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CRETONE AUDIO

AP SERIES

Open Architecture Audio Processor

Opening a new chapter for China's audio DSP

Audio Algorithm Symbiosis Platform
Exclusive custom native UI
Diversified OEM/ODM

CATALOGUE

AP Series Introduction

CRETONE AUDIO's AP series audio processor was born in 2023. It's an audio signal management and processing center that integrates domestic high-performance audio hardware and open software.

The open source series of audio processing products aims at technological innovation, high-quality craftsmanship, perfect functions, and ease of use. By continuously expanding the capabilities of its audio solutions, it provides users with audio processing solutions with outstanding functions, excellent sound quality, and stable performance. Whether it is a small conference room, a medium-sized conference room, a large conference room, a multi-function hall or a super-large stadium, there are corresponding solutions to meet their usage needs.

CRETONE AUDIO is headquartered in Bao'an District, Shenzhen, and conducts regional sales and provides technical support in offices across the country. The open source series of products has an award-winning technical team to provide technical support to customers across the country.

01 Application Scenarios

AP series audio processing solutions can manage audio signals for almost any commercial application, including government and corporate conference rooms, multi-functional halls, shopping malls, exhibition centers, hotels, airports, classrooms, churches, and stadiums. Whether it is voice processing or signal routing, or background music and paging, AP series audio solutions can provide extremely high-quality experience.

02 Hardware Equipment

The hardware equipment is designed and manufactured by CRETONE AUDIO and configured and set up by Open Designer software. Whether it is installed in sports stadiums, exhibition centers, hotels, conference rooms and other places; users, integrators and design consultants rely on the super stability of open source hardware. Its AP series hardware is an open architecture system. This means that the DSP operation module can be placed into the system design and connected using a standard drag-and-drop method. Different models of hardware can also be linked together to provide customers with customized audio solutions. The AP(Dante) series is a network processor. The biggest advantage is that it supports 64x64 channel Dante/AES67, and the system can achieve a 128x128 channel super mixing matrix.

The simple software can provide fast debugging. The dedicated backup protocol port realizes accurate fault hot backup, giving the system super system fault protection capabilities. The entire product series also includes physical channel expanders, telephone expansion devices, control panels and GPIO functions, which can provide users with the required functions and applications through the combination of various devices.

03 Software Algorithms

The AP series audio processors have been developing at an astonishing speed by widely absorbing the suggestions of users and integrators to continuously optimize the system algorithms. Within a year since the product's release, multiple functional algorithms have been optimized and multiple software versions have been updated. This has created the current advanced audio signal processing platform. The AP series includes 20+ different types of DSP processing modules. In addition to the various standard processing modules you expect to find (such as EQ, compression, etc.), there are also some advanced DSP algorithm modules that are not provided by competing products. At the same time, according to the needs of the majority of users and integrators, the Lua language is integrated for secondary development by engineers with development capabilities.

04 Transmission and Routing

The AP series network version provides Dante network digital audio transmission, transmitting up to 128 (64x64) channels of audio signals between Dante devices. Dante network digital audio transmission is the future of digital audio transmission. It is a network audio transmission method that is truly compatible with Ethernet and IP networks and can achieve ultra-low latency, high-precision synchronization, accurate sampling, simple setup, and easy use.

05 Device Control

Audio routing and signal processing are only part of the process. How to achieve real-time control of the system is a critical aspect for the installed audio system. For the end user, the wall panel, tablet, mobile phone or computer screen is the audio system. So we developed touch screen controllers and wall controllers to select sources, zone volume and scene switching. Many controllers can be cascaded for comprehensive configuration to meet the most complex control needs.

In addition, the open source AP series audio processors also support linkage with external third-party control systems, such as AMX, Crestron or other brands of control systems and more advanced integrated control management systems.

Overall, the AP series system is very flexible and stable. Once installed and debugged, the system can run stably and safely for several years.

- 64x64Dante
- Dedicated seamless hot backup protocol
- Automixer with integrated ATS algorithm
- Flexible configuration
- Integrate Lua programming language
- DSP Dynamic Modular Display
- Web-Based User Interface
- Dynamic Signal Line
- 128x128 Intelligent Matrix

AP0808N

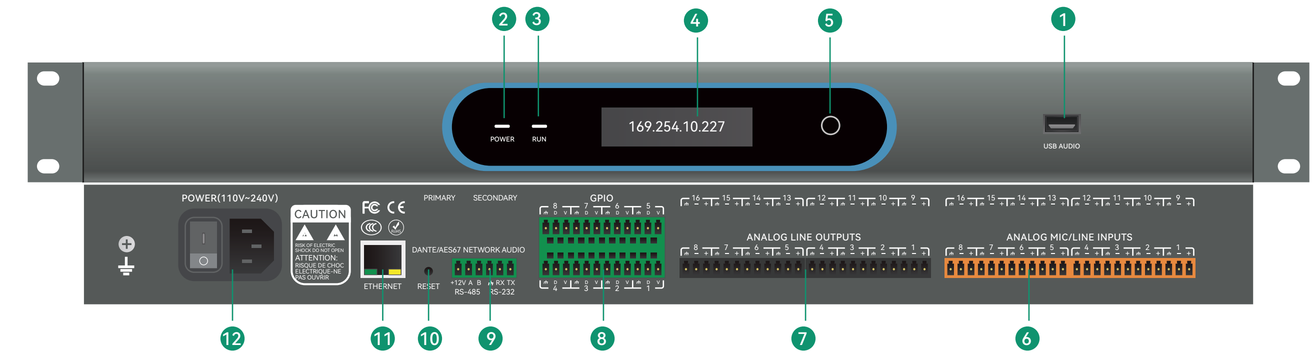
8X8

The AP series audio processor adopts an open modular architecture, supports drag-and-drop programming and intelligent assistance functions, and significantly improves debugging efficiency. Built-in multi-channel AEC, adaptive filtering and other algorithms, with 8-input and 8-output interfaces and 2x2 USB channels, meet the needs of professional scenarios. Innovative three-color status diagnosis, multi-device connection and 999 groups of scene presets, realize cross-device collaboration and seamless scene switching.

Product Features

- Open architecture, drag-and-drop free programming, support module attribute editing and intelligent assistance functions (one-click connection/alignment)
- 8x8 balanced analog input and output, support 8-channel independent AEC echo cancellation (adjustable tail length), built-in multi-channel adaptive filtering/FIR filtering algorithm
- DSP module three-color status diagnosis (signal/on/off), real-time fault location; support multi-device online collaboration, module cross-device replication.
- 999 groups of scene presets, support independent module parameter saving; 2x2 USB channels seamlessly connect to cloud video conferencing.
- Full protocol control (RS485/RS232/TCP IP/UDP), compatible with third-party systems; real-time log management and online firmware upgrade.

Port Introduction



1. USB-B sound card: provides 2x2 audio, used to connect to PC for remote conference or recording and playback of recording and broadcasting software
2. Power indicator: a constant green light indicates normal
3. Device operation status indicator: a constant green light indicates normal
4. OLED display: displays device status information, IP address, overview information, etc
5. Wake-up/navigation button: can switch system dashboard display
6. Analog Mic/line input: 8 balanced analog audio inputs, independent microphone amplifier, phantom power
7. Analog line output: 8 balanced analog outputs, level and mute can be independently controlled by software
8. GPIO: used to connect to control terminal or central control device
9. RS-232/RS-485: used for third-party control of this device or this device outputs RS-232 protocol to control third-party devices
10. RESET: used to restore factory settings
11. Control network: used to connect Open Designer software to program, manage and control the device
12. Power supply: accepts power from detachable IEC cable (100-240VAC, 50-60Hz, 60W maximum)

Specifications

Analog I/O channels	8x8	Equipment noise	≤-95dBu（A-weighted）
Processor	ADI SHARC 21569@1 GHz SIMDx2	Input dynamic range	＞113dB
Input gain	0/3/6/9/12/15/18/21/24/27/30 33/36/39/42/45/48 dB	Output dynamic range	＞113dB
		Input common mode rejection ratio	＞82dB@1kHz，+0dBu）
Phantom power	48Vmax	Output crosstalk	≥93dB@1kHz
Frequency response	20Hz~20kHz（±0.1dB）	Input impedance (balanced connection)	5.4KΩ
Maximum level	+18dBu	Output impedance (balanced connection)	102Ω
Sampling rate	48 kHz	Operating voltage	110-240V AC，50Hz/60Hz
AD\DA bit depth	24Bit	Shipping weight (Net Weight/Gross Weight)	2.32KG/3.23KG
THD+N	≤0.001%（1kHz，+4dBu A-weighted）	Dimensions (W x D x H)	482x262.5x44mm

AP1208N

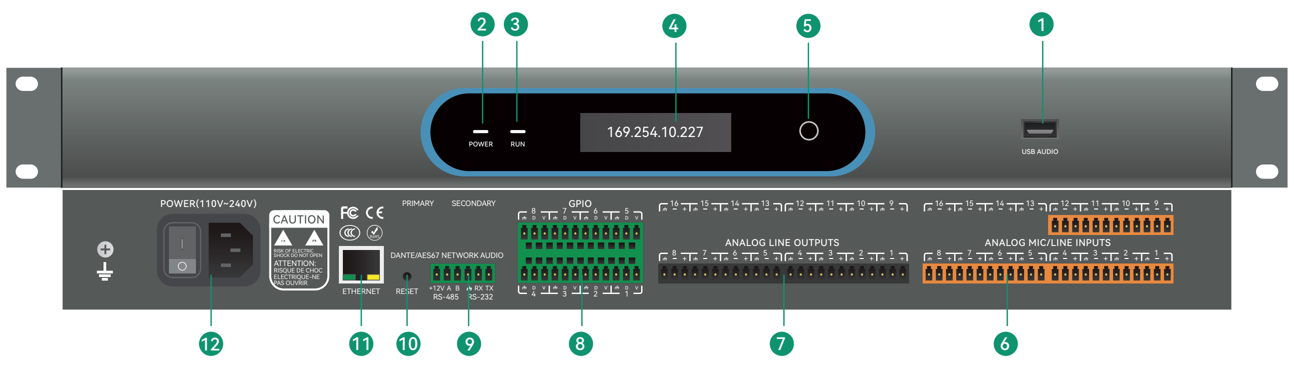
12X8

The AP series audio processor adopts an open modular architecture, supports drag-and-drop programming and intelligent assistance functions, and significantly improves debugging efficiency. Built-in multi-channel AEC, adaptive filtering and other algorithms, with 12-input 8-output interface and 2x2 USB channel, meet the needs of professional scenarios. Innovative three-color status diagnosis, multi-device connection and 999 groups of scene presets, realize cross-device collaboration and seamless scene switching.

Product Features

- Open architecture, drag-and-drop free programming, support module attribute editing and intelligent assistance functions (one-click connection/alignment)
- 12 x8 balanced analog input and output, support 8-channel independent AEC echo cancellation (adjustable tail length), built-in multi-channel adaptive filtering/FR filtering algorithm
- DSP module three-color status diagnosis (signal/on/off), real-time fault location; support multi-device online collaboration, module cross-device replication
- 999 groups of scene presets, support independent module parameter saving; 2x2 USB channel seamlessly connects to cloud video conferencing
- Full protocol control (RS485/RS232/TCP IP/UDP), compatible with third-party systems; real-time log management and online firmware upgrade

Port Introduction



1. USB-B sound card: provides 2x2 audio, used to connect to PC for remote conferencing or recording and playback of recording and broadcasting software
2. Power indicator: a constant green light indicates normal
3. Operation status indicator: a constant green light indicates normal
4. OLED display: displays device status information, IP address, overview information, etc
5. Wake-up/navigation button: this momentary button can switch the system dashboard display
6. Analog Mic/line input: 12 balanced analog audio inputs, independent microphone amplifier, phantom power
7. Analog line output: 8 balanced analog outputs, level and mute can be independently controlled by software
8. GPIO: used to connect to control terminal or central control device
9. RS-232/RS-485: used for third party to control this device or this device to output RS-232 protocol to control third party devices
10. RESET: used to restore factory settings
11. Control network: used to connect Open Designer software to program, manage and control the device
12. Power supply: accepts power supply from detachable IEC cable (100-240VAC, 50-60Hz, 60W maximum)

Specifications

Analog I/O channels	12x8	Equipment noise	≤-95dBu（A-weighted）
Processor	ADI SHARC 21569@1 GHz SIMDx2	Input dynamic range	＞113dB
Input gain	0/3/6/9/12/15/18/21/24/27/30 33/36/39/42/45/48 dB	Output dynamic range	＞113dB
		Input common mode rejection ratio	＞82dB@1kHz，+0dBu）
Phantom power	48Vmax	Output crosstalk	≥93dB@1kHz
Frequency response	20Hz~20kHz（±0.1dB）	Input impedance (balanced connection)	5.4KΩ
Maximum level	+18dBu	Output impedance (balanced connection)	102Ω
Sampling rate	48 kHz	Operating voltage	110-240V AC，50Hz/60Hz
AD\DA bit depth	24Bit	Shipping weight (Net Weight/Gross Weight)	2.41KG/3.32KG
THD+N	≤0.001%（1kHz，+4dBu A-weighted）	Dimensions (W x D x H)	482x262.5x44mm

AP1616N

16X16

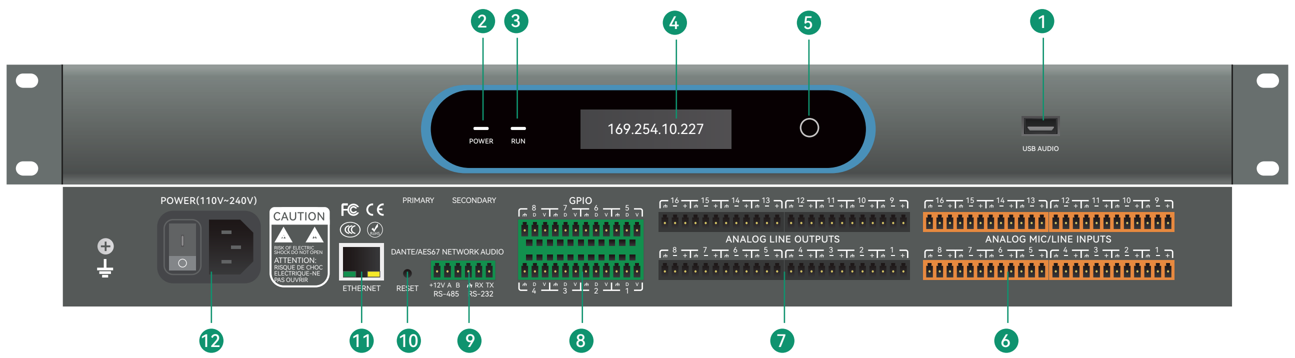
The AP series audio processor adopts an open modular architecture, supports drag-and-drop programming and intelligent assistance functions, and significantly improves debugging efficiency. Built-in multi-channel AEC, adaptive filtering and other algorithms, with 16-input 16-output interface and 2x2 USB channel, meet the needs of professional scenarios. Innovative three-color status diagnosis, multi-device connection and 999 groups of scene presets, realize cross-device collaboration and seamless scene switching

Product Features

- Open architecture, drag-and-drop free programming, support module attribute editing and intelligent assistance functions (one-click connection/alignment)
- 16 x16 balanced analog input and output, support 8-channel independent AEC echo cancellation (adjustable tail length), built-in multi-channel adaptive filtering/FIR filtering algorithm
- DSP module three-color status diagnosis (signal/on/off), real-time fault location; support multi-device online collaboration, module cross-device replication
- 999 groups of scene presets, support independent module parameter saving; 2x2USB channel seamlessly connects to cloud video conferencing
- Full protocol control (RS485/RS232/TCPIP/UDP), compatible with third-party systems; real-time log management and online firmware upgrade

AP SERIES Open Architecture Audio Processor
CRETONE AUDIO

Port Introduction



1. USB-B sound card: provides 2x2 audio, used to connect to a PC for remote conferencing or recording and playback of recording and broadcasting software
2. Power indicator: a steady green light indicates normal
3. Device operation status indicator: a steady green light indicates normal
4. OLED display: displays device status information, IP address, overview information, etc
5. Wake-up/navigation button: this momentary button can switch the system dashboard display
6. Analog Mic/line input: 16 balanced analog audio inputs, independent microphone amplifiers, phantom power
7. Analog line output: 16 balanced analog outputs, level and mute can be independently controlled by software
8. GPIO: used to connect to a control terminal or central control device
9. RS-232/RS-485: used for third-party control of this device or this device outputs RS-232 protocol to control third-party devices
10. RESET: used to restore factory settings
11. Control network: used to connect to Open Designer software to program, manage and control the device
12. Power supply: accepts power from a detachable IEC cable (100-240VAC, 50-60Hz, 60W maximum)

Specifications

Analog I/O channels	16x16	Equipment noise	≤-95dBu（A-weighted）
Processor	ADI SHARC 21569@1 GHz SIMDx2	Input dynamic range	＞113dB
Input gain	0/3/6/9/12/15/18/21/24/27/30 33/36/39/42/45/48 dB	Output dynamic range	＞113dB
		Input common mode rejection ratio	＞82dB@1kHz，+0dBu)
Phantom power	48Vmax	Output crosstalk	≥93dB@1kHz
Frequency response	20Hz~20kHz（±0.1dB）	Input impedance (balanced connection)	5.4KΩ
Maximum level	+18dBu	Output impedance (balanced connection)	102Ω
Sampling rate	48 kHz	Operating voltage	110~240V AC，50Hz/60Hz
AD\DA bit depth	24Bit	Shipping weight (Net Weight/Gross Weight)	2.43KG/3.34KG
THD+N	≤0.001%（1kHz，+4dBu A-weighted）	Dimensions (W x D x H)	482x262.5x44mm

AP0808ND

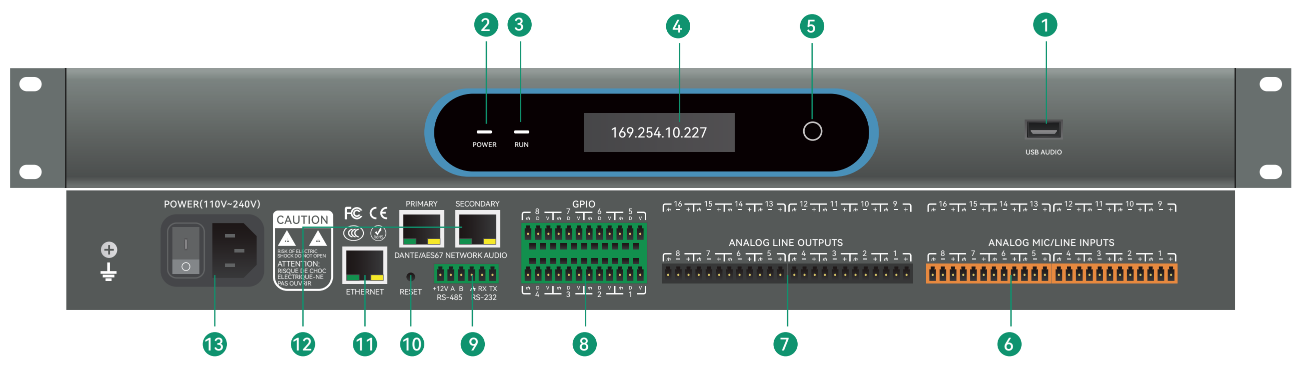
8X8

The AP series audio processor adopts an open modular architecture, supports drag-and-drop programming and intelligent assistance functions, and significantly improves debugging efficiency. Built-in multi-channel AEC, adaptive filtering and other algorithms, with 8-input 8-output interfaces and 2x2 USB channels, meet the needs of professional scenarios. Innovative three-color status diagnosis, multi-device connection and 999 groups of scene presets, realize cross-device collaboration and seamless scene switching.

Product Features

- Open architecture, drag-and-drop free programming, support module attribute editing and intelligent assistance functions (one-click connection/alignment)
- 8x8 balanced analog input and output, support 8-channel independent AEC echo cancellation (adjustable tail length), built-in multi-channel adaptive filtering/FIR filtering algorithm
- DSP module three-color status diagnosis (signal/on/off), real-time fault location; support multi-device online collaboration, module cross-device replication
- 999 groups of scene presets, support independent module parameter saving; 2x2 USB channels seamlessly connect to cloud video conferencing
- Full protocol control (RS485/RS232/TCP IP/UDP), compatible with third-party systems; real-time log management and online firmware upgrade

Port Introduction



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4. OLED display: displays device status information, IP address, overview information, etc
5. Wake-up/navigation button: switches system dashboard display
6. Analog Mic/line input: 8 balanced analog audio inputs, independent microphone amplifier, phantom power
7. Analog line output: 8 balanced analog outputs, level and mute can be independently controlled by software
8. GPIO: used to connect to control terminal or central control device
9. RS-232/RS-485: used by a third party to control this device or this device outputs RS-232 protocol to control third-party devices
10. RESET: used to restore factory settings
11. Control network: used to connect Open Designer software to program, manage and control the device
12. Dante port: 1000 Base-T Ethernet port, providing 128 (64x64) Dante network audio channels
13. Power supply: accepts power from a detachable IEC cable (100-240VAC, 50-60Hz. 60W maximum)

Specifications

Analog I/O channels	8x8	Equipment noise	≤-95dBu（A-weighted）
Dante I/O channels	64x64	Input dynamic range	＞113dB
Processor	ADI SHARC 21569@1 GHz SIMDx2	Output dynamic range	＞113dB
Input gain	0/3/6/9/12/15/18/21/24/27/30 33/36/39/42/45/48 dB	Input common mode rejection ratio	＞82dB@1kHz，+0dBu）
		Output crosstalk	≥93dB@1kHz
Phantom power	48Vmax	Input impedance (balanced connection)	5.4KΩ
Frequency response	20Hz~20kHz（±0.1dB）	Output impedance (balanced connection)	102Ω
Maximum level	+18dBu	Operating voltage	110-240V AC，50Hz/60Hz
Sampling rate	48 kHz	Shipping weight (Net Weight/Gross Weight)	2.38KG/3.29KG
AD\DA bit depth	24Bit	Dimensions (W x D x H)	482x262.5x44mm
THD+N	≤0.001%（1kHz，+4dBu A-weighted）		

AP1208ND

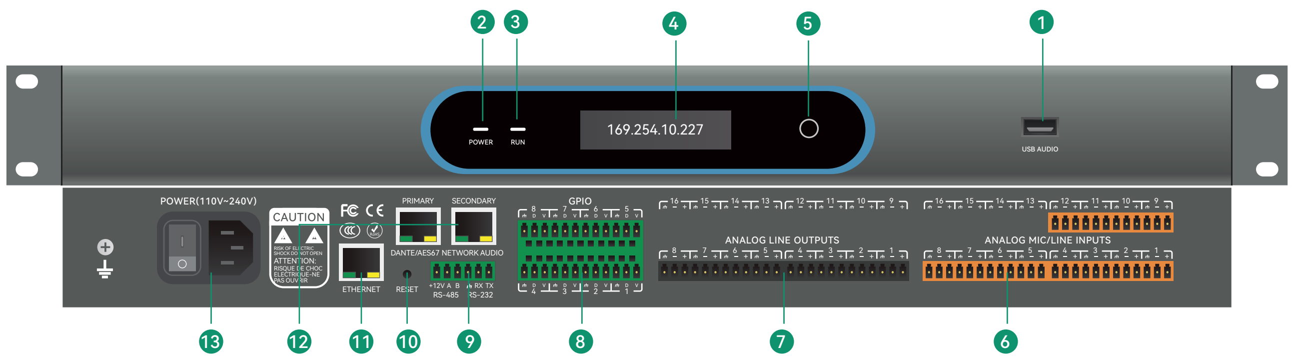
12X8

The AP series audio processor adopts an open modular architecture, supports drag-and-drop programming and intelligent assistance functions, and significantly improves debugging efficiency. Built-in multi-channel AEC, adaptive filtering and other algorithms, with 12-input 8-output interfaces and 2x2 USB channels, meet the needs of professional scenarios. Innovative three-color status diagnosis, multi-device connection and 999 sets of scene presets, realize cross-device collaboration and seamless scene switching.

Product Features

- Open architecture, drag-and-drop free programming, support module attribute editing and intelligent assistance functions (one-click connection/alignment)
- 12 x8 balanced analog input and output, support 8-channel independent AEC echo cancellation (adjustable tail length), built-in multi-channel adaptive filtering/FIR filtering algorithm
- DSP module three-color status diagnosis (signal/on/off), real-time fault location; support multi-device online collaboration, module cross-device replication
- 999 sets of scene presets, support independent module parameter saving; 2x2 USB channel docking-free cloud video conferencing
- Full protocol control (RS485/RS232/TCP IP/UDP), compatible with third-party systems; real-time log management and online firmware upgrade

Port Introduction



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7. Analog line output: 8 balanced analog outputs, level and mute can be independently controlled by software
8. GPIO: used to connect to control terminal or central control device
9. RS-232/RS-485: used for third-party control of this device or this device outputs RS-232 protocol to control third-party devices
10. RESET: used to restore factory settings
11. Control network: used to connect Open Designer software to program, manage and control the device
12. Dante port: 1000 Base-T Ethernet port, providing 128 (64x64) Dante network audio channels
13. Power supply: accepts power from a detachable IEC cable (100-240VAC, 50-60Hz. 60W maximum)

Specifications

Analog I/O channels	12x8	Equipment noise	≤-95dBu（A-weighted)
Dante I/O channels	64x64	Input dynamic range	＞113dB
Processor	ADI SHARC 21569@1 GHz SIMDx2	Output dynamic range	＞113dB
Input gain	0/3/6/9/12/15/18/21/24/27/30 33/36/39/42/45/48 dB	Input common mode rejection ratio	＞82dB@1kHz，+0dBu)
		Output crosstalk	≥93dB@1kHz
Phantom power	48Vmax	Input impedance (balanced connection)	5.4KΩ
Frequency response	20Hz~20kHz（±0.1dB)	Output impedance (balanced connection)	102Ω
Maximum level	+18dBu	Operating voltage	110-240V AC，50Hz/60Hz
Sampling rate	48 kHz	Shipping weight (Net Weight/Gross Weight)	2.47KG/3.38KG
AD\DA bit depth	24Bit	Dimensions (W x D x H)	482x262.5x44mm
THD+N	≤0.001%（1kHz，+4dBu A-weighted)		

AP1616ND

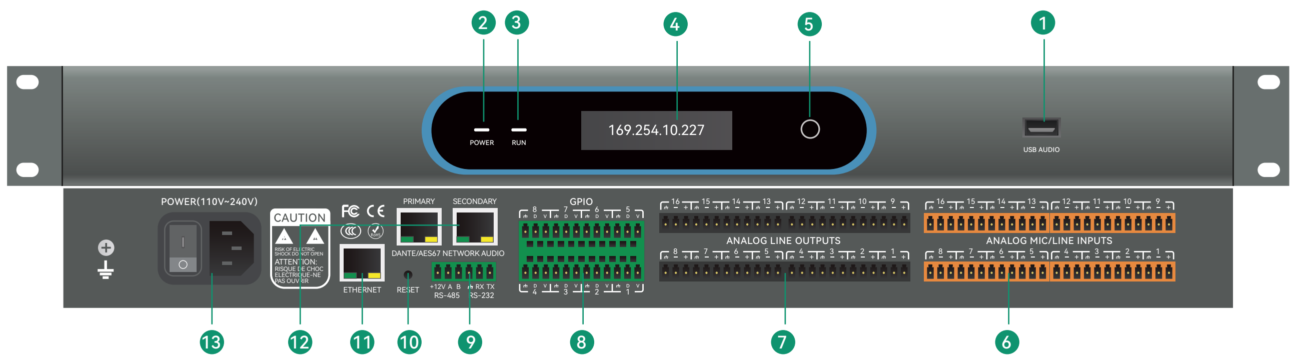
16X16

The AP series audio processor adopts an open modular architecture, supports drag-and-drop programming and intelligent assistance functions, and significantly improves debugging efficiency. Built-in multi-channel AEC, adaptive filtering and other algorithms, with 16-input 16-output interface and 2x2 USB channel, meet the needs of professional scenarios. Innovative three-color status diagnosis, multi-device connection and 999 groups of scene presets, realize cross-device collaboration and seamless scene switching.

Product Features

- Open architecture, drag-and-drop free programming, support module attribute editing and intelligent assistance functions (one-click connection/alignment)
- 16 x16 balanced analog input and output, support 8-channel independent AEC echo cancellation (adjustable tail length), built-in multi-channel adaptive filtering/FIR filtering algorithm
- DSP module three-color status diagnosis (signal/on/off), real-time fault location; support multi-device online collaboration, module cross-device replication
- 999 groups of scene presets, support independent module parameter saving; 2x2 USB channel seamlessly connects to cloud video conferencing
- Full protocol control (RS485/RS232/TCP IP/UDP), compatible with third-party systems; real-time log management and online firmware upgrade

Port Introduction



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3. Device operation status indicator: a constant green light indicates normal
4. OLED display: displays device status information, IP address, overview information, etc.
5. Wake-up/navigation button: can switch the system dashboard display
6. Analog Mic/line input: 16 balanced analog audio inputs, independent microphone amplifier, phantom power
7. Analog line output: 16 balanced analog outputs, level and mute can be independently controlled by software
8. GPIO: used to connect to control terminal or central control device
9. RS-232/RS-485: used for third-party control of this device or this device outputs RS-232 protocol to control third-party devices
10. RESET: used to restore factory settings
11. Control network: used to connect Open Designer software to program, manage and control the device
12. Dante port: redundant 1000 Base-T Ethernet port, providing 128 (64x64) Dante network audio channels
13. Power supply: accepts power from a detachable IEC cable (100-240VAC, 50-60Hz, 60W maximum)

Specifications

Analog I/O channels	16x16	Equipment noise	≤-95dBu（A-weighted）
Dante I/O channels	64x64	Input dynamic range	＞113dB
Processor	ADI SHARC 21569@1 GHz SIMDx2	Output dynamic range	＞113dB
Input gain	0/3/6/9/12/15/18/21/24/27/30 33/36/39/42/45/48 dB	Input common mode rejection ratio	＞82dB@1kHz，+0dBu）
		Output crosstalk	≥93dB@1kHz
Phantom power	48Vmax	Input impedance (balanced connection)	5.4KΩ
Frequency response	20Hz~20kHz（±0.1dB）	Output impedance (balanced connection)	102Ω
Maximum level	+18dBu	Operating voltage	110-240V AC，50Hz/60Hz
Sampling rate	48 kHz	Shipping weight (Net Weight/Gross Weight)	2.5KG/3.41KG
AD\DA bit depth	24Bit	Dimensions (W x D x H)	482x262.5x44mm
THD+N	≤0.001%（1kHz，+4dBu A-weighted）		

AP1208NX

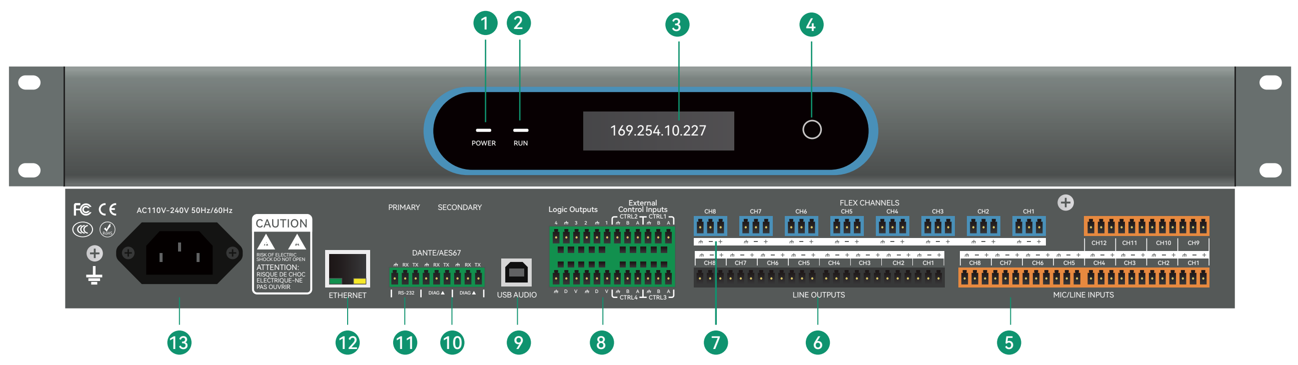
12X8

The AP series audio processor adopts an open modular architecture, supports drag-and-drop programming and intelligent assistance functions, and significantly improves debugging efficiency. Built-in multi-channel AEC, adaptive filtering and other algorithms, with 12-input 8-output interface and 2x2 USB channel, meet the needs of professional scenarios. Innovative three-color status diagnosis, multi-device connection and 999 groups of scene presets, realize cross-device collaboration and seamless scene switching.

Product Features

- Open architecture, drag-and-drop free programming, support module attribute editing and intelligent assistance functions (one-click connection/alignment)
- 12 x8 balanced analog input and output, support 8-channel independent AEC echo cancellation (adjustable tail length), built-in multi-channel adaptive filtering/FIR filtering algorithm
- DSP module three-color status diagnosis (signal/on/off), real-time fault location; support multi-device online collaboration, module cross-device replication
- 999 groups of scene presets, support independent module parameter saving; 2x2 USB channel docking-free cloud video conferencing
- Compatible with third-party systems

Port Introduction



1. Power indicator: Normal if it is always white
2. Device operation status indicator: Slow flashing means it is running normally
3. OLED display: Displays device status information, IP address, overview information, etc.
4. Wake-up/navigation button: switch the system dashboard display
5. Analog Mic/line input: 12 balanced analog audio inputs, independent microphone amplifier, phantom power
6. Analog line output: 8 balanced analog outputs, level and mute can be independently controlled by software
7. Custom channels: 8 customizable input and output port channels, which can be set in the software
8. GPIO/Logic: Used to connect to the control terminal or central control device
9. USB-B sound card: Provides 2x2 audio, used to connect to PC for remote conferencing or recording and playback of recording and broadcasting software
10. DIAG: This function is used for data detection during hot backup of dual machines. The actual port uses the RS232 protocol, and the backup device sends a heartbeat packet to the main device
11. RS-232: Used for third-party control of this device or this device outputs RS-232 protocol to control third-party devices
12. Control network: Used to connect to Open Designer software to program, manage and control the device
13. Power supply: accepts power from a detachable IEC cable (100-240VAC, 50-60Hz, 60W maximum)

Specifications

Analog I/O channels	12x8	Equipment noise	≤-95dBu（A-weighted)
Custom analog variable channel	8	Input dynamic range	＞113dB
Processor	ADI SHARC 21569@1 GHz SIMDx2	Output dynamic range	＞113dB
Input gain	0/3/6/9/12/15/18/21/24/27/30 33/36/39/42/45/48 dB	Input common mode rejection ratio	＞82dB@1kHz，+0dBu)
		Output crosstalk	≥93dB@1kHz
Phantom power	48Vmax	Input impedance (balanced connection)	5.4KΩ
Frequency response	20Hz~20kHz(±0.1dB)	Output impedance (balanced connection)	102Ω
Maximum level	+18dBu	Operating voltage	110-240V AC，50Hz/60Hz
Sampling rate	48 kHz	Shipping weight (Net Weight/Gross Weight)	3.17KG/4.11KG
AD\DA bit depth	24Bit	Dimensions (W x D x H)	482x262.5x44mm
THD+N	≤0.001%(1kHz，+4dBu A-weighted)		

AP1208NXD

12X8

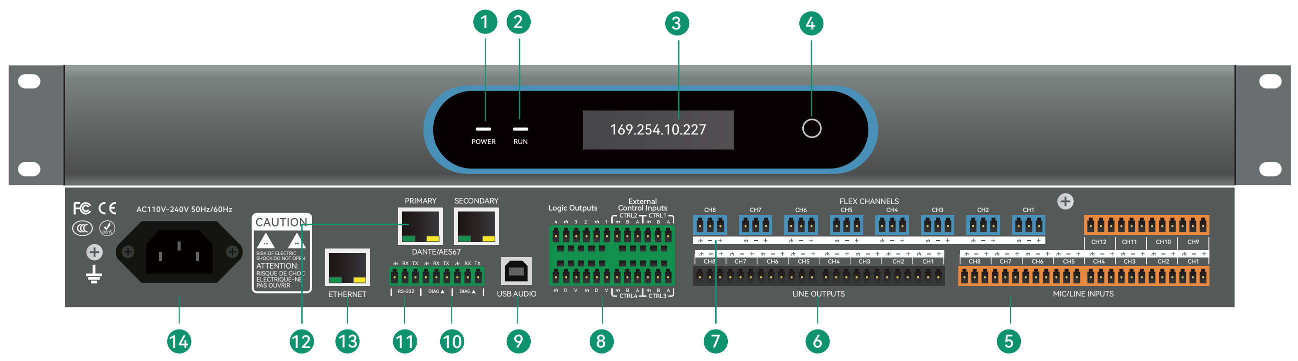
The AP series audio processor adopts an open modular architecture, supports drag-and-drop programming and intelligent assistance functions, and significantly improves debugging efficiency. Built-in multi-channel AEC, adaptive filtering and other algorithms, with 12-input and 8-output interfaces and 2x2 USB channels, meet the needs of professional scenarios. Innovative three-color status diagnosis, multi-device connection and 999 groups of scene presets, realize cross-device collaboration and seamless scene switching.

Product Features

- Open architecture, drag-and-drop free programming, support module attribute editing and intelligent assistance functions (one-click connection/alignment)
- 12 x8 balanced analog input and output, support 8-channel independent AEC echo cancellation (adjustable tail length), built-in multi-channel adaptive filtering/FIR filtering algorithm
- 8-way custom variable analog channels, which can be set as input or output channels according to actual usage requirements
- Dual-host hot backup function, through the detection of heartbeat packet mechanism to mutually judge the status of the master and standby devices, seamless switching
- DSP module three-color status diagnosis (signal/on/off), real-time fault location; support multi-device online collaboration, module cross-device replication
- 999 sets of scene presets, support independent module parameter saving; 2x2 USB channels seamlessly connect to cloud video conferencing
- Full protocol control (RS485/RS232/TCPIP/UDP), compatible with third-party systems; real-time log management and online firmware upgrade

AP SERIES Open Architecture Audio Processor
CRETONE AUDIO

Port Introduction



1. Power indicator: Normal when the white light is on
2. Device operation status indicator: Slow flashing means normal operation
3. OLED display: Displays device status information, IP address, overview information, etc.
4. Wake-up/navigation button: switch the system dashboard display
5. Analog Mic/line input: 12 balanced analog audio inputs, independent microphone amplifier, phantom power
6. Analog line output: 8 balanced analog outputs, level and mute can be independently controlled by software
7. Custom channels: 8 customizable input and output port channels, which can be set in the software
8. GPIO/Logic: Used to connect to the control terminal or central control device
9. USB-B sound card: Provides 2x2 audio, used to connect to the PC for remote conferencing or recording and playback of recording and broadcasting software
10. DIAG: This function is used for data detection during dual-machine hot backup. The actual port uses the RS232 protocol, and the backup device sends a heartbeat packet to the main device
11. RS-232: Used for third-party control of this device or this device outputs RS-232 protocol to control third-party devices
12. Dante port: 1000 Base-T Ethernet port, providing 128 (64x64) Dante network audio channels
13. Control network: used to connect Open Designer software to program, manage and control the device
14. Power supply: accepts power from a detachable IEC power supply (100-240VAC,50-60Hz.60W maximum)

Specifications

Analog I/O channels	12x8	THD+N	≤0.001% (1kHz，+4dBu A-weighted)
Dante I/O channel	64x64	Equipment noise	≤-95dBu(A-weighted)
Custom analog variable channel	8	Input dynamic range	＞113dB
Processor	ADI SHARC 21569@1 GHz SIMDx2	Output dynamic range	＞113dB
Input gain	0/3/6/9/12/15/18/21/24/27/30 33/36/39/42/45/48 dB	Input common mode rejection ratio	＞82dB@1kHz，+0dBu
		Output crosstalk	≥93dB@1kHz
Phantom power	48Vmax	Input impedance (balanced connection)	5.4KΩ
Frequency response	20Hz~20kHz（±0.1dB）	Output impedance (balanced connection)	102Ω
Maximum level	+18dBu	Operating voltage	110-240V AC，50Hz/60Hz
Sampling rate	48 kHz	Shipping weight (Net Weight/Gross Weight)	3.23KG/4.14KG
AD\DA bit depth	24Bit	Dimensions (W x D x H)	482x262.5x44mm

AP SERIES Open Architecture Audio Processor
CRETONE AUDIO

Control Panel

Control panel series, tcustomized for your processor!

We know that every detail matters to the user experience. Therefore, we have specially equipped the NLP series with a variety of advanced and easy-to-use control panels to meet your diverse operating needs. This series covers the 8-button control panel NCP-1, the 1.13-inch control panel NCP-3, the 4-inch touch screen NCP-4, the 8-inch Android touch screen NCP-8, and the 4-inch touch serial screen CCP-4 with serial communication function, providing you with a comprehensive choice.

These control panels are not only a perfect match for the NLP series, but can also be customized according to your specific needs. Choose our control panel series to make your processor better and easier to operate.

8-button control panel

NCP-1

Type	Wall-mounted
Programmable buttons	8
Knob	1
Control protocol	UDP
Function	User-defined
Network interface	1, 100Mbps RJ45, communication distance 100 meters
Power supply method	PoE
Appearance size (HxW)	156X96mm
Bottom box size (WxDxH)	152X84X50mm
Material color	Silver matte

1.13″ screen knob control panel

NCP-3

Type	Wall-mounted
Display	1.13″ OLED display
Knob	1 knob, supports up to 32 menu functions
Control protocol	UDP
Function	Supports user customization
Network interface	1, 100Mbps RJ45, communication distance 100 meters
Power supply method	PoE
Appearance size (HxW)	86X86mm
Bottom box size (WxDxH)	82X82X31mm
Material color	Silver matte

4″TFT touch screen

NCP-4

Resolution	480x480
Operating system	Android 10
Processor	Quad-core ARMCortexA53CPU1.6GHz, 64-bit
Control protocol	UDP
Function	Supports user-defined operation interface
Network interface	1, 100Mbps RJ45, communication distance 100 meters
Power supply method	PoE & +12VDC
Power consumption	<8W
Product size (WxH)	87.2X87.2mm
Opening size (WXDXH)	86.2 x40.5x 86.2mm

8″ Android touch screen

NCP-8

Resolution	1280x800
Operating system	Android 8.1 Memory: 2GB Storage: 16GB
Screen type	IPS
Size	8 inches
Control protocol	UDP
Function	Supports customizable user interface
Main chip	Quad-core cortex-A17, RockChip RK3288
Contrast	800:1
Power input	DC 12V 1.5A &PoE IEEE802.3AT
Product size	212.2x147.6x31

4″ serial port screen

CCP-4

Resolution	480x480
Control protocol	RS-485
Function	Built-in operation interface
Signal line length	100 meters
Connection port	4pin Phoenix terminal
Power supply method	+12VDC
Power consumption	＜100mW
Product size (WxH)	88X88mm
Opening size (WXDXH)	82X18.1X78mm
Color	Gray