

# SIP Voice Paging Solution



**SIP based Paging Solution**

- AP1602 SIP Audio Broadcasting Terminal
- AP1605 SIP Audio Broadcasting Terminal
- AP2620 VoIP Gateway
- AP-SPS2000 SIP Paging Server
- AP-SPS210000 SIP Paging Server
- AP-IP300 IP Phone
- AP-IP20 IP Presence Terminal
- AP-SPMS SIP Paging Management S/W

[Learn More >](#)

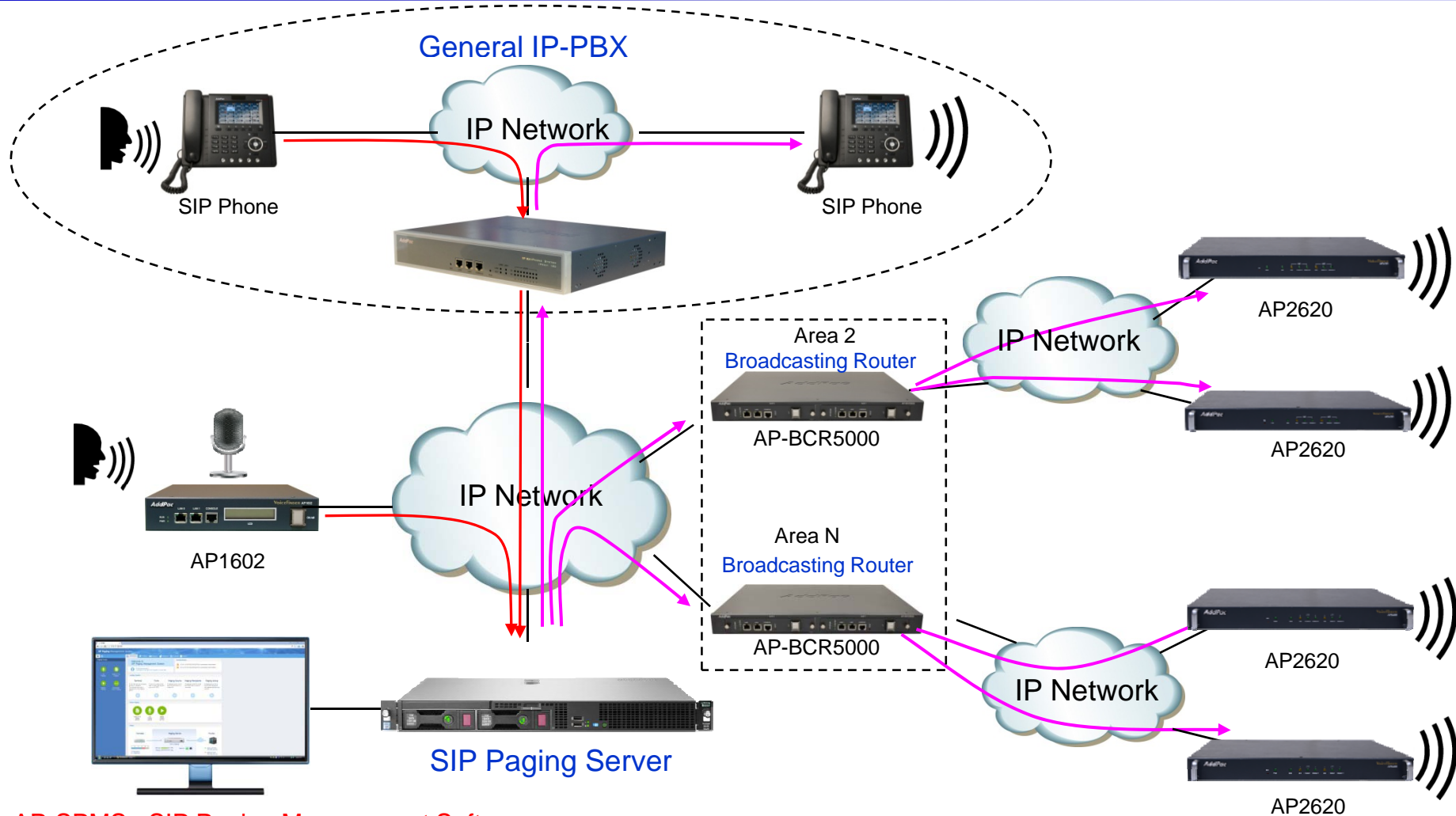
The advertisement features a collection of AddPac hardware components including a JBL speaker, a rack-mounted server, a VoIP gateway, and several IP phones. A computer monitor displays a web-based management interface for the paging system.

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- SIP Paging Call Service Diagram
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  - SIP Phones for Paging Service : AP-IP300, AP-IP120
  - Broadcasting Router : AP-BCR5000
  - SIP Paging Management Software : AP-SPMS

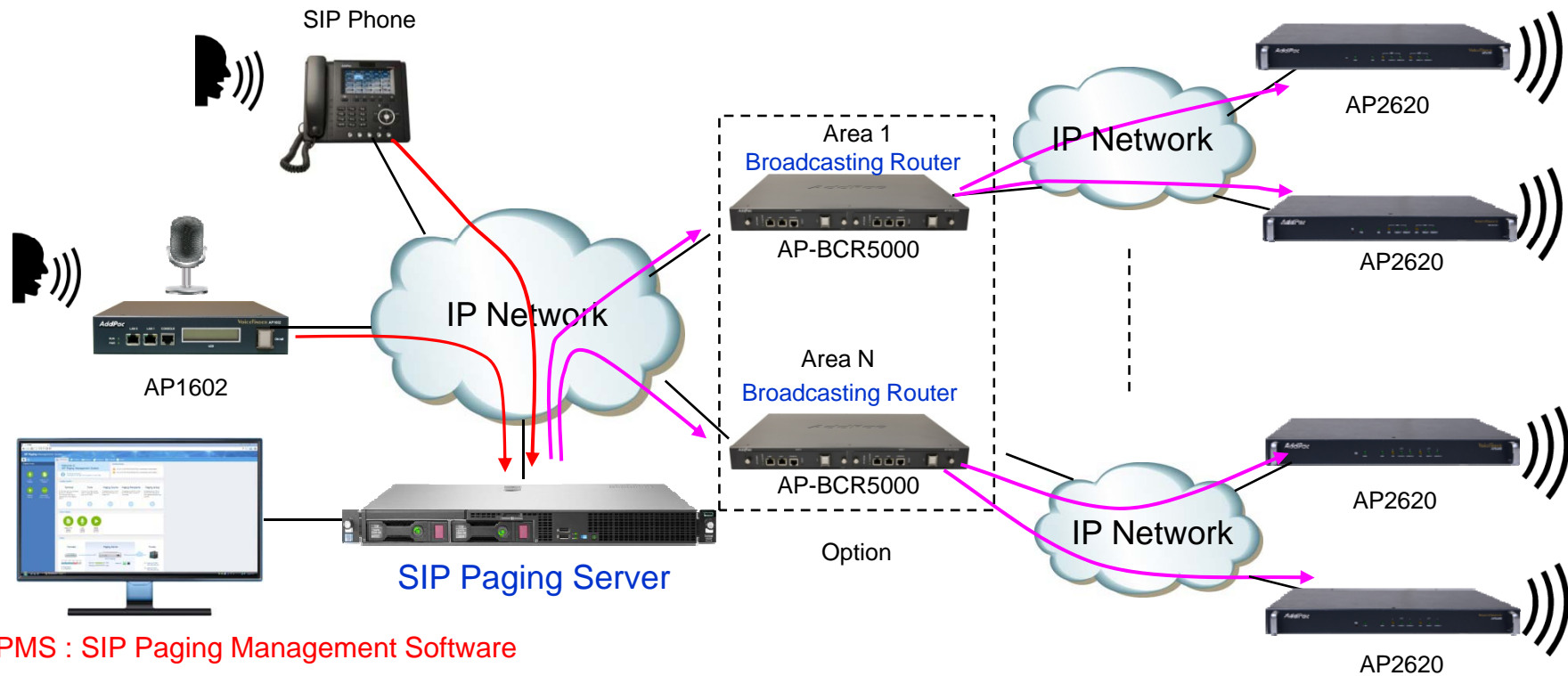
# SIP Paging Call Service Network Diagram

## - IP-PBX Clone Service Mode






AP-SPMS : SIP Paging Management Software

# SIP Paging Call Service Network Diagram - Standalone Paging Service Mode



# SIP Paging Product Solution

SIP Paging Management S/W	SIP Paging Server	SIP Paging Terminal	Broadcasting Router (Relay Server)	SIP Phone for Paging Service
 <p>AP-SPMS</p>	 <p>AP-SPS10000</p>	 <p>AP2620</p>	 <p>AP-BCR5000</p>	 <p>AP-IP300 + AP-PT20x2</p>
	 <p>AP-SPS2000</p>	 <p>AP1605</p>		 <p>AP-IP120 + AP-PT20x2</p>
	 <p>AP-SPS600</p>	 <p>AP1602</p>		

# SIP Paging Call Sources

- Real-Time Paging
  - IP Phones or Terminals Registered to Paging Server
  - IP Phones Registered to IP PBX
  - PSTN Phones via VoIP Gateway
  - Mobile Phones via VoIP Gateway or Mobile Gateway
- Recorded Paging
  - Record from IP Phone, PSTN Phone, Mobile Phone
  - Record from TTS
  - Any Wave Audio File
- HTTP API Paging
  - 3<sup>rd</sup> Party System via Restful HTTP API
- TTS Paging

# SIP Paging Call Destinations

- IP Phones or Terminals Registered to Paging Server
- IP Phones Registered to IP PBX
- PSTN Phones via VoIP Gateway
- Mobile Phones via VoIP Gateway or Mobile Gateway
- SMS via Mobile Gateway
- General PC based SIP Phone



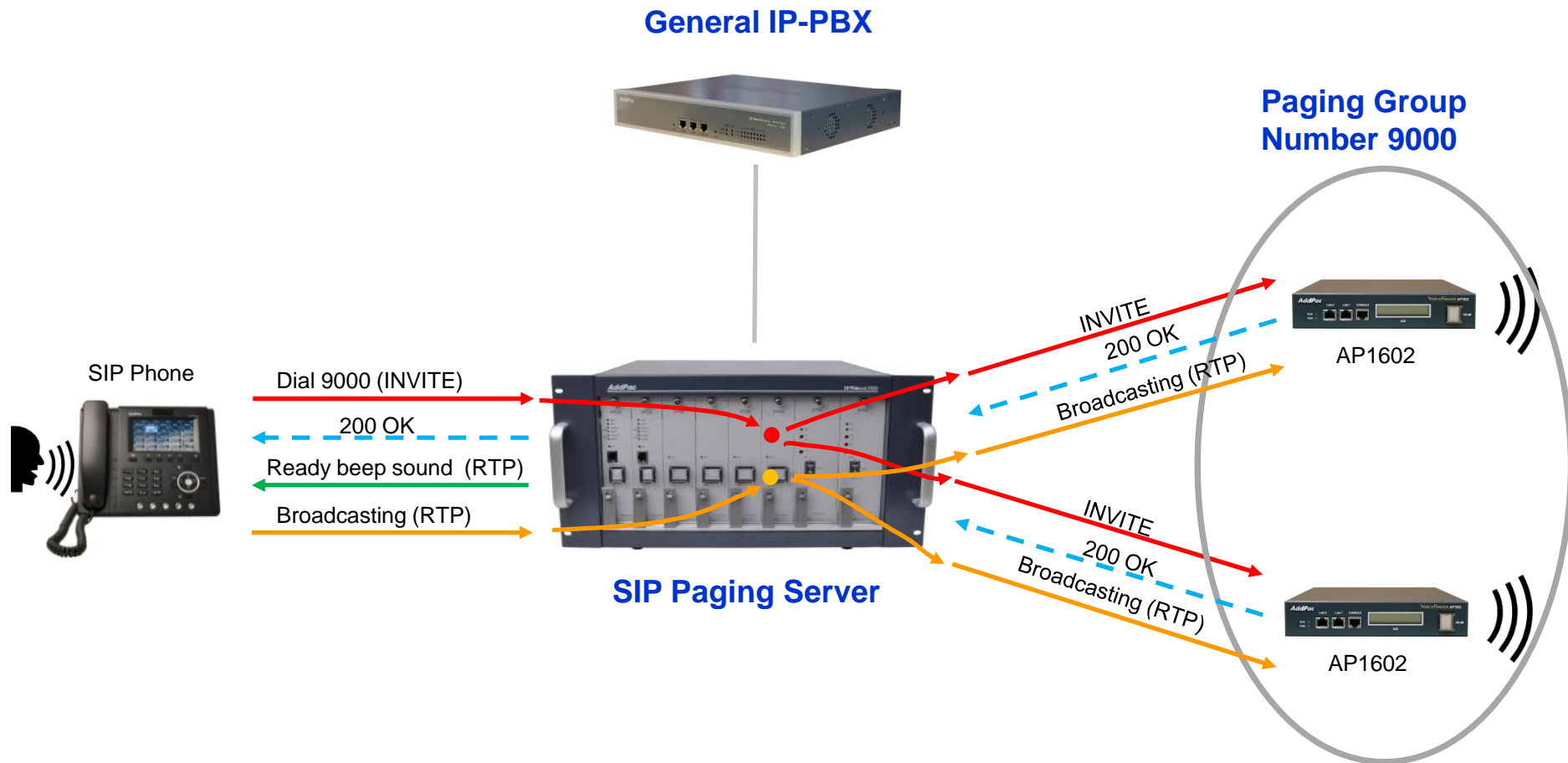
# SIP Paging Call Scenario



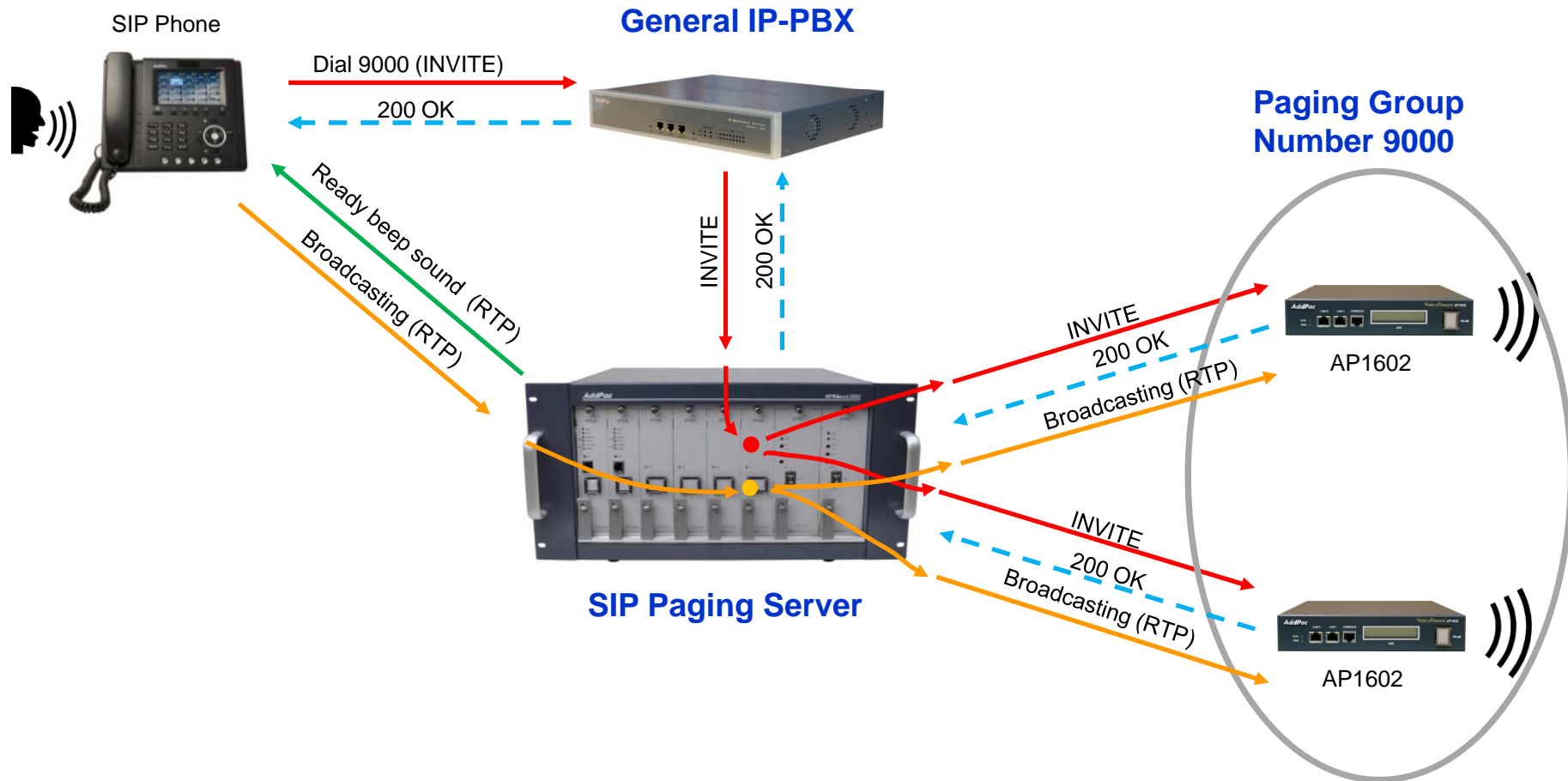
# Paging Call Scenarios

- SIP Phone Initiated Real-Time Paging
- SIP Phone Initiated Real-Time Paging via PBX
- SIP Phone Initiated Real-Time Paging to PBX
- Paging to and from PSTN/Mobile Network
- Web Controller Initiated Real-Time Paging
- Add-hoc Paging with IVR
- Recorded Paging with Simultaneous Casting
- Recorded Paging with Individual Casting
- TTS (Text to Speech) Paging
- Audio and SMS Paging by HTTP API
- Scheduled Paging

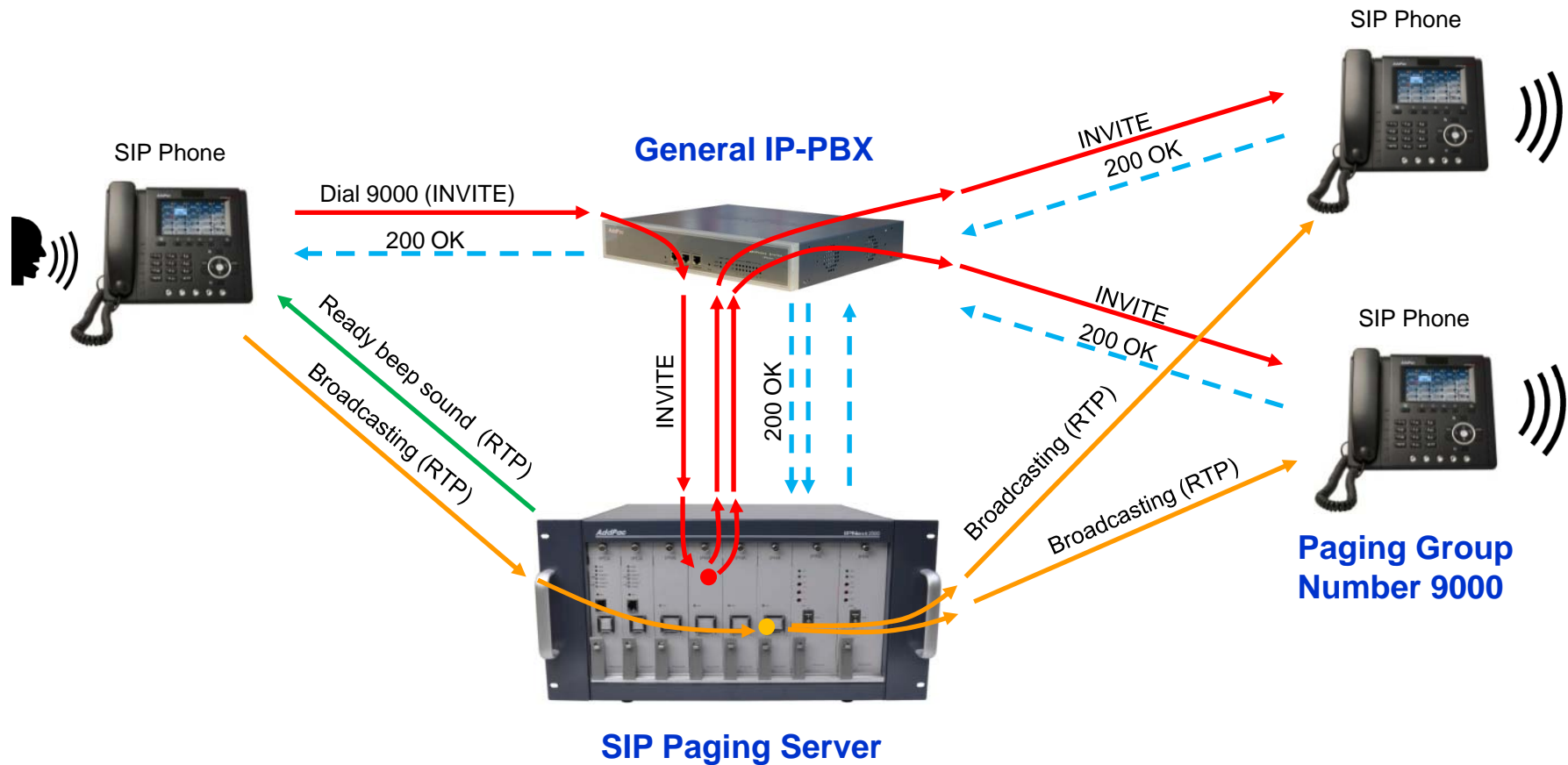
# SIP Phone Initiated Real-Time Paging



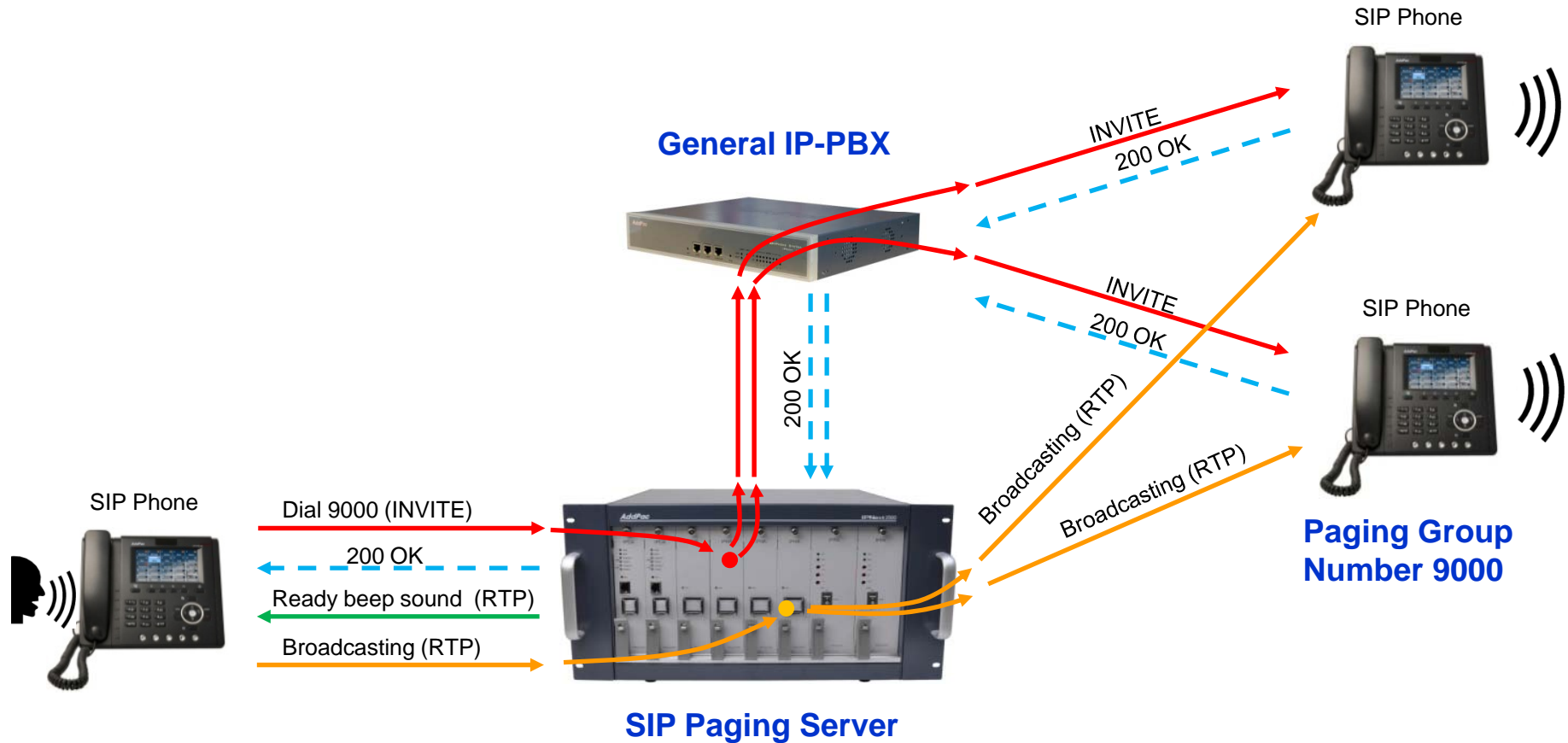
# SIP Phone Initiated Real-Time Paging via PBX



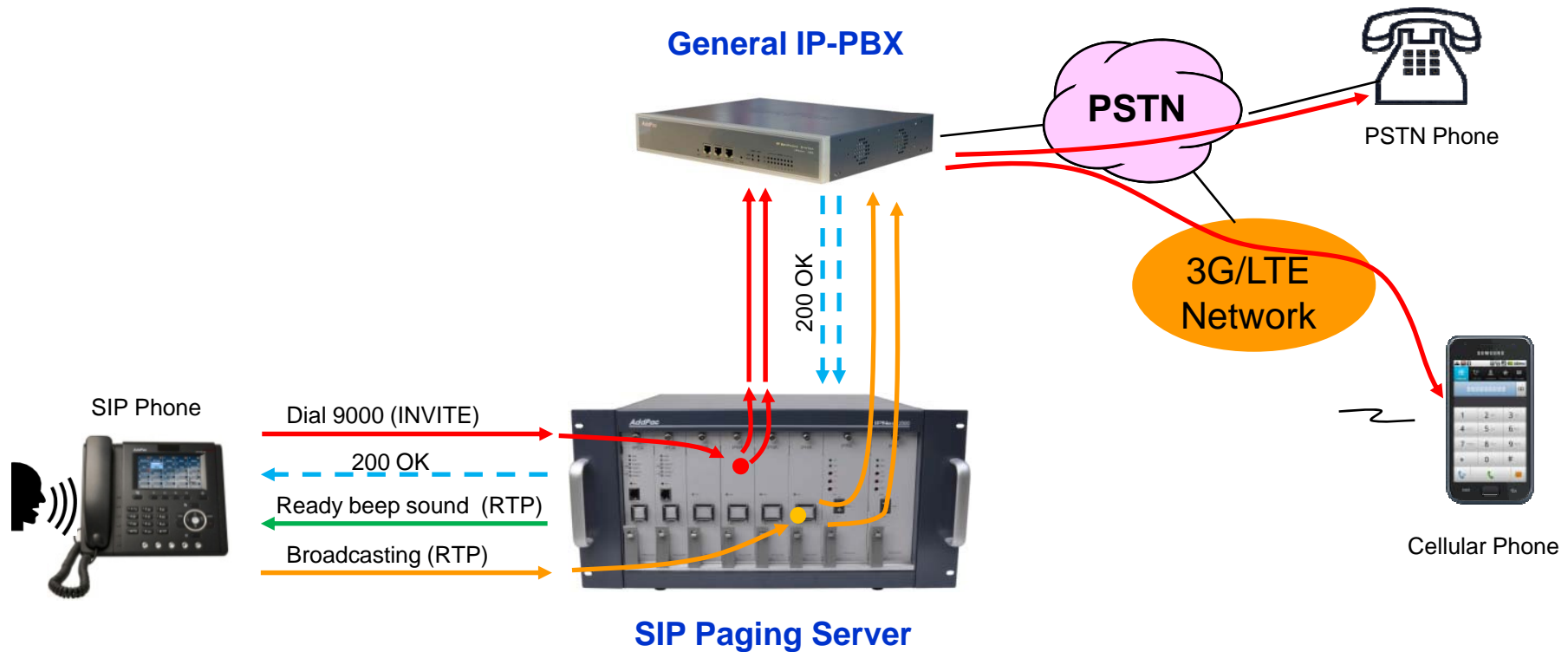
# SIP Phone Initiated Real-Time Paging to PBX



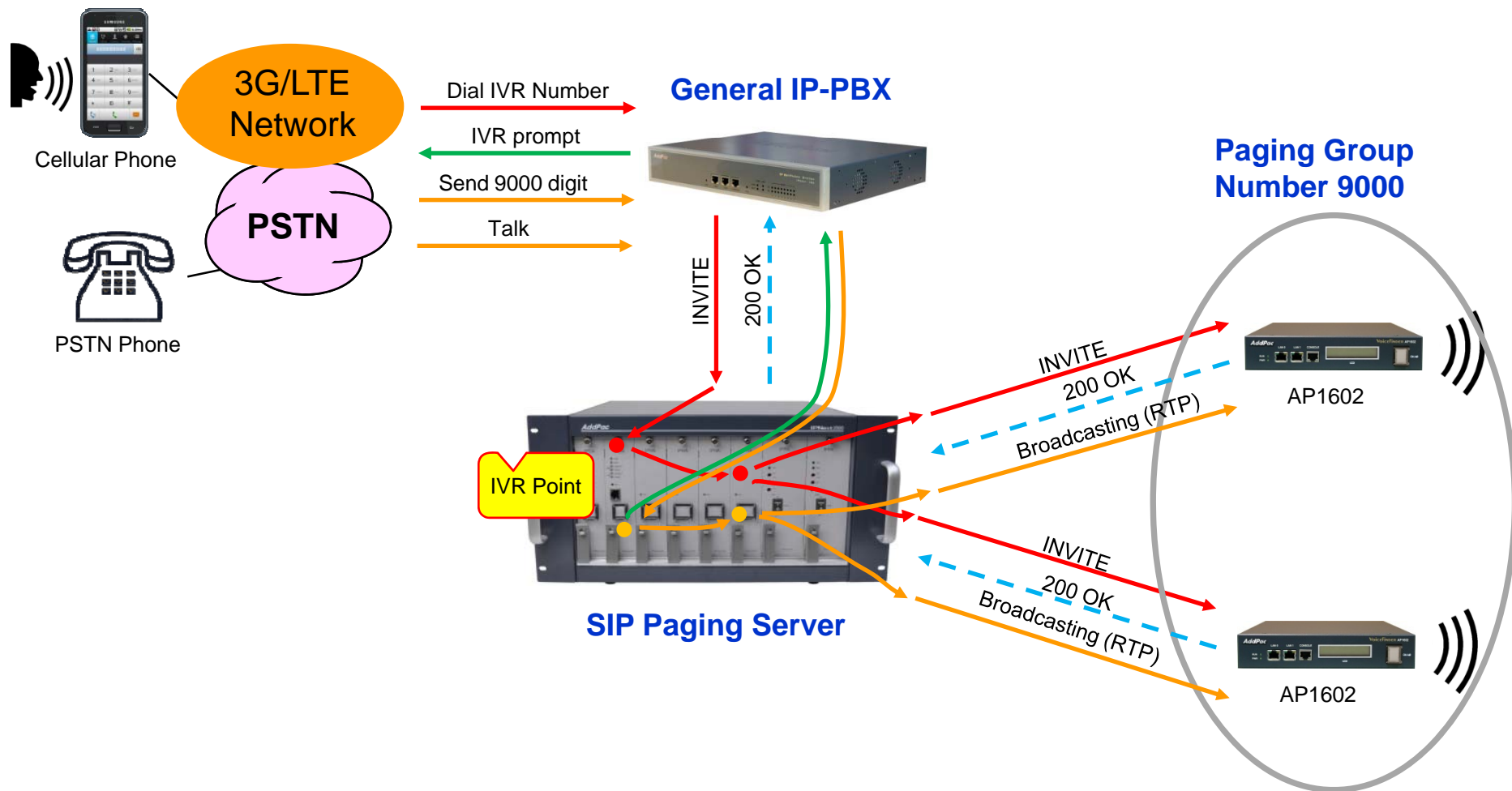
# SIP Phone Initiated Real-Time Paging to PBX



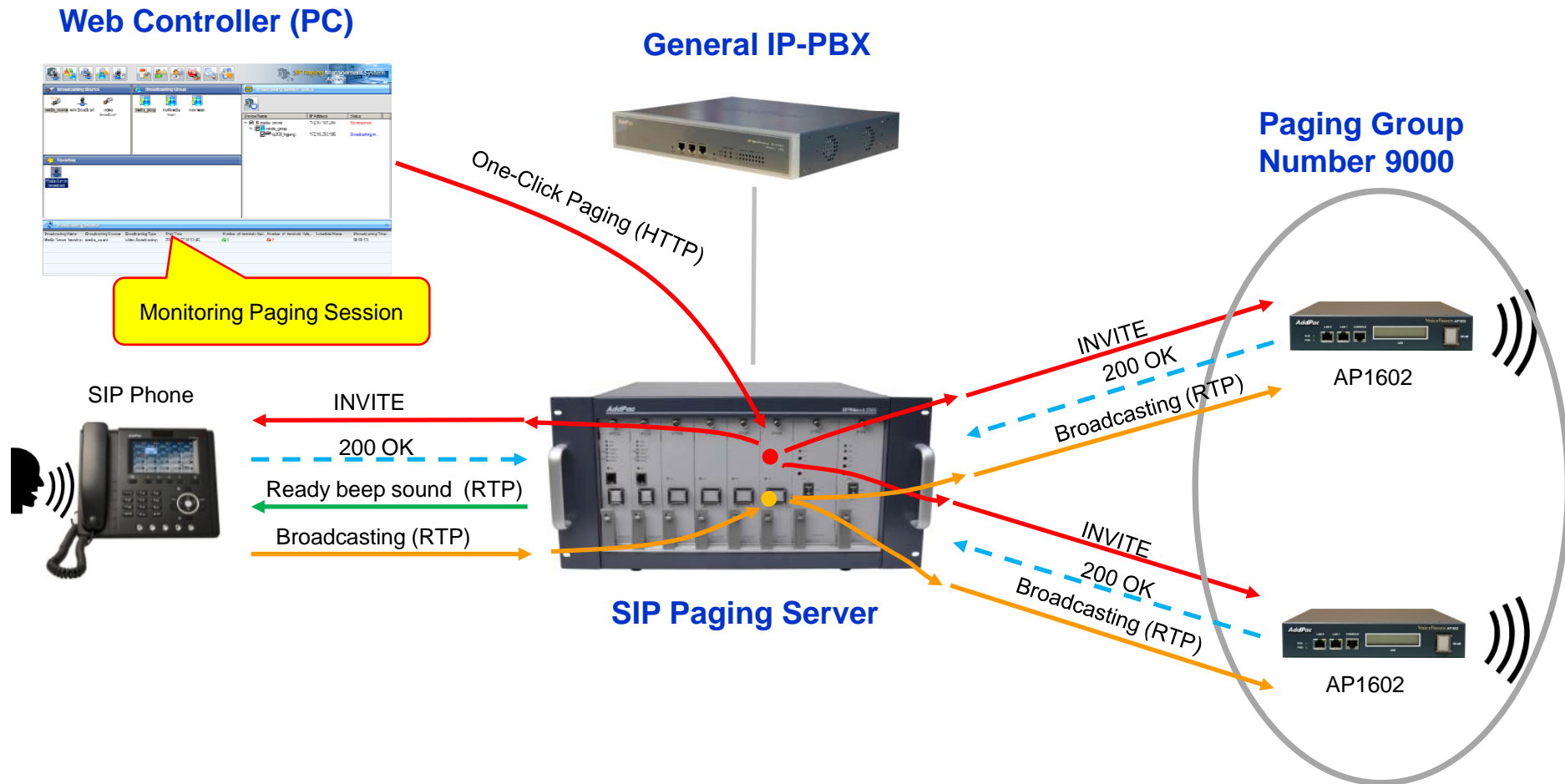
# SIP Phone Initiated Real-Time Paging to PSTN and Mobile Network



# PSTN or Mobile Network Initiated Paging with IVR

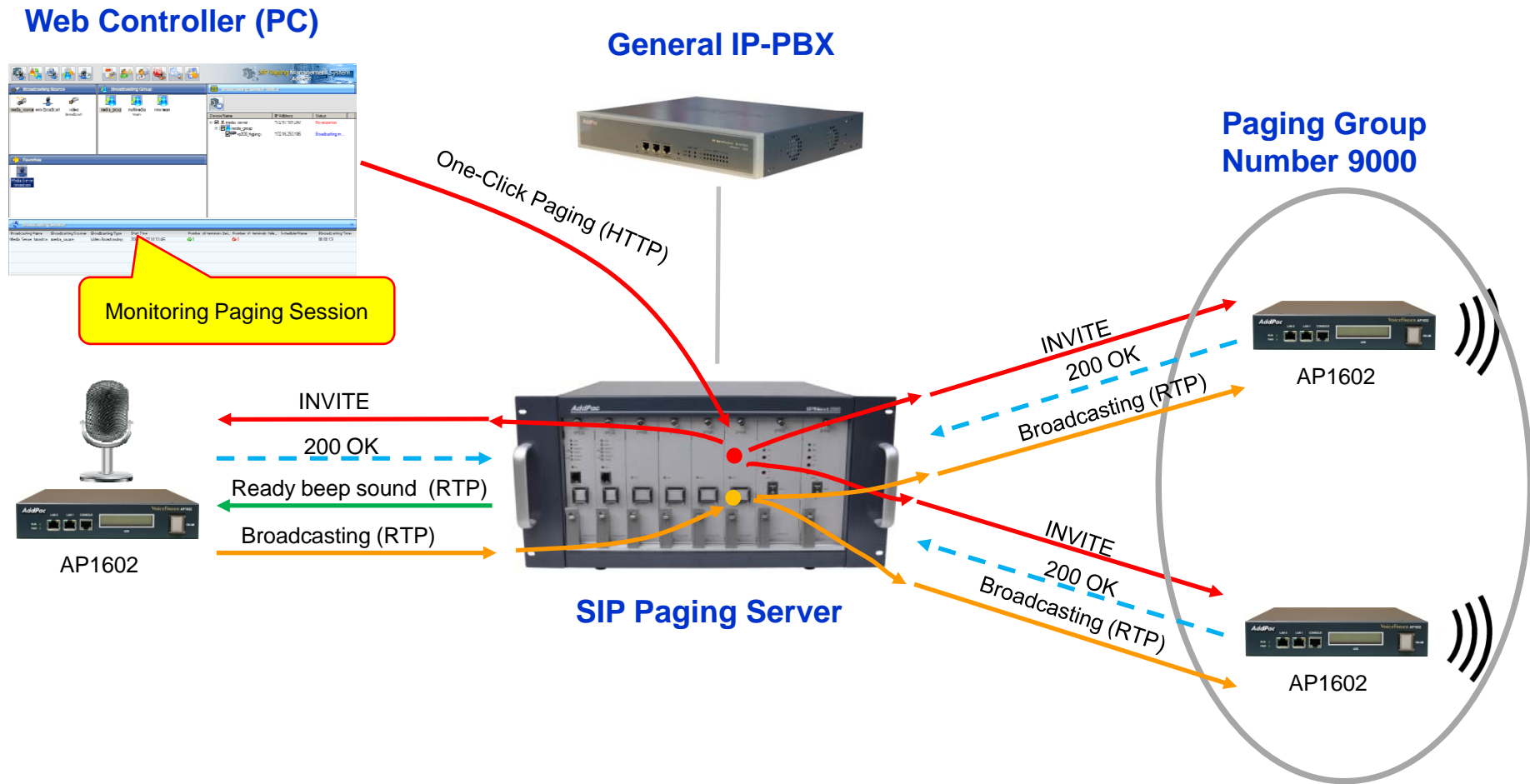


# Web Controller Initiated Real-Time Paging (1)

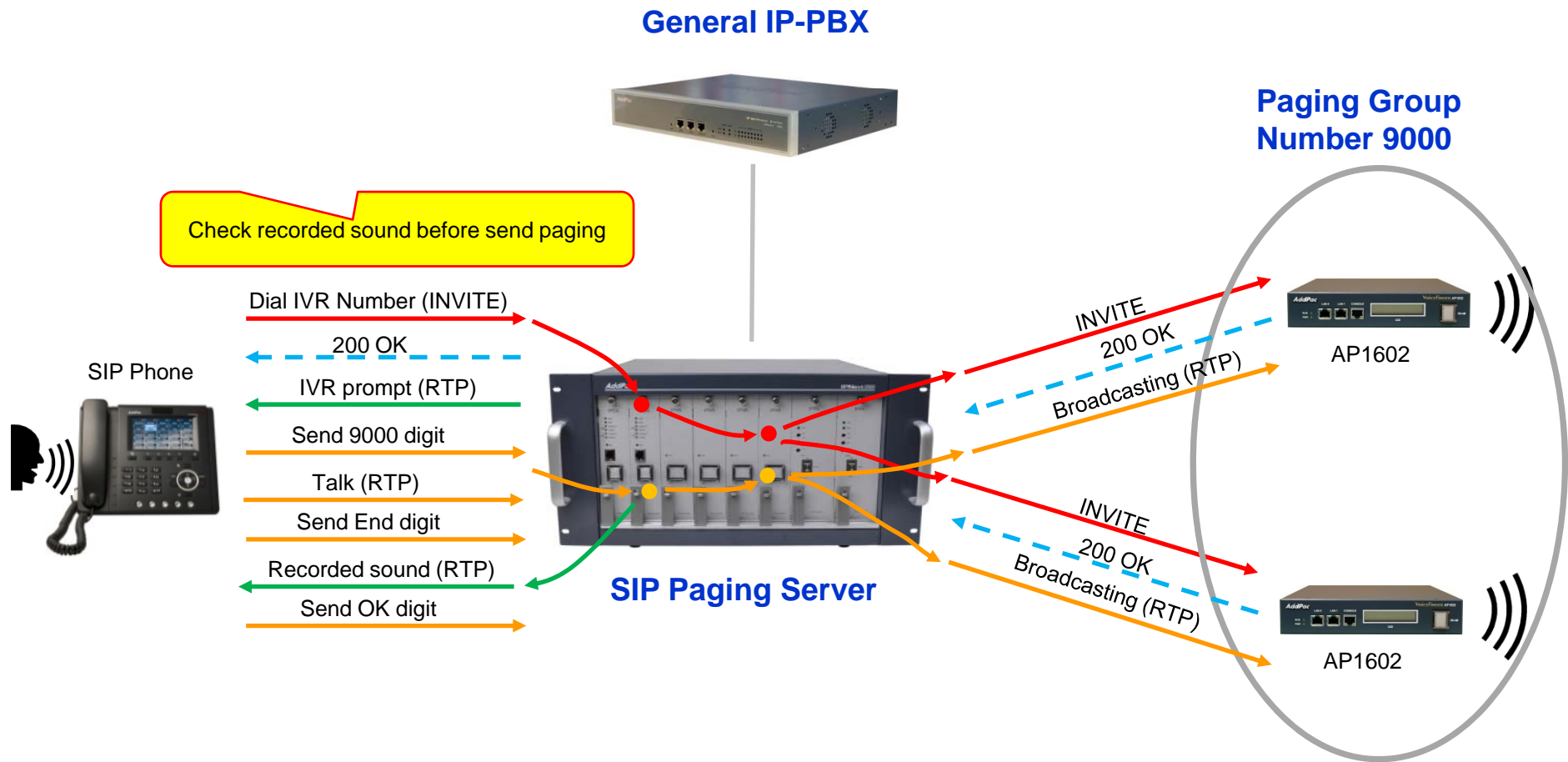




# Web Controller Initiated Real-Time Paging (2)

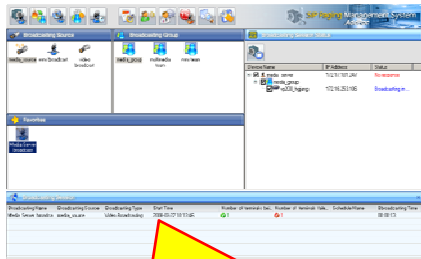


# Add-hoc Paging with IVR



# Recorded Paging with Simultaneous Casting

## Web Controller (PC)



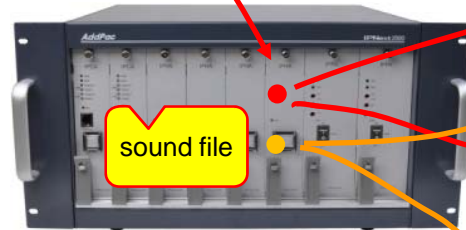
Recorded or uploaded sound file

## General IP-PBX



Start Paging (HTTP)

## Paging Group Number 9000



## SIP Paging Server

Send RTP simultaneously



AP1602

INVITE  
200 OK  
Broadcasting (RTP)

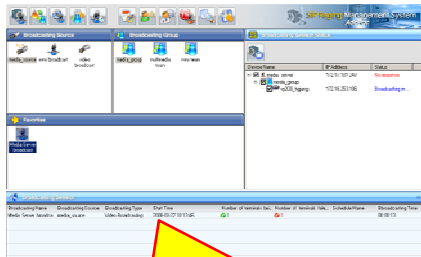


AP1602

INVITE  
200 OK  
Broadcasting (RTP)

# Recorded Paging with Individual Casting

## Web Controller (PC)



Recorded or uploaded sound file

## General IP-PBX



Start Paging (HTTP)

## Paging Group Number 9000

SIP Phone



Accept call individually

SIP Phone



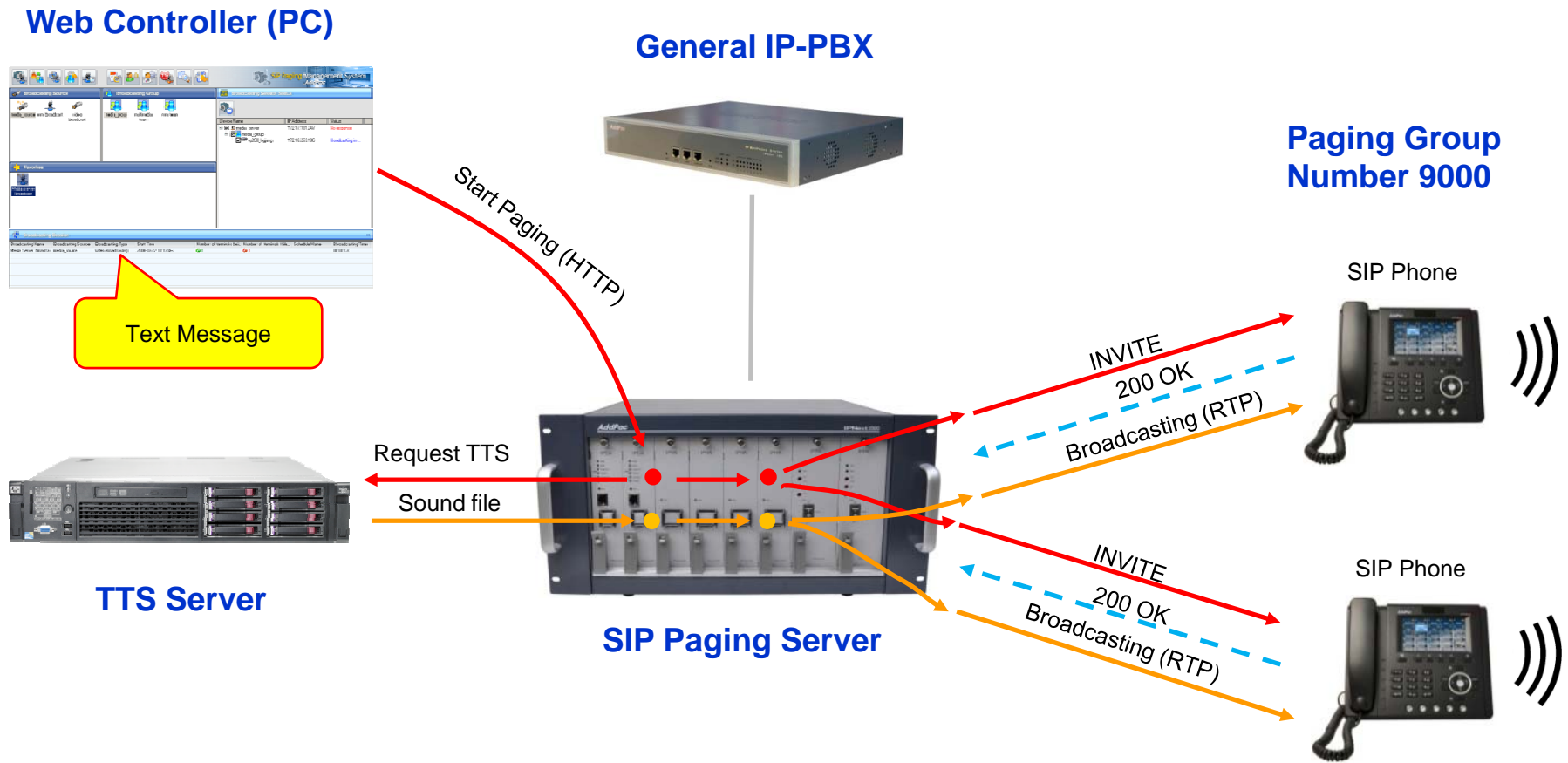
## SIP Paging Server

Send RTP individually

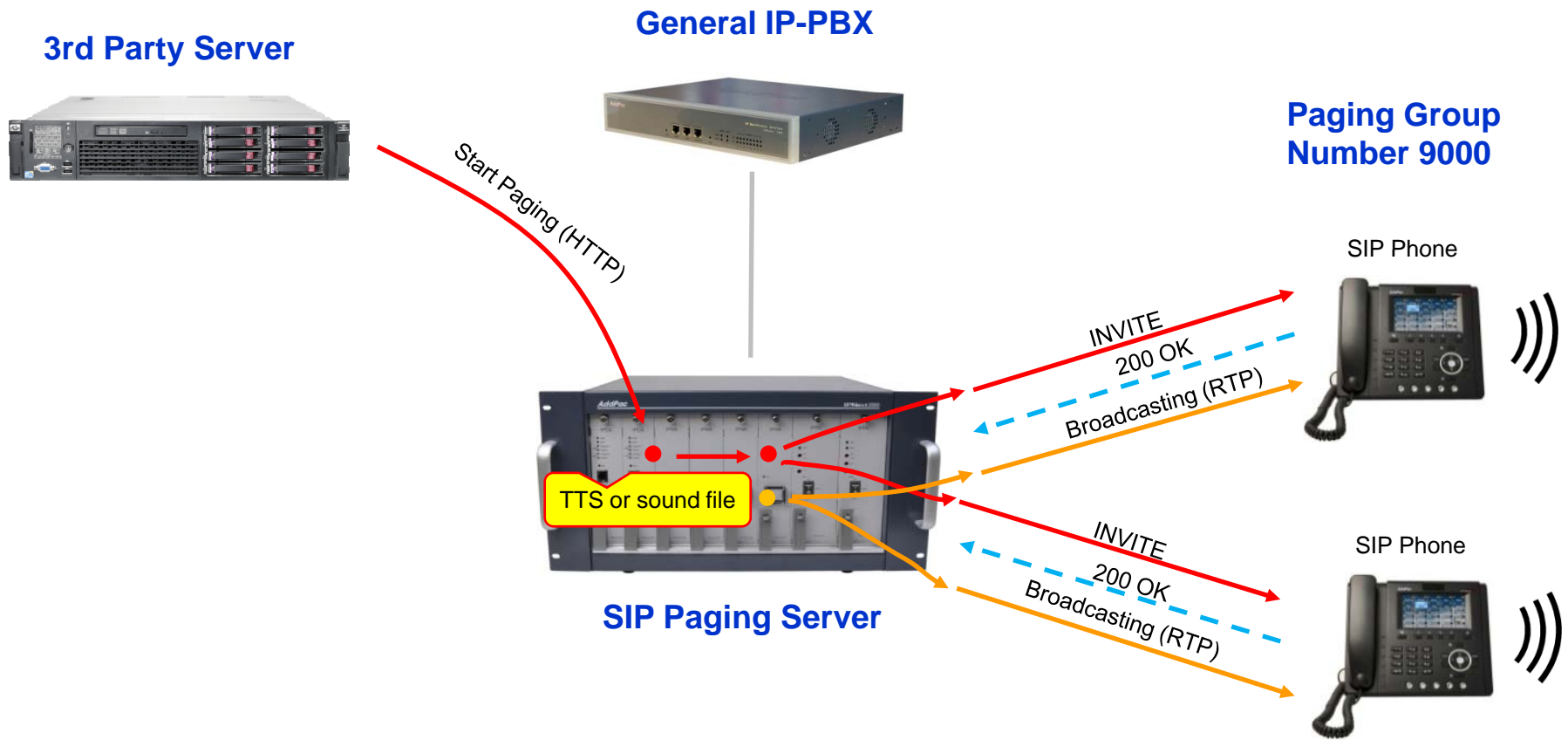
INVITE  
200 OK  
Broadcasting (RTP)

INVITE  
200 OK  
Broadcasting (RTP)

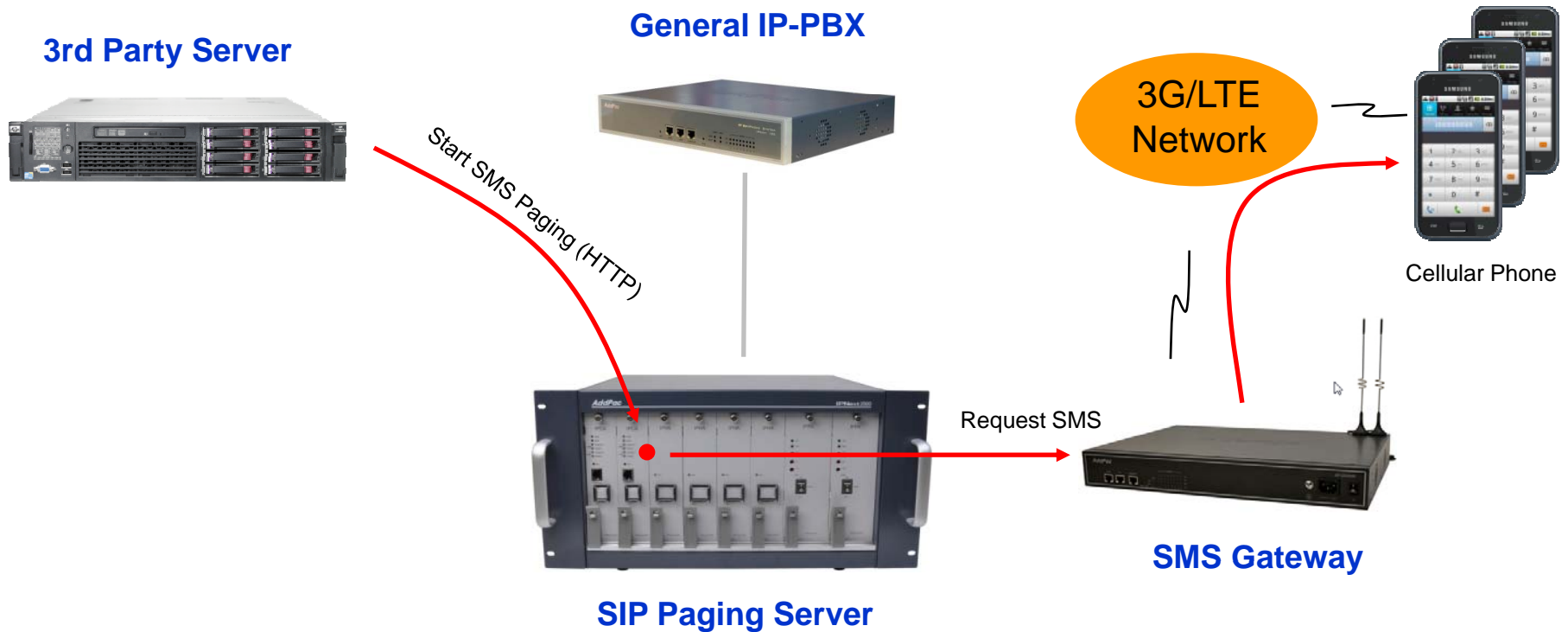
# TTS (Text to Speech) Paging



# Audio Paging by HTTP API

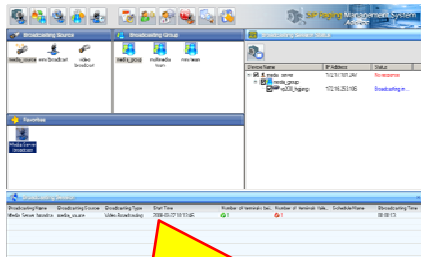


# SMS Paging by HTTP API



# Scheduled Paging

## Web Controller (PC)



Make a schedule

## General IP-PBX



Add a Schedule (HTTP)



## SIP Paging Server

## Paging Group Number 9000



AP1602

INVITE  
200 OK  
Broadcasting (RTP)



AP1602

INVITE  
200 OK  
Broadcasting (RTP)





# SIP Paging Server

# AP-SPS10000 SIP Paging Server (Commercial Server)



# Product Overview

## AP-SPS10000 SIP Paging Server

- Commercial Server Hardware Platform
- Linux Operating System
- SIP Application Server, Proxy, Registrar and Location Server
- SIP Paging Server Service
- Standalone SIP Paging & Broadcasting Service Support
- Legacy IP-PBX Clone Mode Support (Trunk, etc)
- RTP(Real-time Transport Protocol) Support for Unicast and Multicast Paging Service
- Internal & External RTP Routing Service Support
- Web based SPMS(SIP Paging Management Software) Support
- Various Paging Service Mode Support (Ex : Scheduled Broadcasting, etc)
- Various SIP Paging Terminal Support (SIP Phone, etc) Beside AddPac Product
- Paging Service Support via SIP IP Phone
- Paging Service Support via AddPac Smart Communicator (PC based AddPac Multimedia Manager)
- TTS Paging Service Support
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture

# AP-SPS2000 SIP Paging Servers



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

# Product Overview

## AP-SPS2000 SIP Paging Server

- SIP Application Server, Proxy, Registrar and Location Server
- SIP Paging Server Service
- Standalone SIP Paging & Broadcasting Service Support
- Legacy IP-PBX Clone Mode Support (Trunk, etc)
- RTP(Real-time Transport Protocol) Support for Unicast and Multicast Paging Service
- Internal & External RTP Routing Service Support
- Web based SPMS(SIP Paging Management Software) Support
- Various Paging Service Mode Support (Ex : Scheduled Broadcasting, etc)
- Various SIP Paging Terminal Support (SIP Phone, etc) Beside AddPac Product
- Paging Service Support via SIP IP Phone
- Paging Service Support via AddPac Smart Communicator (PC based AddPac Multimedia Manager)
- TTS Paging Service Support
- Dual System Redundancy Architecture
  - Two(2) Gigabit Ethernet Interface / System
- High Performance RISC Architecture
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- Dual Redundancy Power Module

# Hardware Specification

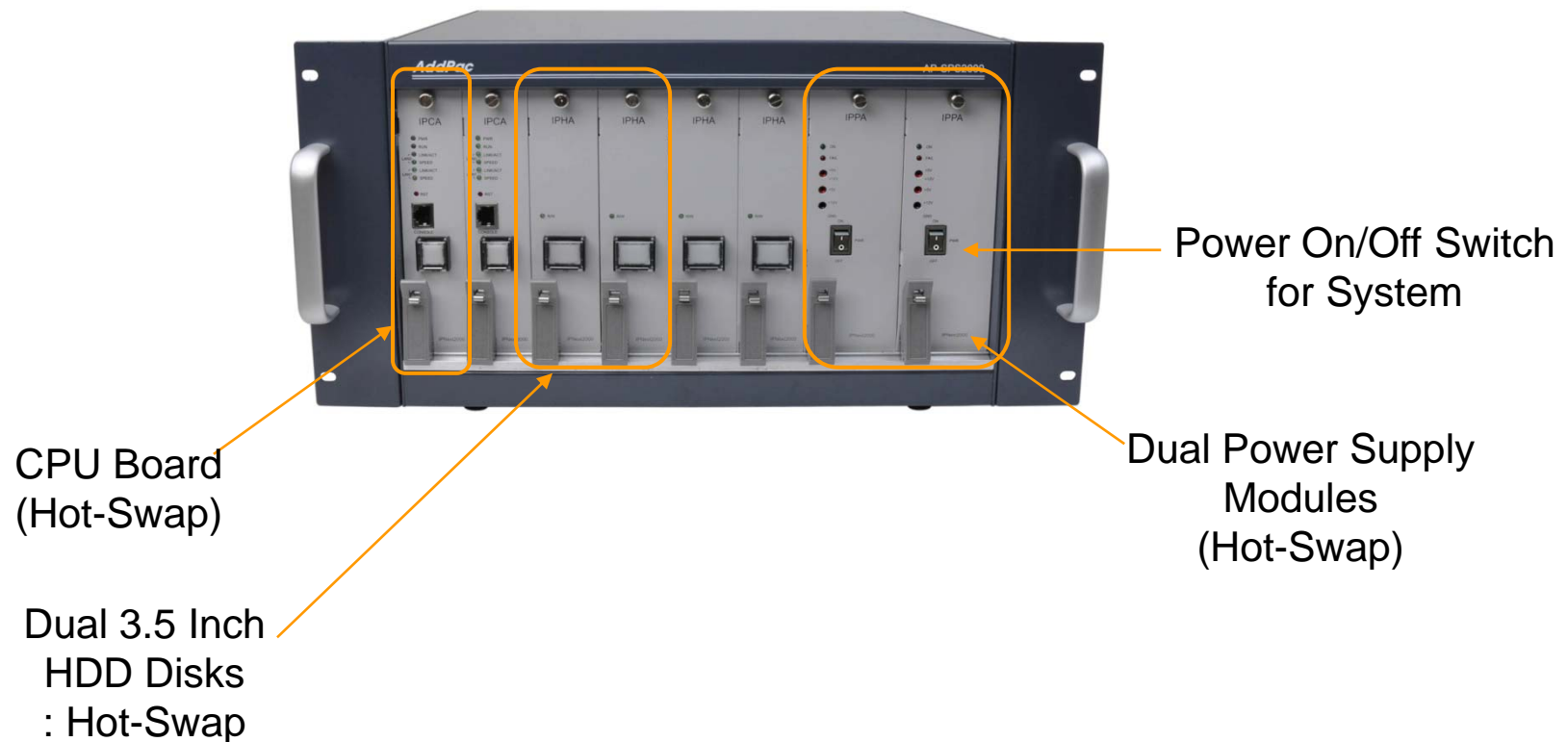
## AP-SPS2000 SIP Paging Server

- 64bit High-End Microprocessor Computing Power
- Main Chassis
  - Dual Redundancy CPU Boards for System Fault Tolerant
    - Two(2) 10/100/1000Mbps Gigabit Ethernet
    - One(1) RS-232C Console (RJ45)
    - Two(2) 3.5 Inch Hard Disk Interface Slot (RAID 1)
  - Dual Redundancy Power Supply Module
  - Hot-Swap Features

# Hardware Specification

## AP-SPS2000 SIP Paging Server

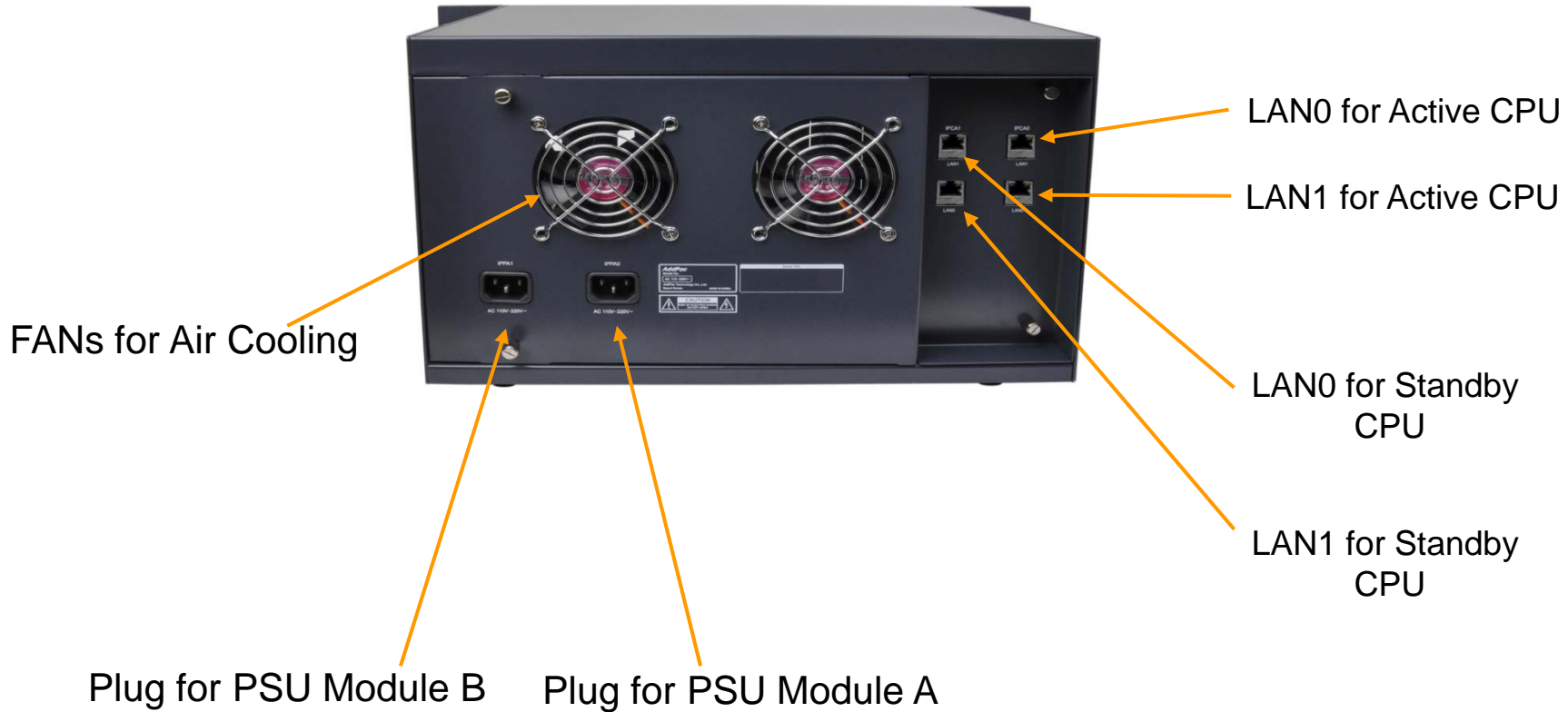
### AP-SPS2000 Front Side



# Hardware Specification

## AP-SPS2000 SIP Paging Server

### AP-SPS2000 Back Side

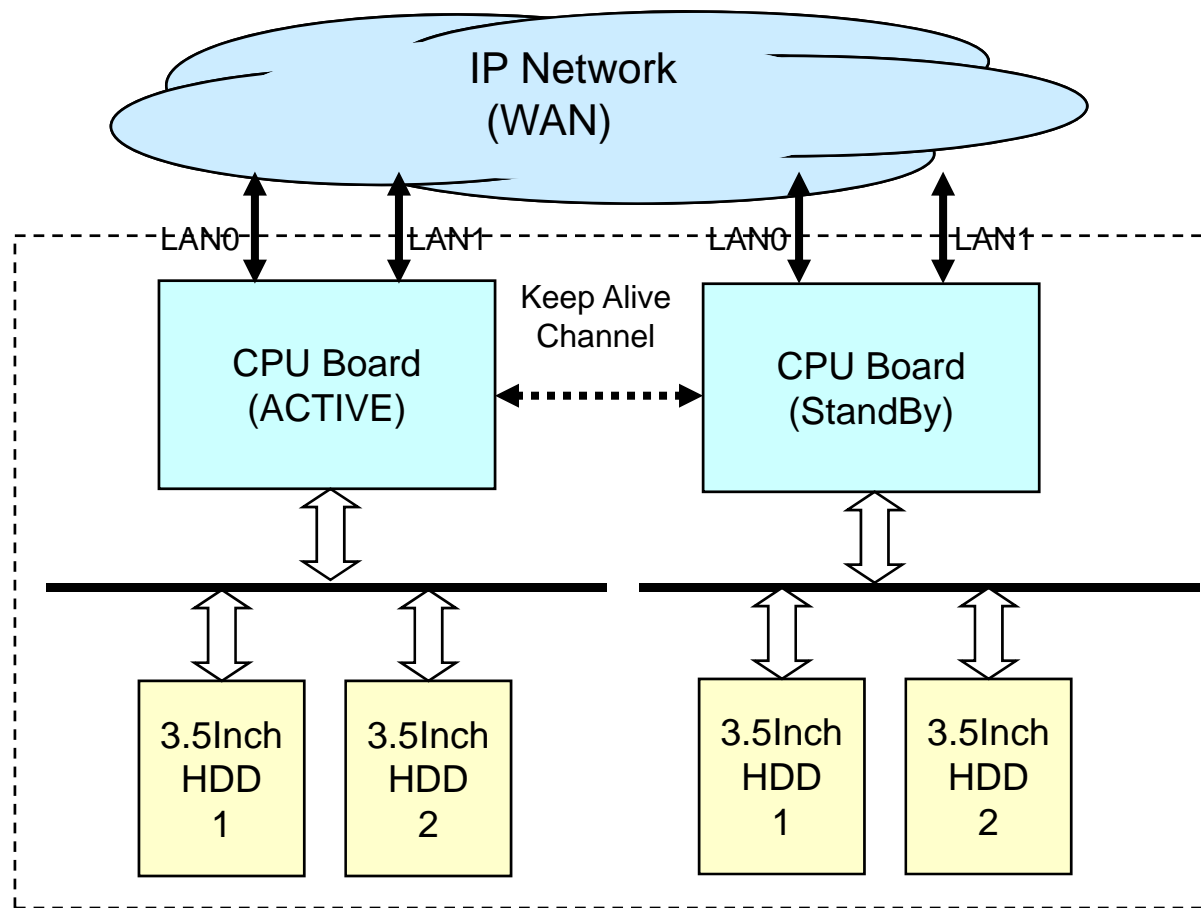




# System Redundancy Features

AP-SPS2000 SIP Paging Server

AP-SPS2000 System Block Diagram



# AP-SPS600 SIP Paging Server



# Product Overview

## AP-SPS600 SIP Paging Server

- SIP Application Server, Proxy, Registrar and Location Server
- SIP Paging Server Service
- Standalone SIP Paging & Broadcasting Service Support
- Legacy IP-PBX Clone Mode Support (Trunk, etc)
- RTP(Real-time Transport Protocol) Support for Unicast and Multicast Paging Service
- Internal & External RTP Routing Service Support
- Web based SPMS(SIP Paging Management Software) Support
- Various Paging Service Mode Support (Ex : Scheduled Broadcasting, etc)
- Various SIP Paging Terminal Support (SIP Phone, etc) Beside AddPac Product
- Paging Service Support via SIP IP Phone
- Paging Service Support via AddPac Smart Communicator (PC based AddPac Multimedia Manager)
- TTS Paging Service Support
- Dual System Redundancy Architecture
  - Two(2) Fast Ethernet Interface / System
- High Performance RISC Architecture
- Powerful Network Protocols (PPPoE, DHCP, Static Routing, etc)
- Firmware Upgradeable Architecture
- Dual Redundancy Power Module

# Hardware Specification

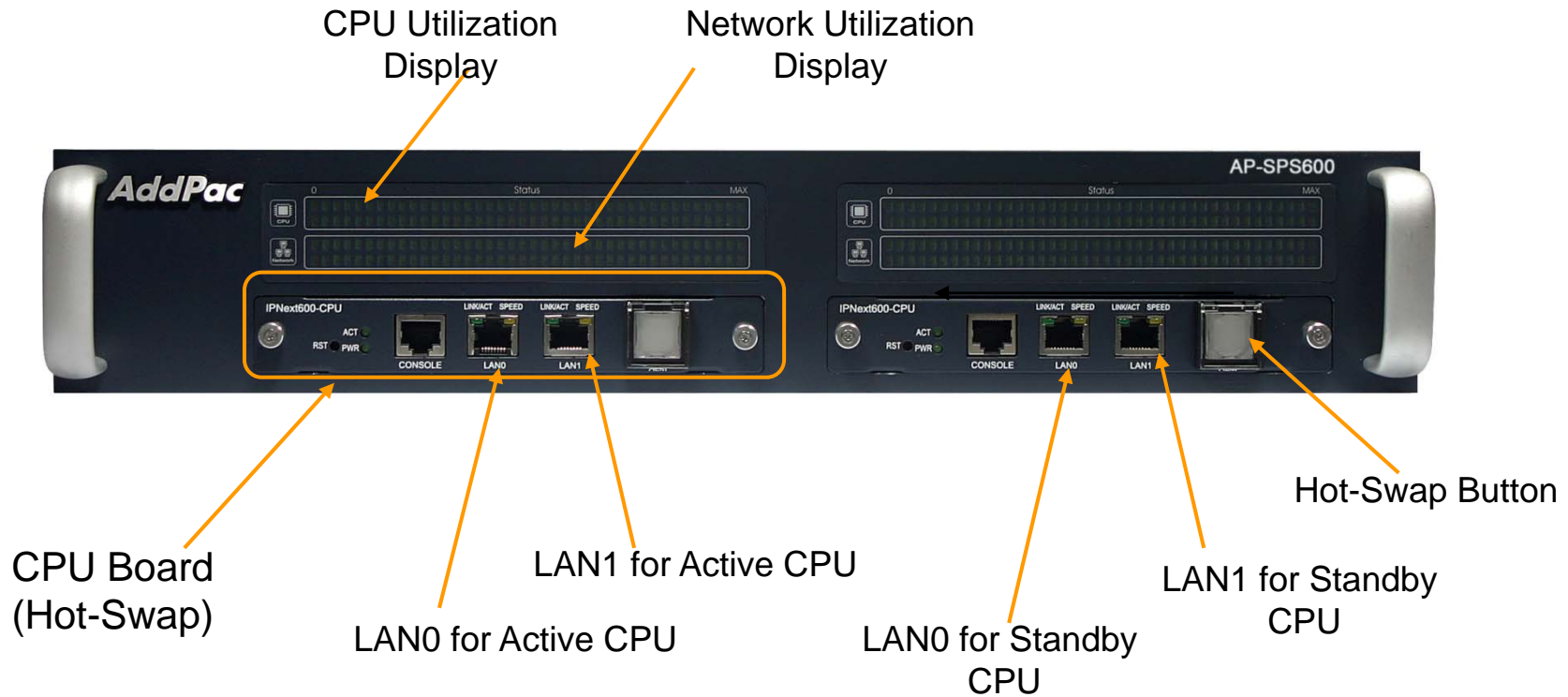
## AP-SPS600 SIP Paging Server

- 64bit High-End Microprocessor Computing Power
- Main Chassis
  - Dual Redundancy CPU Boards for System Fault Tolerant
    - Two(2) 10/100Mbps Fast Ethernet
    - One(1) RS-232C Console (RJ45)
  - Dual Redundancy Power Supply Module
  - Hot-Swap Features

# Hardware Specification

## AP-SPS600 SIP Paging Server

### AP-SPS600 Front Side



# Hardware Specification

AP-SPS600 SIP Paging Server

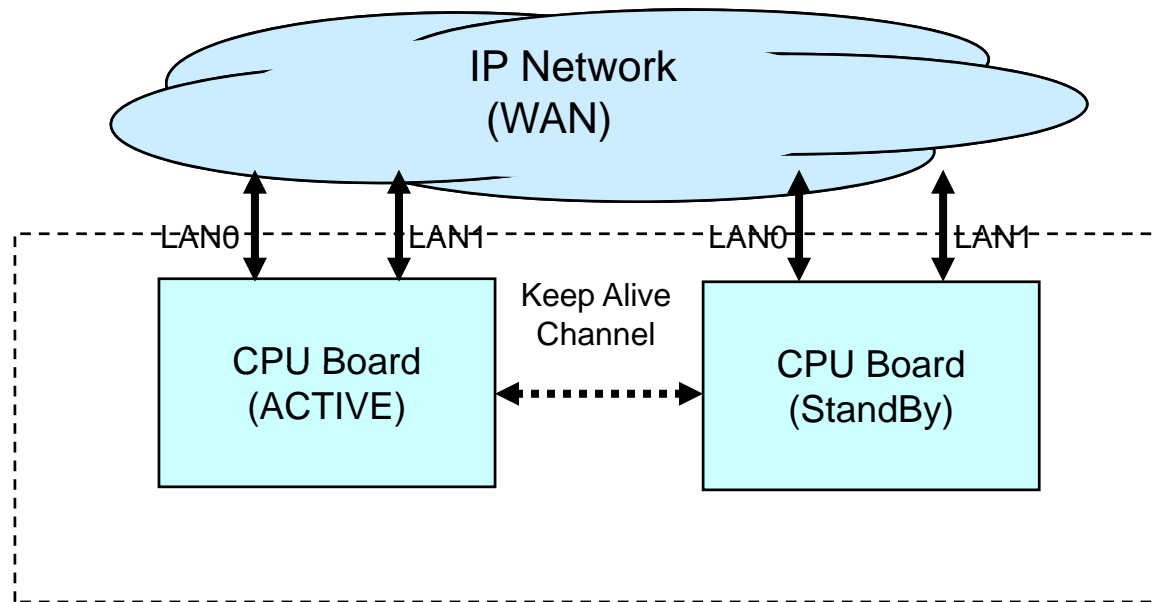
## AP-SPS600 Back Side



# System Redundancy Features

AP-SPS600 SIP Paging Server

AP-SPS600 System Block Diagram





# SIP Paging Terminals



# AP2620

## SIP VoIP Gateway + SIP Audio Broadcasting Terminal



# 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

# 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

# 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

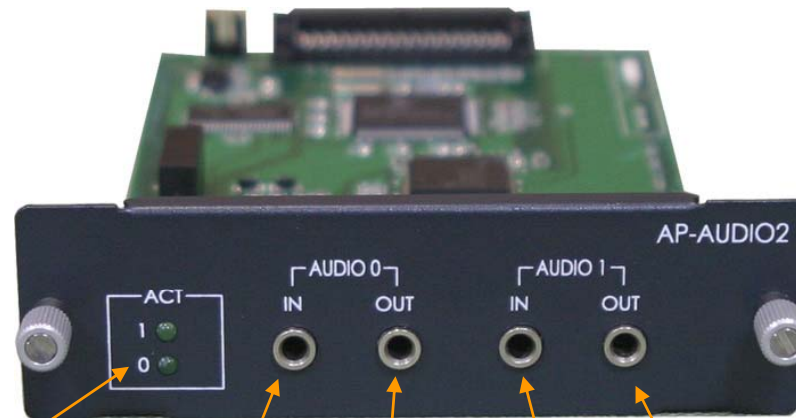
# Hardware Specification

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

**AddPac**

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

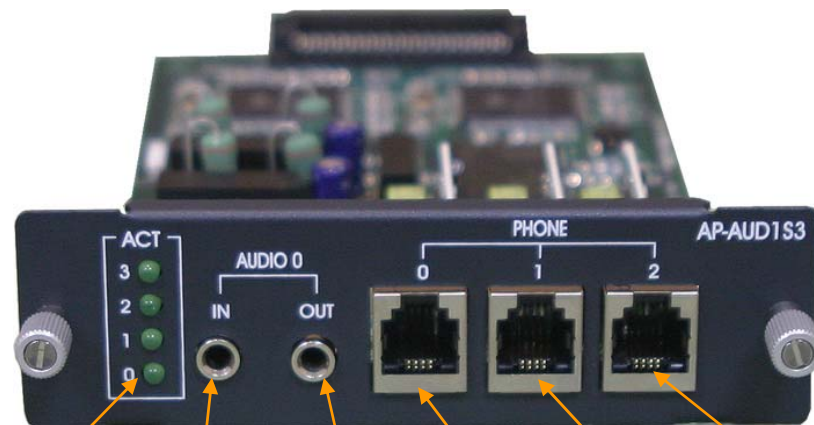
# Hardware Specification

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

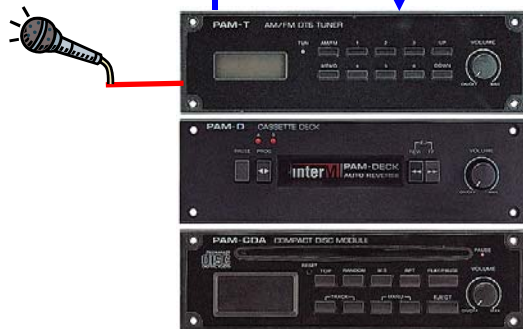
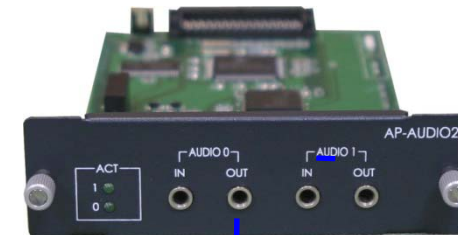
Audio Module Type	Audio Module Features
<p data-bbox="562 756 804 794">AP-AUD1S3</p>  <p>The image shows the AP-AUD1S3 audio module, a black rectangular device. On the left side, there are three green LEDs labeled 'ACT' with numbers 3, 2, 1, and 0. Below the LEDs are two circular ports labeled 'IN' and 'OUT'. In the center, there are three RJ11 ports labeled '0', '1', and '2'. On the right side, there are two circular ports labeled 'PHONE' and 'AP-AUD1S3'.</p>	<p data-bbox="1178 751 1742 790">One(1)-Channel Audio In/Out Port</p> <p data-bbox="1178 842 1742 880">Audio Encoding/Decoding Service</p> <p data-bbox="1178 906 1469 944">Audio IN : MIC IN</p> <p data-bbox="1178 962 1541 1000">Audio OUT :Line OUT</p> <p data-bbox="1178 1018 1518 1056">3.5mm Stereo JACK</p> <p data-bbox="1178 1082 1816 1120">Three(3) FXS Port Interface (RJ11 x 3)</p> <p data-bbox="1178 1169 1816 1249">G.711, G.726, G.729A, G.723.1 Audio Codec</p>

# 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

Headphon  
e

# AP1602

## SIP Audio Broadcasting Terminal



# 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

# 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

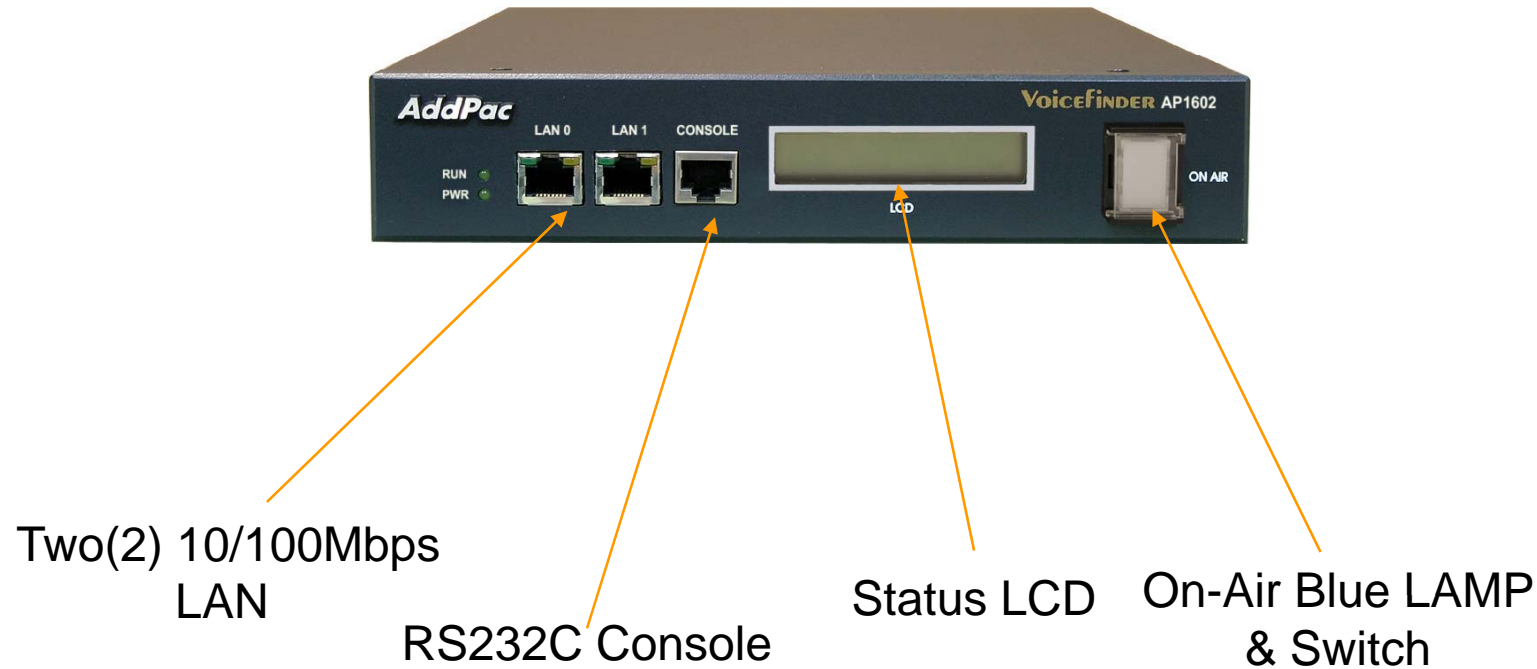
# Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP

## AP1602 Front Side



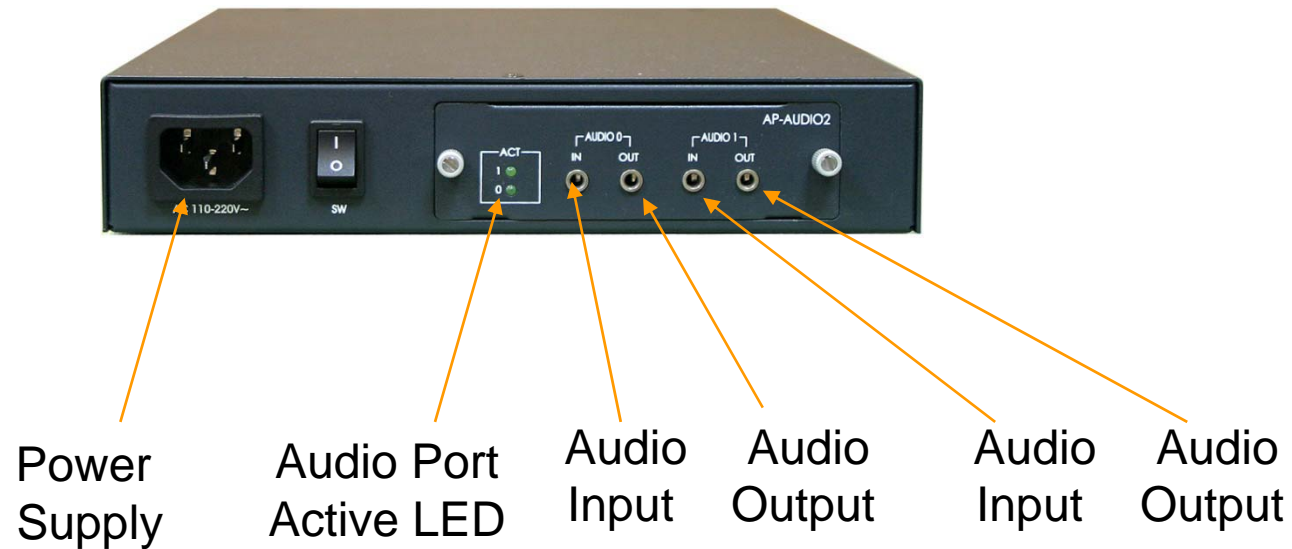
# Hardware Specification

AP1602 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP

## AP1602 Back Side



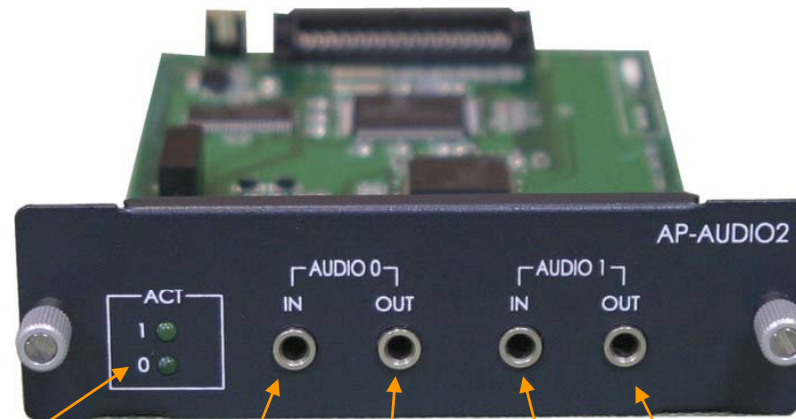
# Hardware Specification

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**

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



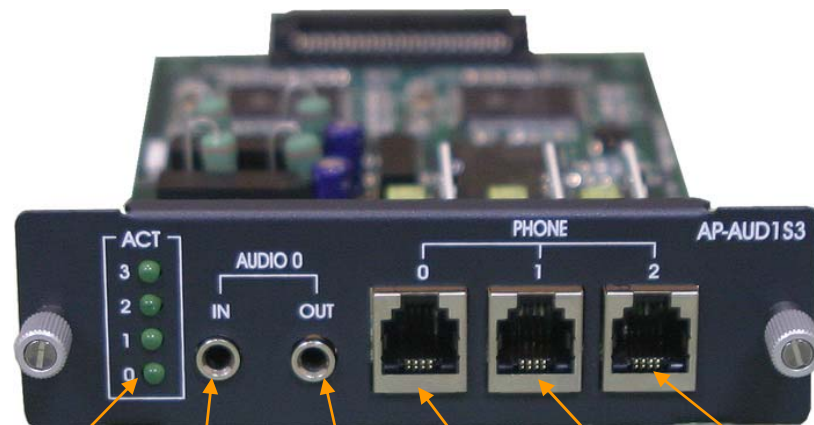
# Hardware Specification

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

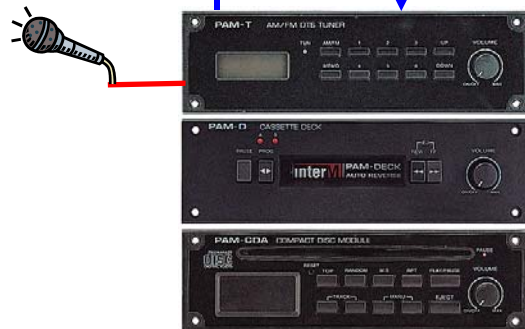
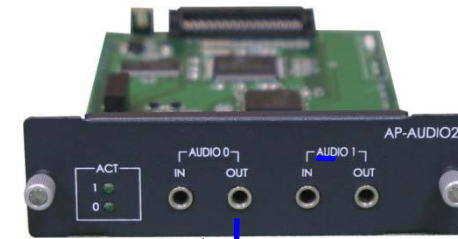
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<p data-bbox="562 756 801 794">AP-AUD1S3</p>  <p>The image shows the AP-AUD1S3 audio module, a black rectangular device with various ports and indicators. On the left side, there are three green LEDs labeled 'ACT' with numbers 3, 2, 1, and 0. Next to them are 'IN' and 'OUT' ports. In the center, there are three RJ11 ports labeled '0', '1', and '2' under the heading 'PHONE'. On the right side, there are two 3.5mm stereo jacks. The model number 'AP-AUD1S3' is printed on the top right of the device.</p>	<p data-bbox="1178 751 1742 790">One(1)-Channel Audio In/Out Port</p> <p data-bbox="1178 842 1742 880">Audio Encoding/Decoding Service</p> <p data-bbox="1178 906 1469 944">Audio IN : MIC IN</p> <p data-bbox="1178 962 1541 1000">Audio OUT :Line OUT</p> <p data-bbox="1178 1018 1518 1056">3.5mm Stereo JACK</p> <p data-bbox="1178 1082 1816 1120">Three(3) FXS Port Interface (RJ11 x 3)</p> <p data-bbox="1178 1169 1816 1249">G.711, G.726, G.729A, G.723.1 Audio Codec</p>

# AP-AUDIO2 Module

AP1602 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

Headphon  
e

# AP1605

## SIP Audio Broadcasting Terminal



# Product Overview

## AP1605 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
- High Quality Audio Codec Support (High Quality Codec, G.711, G.726, etc)
- Unicast and Multicast Broadcasting Scheme
- SPMS (SIP Paging Management Software) Support
- **Optional Built-In Digital AMP.**
- On-AIR Blue LAMP
- **Volume Control Rotary Switch at front panel**
- High-Quality Audio/Voice Service
- Firmware Upgradeable Architecture
- Broadcasting Solution with Outstanding Network Service Capability

# Hardware Specification

## AP1605 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
- High Quality Audio Encoding/Decoding Service
- ON-AIR Blue LAMP
- Rotary Volume Control Switch
- Option Module : AP-N3-HQA1000
  - One(1) 10/100Mbps Fast Ethernet (RJ45)
  - One(1) RS-232C Interface (RJ45) for Command Line Interface
  - Stereo Audio Input/Output Connector
- Option Module : AP-N3-HQA1000A
  - One(1) 10/100Mbps Fast Ethernet (RJ45)
  - One(1) RS-232C Interface (RJ45) for Command Line Interface
  - Stereo Audio Input/Output Connector
  - Built-in Audio AMP.

# Hardware Specification

AP1605 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP

## AP1605 Front Side



Status LCD

Rotary Volume  
Control Switch

On-Air Blue LAMP



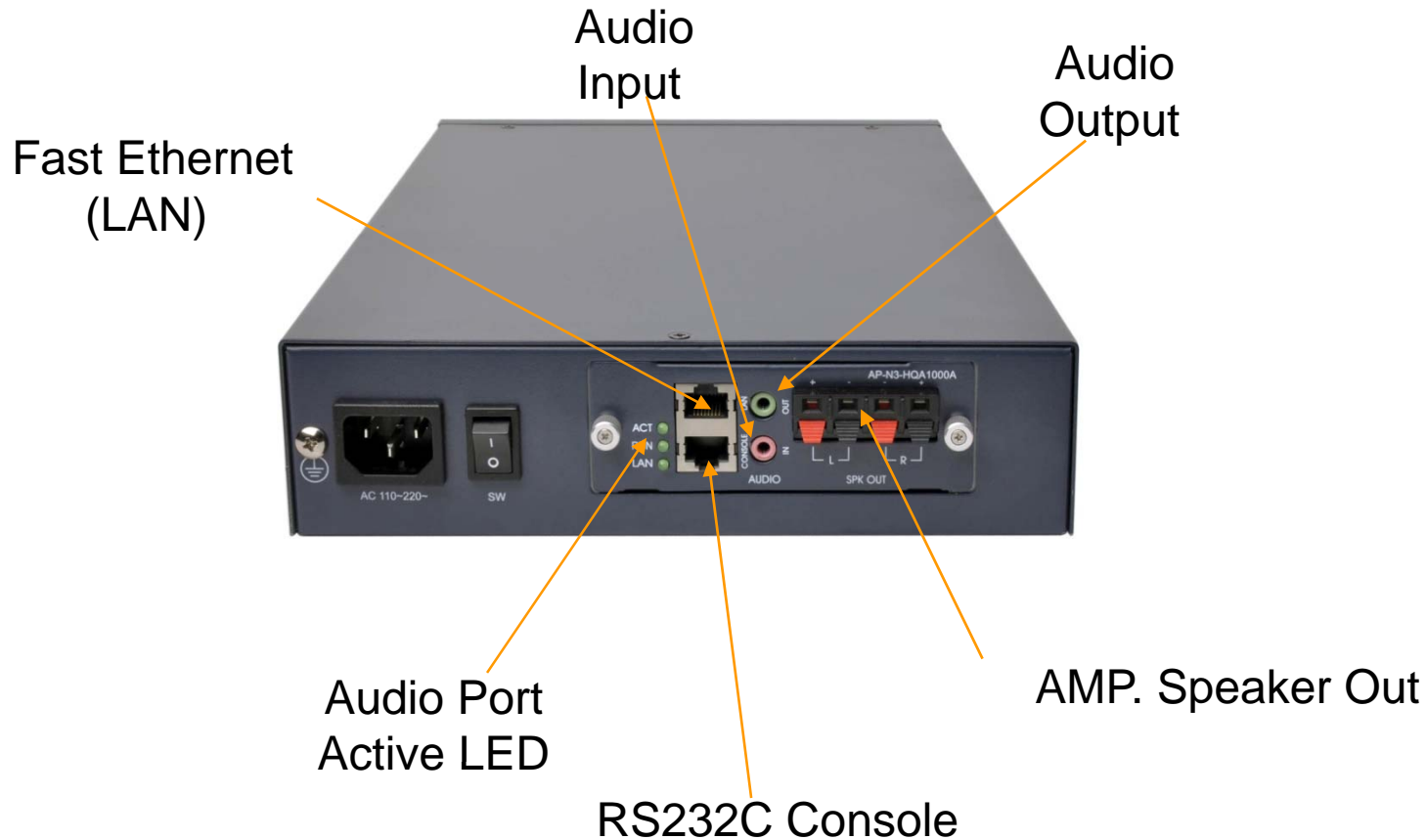
# Hardware Specification

AP1605 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP

## AP1605 Back Side



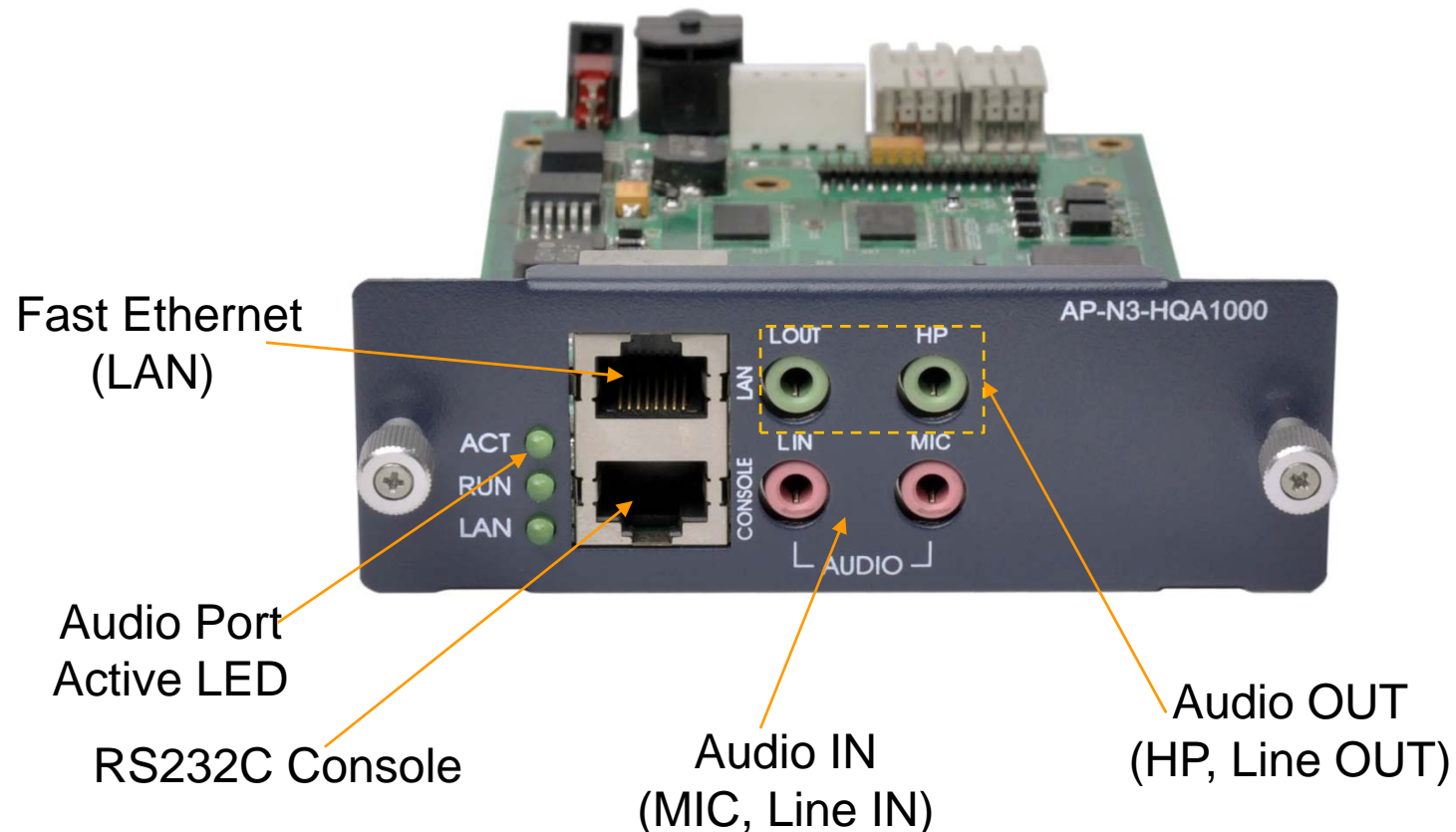
# Hardware Specification

AP1605 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP

## AP-N3-HQA1000 Board



**AddPac**

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

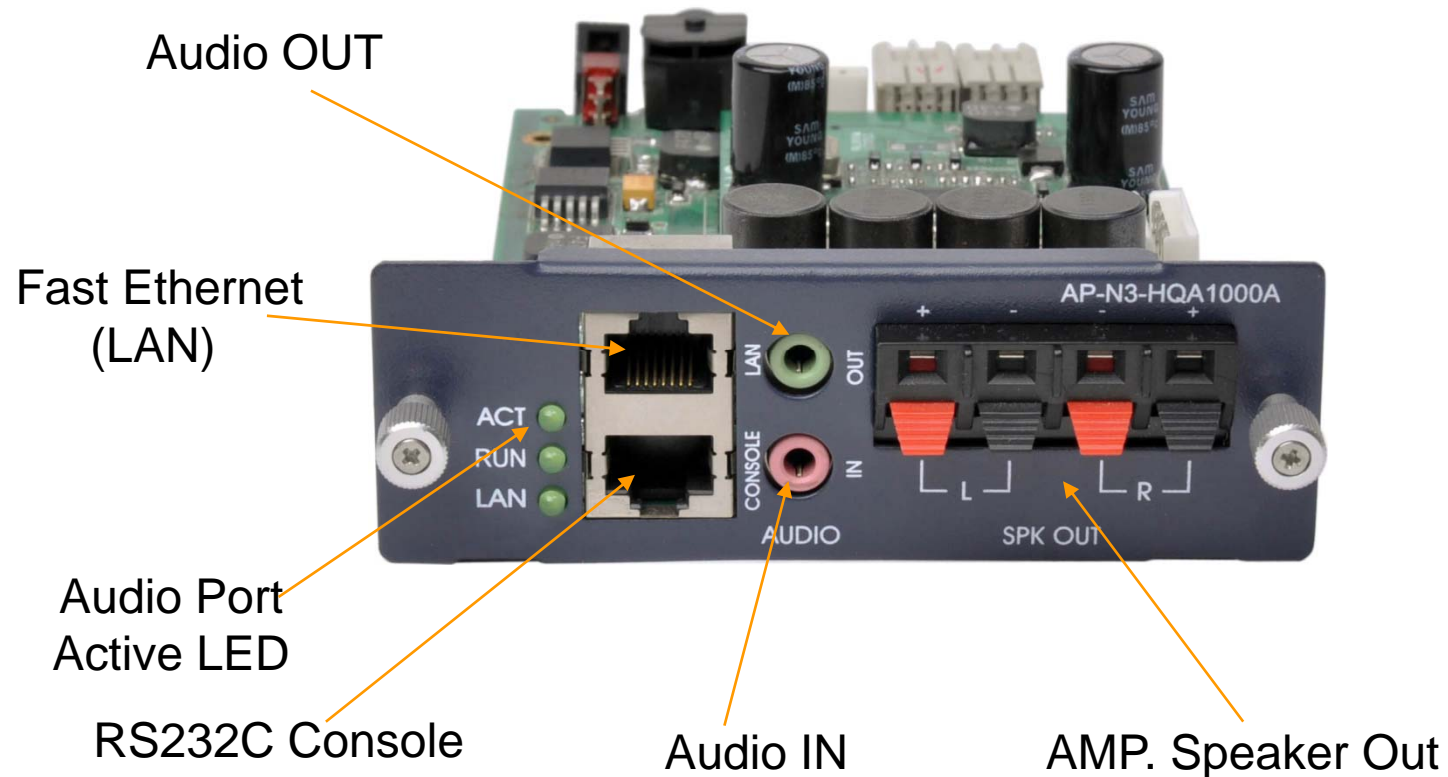
# Hardware Specification

AP1605 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP

## AP-N3-HQA1000A Board



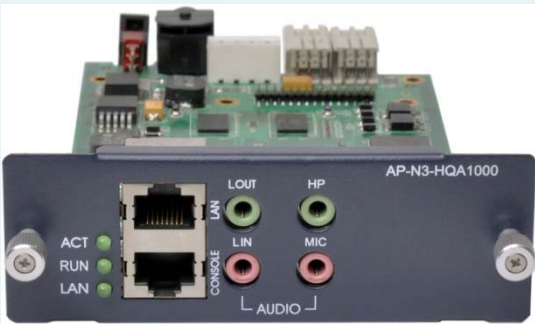
# Hardware Specification

AP1605 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP

## AP1605 Audio Module

Audio Module Type	Audio Module Features
<p data-bbox="562 703 893 743">AP-N1-HQA1000</p>  <p>The image shows a dark grey audio module with a green PCB. On the front panel, there are three status LEDs labeled ACT, RUN, and LAN. To the right of the LEDs are a LAN port, a console port, and an audio jack labeled 'AUDIO'. Further right are two pairs of ports: LOUT and HP (green), and LIN and MIC (red). The model number 'AP-N3-HQA1000' is printed on the top right of the front panel.</p>	<ul style="list-style-type: none"><li data-bbox="1178 699 1742 738">One(1)-Channel Audio In/Out Port</li><li data-bbox="1178 762 1608 802">One(1) Fast Ethernet Port</li><li data-bbox="1178 826 1525 866">One(1) RS232C Port</li><li data-bbox="1178 890 1742 930">Audio Encoding/Decoding Service</li><li data-bbox="1178 954 1559 994">Audio IN : MIC, Line IN</li><li data-bbox="1178 1010 1765 1050">Audio OUT : Headphone, Line OUT</li><li data-bbox="1178 1066 1518 1106">3.5mm Stereo JACK</li><li data-bbox="1178 1153 1843 1193">High Quality Codec, G.711 Audio Codec</li></ul>

# Hardware Specification

AP1605 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP

## AP1605 Audio Module

Audio Module Type	Audio Module Features
AP-N1-HQA1000A	One(1)-Channel Audio In/Out Port
	One(1) Fast Ethernet Port
	One(1) RS232C Port
	Audio Encoding/Decoding Service
	Audio IN
	Audio OUT
	AMP. Built-in Speaker Out (Left, Right)
	4ohm Speaker : 50Watt
	8ohm Speaker : 30Watt
	High Quality Codec, G.711 Audio Codec

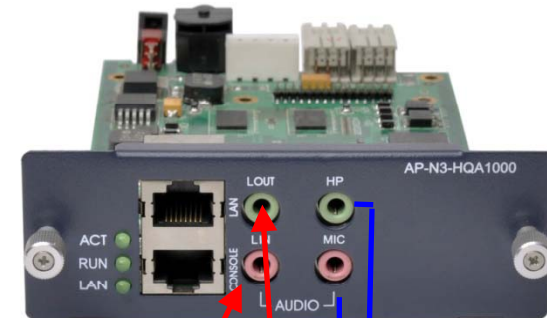


# AP-N3-HQ1000 Module

AP1605 SIP Audio Broadcasting Terminal

RISC  
CPU

High-end  
DSP



Connect with 1:2 Stereo Jack

Connect with 2:1 Stereo Jack



- High Quality audio codec realizes high quality audio service
- Direct MIC-In/ Headphone In Port for monitoring
- Real time High Quality Audio Band Broadcasting



Headphon  
e



# SIP Phones for Paging Service

# AP-IP300

## SIP Broadcasting Phone



AP-IP300

AP-PT20 (40 Speed Dial Key)



# Product Overview

## AP-IP300 SIP Broadcasting Phone

- Premium IP Phone Solution
- SIP, H.323 Dual VoIP Signaling Stack
- SIP Paging Service Solution
- 25 Speed-Dial Button for Group Paging Service
- External Speed-Dial Extend Pack Support (AP-PT20, etc)
- Various VoIP Voice Codec Support (G.711,G.726, G.729A,G.7231.1,etc)
- High-performance Audio, and Voice Service
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection

# Hardware Specification

## AP-IP300 SIP Broadcasting Phone

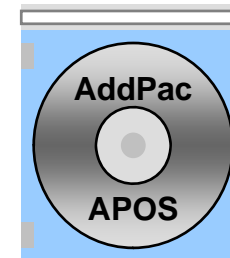
- RISC+DSP Microprocessor Computing Power (Dual Processor Architecture)
- High Quality 4.3 Inch Color LCD Panel
- 25 Speed Dial Key & User Presence Indication LED
- Optional PSTN Backup Interface
  - FXO Interface
- High quality Audio and Voice Interface
  - Stereo Audio Input Connector
  - Stereo Audio Output Connector
- Network Interface
  - Two(2) 10/100Mbps Fast Ethernet
- USB Host Mode Interface
  - USB Memory(Flash, HDD), USB Keyboard, USB Mouse, USB Wifi
- Power Supply
  - Power over Ethernet
  - External Power Adaptor (5V, 3A)



# Software Service

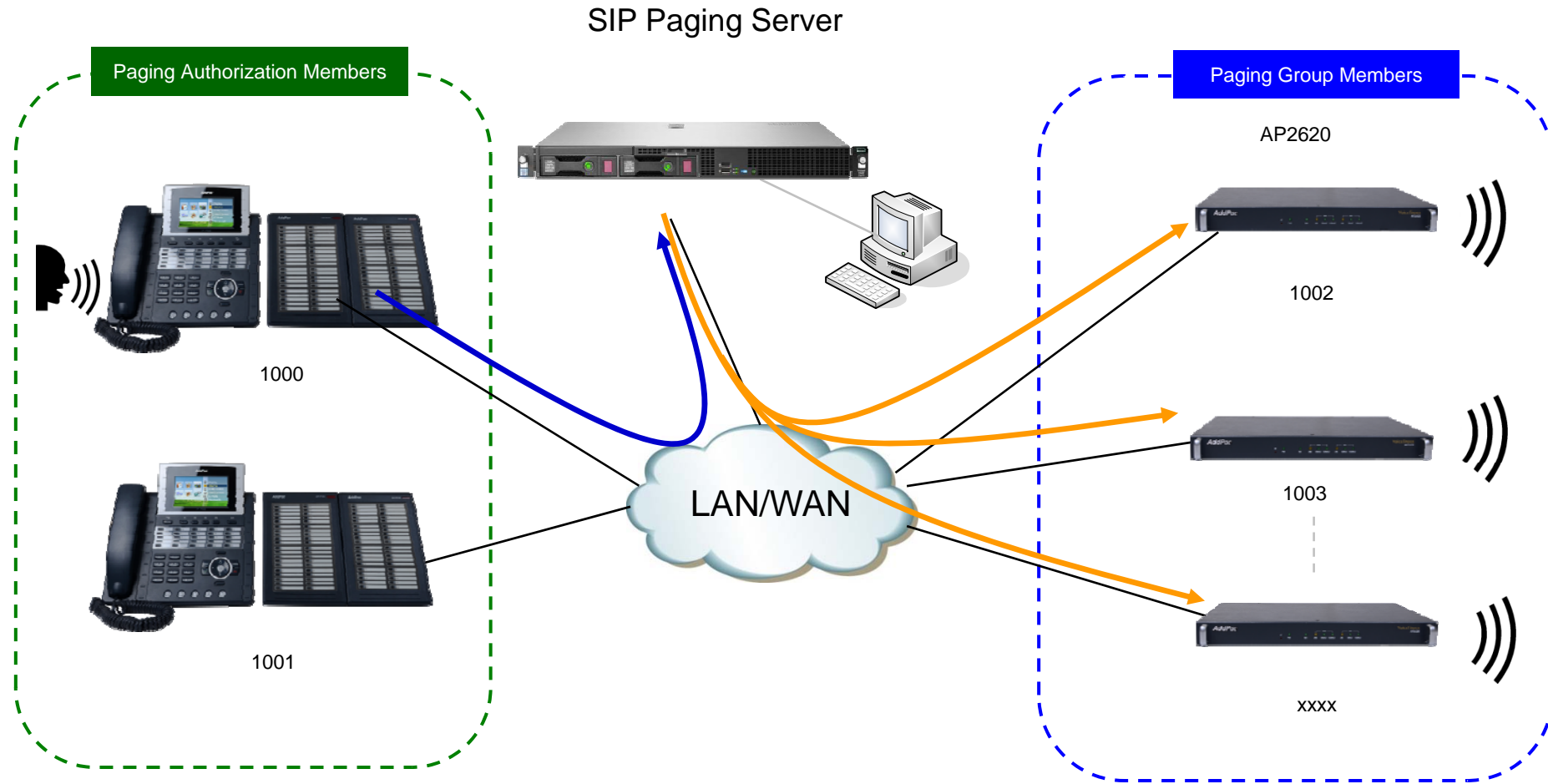
## AP-IP300 SIP Broadcasting Phone

- Built-in AddPac APOS Internetworking Software
  - Scalability, Functionality, and Stability Features
  - Audio Traffic QoS Control
- Programmable Video, Audio, and Voice Services
  - Audio, and Voice Codec
- Firmware Upgradeable Architecture
- Industry Standard IP based Network Protocol Features



# AP-IP300 SIP Broadcasting Phone

## Application Area



# AP-IP120

## SIP Broadcasting Phone



AP-IP120

AP-PT20 (40 Speed Dial Key)

# Product Overview

## AP-IP120 SIP Broadcasting Phone

- IP Phone Solution
- SIP, H.323 Dual VoIP Signaling Stack
- SIP Paging Service Solution
- 12 Speed-Dial Key with Presence Indication Lamp
- External Speed-Dial Extend Pack Support (AP-PT20, etc)
- Various VoIP Voice Codec Support (G.711,G.726, G.729A,G.7231.1,etc)
- High-performance Audio, and Voice Service
- Firmware Upgradeable Architecture
- VoIP Solution with Outstanding Network Service Capability
- Audio Privacy Protection

# Hardware Specification

## AP-IP120 SIP Broadcasting Phone

- RISC+DSP Microprocessor Computing Power (Dual Processor Architecture)
- Optional PSTN Backup (FXO) Interface
- Optional PoE (Power over Ethernet)
- High quality Audio and Voice Interface
  - Stereo Audio Input Connector
  - Stereo Audio Output Connector
- Network Interface
  - Two(2) 10/100Mbps Fast Ethernet
- LCD Window : Graphic LCD (4 Line Text)
- 12 Speed-Dial Key with Presence Indication LAMP
- Power Supply
  - External Power Adaptor (5V, 2A)

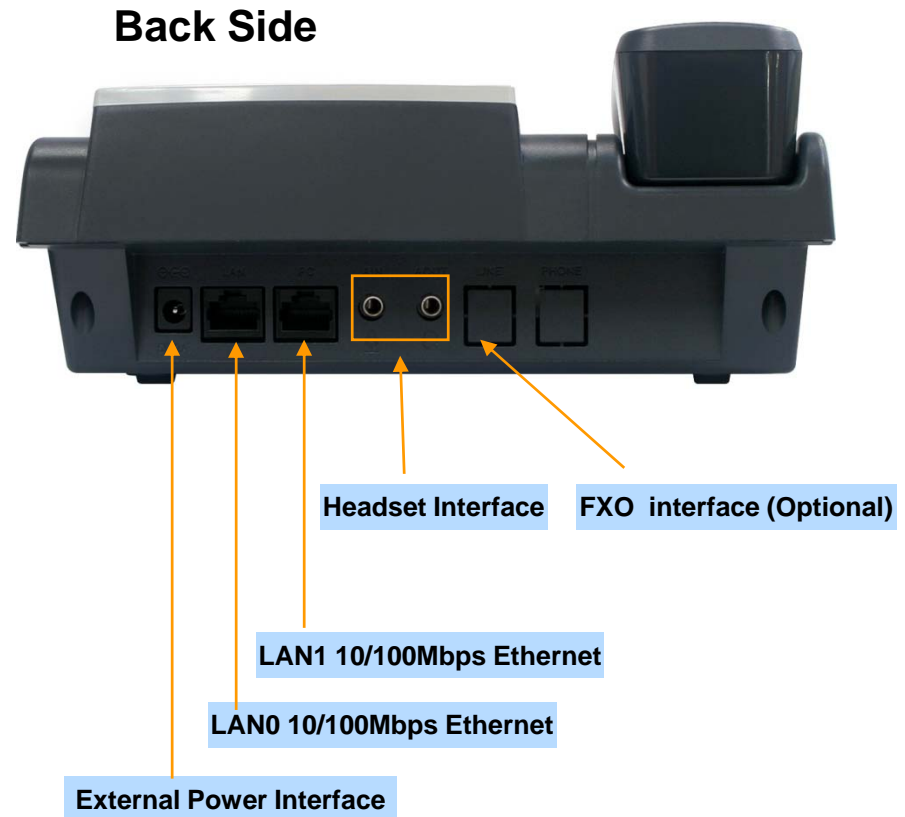
# Hardware Specification

## AP-IP120 SIP Broadcasting Phone

### Hardware Specifications

AP-IP120 SIP Broadcasting Phone	Basic Specifications
CPU	RISC Microprocessor
Ethernet Interface	2-Ports 10/100Mbps Ethernet Interface(RJ-45)
PSTN Backup Port (Optional)	1-Port PSTN Backup Port(RJ-11)
Flash Memory	4Mbyte High-speed Flash Memory
Base Memory	16Mbyte High-speed SDRAM
Power Requirement	External Power Supply Adaptor / VAC 110~220V, 50/60Hz, 10Watt(5V,2A)
	Power over Ethernet (option)
Operating Temperature	0°C ~ 45°C (32 °F ~ 122°F)
Storage Temperature	-40°C ~ 85°C (-40°C ~ 185°F)
Relative Humidity	5% ~ 95% (Non-condensing)
Dimensions	H x W x D ( 70mm x 200mm x 210mm)
Weight (g)	1Kg

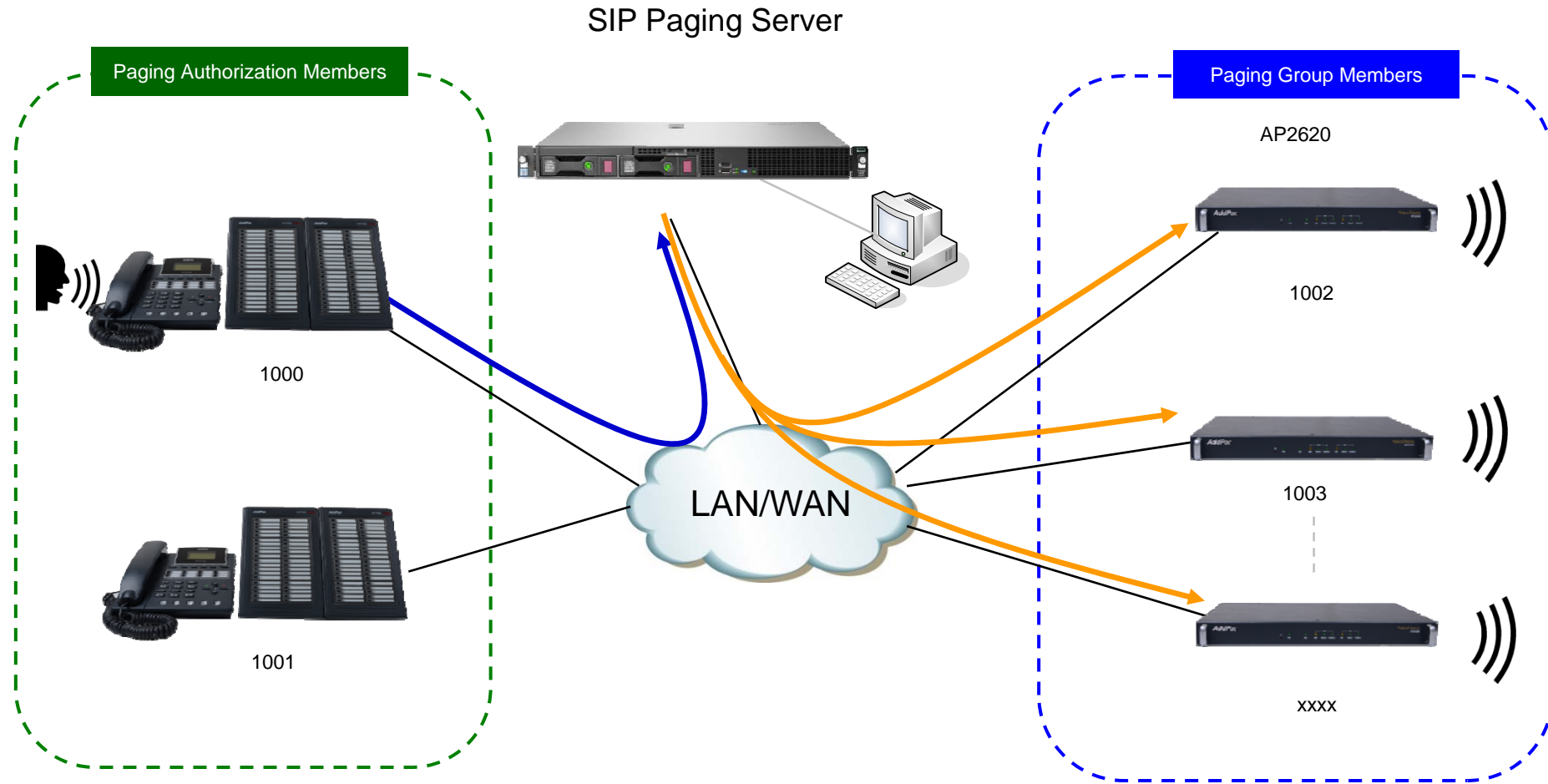
### Network interface Configurations





# AP-IP120 SIP Broadcasting Phone

## Application Area





# AP-BCR5000 Broadcasting Router

# AP-BCR5000

## Real-time IP Audio Broadcasting Router



# Product Overview

## AP-BCR5000 IP Audio Broadcasting Router

- IP based Voice/Audio Broadcasting Router Solution
- High Performance Real-time Voice/Audio Broadcasting Service
- Audio(Voice) Broadcasting, Multicasting, Relay Broadcasting
- Multi-Session Voice/Audio Broadcasting Scheme
- High-performance Embedded System Architecture
- Firmware Upgradeable Architecture
- Multimedia Broadcasting Management Software (Client/Server Architecture)

# Benefits and Features

## AP-BCR5000 IP Audio Broadcasting Router

- RTP & RTSP based Streaming Service
- TCP/UDP Transport Protocol Support
- Two(2) Module Slots for Voice/Audio Streaming
- Embedded Streaming Service Module
- Fault Tolerant and High Reliability Service using Dual Streaming Service CPU Module
- Load Balance and Hot-Swap Service
- Compact Size, Low Noise and Low Power Consumption compare with Commercial Server
- IP based Network Protocol Support

# Hardware Specification

AP-BCR5000 IP Audio Broadcasting Router

High-end  
DSP

- High-end Programmable RISC Hardware Architecture
- Two(2) Module Slots for Audio Streaming Service
- Voice/Audio Streaming Service Module
  - Network Interface
    - Two(2) 10/100/1000Mbps Gigabit Ethernet
    - One(1) Console Port

# Hardware Specification

AP-BCR5000 IP Audio Broadcasting Router

High-end  
RISC

AP-BCR5000 Front View



Audio/Voice Streaming Service Module

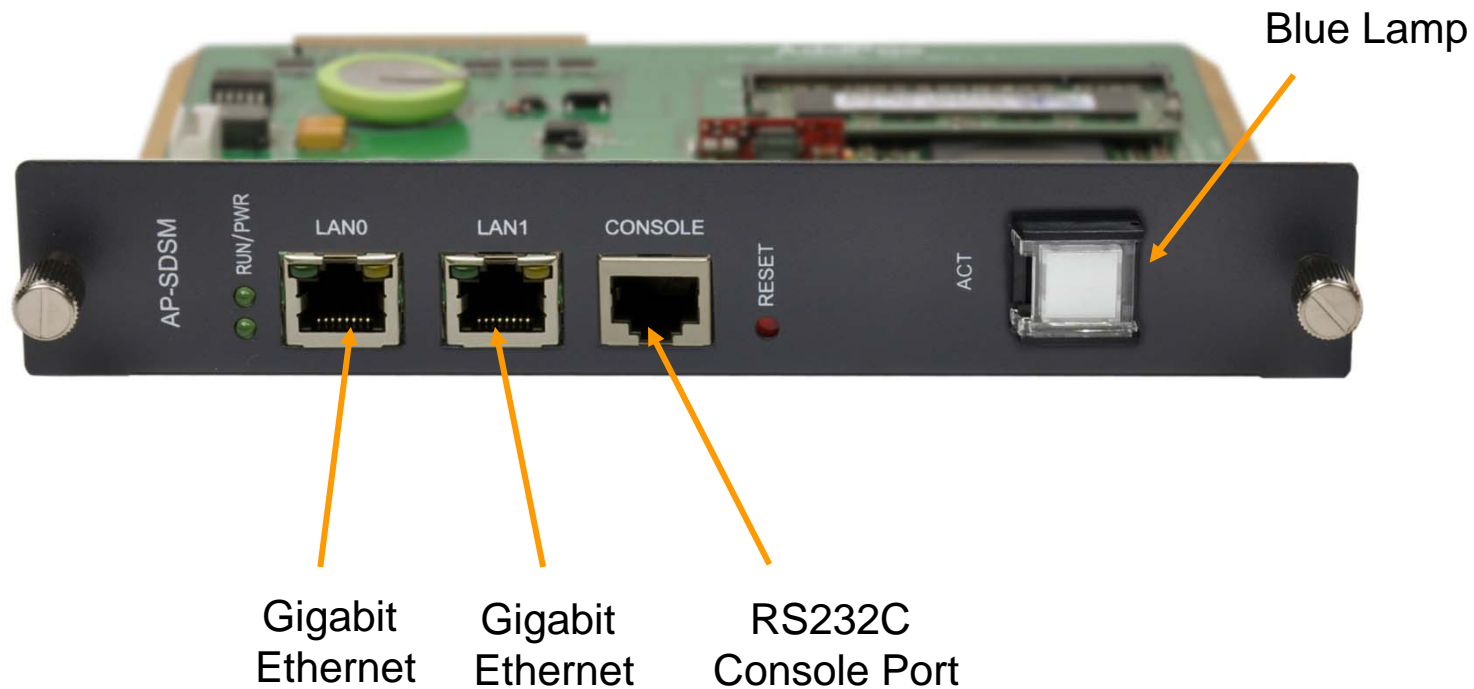


# Hardware Specification

AP-BCR5000 IP Audio Broadcasting Router

High-end  
DSP

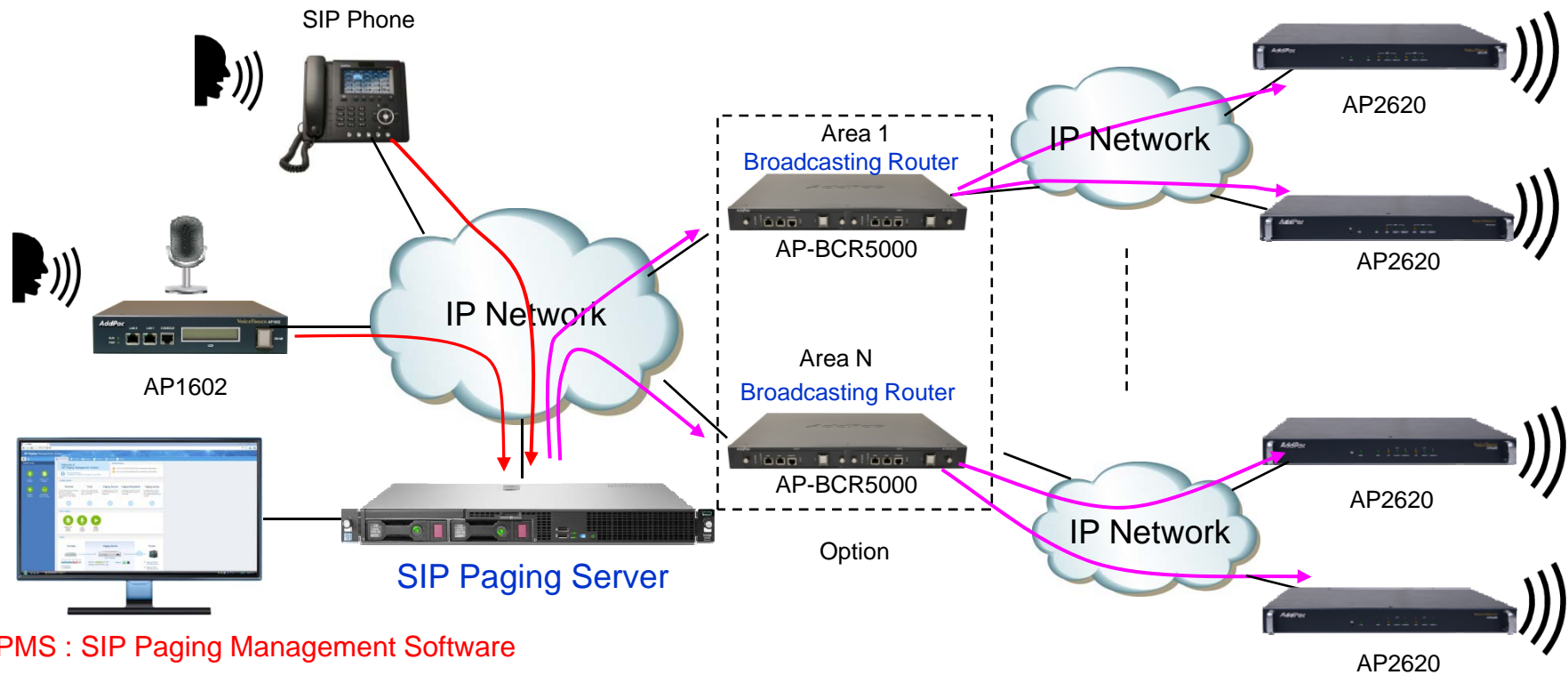
## Audio/Voice Streaming Service Module





# SIP Paging Call Service Network Diagram

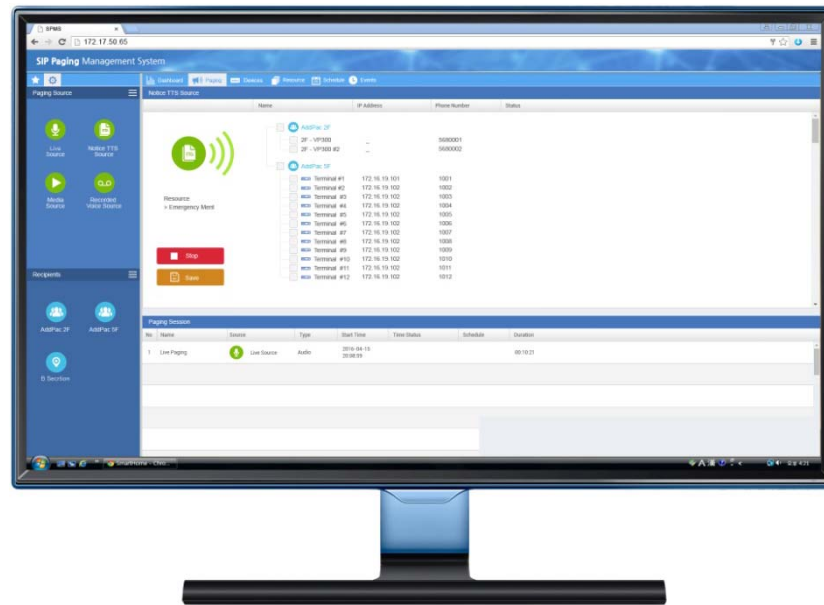
AP-BCR5000 IP Audio Broadcasting Router



AP-SPMS : SIP Paging Management Software

# AP-SPMS

## SIP Paging Management Software



# SIP Paging Management Software Overview

- Paging Device Management
- Paging Group Management
- Access Authority Management
- Real-time and Recorded Paging
- One Click Paging
- Scheduled Paging
- HTTP API Paging
- TTS Paging
- SMS Paging

# SIP Paging Management System GUI Initial Screen

**SIP Paging Management System**

Dashboard | Paging | Devices | Resource | Schedule | Events

Paging Group

- Live Paging
- Notice TTS Paging
- Media Paging
- Recorded Voice Paging

Welcome to SIP Paging Management System

Root (Administrator)  
Last login at June 06 10:37:20 AM (172.16.63.150)

Unread Alarms

- 12:34:10 AP1602 #23(2918) connection terminated.
- 10:12:45 AP1602 #25(2915) connection terminated.

Getting Started

- Terminal**  
A terminal can be a paging source or recipient. The internal terminal is registered to this paging server.
- Trunk**  
A trunk is a call routing path to other SIP servers such as IP-PBX.
- Paging Source**  
A paging source could be a live terminal or a media file.
- Paging Recipients**  
A paging recipients could be a terminal or a set of terminals.
- Paging Group**  
A paging group has a source and recipients with paging settings and number.

Active Paging

- Notice TTS paging (05:12)
- Live Paging (00:38)
- Media Paging (07:12)

Status

Terminals | Paging Server | Trunks

172.17.50.65

Memory 80%  
Storage 60%  
Network

Registered  
Unregistered

External IP-PBX 192.168.100.102  
External Trunk 192.168.100.102

Favorite Paging Group with Source

Active Paging Sessions

# Paging Device Management

**SIP Paging Management System**

Dashboard Paging **Devices** Resource Schedule Events

Terminals Trunks CTI Port

Select item  🔍 🔄 +

No	Name	Type	Phone Number	SIP User Name	IP Address	Status	Modify	Delete
1	Front Door AP-VAC50	Internal	1001	1001	172.16.114.200	● Registered	⚙️	🗑️
2	Secretary IP Phone	Internal	1000	1001	172.16.114.201	● Registered	⚙️	🗑️
3	IP Phone #1	External	2564120	2564120	-	● Unregistered	⚙️	🗑️

# Paging Group Management

The screenshot displays the SIP Paging Management System interface. The top navigation bar includes Dashboard, Paging, Devices, Resource, Schedule, and Events. The main content area is divided into several sections:

- Paging Source:** A sidebar on the left contains icons for Live Source, Notice TTS Source, Media Source, and Recorded Voice Source. A yellow callout box labeled "Paging source" points to this sidebar.
- Notice TTS Source:** A table with columns for Name, IP Address, Phone Number, and Status. It shows a tree structure for "AddPac 2F" and "AddPac 5F".
 

Name	IP Address	Phone Number	Status
<b>AddPac 2F</b>			
2F - VP300	-	5680001	
2F - VP300 #2	-	5680002	
<b>AddPac 5F</b>			
Terminal #1	172.16.19.101	1001	
Terminal #2	172.16.19.102	1002	
Terminal #3	172.16.19.102	1003	
Terminal #4	172.16.19.102	1004	
Terminal #5	172.16.19.102	1005	
Terminal #6	172.16.19.102	1006	
Terminal #7	172.16.19.102	1007	
Terminal #8	172.16.19.102	1008	
Terminal #9	172.16.19.102	1009	
Terminal #10	172.16.19.102	1010	
Terminal #11	172.16.19.102	1011	
Terminal #12	172.16.19.102	1012	

 A yellow callout box labeled "Paging Organizer" points to this tree view.
- Resource:** Shows "Emergency Ment" with "Stop" and "Save" buttons.
- Paging Session:** A table at the bottom showing session details.
 

No	Name	Source	Type	Start Time	Time Status	Schedule	Duration
1	Live Paging	Live Source	Audio	2016-04-15 20:08:59			00:10:21
- Recipients:** A sidebar on the left shows "AddPac 2F", "AddPac 5F", and "B Section". A yellow callout box labeled "Paging Groups" points to this sidebar.

# TTS Paging

**SIP Paging Management System**

Dashboard Paging Devices Resource Schedule Events

Media TTS Recorded

No	Name	Play	Size	Update Time	Delete
1	Announce.wav		3,758 KB	2016-04-14 13:23:12	
2	Secretary IP Phone		12,234 KB	2016-04-15-16:19:23	

# Scheduled Paging Management

The screenshot displays the 'SIP Paging Management System' interface. At the top, there is a navigation bar with tabs for Dashboard, Paging, Devices, Resource, Schedule (selected), and Events. Below this is a sub-menu with 'Calendar' and 'List'. The main area shows a calendar for April 2016, with navigation options for Day, Week, and Month. The calendar grid shows dates from March 27 to April 30. A yellow event is scheduled for April 20th at 2:44 pm.

Sun	Mon	Tue	Wed	Thu	Fri	Sat
Mar 27, 2016	28	29	30	31	Apr 1	2
3	4	5	6	7	8	9
10	11	12	13	14	15	16
17	18	19	20 Today 2:44 pm	22	23	
24	25	26	27	28	29	30



# Events

SIP Paging Management System					
<a href="#">Dashboard</a> <a href="#">Paging</a> <a href="#">Devices</a> <a href="#">Resource</a> <a href="#">Schedule</a> <a href="#">Events</a>					
<input type="text" value="Latest 1day"/> <input type="text" value="Select item"/> <input type="button" value="Search"/> <input type="button" value="Refresh"/>					
ID	Date Time	Level	Category	Message	
1	2016-04-16 09:10:16	Information	Paging	Paging ' Notice TTS Paging ' Started	
2	2016-04-16 14:12:16	Warning	Paging	5F VP300 connection terminated	
3	2016-04-17 11:43:22	Warning	Paging	System Start up	
4	2016-04-19 15:10:59	Critical	Paging	System Shutdown	



# Thank you!

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# Thank you!

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