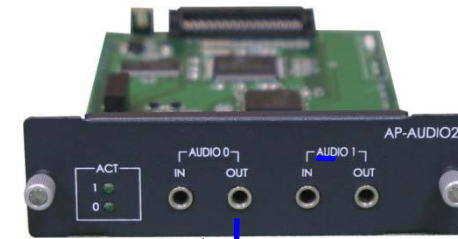
| Audio Module Type | Audio Module Features |
|--|---|
| AP-AUD1S3 | One(1)-Channel Audio In/Out Port |
|  | Audio Encoding/Decoding Service |
| | Audio IN : MIC IN |
| | Audio OUT :Line OUT 3.5mm Stereo JACK |
| | Three(3) FXS Port Interface (RJ11 x 3) |
| | G.711, G.726, G.729A, G.723.1 Audio Codec |

AP-AUDIO2 Module

AP1602 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP



- G.7xx Voice codec realizes IP voice broadcasting service
- Real time VoIP Broadcasting Service using RTP (Real-time Transport Protocol) Protocol

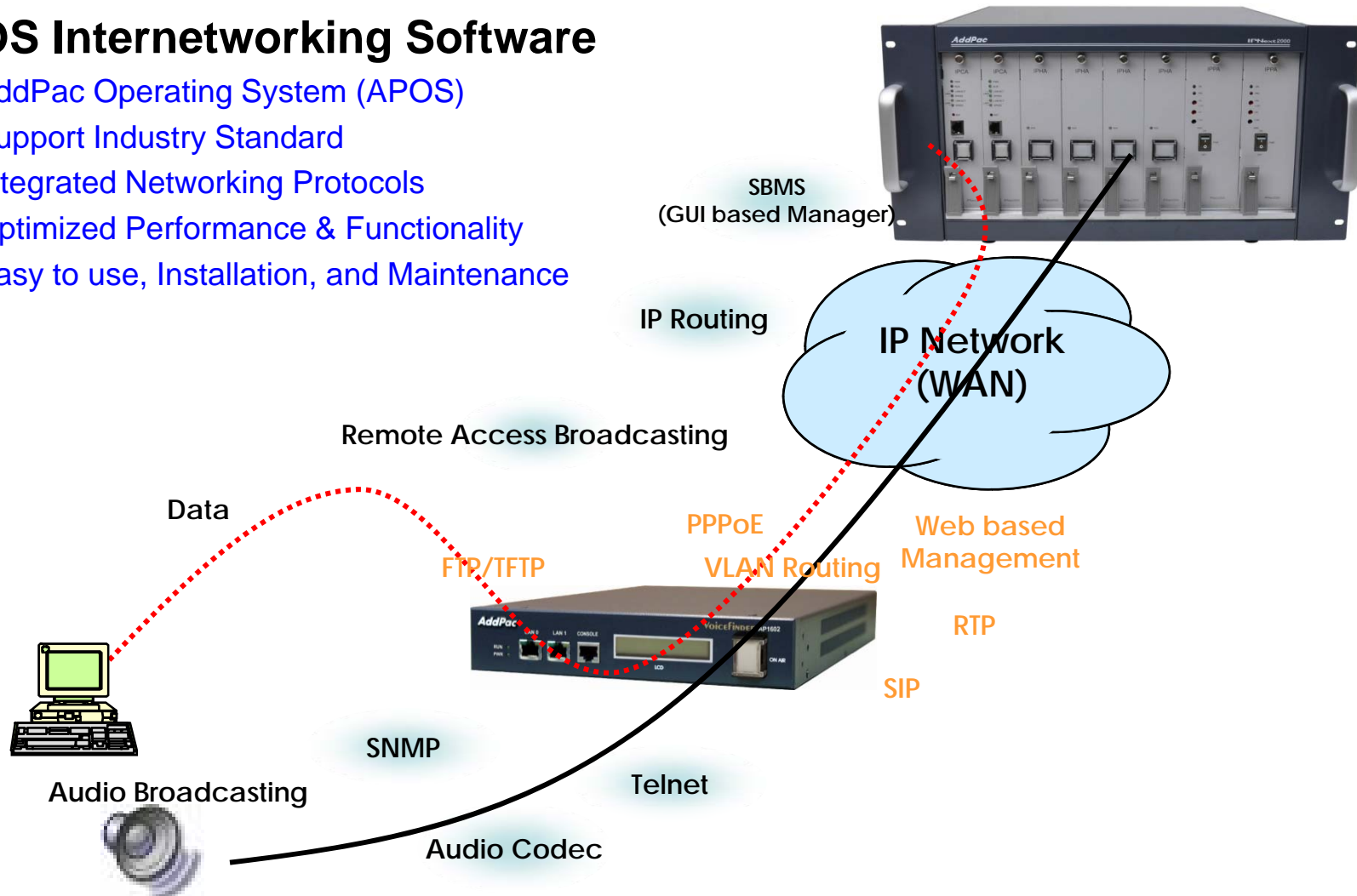
Headphone

APOS™ Service Features

AP1602 SIP Audio Broadcasting Terminal

- **APOS Internetworking Software**

- AddPac Operating System (APOS)
- Support Industry Standard
- Integrated Networking Protocols
- Optimized Performance & Functionality
- Easy to use, Installation, and Maintenance



APOS™ Service Features

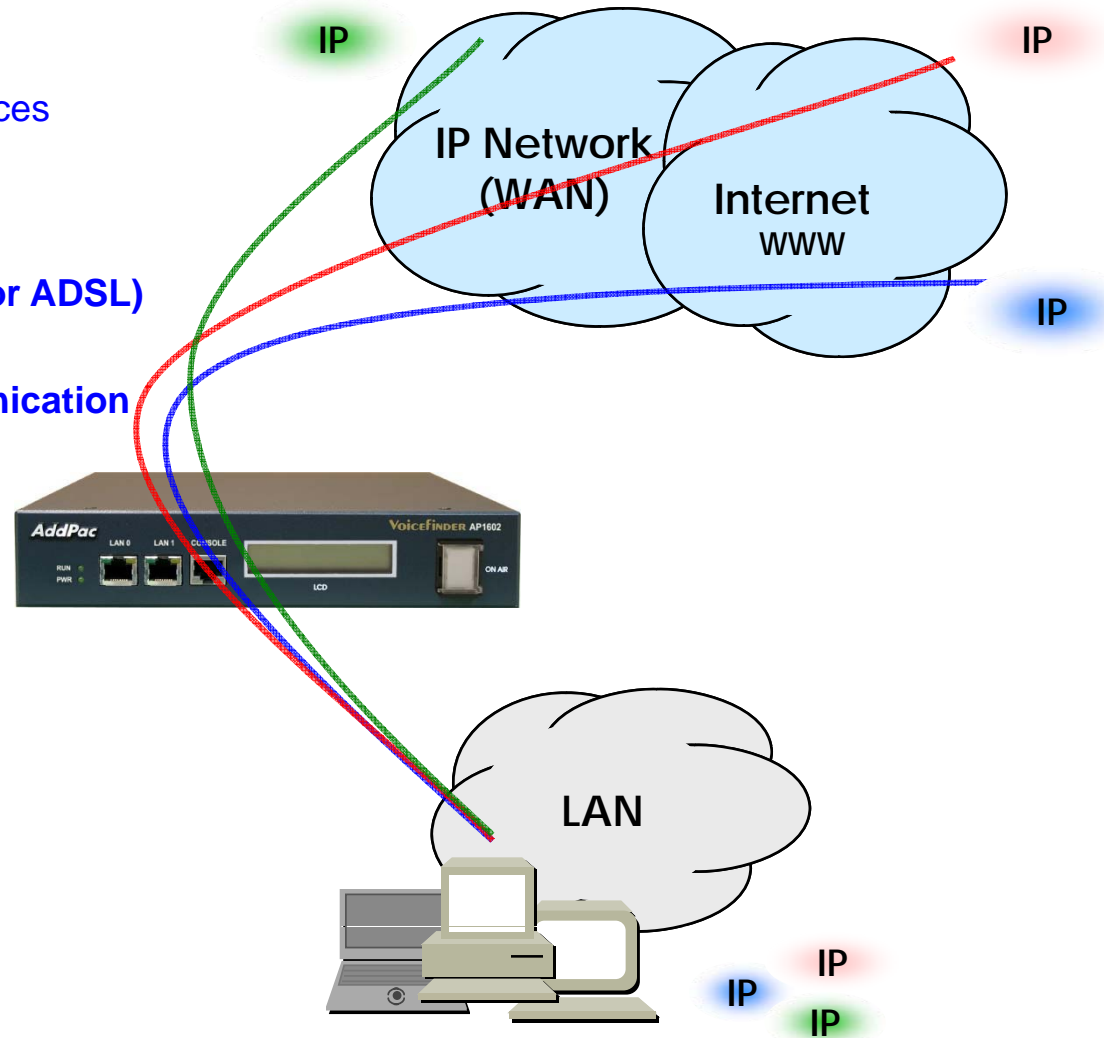
AP1602 SIP Audio Broadcasting Terminal

- **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

AP1602 SIP Audio Broadcasting 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

AP1602 SIP Audio Broadcasting 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

AP1602 SIP Audio Broadcasting 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

AP1602 SIP Audio Broadcasting Terminal

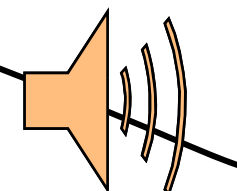
- Voice Codec

- 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

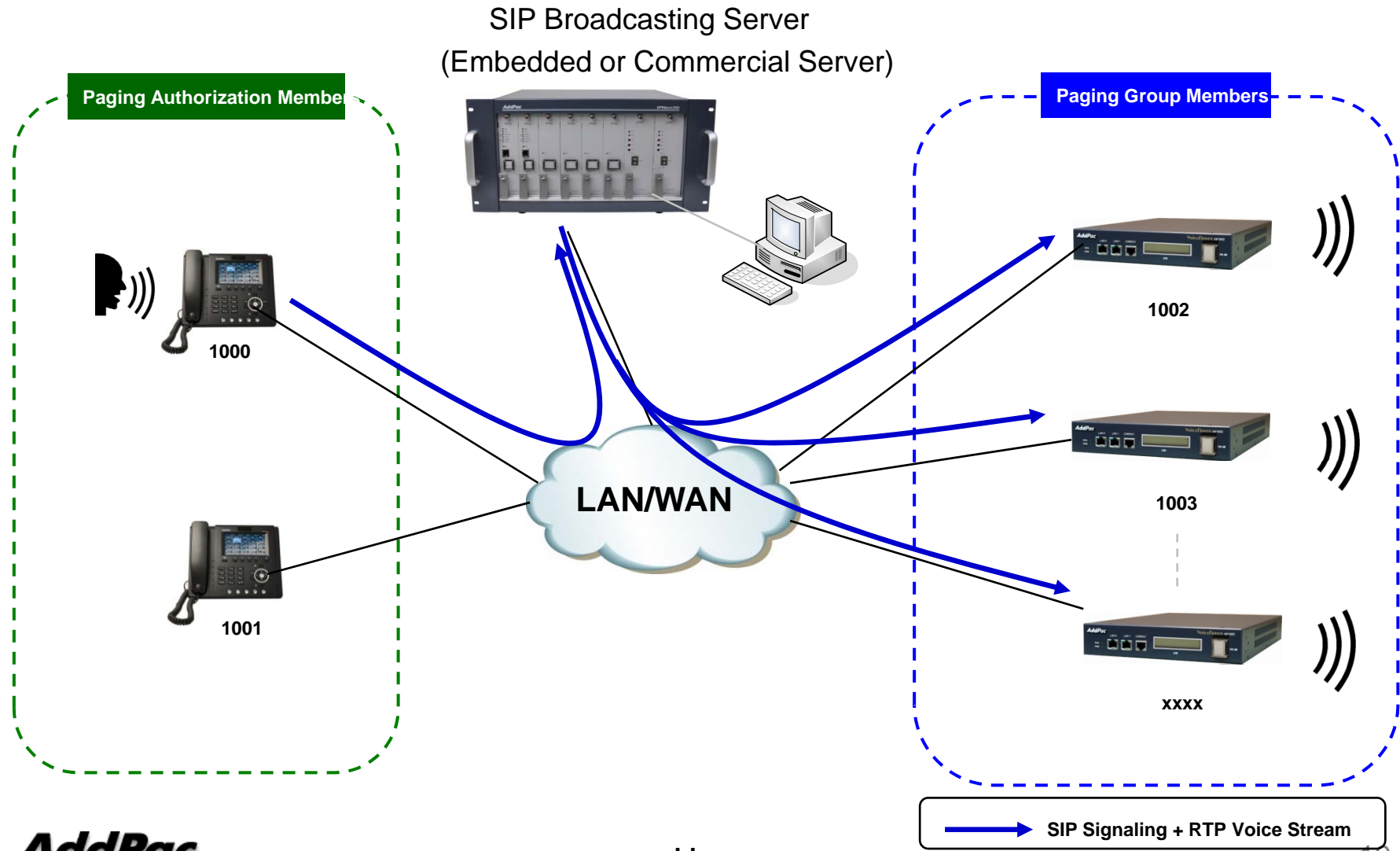
- Redundant RTP packet transmission in case of severe packet loss
- Dynamic jitter buffer management and RTP packet jitter and loss compensation with heuristic & DSP error concealment
- Static jitter buffer setting support
- Voice frame per RTP packet number control for each codec

IP Broadcasting



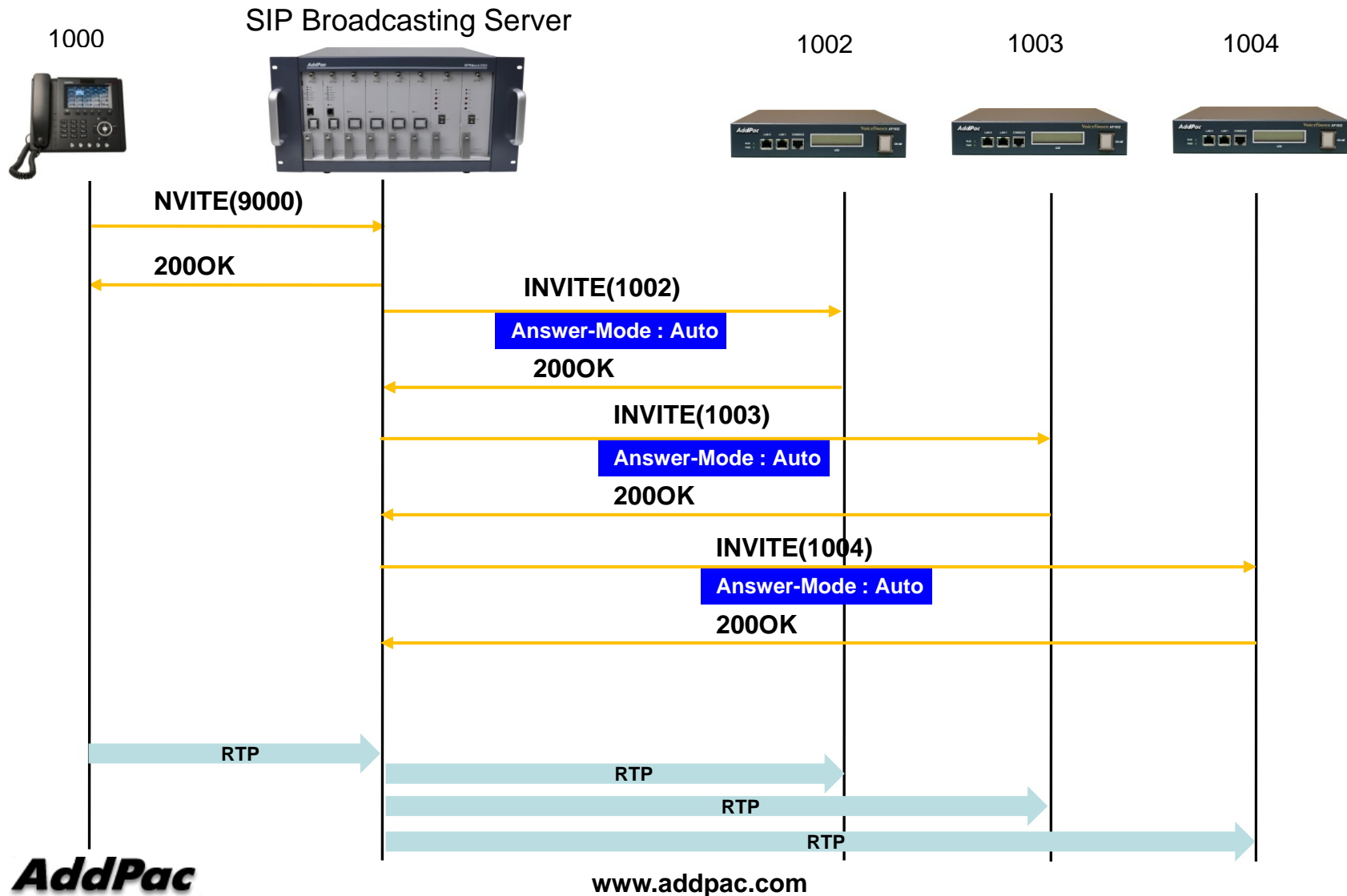
SIP Broadcasting Network Diagram

AP1602 SIP Audio Broadcasting Terminal



Paging Group Signaling Flow

AP1602 SIP Audio Broadcasting Terminal





WSMM Configuration for Paging Group

(WSMM : Web based Smart Multimedia Manager)

Extension – Paging Group

Smart Multimedia Manager
www.addpac.com

Extensions

| Modify | Delete | User Portal | Extension Number | Type | Name | Date Created |
|--------|--------|-------------|------------------|----------------|-----------------|---------------------|
| 1 | | | 1000 | User Extension | Ashley Allen | 2015-07-28 12:39:51 |
| 2 | | | 1001 | User Extension | Mary Moore | 2015-07-28 12:39:55 |
| 3 | | | 1002 | User Extension | Thomas Taylor | 2015-07-28 12:40:00 |
| 4 | | | 1003 | User Extension | Victoria Valdez | 2015-07-28 12:40:04 |
| 5 | | | 1004 | User Extension | Olivia Orth | 2015-07-28 12:40:08 |
| 6 | | | 1005 | User Extension | Linda Lewis | 2015-07-28 12:40:12 |
| 7 | | | 1006 | User Extension | George Gale | 2015-07-28 12:40:17 |
| 8 | | | 1007 | User Extension | Isabel Irwin | 2015-07-28 12:40:21 |
| 9 | | | 1008 | User Extension | William Watson | 2015-07-28 12:40:25 |
| 10 | | | 1009 | User Extension | Sarah Scott | 2015-07-28 12:40:30 |
| 11 | | | 1010 | User Extension | Nicolas Nelson | 2015-07-28 12:40:34 |
| 12 | | | 1011 | User Extension | Emma Evans | 2015-07-28 12:40:38 |
| 13 | | | 1012 | User Extension | Rachel Ross | 2015-07-28 12:40:43 |

Add a Paging Group

Extension * (2~12 digits)

Name *

Audio Codec

Play beep at start

Play Announcement

Extensions

| Extension | Name |
|----------------------|----------------------|
| <input type="text"/> | <input type="text"/> |

Paging Group Members

| Name | Extension | Display Name | Multicast |
|----------------------|----------------------|----------------------|----------------------|
| <input type="text"/> | <input type="text"/> | <input type="text"/> | <input type="text"/> |

Paging Group

Group Members

Getting Started **Clustering Guide** **Partitioning Guide**

Linked in facebook YouTube

IVR Extension
An IVR (Interactive Voice Response) extension has a role of auto attendant for incoming calls from trunks. If incoming calls from trunk are routed to an IVR extension by incoming call rule, the interactive scenario will proceed to transfer the call to a proper user extension.

Push-to-Talk Group
A PTT (Push to Talk) group has members of user extensions who will receive broadcasting announcement with auto answering and also can be a floor (speaker role) by pushing the talk button. This is half-duplex two-way broadcasting.

Paging Group
A paging group has members of user extensions who will receive broadcasting announcement with auto answering by speaker phone. This is half-duplex one-way broadcasting.

Attendant Queue
When a call is inbound from trunk or extensions to this queue member and handled by them. Currently, the queue member

Paging Group
A paging group has members of user extensions who will receive broadcasting announcement with auto answering by speaker phone. This is half-duplex one-way broadcasting.

Paging Group Configuration

Smart Multimedia Manager
www.addpac.com

Add a Paging Group

Extension * 9000 (2~12 digits) Check Extension Extension number is valid.

Name *

Audio Codec G.711U

Play beep at start Beep Sound 3

Play Announcement

Group Members

| Extension | Name |
|-----------|----------------|
| 1000 | Ashley Allen |
| 1001 | Mary Moore |
| 1007 | Isabel Irwin |
| 1008 | William Watson |
| 1009 | Sarah Scott |
| 1010 | Nicolas Nelson |
| 1011 | Emma Evans |
| 1012 | Debra Depp |

Paging Group Members

| Name | Extension | Display Name | Multicast |
|-----------------|-----------|--------------|-----------|
| Thomas Taylor | 1002 | | Off |
| Victoria Valdez | 1003 | | Off |
| Olivia Ortiz | 1004 | | Off |
| Linda Lewis | 1005 | | Off |
| George Gale | 1006 | | Off |

Authorization Members

| Name | Extension | Display Name | Multicast |
|--------------|-----------|--------------|-----------|
| Ashley Allen | 1000 | | Off |
| Mary Moore | 1001 | | Off |

Play Announcement

Play Announcement

Announcement Closing Notification

Repeat Count 1

Retry Count 2

Retry Interval 3 sec

Close on Caller Drop Call

Paging Extension
This is paging extension number to make the paging by dialing digits.

Group Members
These members can receive broadcasting announcement.

Play Announcement
If enabled, group members will hear announcement at broadcasting. The announcement can be selected among announcement files and can be uploaded at Announcements and Tones menu.

Authorization Members
Only these authorized member can start this paging by dialing the paging extension digits.

Ordering Information

- **AP1602 SIP Broadcasting Terminal Hardware**
 - AP1602 Main Body
 - RISC Microprocessor with High-end Programmable DSP Architecture
 - Option : AP-AUDIO2 Module , AP-AUD1S3 Module
 - Including Network Cable Set & Power Supply, etc.
- **Built-in APOS Internetworking Software for AP1602**
- **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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