AP1602

SIP Audio Broadcasting Terminal

High Performance SIP Audio Broadcasting Terminal

AddPac Technology

2015, Sales and Marketing

www.addpac.com

Contents

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

SIP based Broadcasting Terminal

Unicast & Multicast Broadcasting Service Scheme

SPMS (SIP Paging Management Software) Support

VoIP Codec Support (G711,G.726,etc) for Broadcasting

High Performance RISC CPU & Programmable DSP

One(1) Audio Module Slot for Broadcasting VoIP Module

Module type Audio Card for Flexibility

APOS™Technology
Multimedia Network Protocol

AddPac Proprietary IP Broadcasting Scheme Support
.

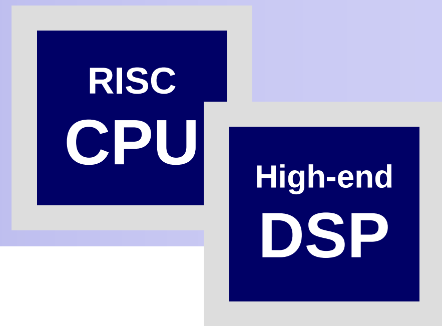
IP based Network Protocol & Audio/Voice Traffic QoS

Broadband IP Networking With Two(2)10/100Mbps Fast Ethernet

RS232C Console Interface Support for Command Line Interface

Hardware Specification

AP1602 SIP Audio Broadcasting Terminal



- 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

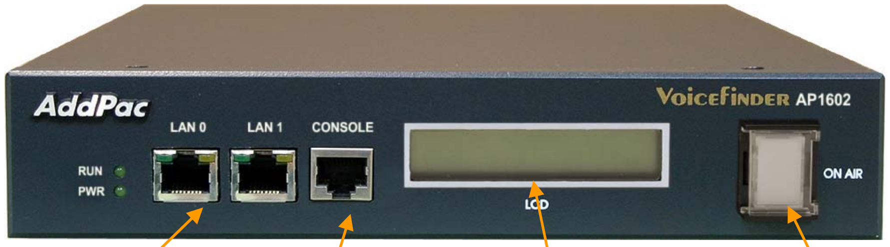
Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC CPU

High-end DSP

AP1602 Front Side



Two(2) 10/100Mbps LAN

RS232C Console

Status LCD

On-Air Blue LAMP & Switch

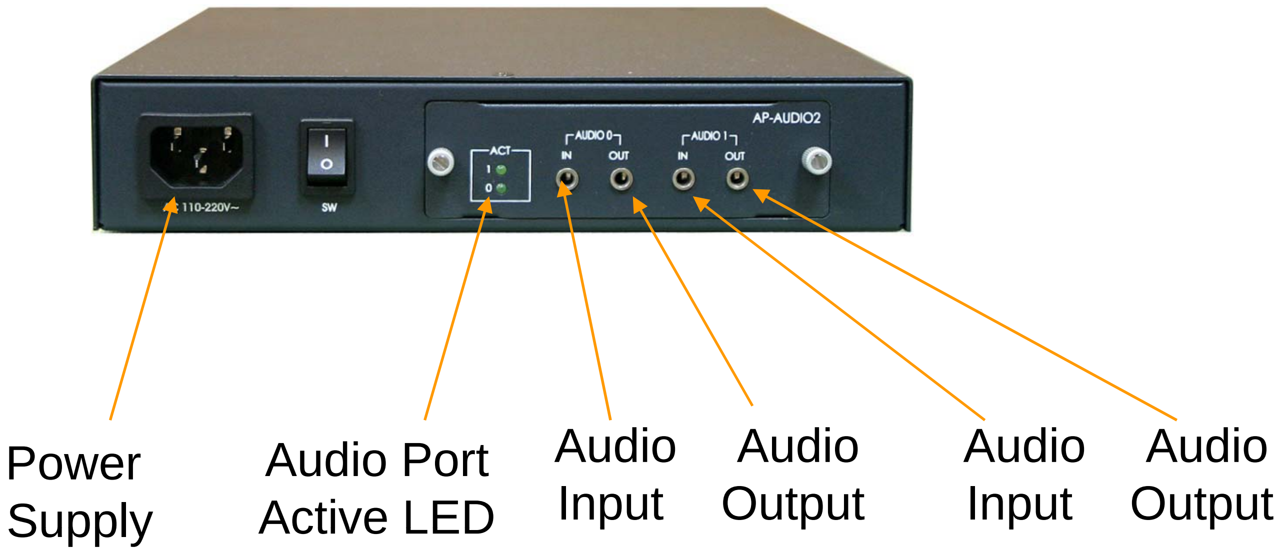
Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC CPU

High-end DSP

AP1602 Back Side



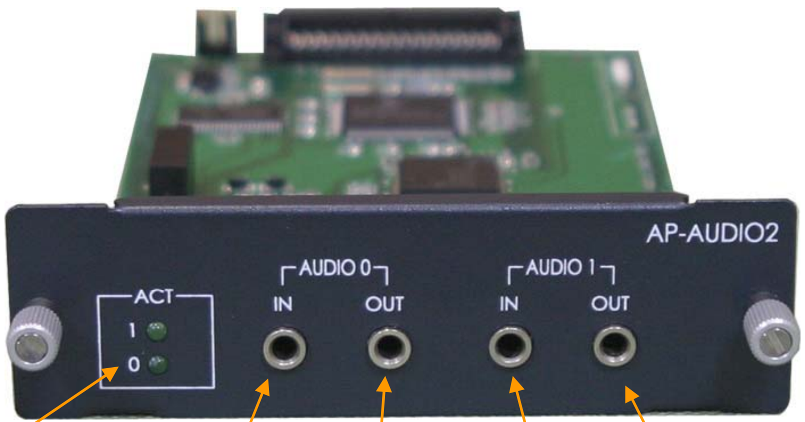
Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC CPU

High-end DSP

AP-AUDIO2 Board



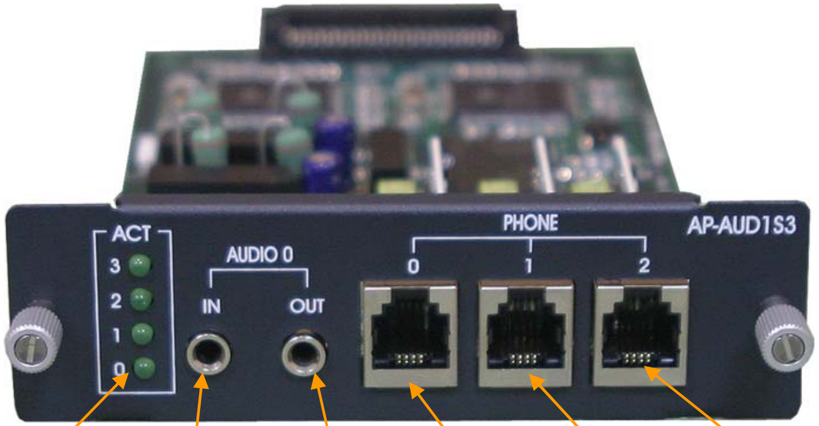
Audio Port Active LED

Audio 0 Channel		Audio 1 Channel	
Audio Input	Audio Output	Audio Input	Audio Output

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

RISC CPU

High-end DSP

AP1602 Audio Module

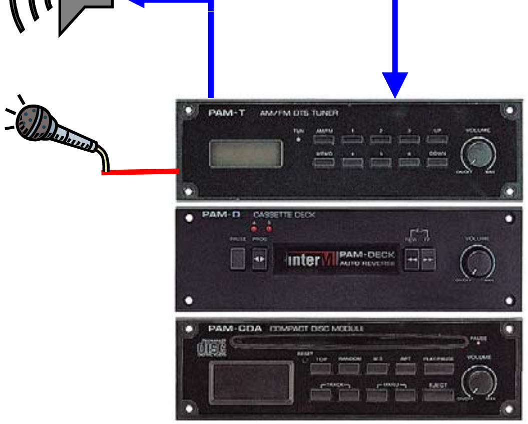
Audio Module Type	Audio Module Features
AP-AUD1S3	One(1)-Channel Audio In/Out Port
	Audio Encoding/Decoding Service
	Audio IN : MIC IN Audio OUT :Line OUT 3.5mm Stereo JACK
	Three(3) FXS Port Interface (RJ11 x 3)
	G.711, G.726, G.729A, G.723.1 Audio Codec

AP-AUDIO2 Module

AP1602 SIP Audio Broadcasting Terminal

RISC CPU

High-end DSP



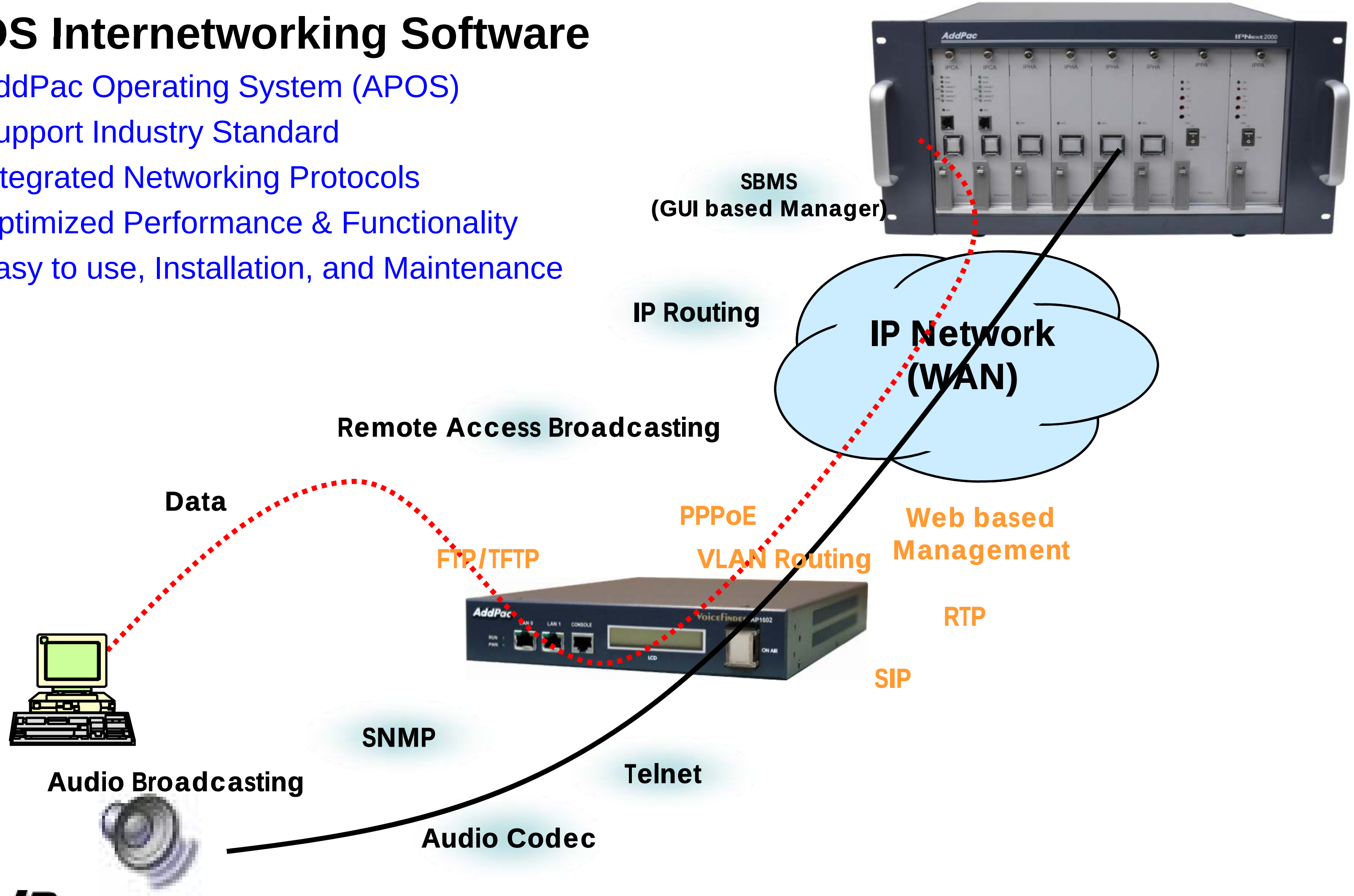
•G.7xx Voice codec realizes IP voice broadcasting service
•Real time VoIP Broadcasting Service using RTP (Real-time Transport Protoco) Protocol

Headphone

APOS™ Service Features

AP1602 SIP Audio Broadcasting Terminal

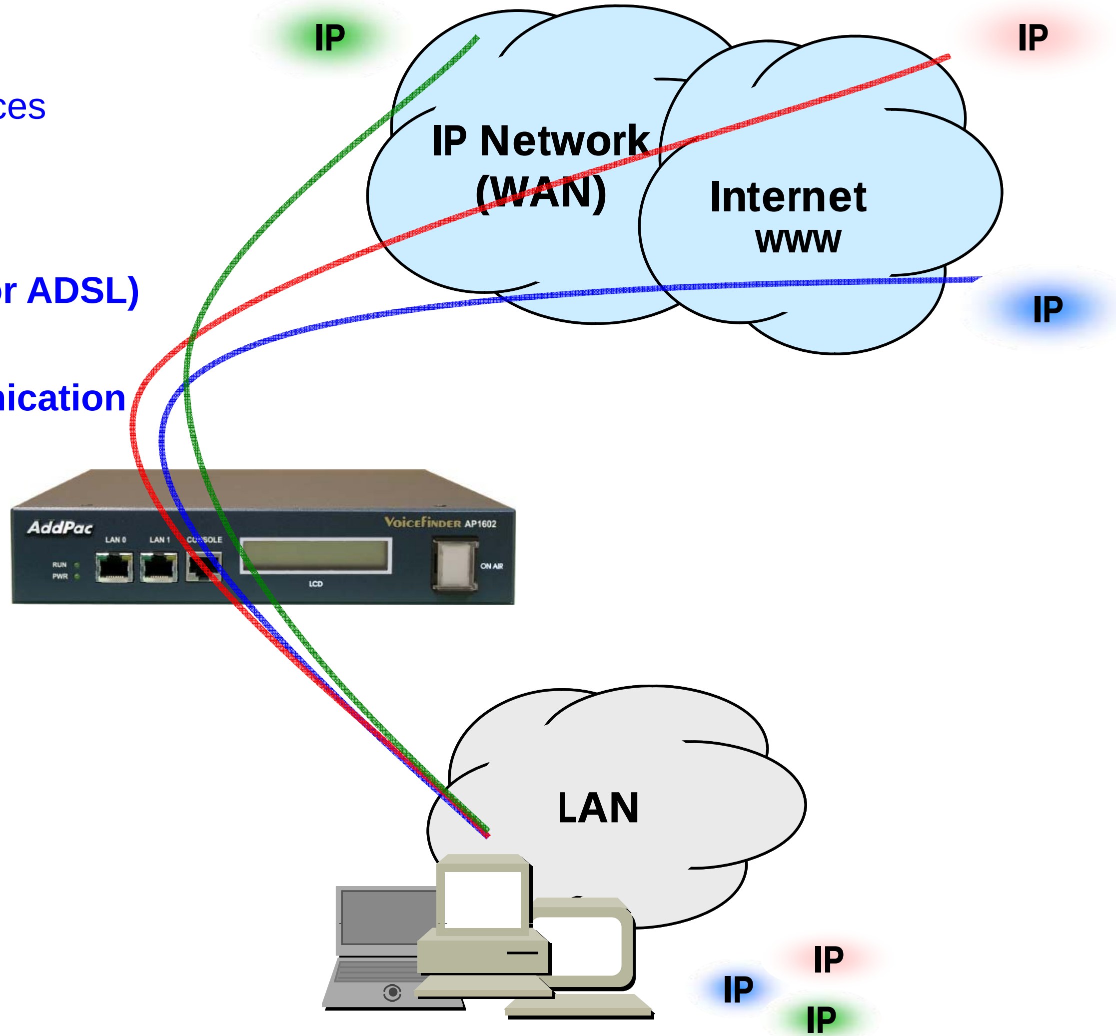
- **APOS Internetworking Software**
  - AddPac Operating System (APOS)
  - Support Industry Standard
  - Integrated Networking Protocols
  - Optimized Performance & Functionality
  - Easy to use, Installation, and Maintenance



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^{TM} 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

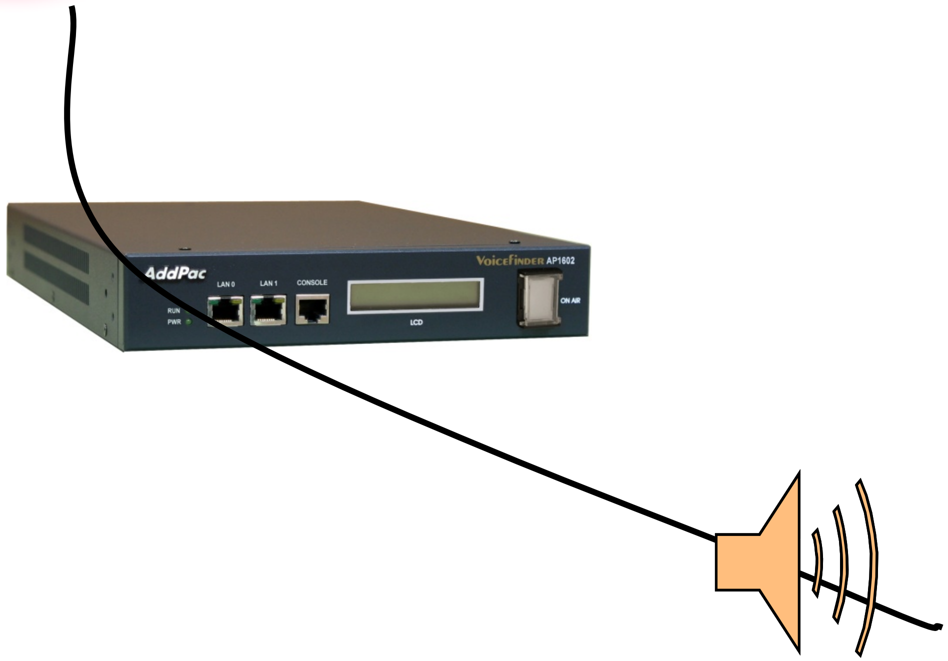
- SIP Protocol Service
  - Gateway-based / Endpoint-based registration support
  - Secondary proxy-server assignment function
  - SIP signaling port change function
  - SIP proxy server assignment according to the domain name
  - T.38 real-time fax relay support
  - DTMF relay support with RFC2833 /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

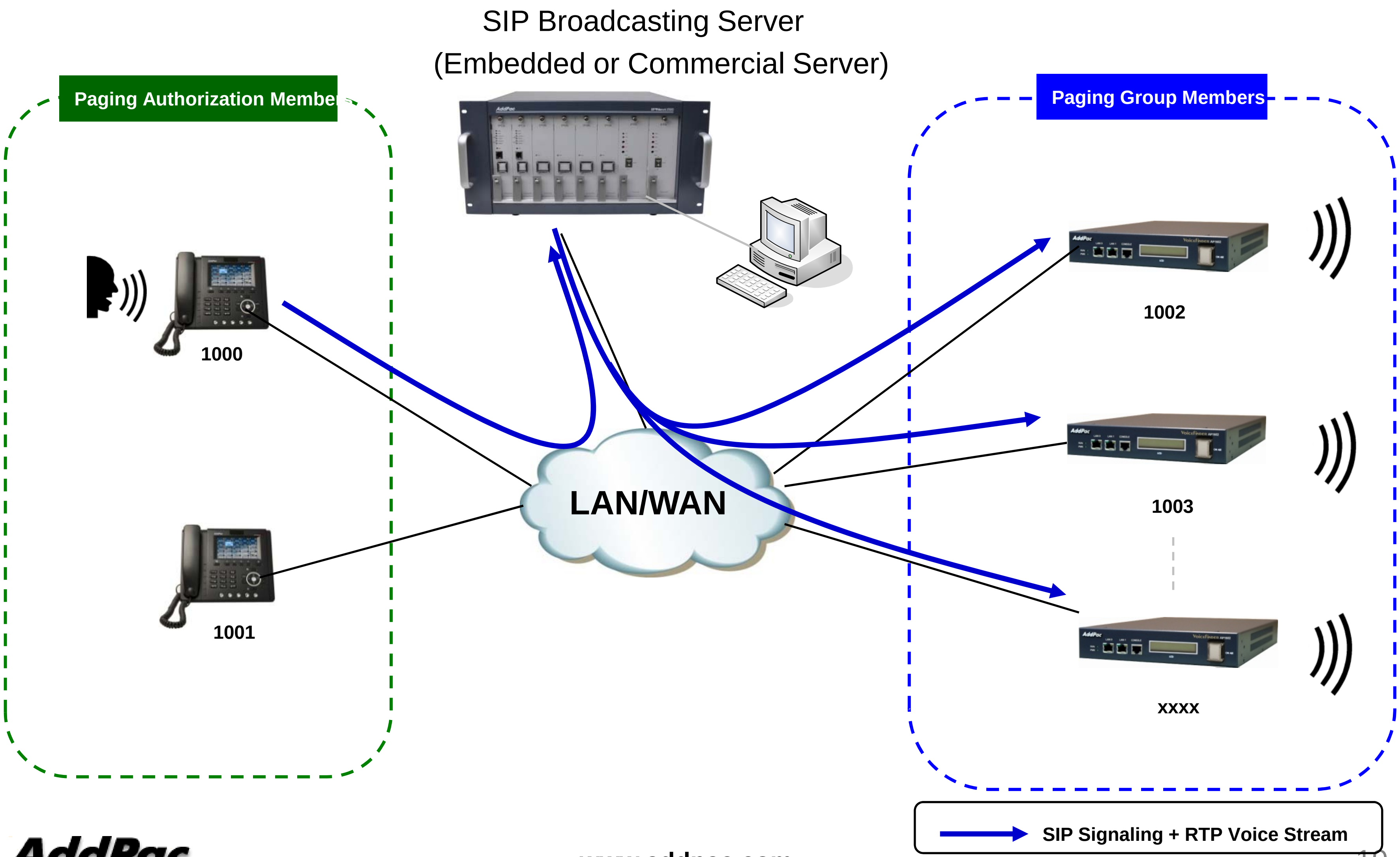
IP Broadcasting



- 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

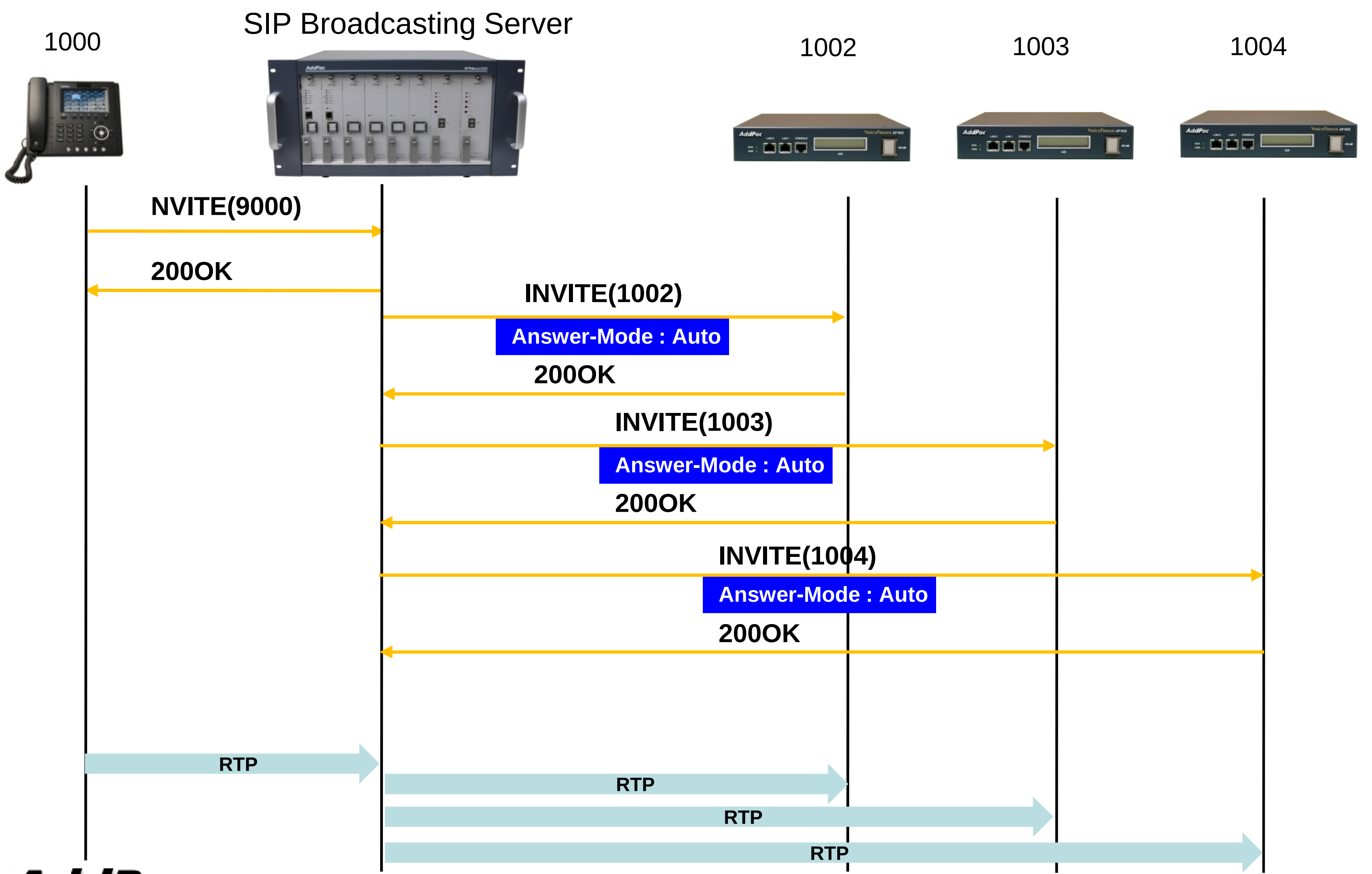
SIP Broadcasting Network Diagram

AP1602 SIP Audio Broadcasting Terminal



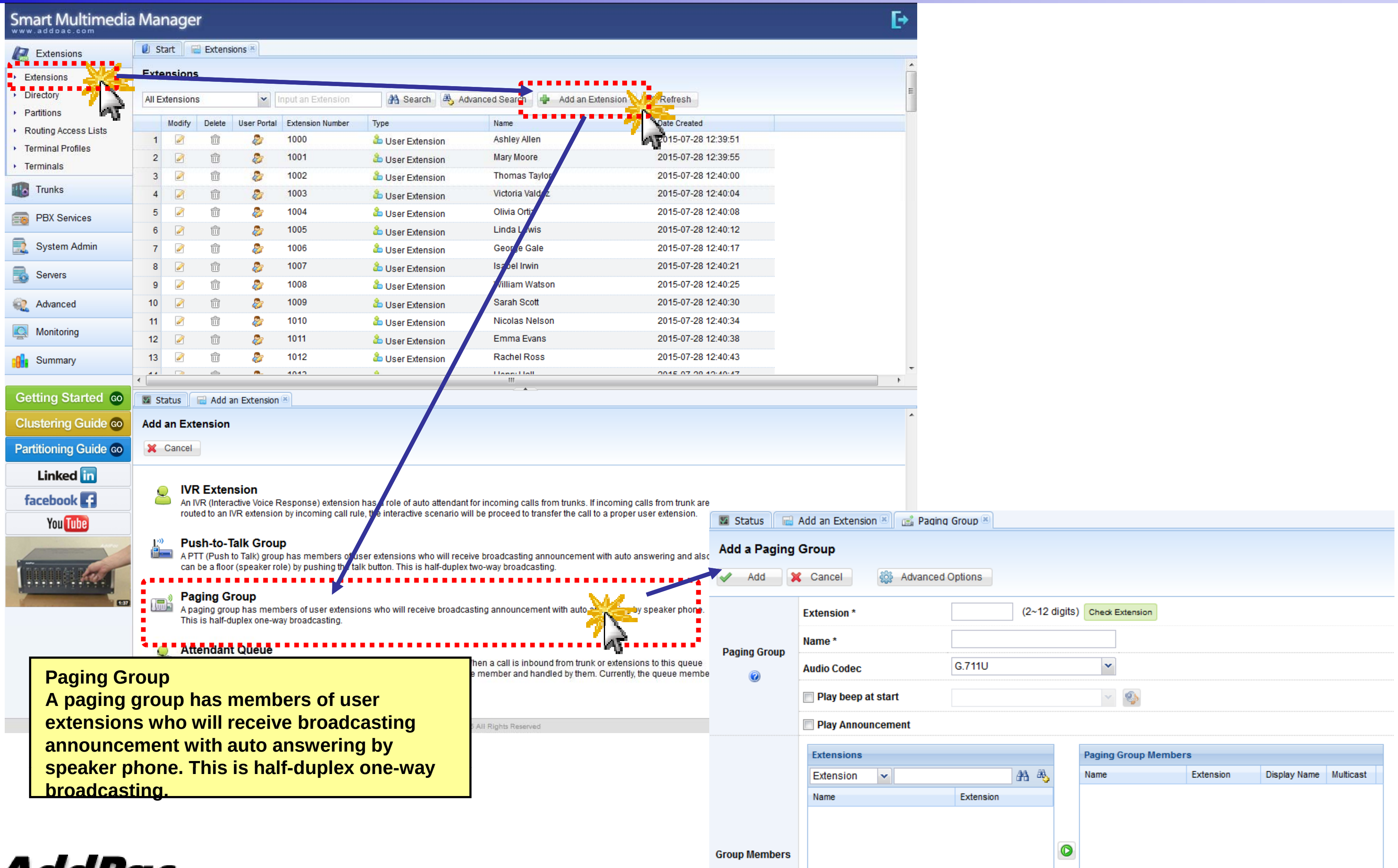
Paging Group Signaling Flow

AP1602 SIP Audio Broadcasting Terminal



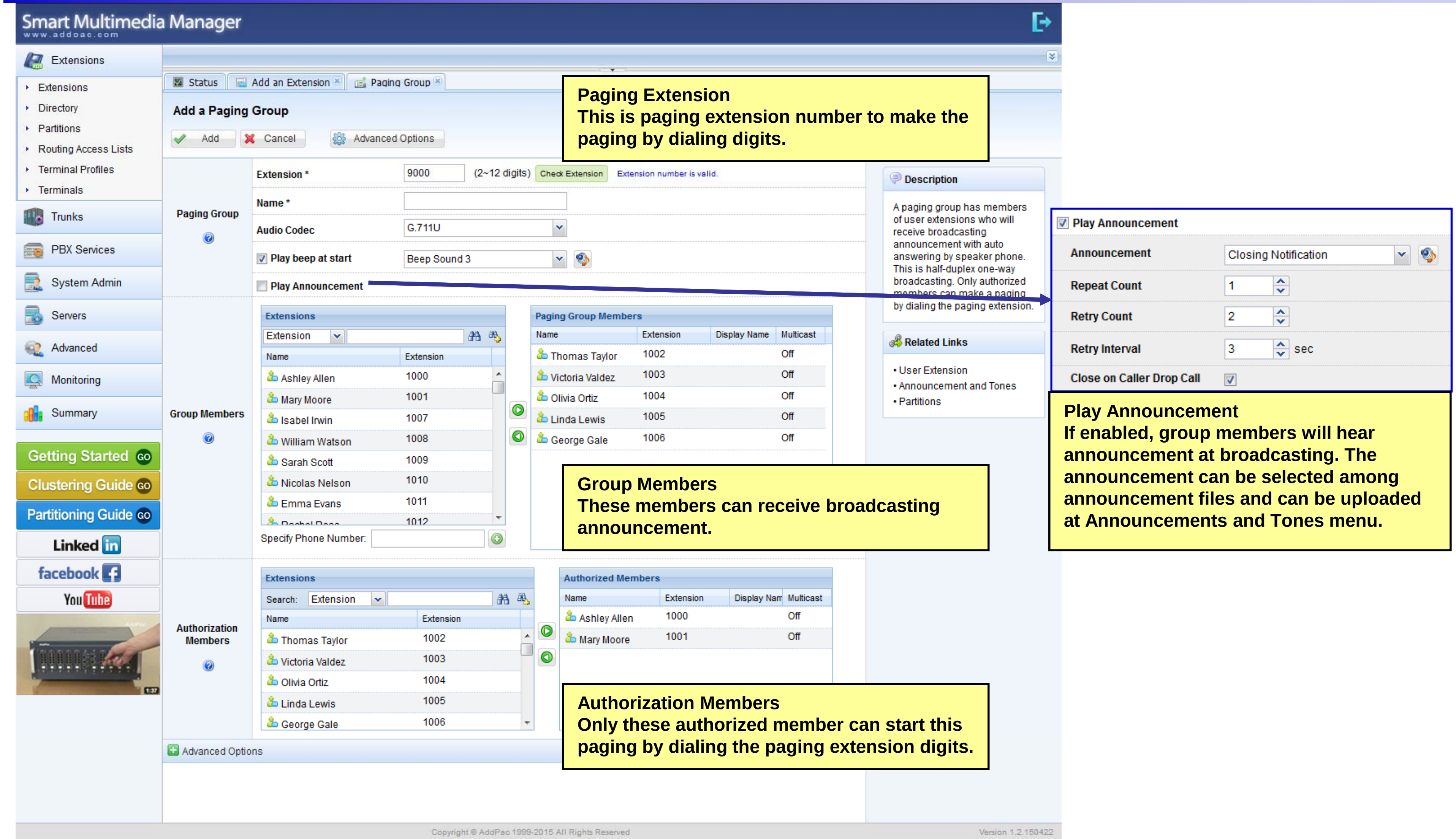
WSMM Configuration for Paging Group
(WSMM : Web based Smart Multimedia Manager)

Extension – Paging Group



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



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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Sales and Marketing

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