

AP2620

SIP VoIP Gateway + SIP Audio Broadcasting Terminal

High Performance SIP Audio Broadcasting Terminal



AddPac

AddPac Technology

Sales and Marketing

www.addpac.com

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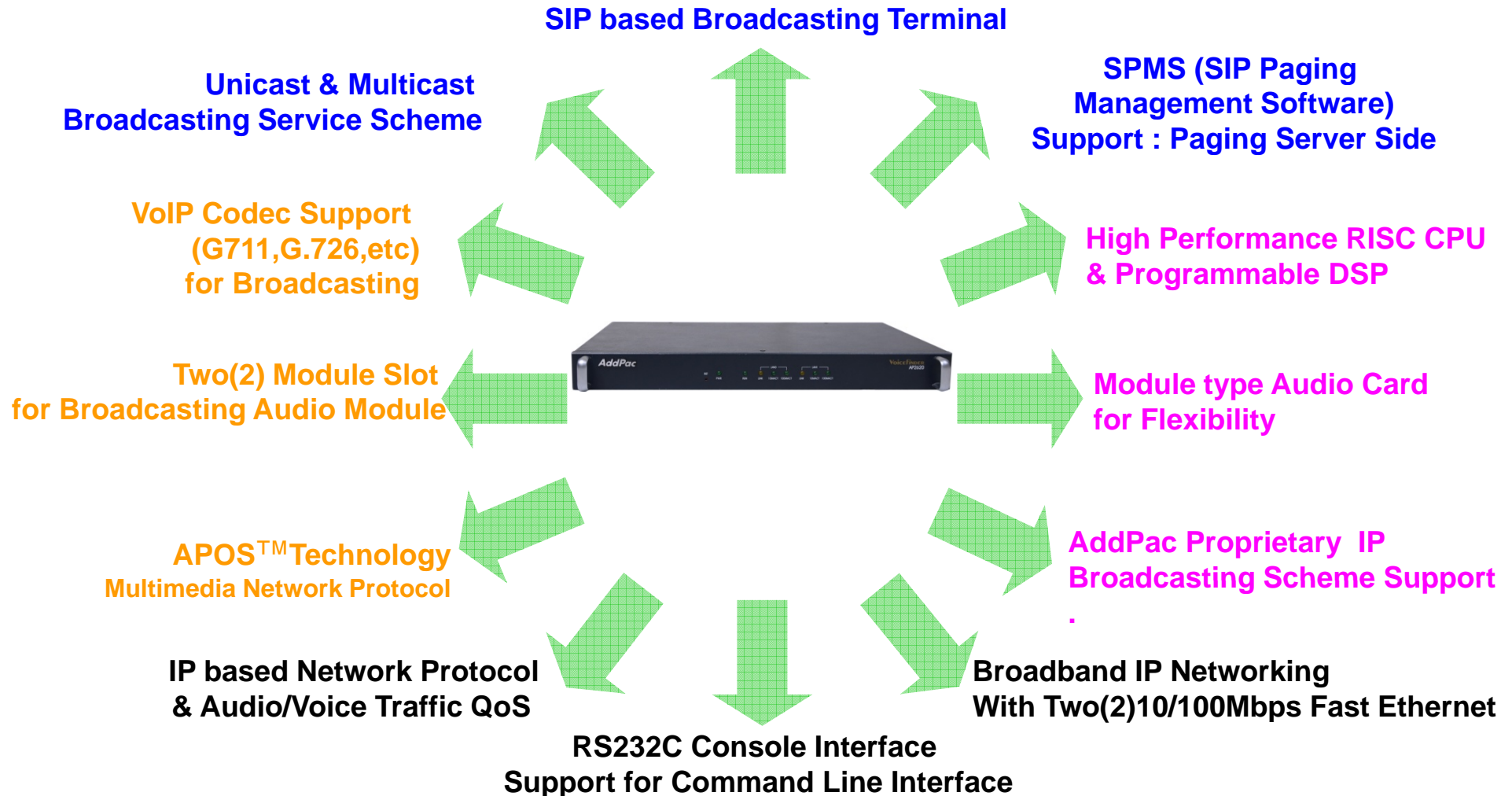
Product Overview

AP2620 SIP Audio Broadcasting Terminal

- High Performance VoIP Gateway Solution
- 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
- Two(20 Module Slot for VoIP Gateway + 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 (Paging Server Side)
- Various Audio Broadcasting Module Support
- Firmware Upgradeable Architecture
- Broadcasting Solution with Outstanding Network Service Capability

Product Highlights

AP2620 SIP Audio Broadcasting Terminal



Hardware Specification

AP2620 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

- RISC Microprocessor Computing Power
- High-end Programmable DSP Hardware Architecture
- Two(2) Module Slot for Audio Broadcasting Codec Module
- VoIP Audio Encoding/Decoding Service
- 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

AP2620 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP2620 Front Side



Two(2) 10/100Mbps
LAN

RS232C Console

AddPac

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Hardware Specification

AP2620 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP2620 Back Side



RS232C Console

Two(2) 10/100Mbps
LAN

Audio Port
Active LED

Audio
Input

Audio
Output

Audio
Input

Audio
Output

Power
Supply

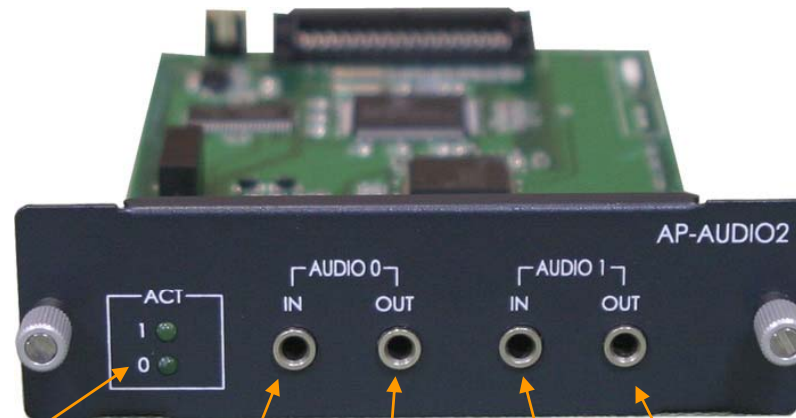
Hardware Specification

AP2620 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

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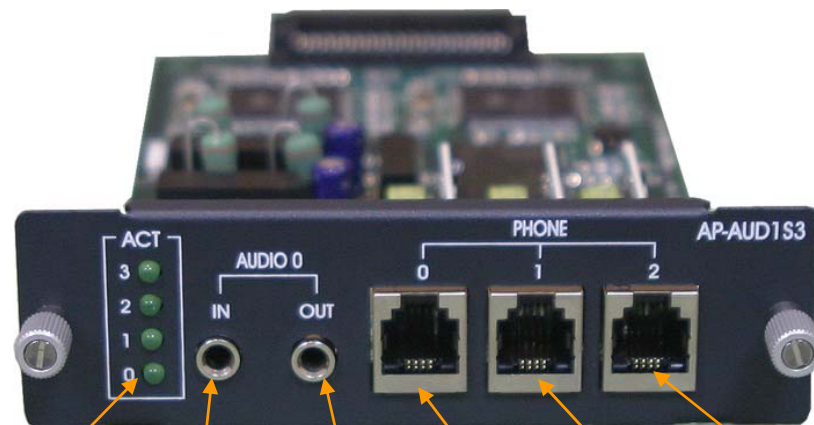
Hardware Specification

AP2620 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

AP2620 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP2620 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

AP2620 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP

AP2620 Audio Module

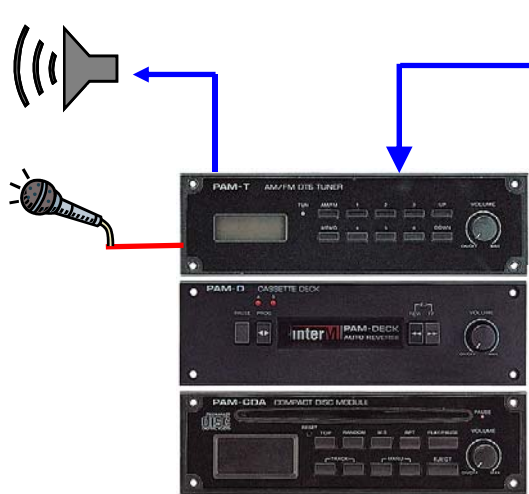
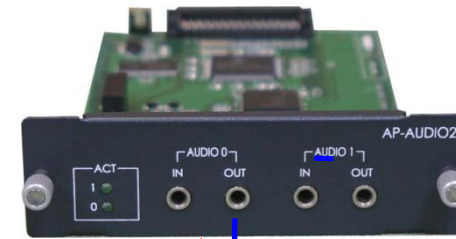
Audio Module Type	Audio Module Features
AP-AUD1S3	One(1)-Channel Audio In/Out Port
	Audio Encoding/Decoding Service
	Audio IN : MIC IN
	Audio OUT :Line OUT 3.5mm Stereo JACK
	Three(3) FXS Port Interface (RJ11 x 3)
	G.711, G.726, G.729A, G.723.1 Audio Codec

AP-AUDIO2 Module

AP2620 SIP Audio Broadcasting Terminal

RISC
CPU

High-end
DSP



3.5mm Line Out JACK

Direct MIC

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

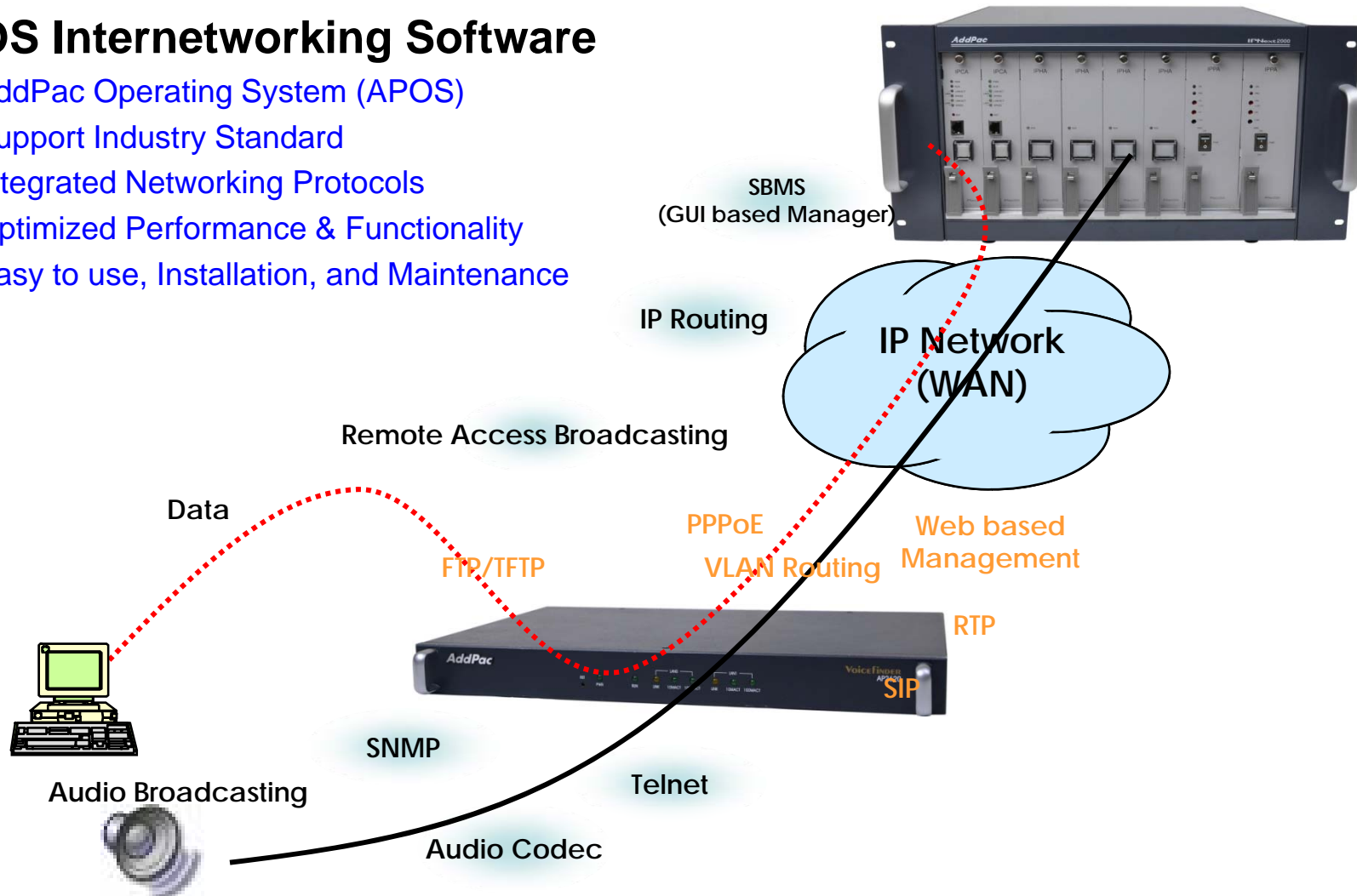
Headphone

APOS™ Service Features

AP2620 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

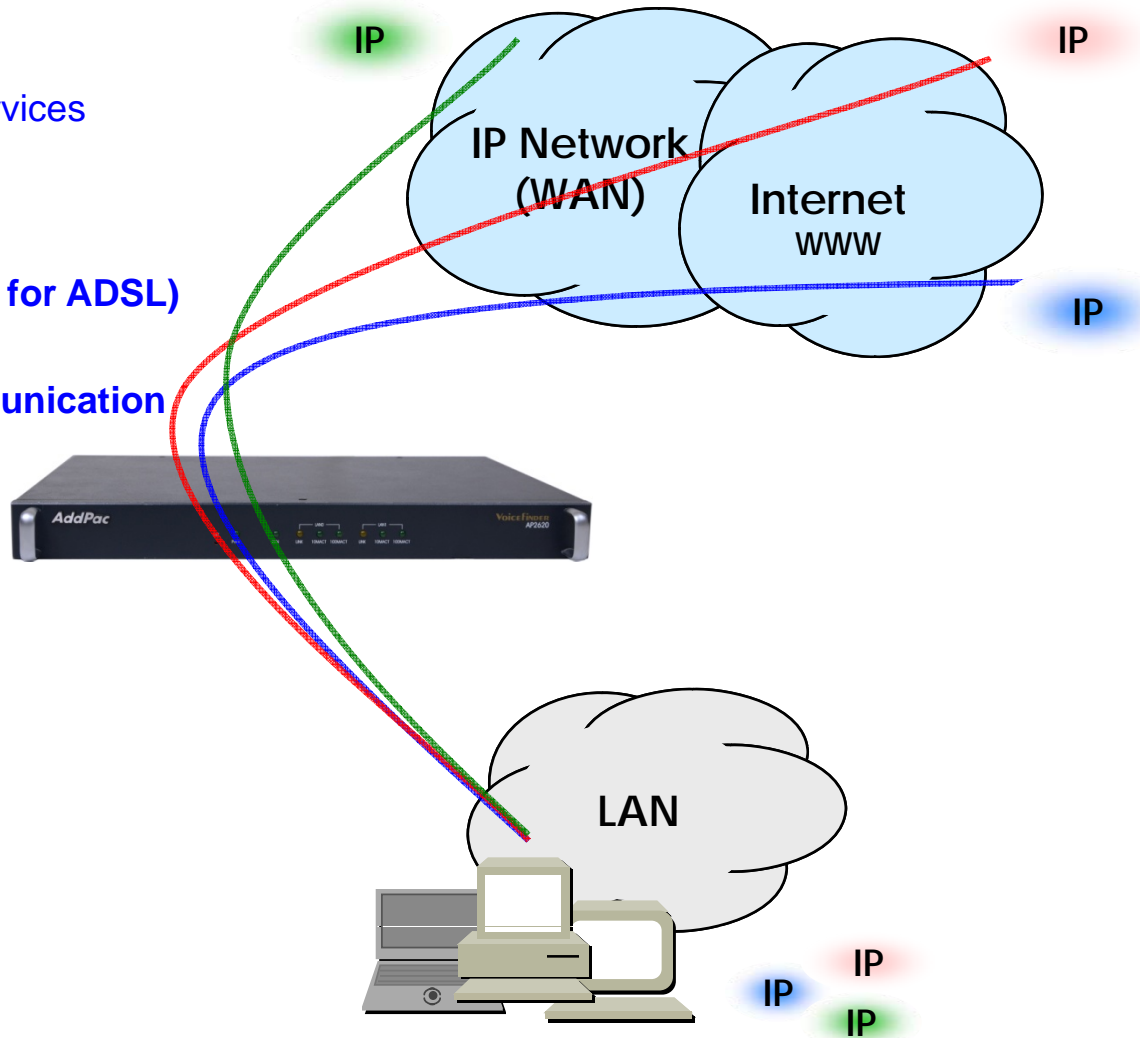
AP2620 SIP Audio Broadcasting Terminal

- **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

AP2620 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

AP2620 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

AP2620 SIP Audio Broadcasting Terminal

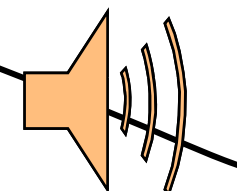
- SIP Protocol Service
 - Gateway-based / Endpoint-based registration support
 - Secondary proxy-server assignment function
 - SIP signaling port change function
 - SIP proxy server assignment according to the domain name
 - T.38 real-time fax relay support
 - DTMF relay support with RFC2833 / OPTION message
 - Re-INVITE support

VoIP Service Features

AP2620 SIP Audio Broadcasting Terminal

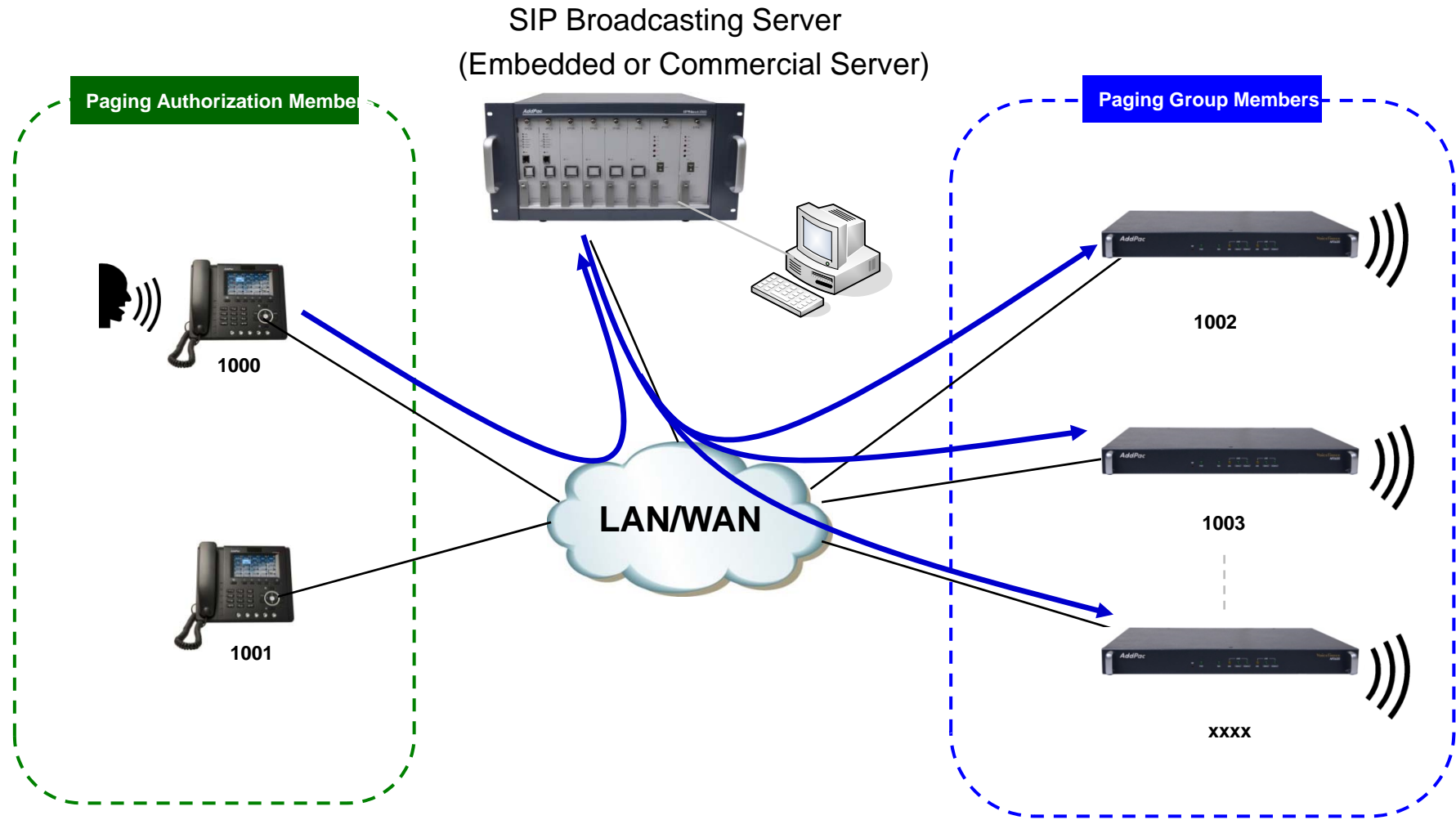
- Voice Codec
 - G.711 A-Law, G.711 U-Law
 - G.726 r16, G.726 r32
 - G.729A
 - G.723.1 r63, G.723.1 r53
 - VAD (Voice Activity Detection) function support
 - DTMF relay support (H.323, SIP, MGCP common) based on RFC2833
- RTP Protocols
 - Redundant RTP packet transmission in case of severe packet loss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

IP Broadcasting



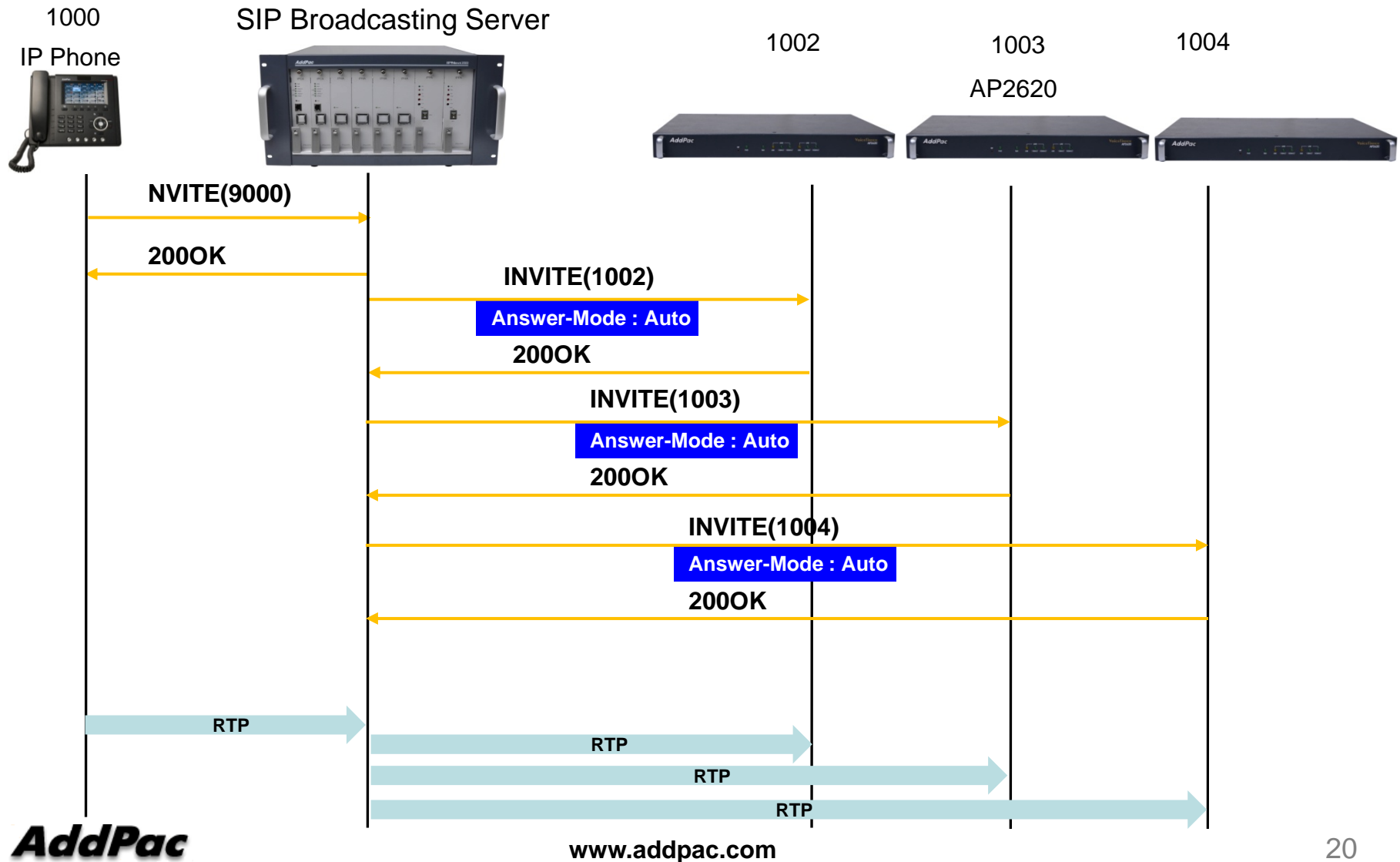
SIP Broadcasting Network Diagram

AP2620 SIP Audio Broadcasting Terminal



Paging Group Signaling Flow

AP2620 SIP Audio Broadcasting Terminal





WSMM Configuration for Paging Group

(WSMM : Web based Smart Multimedia Manager)

Extension – Paging Group

Smart Multimedia Manager
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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 Ortiz	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

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

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

Group Members
These members can receive broadcasting announcement.

Authorization Members

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

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

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

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.

Description
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. Only authorized members can make a paging by dialing the paging extension.

Play Announcement Configuration

- Play Announcement
- Announcement: Closing Notification
- Repeat Count: 1
- Retry Count: 2
- Retry Interval: 3 sec
- Close on Caller Drop Call:

Ordering Information

- **AP2620 SIP Broadcasting Terminal Hardware**
 - AP2620 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 AP2620**
- **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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