



AES67

Network Audio Ecosystem

A glowing blue AES67 chip is the central focus, resting on a dark, textured surface that resembles a circuit board. The chip is illuminated from below, casting a bright blue glow. The text 'AES67' is printed in a bold, white, sans-serif font on the top surface of the chip. The background is a dark, grid-like pattern with glowing blue lines and shapes, suggesting a complex digital or network environment.

AES67

R&D Background

Currently, there are numerous types of audio protocols on the market, but their development concepts fail to effectively respond to the evolving trends of platformization, digitalization, networking, professionalization, and ecologicalization, let alone achieving efficient docking and centralized control. Closed and semi-closed proprietary protocols have emerged in large numbers in the market, but due to their limited service scope, user satisfaction remains low. Meanwhile, integrated chips spanning audio and video face technical barriers in protocol docking, and their prices remain high.

To overcome the long-standing technical challenges in the industry, such as incompatible audio protocols, difficult network adaptation, lack of ecological deployment, low interface openness, and dependence on professionals, and to quickly penetrate both existing and growing markets, Soundking relies on the industry-standard RAVENNA/AES67 as its foundation to drive innovation and upgrading of system products. With core technologies such as modular channel scalability, autonomous DSP redundancy backup algorithm mechanism, voiceprint large model I/O separation, and IoT visualization as its mission for industry development, Soundking also ensures good compatibility with other mainstream audio and video protocols worldwide. Through integration with IoT, it can achieve technical control of the overall audio ecosystem, opening up broader market space for the industry.

This series of products is widely applicable to various settings such as government agencies, radio and television stations, transportation hubs, sports venues, theaters, military units, schools, and theme parks.

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AES67 Audio Workstation

Naming terminology: S stands for workstation management, I for audio ecosystem, PT for extended version, P for flagship version, U for redundant flagship version, and the number represents the number of channels

The Soundking audio workstation is a professional large-scale network audio processing platform built on the RAVENNA/AES67 standard. High-performance hardware and operating system integrate core technologies such as professional network audio high-speed transmission and reception, and DSP 6A+ processing. It supports a single network cable and a single motherboard, and supports 4096-level FIR (filtering) and 192kHz high sampling rate. AES67 local and network audio control software (SK AES67 Controller and VSC virtual sound card) enables more flexible project coordination. This series of products adopts redundant backup design for core interfaces such as power supply and DA network audio, enhancing the reliability of large projects and suitable for large conferences, airports and railways, radio and television, theater sound reinforcement, and other scenarios.

DSP function can be customized freely

- DSP 6A+ processing, supporting "unlimited" combination expansion.

Ultra-large AES67 audio matrix

- 1 network cable, 1 motherboard (non-spliced) for transmission. It can handle up to 1536x1520 channels.

Ultra-large channel FIR processor

- 4096-level, 192-channel FIR processing simultaneously, with an ultra-high sampling rate of 192kHz.

Ultra-large channel audio processing workstation

- 1536x1520 channels with full audio mixing, each channel equipped with DSP functions such as EQ, delay, high and low pass filters, fully enabled. Protocol compatibility standard: RAVENNA_AES67.
- Digital and Analog interface expansion: 32 x 32, GPIO, HDMI.

Compatible with mainstream chips and expansion modules in the industry

- Self-developed high-performance audio operating system, AES67 local and network audio control software (SK AES67 Controller and VSC virtual sound card), with optional redundant backup expansion design for more flexible project coordination.
- Customizable expansion, with a thousand workstations and a thousand different configs.



Audio Workstation

SI-256PT



Features

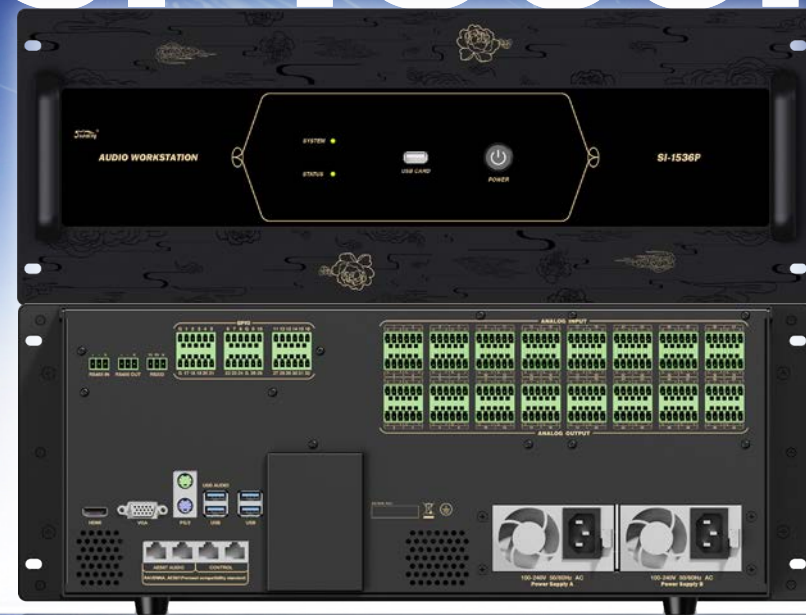
1. DSP functions can be freely customized for each channel.
2. Up to 256x256 channels of network audio.
3. Supports up to 180 4096-tap FIRs.
4. It can be equipped with up to 32x32 analog channels (with the ability to expand to 32 analog microphone channels).
5. GPIO can be optionally equipped with up to 16 groups (32, with customizable input and output).

Specifications

Power supply	100-240VAC 50/60HZ
Rack size	4U
Front panel	Power switch: 1; System indicator light: 1; Status indicator light: 1; USB CARD audio interface: 1
Rear panel	AC power interface: 2; RJ45 2.5G network audio interface: 2; RJ45 network control interface: 2; USB Audio interface: 1; USB debug port: 3; HDMI interface: 1; VGA interface: 1; PS/2 mouse and keyboard interface: 1; RS232 interface: 1; RS485 IN: 1; RS485 OUT: 1; GPIO: 16 groups (32, customizable input/output); Analog input interface: 32 (customizable); Analog output interface: 32 (customizable)
USB Audio channel	8x8
Protocol compatibility standard	RAVENNA_AES67
SND Network channel input	256@48k
SND Network channel output	256@48k
Maximum sampling rate	192kHz
Number of automatic mixing groups	16
Max channel count per group for automatic mixing	64
Remix	Fully mixed audio, supporting independent GainSharing for output channels
DSP function supports customization	Echo cancellation (up to 200 channels), FIR (4096-tap, up to 180), feedback suppression, delay unit, expander, Compressor, limiter, EQ, high-pass filter, low-pass filter
Accessories	2 AC power cords, Phoenix terminal connectors
Product dimensions (L x W x H)	482 x 414 x 188mm
Packaging size (L x W x H)	595 x 520 x 310mm=0.095m ³
Net weight/Gross weight of product	13.22kg/17.5kg

Audio Workstation

SI-1536P



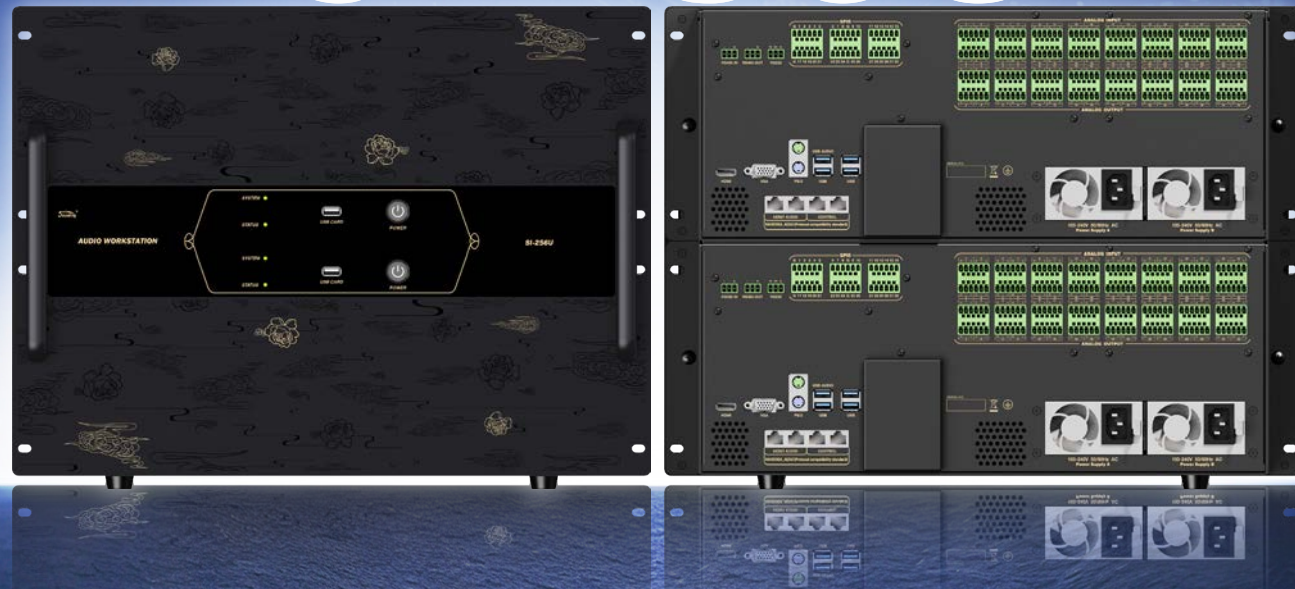
Features

1. DSP functions can be freely customized for each channel.
2. Capable of processing up to 1536x1520 network audio channels.
3. Supports up to 160-channel echo cancellation or 192 4096-tap FIRs.
4. It can be equipped with up to 32x32 analog channels (with the ability to expand to 32 analog microphone channels).
5. Up to 16 sets of GPIO interfaces are available for option.

Specifications

Power supply	100-240VAC 50/60HZ
Rack size	4u
Front panel	Power switch: 1; System indicator light: 1; Status indicator light: 1; USB CARD audio interface: 1;
Rear panel	AC power interface: 2; RJ45 2.5G network audio interface: 2; RJ45 network control interface: 2; USB Audio interface: 1; USB debugging port: 3; HDMI interface: 1; VGA interface: 1; PS/2 mouse and keyboard interface: 1; RS232 interface: 1; RS485 IN: 1; RS485 OUT: 1; GPIO: 16 groups (32, customizable input/output); Analog input interface: 32 (customizable); Analog output interface: 32 (customizable)
USB Audio channel	8x8
Protocol compatibility standard	RAVENNA_AES67
Total input / output channels of matrix channels	1536/1520
SND Network channel input	1456@48k (Net channel 1456 + USB input 24 + analog 32, total 1512, customizable with different ratios)
SND Network channel output	1456@48k (Net channel 1456 + USB input 24 + analog 32, total 1512; customizable with different ratios)
Internal matrix	1536x1520@48k
Maximum sampling rate	192kHz
Number of automatic mixing groups	16
Max channel count per group for automatic mixing	64
AEC (Acoustic Echo Cancellation)	160
ANS background noise reduction	160
Remix	Fully mixed audio, supporting independent GainSharing for output channels
EQ (input/output)	1520
FIR	4096 steps, 192 options (choose one from AEC/ANS functions)
Feedback inhibition (input)	1520
Delay device (input, output)	1520
Expander (input, output)	1520
Compression (input, output)	1520
Limiting (output)	1520
High-pass, low-pass (input, output)	1520
Accessories	2 AC power cords, Phoenix terminal connectors
Product dimensions (L x W x H)	482 x 414 x 188mm
Packaging size (L x W x H)	595 x 520 x 310mm=0.095m ³
Net weight/Gross weight of product	13.22kg/17.5kg

Audio Workstation SI-256U



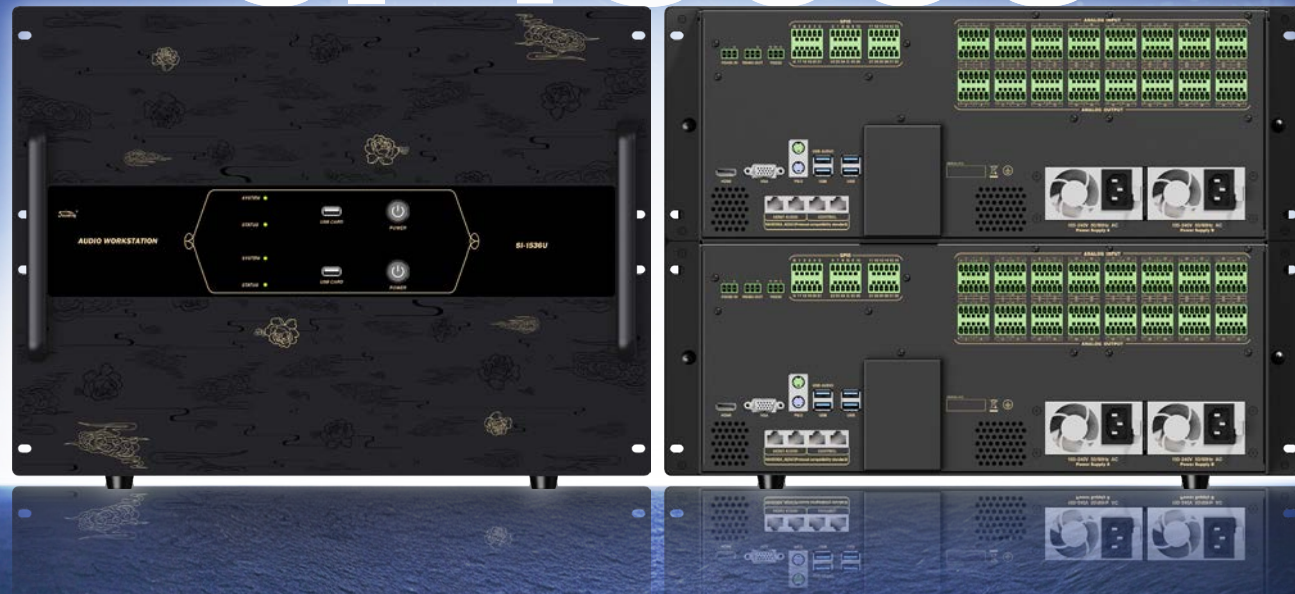
Features

1. Redundancy mode: Pioneering quadruple ultra-redundancy: dual analog backups, dual network backups, dual-machine independent operation mode, capable of dual-core stacking, doubling the usage for two service stations, and doubling the total number of DSP processing and I/O.
2. Customizable DSP functions for each channel.
3. Maximum processing (256x256) * 2-channel network audio.
4. Supports up to 180*2 4096-order FIRs.
5. Optional up to (32x32) *2 analog channels (expandable to 32*2 analog microphones).
6. GPIO can be optionally equipped with up to 16*2 groups (32*2, with customizable input and output).

Specifications

Power supply	100-240VAC 50/60HZ
Rack size	8U
Dual-machine version	The following functions/interfaces are provided in two copies, which can serve as backups for a dual-network system.
Front panel	Power switch: 2; System indicator light: 2; Status indicator light: 2; USB CARD audio interface: 2;
Rear panel	AC power interface: 2; RJ45 2.5G network audio interface: 2; RJ45 network control interface: 2; USB Audio interface: 1; USB debugging port: 3; HDMI interface: 1; VGA interface: 1; PS/2 mouse and keyboard interface: 1; RS232 interface: 1; RS485IN: 1; RS485 OUT: 1; GPIO: 16 groups (32, customizable input/output); Analog input interface: 32 (customizable); Analog output interface: 32 (customizable); PS: All corresponding interfaces on the back panel can be used individually; the quantity is doubled.
USB Audio channel	8x8
Protocol compatibility standard	RAVENNA_AES67
SND Network channel input	256@48k
SND Network channel output	256@48k
Maximum sampling rate	192kHz
Number of automatic mixing groups	16
Max channel count per group for automatic mixing	64
Remix	Fully mixed audio, supporting independent GainSharing for output channels
DSP function supports customization	Echo cancellation (up to 200 channels), FIR (4096-tap, up to 180), feedback suppression, Delay, expander, compressor, limiter, EQ, high-pass filter, low-pass filter
Accessories	4 AC power cords, Phoenix terminal connectors
Product dimensions (L x W x H)	482 x 438 x 364mm
Packaging size (L x W x H)	585 x 535 x 445mm=0.139m ³
Net weight/Gross weight of product	28.1kg/33.28kg

Audio Workstation SI-1536U



Features

1. Redundancy mode: Pioneering quadruple ultra-redundancy: dual analog backups, dual network backups, dual-machine independent operation mode, capable of dual-core stacking, doubling the usage for two service stations, and doubling the total number of DSP processing and I/O.
2. Customizable DSP functions for each channel.
3. Maximum processing (1536x1520) * 2-channel network audio.
4. It supports up to 160*2 channels of echo cancellation or 192*2 4096-tap FIRs.
5. Optional up to (32X32) * 2 analog channels (expandable to 32*2 analog microphones).
6. Up to 16 sets of *2-channel GPIO interfaces are available for option.

Specifications

Power supply	100-240VAC 50/60HZ
Rack size	8u
Dual-machine version	The following functions/interfaces are provided in two copies, which can serve as backups for a dual-network system.
Front panel	Power switch: 2; System indicator light: 2; Status indicator light: 2; USB CARD audio interface: 2;
Rear panel	AC power interface: 2; RJ45 2.5G network audio interface: 2; RJ45 network control interface: 2; USB Audio interface: 1; USB debugging port: 3; HDMI interface: 1; VGA interface: 1; PS/2 mouse and keyboard interface: 1; RS232 interface: 1; RS485IN: 1; RS485 OUT: 1; GPIO: 16 groups (32, customizable input/output); Analog input interface: 32 (customizable); Analog output interface: 32 (customizable) PS: All corresponding interfaces on the back panel can be used individually; the quantity is doubled.
USB Audio channel	8x8
Protocol compatibility standard	RAVENNA_AES67
Localization status of core components	Import
Total input / output channels of matrix channels	1536/1520
SND Network channel input	1456@48k (Net channel 1456 + USB input 24 + analog 32, total 1512, customizable with different ratios)
SND Network channel output	1456@48k (Net channel 1456 + USB input 24 + analog 32, total 1512, customizable with different ratios)
Internal matrix	1536x1520@48k
Maximum sampling rate	192kHz
Number of automatic mixing groups	16
Max channel count per group for automatic mixing	64
AEC (Acoustic Echo Cancellation)	160
ANS background noise reduction	160
Remix	Fully mixed audio, supporting independent GainSharing for output channels
EQ (input/output)	1520
FIR	4096 steps, 192 options (choose one from AEC/ANS functions)
Feedback inhibition (input)	1520
Delay device (input, output)	1520
Expander (input, output)	1520
Compression (input, output)	1520
Limiting (output)	1520
High-pass, low-pass (input, output)	1520
Accessories	4 AC power cords, Phoenix terminal connectors
Product dimensions (L x W x H)	482 x 438 x 364mm
Packaging size (L x W x H)	585 x 535 x 445mm=0.139m ³
Net weight/Gross weight of product	28.1kg/33.28kg

System Topology Diagram



AES67 Network Audio Matrix

A standard network audio matrix processor, compatible with all genuine AES67 audio ecosystem products for interoperability, High-standard RAVENNA-AES67 protocol interface. Supports full-featured DSP, with each channel allowing for free definition.

DSP function can be customized freely

- Support "unlimited" combination expansion.

Portable processor

- It can be made as small as 1/4 rack, with a height of 1U, and supports multi-unit hand-in-hand serial connection.

Multiple ways of using electricity

- NETA port supports POE power supply.

Small size, big function

- It can achieve 4-32 channels of input and output.
- The channel supports full mixing, with DSP functions such as EQ, delay, high and low pass, etc., available for all channels.

Support specific chip

- Chinese-made chips.



AMP-44T

Digital Audio Processor

AMP-44T is a 1U-wide network audio processor that adopts the AES67 network audio transmission protocol, supports Dante multicast compatibility, and supports 4 analog + 4 network audio input and output. It supports rich DSP functions such as automatic mixing and echo cancellation, and supports GPIO control, making it suitable for network node audio transmission processing.



Functional features

- Sampling rate is 48kHz.
- High-performance AD/DA, low distortion, wide dynamic range, and pure sound quality.
- Online, efficient, and user-friendly operation interface software for engineering, providing real-time control of all functions and monitoring of machine operating status.
- 4-channel MIC/LINE input, 4-channel balanced output.
- 4-channel AES67 network input, 4-channel AES67 network output.
- Each input features adjustable hardware gain (0dB/15dB/20dB/24dB) and independent 48V phantom power control.
- The volume of input and output audio signals can be independently adjusted, with a range of mute, -60.0 to +12.0 dB.
- 2 Gigabit Ethernet ports, capable of being connected in series for networking. The NETA port supports POE power supply.
- Supports a power LINK interface for hand-in-hand power supply during multi-machine expansion, facilitating engineering installation.
- The panel features 14 LEDs indicating audio and system status.
- Support automatic audio mixing function.
- Supports 8 GPIO interfaces.
- Supports 1-channel echo cancellation, background noise reduction, and automatic gain control functions.

Specifications

Input voltage	DC12V
Front panel	Channel level indicator lights x8; [POWER] power indicator light x1; [SIGNAL] network communication status light x1; [MIC] indicator lights x4
Rear panel	【DC12V/1.5A】 Power input socket x1; 【NETA(POE)】 Gigabit network interface x1; 【NETB】 Gigabit network interface x1; 【POWER IN】 Power LINK input interface x1; 【POWER OUT】 Power LINK output interface x1; Audio input channels x4; Audio output channels x4; GPIO control ports x8;
Analog audio channel	4 inputs and 4 outputs
Network audio channel	4 inputs and 4 outputs
Network audio delay	1/2/5/6ms
Sampling rate	48kHz
Signal-to-noise ratio	≥99dB (1kHz+4dBu, A-weighted, 0dB microphone gain); ≥85dB (1kHz+4dBu, A-weighted, 24dB microphone gain)
Harmonic distortion	≤0.005% (1kHz+4dBu, A-weighted, unity gain)
Channel isolation	≤-90dB, (1kHz+4dBu, A-weighted, unity gain)
Frequency response	±0.2dB (20Hz~20kHz)
Maximum input level	14dBu±1dBu (1kHz, THD+N≤1%)
Maximum output level	14dBu±1dBu (1kHz, THD+N≤1%)
Dynamic range	≥109dB (1kHz maximum output level, unity gain, A-weighted)
Microphone preamp gain	0/15/20/24dB, 4 levels in total
Power supply	Supports adapters and PoE-AF power supply, with an adapter power supply rated at 12VDC, 1.5A; supports daisy-chained power supply (requires adapter power supply, up to 2 devices can be daisy-chained)
Accessories	1 power adapter, Phoenix terminal connector
Product dimensions (L x W x H)	482 x 164 x 48.5mm
Packaging size (L x W x H)	250 x 80 x 520mm=0.01m ³
Net weight/Gross weight	1.90kg/2.35kg

AMP-3232

Digital Audio Processor

AMP-3232 is a 1U high rack-mounted network audio matrix that utilizes the AES67 network audio transmission protocol. It supports 4 analog + 28 network audio inputs and outputs, automatic mixing, GPIO control, and multiple network channels, making it suitable for mixing processing across multiple nodes.



Functional features

- The sampling rate is 48kHz.
- High-performance AD/DA, low distortion, wide dynamic range, and pure sound quality.
- Network connection, efficient and concise, engineering-friendly operation interface software, real-time control of all functions and monitoring of machine working status.
- 4-channel MIC/LINE input, 4-channel balanced output.
- 28-channel AES67 network input, 28-channel AES67 network output.
- Each input features adjustable hardware gain (0dB/15dB/20dB/24dB) and independent 48V phantom power control.
- The volume of input and output audio signals can be independently adjusted, with a range of mute, -60.0 to +12.0 dB.
- 2 Gigabit Ethernet ports, capable of being connected in series for networking. The NETA port supports POE power supply.
- Supports a power LINK interface for hand-in-hand power supply during multi-machine expansion, facilitating engineering installation.
- The panel features 14 LEDs indicating audio and system status.
- Support automatic audio mixing function.
- Supports 8 GPIO interfaces.

Specifications

Input voltage	DC12V
Front panel	Channel level indicator lights x8; [POWER] power indicator light x1; [SIGNAL] network communication status light x1; [MIC] indicator lights x4
Rear panel	[DC12V/1.5A] power input socket x1; [NETA(POE)] Gigabit network interface x1; [NETB] Gigabit network interface x1; [POWER IN] power LINK input interface x1; [POWER OUT] power LINK output interface x1; audio input channels x4; audio output channels x4; GPIO control ports x8;
Analog audio channel	4 inputs and 4 outputs
Network audio channel	28 in and 28 out
Network audio delay	1/2/5/6ms
Sampling rate	48kHz
Signal-to-noise ratio	≥99dB (1kHz+4dBu, A-weighted, 0dB microphone gain); ≥85dB (1kHz+4dBu, A-weighted, 24dB microphone gain)
Harmonic distortion	≤0.005% (1kHz+4dBu, A-weighted, unity gain)
Channel isolation	≤-90dB, (1kHz+4dBu, A-weighted, unity gain)
Frequency response	±0.2dB (20Hz~20kHz)
Maximum input level	14dBu±1dBu (1kHz, THD+N≤1%)
Maximum output level	14dBu±1dBu (1kHz, THD+N≤1%)
Dynamic range	≥109dB (1kHz maximum output level, unity gain, A-weighted)
Microphone preamp gain	0/15/20/24dB, 4 levels in total
Power supply	Supports adapters and PoE-AF power supply, with an adapter power supply rated at 12VDC, 1.5A; supports daisy-chained power supply (requires adapter power supply, up to 2 devices can be daisy-chained)
Accessories	1 power adapter, Phoenix terminal connector
Product dimensions (L x W x H)	482 x 164 x 48.5mm
Packaging size (L x W x H)	250 x 80 x 520mm=0.01m ³
Net weight/Gross weight	1.90kg/2.35kg

AMP-44TM

Digital Audio Processor

AMP-44TM is a 1/4 rack-width network audio processor that utilizes the AES67 network audio transmission protocol. It supports Dante multicast compatibility and offers 4 analog + 4 network audio input and output. With rich DSP functions such as automatic mixing and echo cancellation, it boasts a compact and flexible design, making it ideal for network node audio transmission processing.



Functional features

- The sampling rate is 48kHz.
- High-performance AD/DA, low distortion, wide dynamic range, and pure sound quality.
- Online connectivity, efficient and concise, engineering-friendly user interface software, real-time control of all functions and monitoring of machine operating status.
- 4-channel MIC/LINE input, 4-channel balanced output.
4-channel AES67 network input, 4-channel AES67 network output.
- Each input features adjustable hardware gain (0dB/15dB/20dB/24dB) and independent 48V phantom power control.
- The volume of input and output audio signals can be independently adjusted, with a range of mute, -60.0 to +12.0 dB.
- 2 Gigabit Ethernet ports, capable of being connected in series for networking. NETA port supports POE power supply.
- Supports a power LINK interface for hand-in-hand power supply during multi-machine expansion, facilitating engineering installation.
- The panel features 14 LEDs indicating audio and system status, along with a phantom power switch button, enabling user-friendly human-machine interaction.
- Support the expansion from 2 machines to 1 1U machine cabinet installation, with panel design featuring bracket mounting holes.
- Support automatic audio mixing function.
- Open DSP functionality, supporting 5A algorithm.
- Supports 1-channel echo cancellation, background noise reduction, and automatic gain control functions.

Specifications

Input voltage	DC12V
Front panel	Channel level indicator lights x8; [POWER] power indicator light x1; [Phantom Power] 48V phantom power x1; [SIGNAL] network communication status light x1; phantom power indicator lights x4;
Rear panel	[DC12V/1.5A] power input socket x1; [NETA(POE)] Gigabit network interface x1; [NETB] Gigabit network interface x1; [POWER IN] power LINK input interface x1; [POWER OUT] Power LINK output interface x1; Audio input channels x4; Audio output channels x4
Analog audio channel	4 in 4 out
Network audio channel	4 in 4 out
Network audio delay	1/2/5/6ms
Sampling rate	48kHz
Signal-to-noise ratio	≥99dB (1kHz+4dBu, A-weighted, 0dB microphone gain); ≥85dB (1kHz+4dBu, A-weighted, 24dB microphone gain)
Harmonic distortion	≤0.005% (1kHz+4dBu, A-weighted, unity gain)
Channel isolation	≤-90dB (1kHz+4dBu, A-weighted, unity gain)
Frequency response	±0.2dB (20Hz~20kHz)
Maximum input level	14dBu±1dBu (1kHz, THD+N≤1%)
Maximum output level	14dBu±1dBu (1kHz, THD+N≤1%)
Dynamic range	≥109dB (1kHz maximum output level, unity gain, A-weighted)
Microphone preamp gain	0/15/20/24dB, 4 levels in total
Power supply	Supports adapters and PoE-AF power supply. The adapter power supply is rated at 12VDC, 1.5A and supports daisy-chaining (requires adapter power supply, up to 2 devices can be daisy-chained)
Accessories	1 power adapter, Phoenix terminal connector
Product dimensions (L x W x H)	108 x 149 x 45mm
Packaging size (L x W x H)	275 x 275 x 95mm=0.007m ³
Net weight/gross weight	0.65kg/1.18kg

AMP-3232M

Digital Audio Processor

AMP-3232M is a 1/4 rack-width network audio matrix that utilizes the AES67 network audio transmission protocol. It supports 4 analog + 28 network audio inputs and outputs. With automatic mixing capabilities, compact and flexible size, and multiple network channels, it can be used for mixing processing across multiple nodes.



Functional features

- The sampling rate is 48kHz.
- High-performance AD/DA, low distortion, wide dynamic range, and pure sound quality.
- Online connectivity, efficient and concise, engineering-friendly operation interface software, real-time control of all functions and monitoring of machine working status.
- 4-channel MIC/LINE input, 4-channel balanced output.
- 28-channel AES67 network input, 28-channel AES67 network output.
- Each input features adjustable hardware gain (0dB/15dB/20dB/24dB) and independent 48V phantom power control.
- The volume of input and output audio signals can be independently adjusted, with a range of mute, -60.0 to +12.0 dB.
- 2 Gigabit Ethernet ports, capable of being connected in series for networking. The NETA port supports POE power supply.
- Support power LINK interface for hand-in-hand power supply during multi-machine expansion, facilitating engineering installation.
- The panel features 14 LEDs indicating audio and system status, along with a phantom power switch button, enabling user-friendly human-machine interaction.
- Support the expansion from 2 machines to 1 1U machine cabinet installation, with panel design featuring bracket mounting holes.
- Support automatic audio mixing function.

Specifications

Input voltage	DC12V
Front panel	Channel level indicator lights x8; [POWER] power indicator light x1; [Phantom Power] 48V phantom power x1; [SIGNAL] network communication status light x1; phantom power indicator lights x4
Rear panel	【DC12V/1.5A】 Power input socket x1; 【NETA(POE)】 Gigabit network interface x1; 【NETB】 Gigabit network interface x1; 【POWER IN】 Power LINK input interface x1; 【POWER OUT】 Power LINK output interface x1; Audio input channels x4; Audio output channels x4
Analog audio channel	4 inputs and 4 outputs
Network audio channel	28 in, 28 out
Network audio delay	1/2/5/6ms
Sampling rate	48kHz
Signal-to-noise ratio	≥99dB (1kHz+4dBu, A-weighted, 0dB microphone gain); ≥85dB (1kHz+4dBu, A-weighted, 24dB microphone gain)
Harmonic distortion	≤0.005% (1kHz+4dBu, A-weighted, unity gain)
Channel isolation	≤-90dB (1kHz+4dBu, A-weighted, unity gain)
Frequency response	±0.2dB (20Hz~20kHz)
Maximum input level	14dBu±1dBu (1kHz, THD+N≤1%)
Maximum output level	14dBu±1dBu (1kHz, THD+N≤1%)
Dynamic range	≥109dB (1kHz maximum output level, unity gain, A-weighted)
Microphone preamp gain	0/15/20/24dB, a total of 4 levels
Power supply	Supports adapters and PoE-AF power supply. The adapter power supply has a rated voltage of 12VDC and 1.5A, and supports daisy-chaining (requires adapter power supply, up to 2 devices can be daisy-chained)
Accessories	1 power adapter, Phoenix terminal connector
Product dimensions (L x W x H)	108 x 149 x 45mm
Packaging size (L x W x H)	275 x 275 x 95mm=0.007m ³
Net weight/gross weight	0.65kg/1.18kg

AMP-88AT

Digital Audio Matrix Processor

AMP-88AT is a 1U-wide network audio processor that adopts the AES67 network audio transmission protocol, supports Dante multicast compatibility, and supports 8 analog + 8 network audio input and output. It supports rich DSP functions such as automatic mixing and echo cancellation, supports GPIO control, and can be used for network node audio transmission processing.



Functional features

- Sampling rate is 48kHz.
- High-performance AD/DA, low distortion, wide dynamic range, and pure sound quality.
- Online, efficient, concise, and user-friendly operation interface software for engineering, enabling real-time control of all functions and monitoring of machine working status.
- 8-channel MIC/LINE input, 8-channel balanced output.
- 8-channel AES67 network input, 8-channel AES67 network output.
- 3-channel automatic mixing; supports 1-channel echo cancellation/background noise reduction/automatic gain control.
- Each input features adjustable sensitivity (0dB/15dB/20dB/24dB) and independent 48V phantom power control.
- Supports up to 24 internal matrix routing mixing channels (analog + network + USB + automatic mixing + echo cancellation)
- The volume of input and output audio signals can be independently adjusted, with a range of mute, -60.0 to +12.0 dB.
- 2 Gigabit Ethernet ports, each capable of simultaneously transmitting both communication data and audio data.
- The panel features 14 LEDs indicating audio and system status.
- Support automatic audio mixing function.
- One RS232 interface supporting bidirectional data transmission and reception.
- 2 RS485 interfaces, which can be used in series (RS485 IN for input, and RS485 OUT connected to another RS485 IN for simultaneous control).
- 16 GPIO interfaces, with customizable input/output directions.
- Input/output channels provide optional DSP functions such as signal generator, noise gate, feedback suppression, equalization, high-pass, low-pass, delay, expander, compressor, limiter, duck, FIR, etc.

Specifications

Input voltage	90V-260Vac 50/60Hz
Front panel	[INPUTS] Channel level indicator lights x8; [OUTPUTS] Channel level indicator lights x8; [SYSTEM] System status light x1; [STATUS] abnormal status light x1; [UAC/PLAYBACK] x1; [USB AUDIO] x1;
Rear panel	[90V-260Vac 50/60Hz 40W] Power input socket (with switch, fuse) x1; [NET-AES67/RAVENNA] Gigabit network interface x2; [RS232] x1; [RS485 IN] x1; [RS485 OUT] x1; [GPIO] GPIO control port x16; [ANALOG LINE/MIC INPUTS] analog audio input x8; [ANALOG LINE OUTPUTS] analog audio output x8;
Analog audio channel	8 inputs and 8 outputs
Network audio channel	8 inputs and 8 outputs
Network audio delay	1/2/5/6ms
Sampling rate	48kHz
Signal-to-noise ratio	≥96dB (1kHz+4dBu, A-weighted, 0dB microphone gain); ≥85dB (1kHz+4dBu, A-weighted, 24dB microphone gain)
Harmonic distortion	≤0.005% (1kHz+4dBu, A-weighted, unity gain)
Channel isolation	≤-90dB, (1kHz+4dBu, A-weighted, unity gain)
Frequency response	±0.3dB (20Hz~20kHz)
Maximum input level	14dBu±1dBu (1kHz, THD+N≤1%)
Maximum output level	14dBu±1dBu (1kHz, THD+N≤1%)
Dynamic range	≥106dB (1kHz maximum output level, unity gain, A-weighted)
Microphone preamp gain	0/15/20/24dB, 4 levels in total
Accessories	Power cord, Phoenix terminal connector
Product dimensions (L x W x H)	482 x 231 x 45mm
Packaging size (L x W x H)	520 x 340 x 90mm=0.016m ³
Net weight/Gross weight	2.7kg/3.7kg

AMP-1616AT

Digital Audio Matrix Processor

AMP-1616AT is a 1U high rack-mounted network audio matrix that utilizes the AES67 network audio transmission protocol. It supports Dante multicast compatibility and can handle 16 analog + 16 network audio inputs and outputs. With rich DSP functions such as automatic mixing and echo cancellation, as well as GPIO control, it is ideal for network node audio transmission processing.



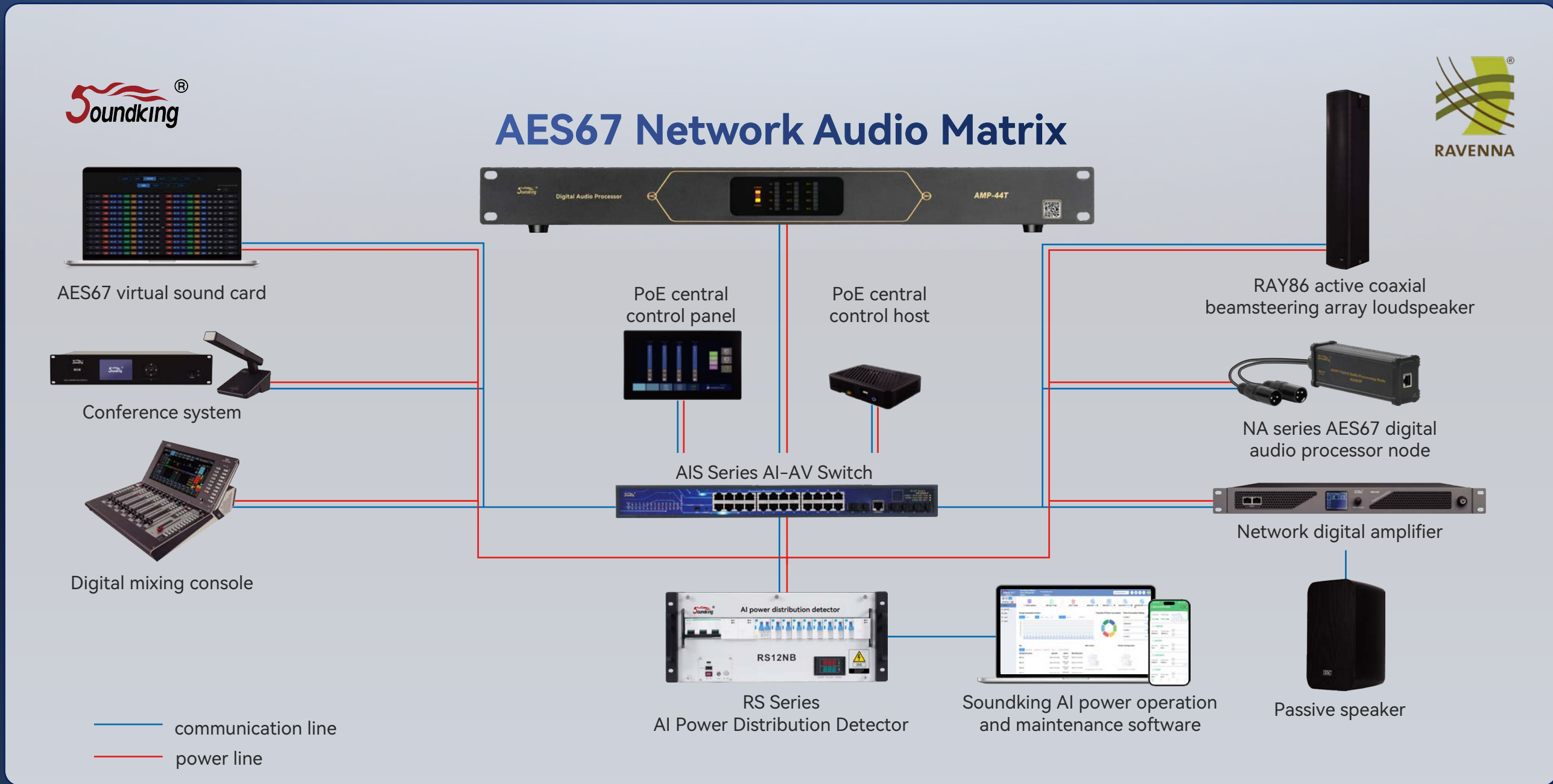
Functional features

- The sampling rate is 48kHz.
- High-performance AD/DA, low distortion, wide dynamic range, and pure sound quality.
- Online, efficient, concise, and user-friendly operation interface software for engineering, enabling real-time control of all functions and monitoring of machine operating status.
- 16-channel MIC/LINE input, 16-channel balanced output.
- 16-channel AES67 network input, 16-channel AES67 network output.
- 3-channel automatic mixing; supports 1-channel echo cancellation/background noise reduction/automatic gain control.
- Each input features adjustable sensitivity (0dB/15dB/20dB/24dB) and independent 48V phantom power control.
- Supports up to 40 internal matrix routing mixing channels (analog + network + USB + automatic mixing + echo cancellation)
- The volume of input and output audio signals can be independently adjusted, with a range of mute, -60.0 to +12.0 dB.
- 2 Gigabit Ethernet ports, each capable of simultaneously transmitting both communication data and audio data.
- The panel features 14 LEDs indicating audio and system status.
- Support automatic audio mixing function.
- One RS232 interface supporting bidirectional data transmission and reception.
- 2 RS485 interfaces, which can be used in series (RS485 IN for input, and RS485 OUT connected to another RS485 IN for simultaneous control)
- 16 GPIO interfaces, with customizable input/output directions
- Input/output channels provide optional DSP functions such as signal generator, noise gate, feedback suppression, equalization, high pass, low pass, delay, expander, compressor, limiter, duck, FIR, etc

Specifications

Input voltage	90V-260Vac 50/60Hz
Front panel	[INPUTS] Channel level indicator lights x16; [OUTPUTS] Channel level indicator lights x16; [SYSTEM] System status light x1; [STATUS] Abnormal status light x1; [UAC/PLAYBACK] x1; [USB AUDIO] x1;
Rear panel	[90V-260Vac 50/60Hz 40W] Power input socket (with switch, fuse) x1; [AES67/RAVENNA] Gigabit network interface x2; [RS232] x1; [RS485 IN] x1; [RS485 OUT] x1; [GPIO] GPIO control port x16; [ANALOG LINE/MIC INPUTS] analog audio input x16; [ANALOG LINE OUTPUTS] analog audio output x16;
Analog audio channel	16 inputs and 16 outputs
Network audio channel	16 inputs and 16 outputs
Network audio delay	1/2/5/6ms
Sampling rate	48kHz
Signal-to-noise ratio	≥96dB (1kHz+4dBu, A-weighted, 0dB microphone gain); ≥85dB (1kHz+4dBu, A-weighted, 24dB microphone gain)
Harmonic distortion	≤0.005% (1kHz+4dBu, A-weighted, unity gain)
Channel isolation	≤-90dB, (1kHz+4dBu, A-weighted, unity gain)
Frequency response	±0.3dB (20Hz~20kHz)
Maximum input level	14dBu±1dBu (1kHz, THD+N≤1%)
Maximum output level	14dBu±1dBu (1kHz, THD+N≤1%)
Dynamic range	≥106dB (1kHz maximum output level, unity gain, A-weighted)
Microphone preamp gain	0/15/20/24dB, 4 levels in total
Accessories	Power cord, Phoenix terminal connector
Product dimensions (L x W x H)	482 x 231 x 45mm
Packaging size (L x W x H)	520 x 340 x 90mm=0.016m ³
Net weight/Gross weight	2.8kg/3.8kg

System Topology Diagram



AES67 Peripherals and Software

AES67 Digital Audio Processing Node

A small portable network audio processor compatible with all genuine AES67 audio ecosystem products for interoperability,
High-standard RAVENNA/AES67 protocol interface.

Supports full-featured DSP, with each channel allowing for free definition.

Suitable for analog audio, analog mixer, analog amplifier, analog microphone, digital conversion device, etc.



AES67 Digital Audio Processing Node **VS** Traditional Equipment

AES67 Digital Audio Processing Node



- Good waterproof performance and stable connection.
- Convenient maintenance and powerful software performance.
- Strong anti-interference capability and dedicated audio wiring harness.
- EQ: Selectable from 1-16 segments, with network channels freely combinable.
- Powerful DSP function, with each channel supporting free definition.
- High-performance 5A and algorithms: automatic gain control, noise reduction, feedback suppression, echo cancellation, and audio mixing.

=

Audio stage box



- The network channel functionality is restricted
- Poor waterproof performance
- The connection is not portable
- Inconvenient for maintenance
- The software has limited and restricted functionality

+

Audio DSP processor



- Loss of sound quality
- Design constraints
- Poor compatibility of processing algorithms
- Interface expansion is limited

+

Network DA audio cable



- Signal loss and interference
- Bandwidth limit
- The transmission impedance is high
- The wire material is not standard

AES67 Digital Audio Processor Node

NA02D



Introduction

Supports full-featured DSP, with each channel allowing for free definition.

EQ: Optional 1-16 segments, delay, compression, feedback suppression, high and low pass, HR, limiter, expander, noise gate, signal generator, automatic mixer (threshold type, gain sharing type), matrix mixer.

Features

1. Supports two-channel analog audio signal input and two-channel AES67 network audio signal output;
2. Two XLR female connectors for inputting analog audio signals;
3. One RJ45 network port connected to a switch for outputting AES67 network audio signals;
4. Convert the analog audio signal into an AES67 network audio signal for output.

Specifications

Maximum signal level (balanced)	18dBu
Frequency response	20Hz-20kHz, ±0.5dB
Impedance	20kΩ (balanced); 10kΩ (unbalanced)
Dynamic range	≥110dB @+18dBu, A-weighted
Signal-to-noise ratio	≥99dB@+4dBu
THD+N	<0.005%@1kHz, A-weighted
Connector	RJ45 and 2 XLR-F
Power supply	Class 802.3 af PoE
Sampling rate	48 kHz
Bit depth	24 bit
AES67 device latency	1, 2, 5, or 6ms (configurable using skwkaesm win)
Network transmission	AES67 RTP audio
Product dimensions (L x W x H)	176 x 42.9 x 57.4mm / 6.9 x 1.7 x 2.3in
Product weight	0.3kg
1 Inner box packaging dimensions (L x W x H)	235 x 125 x 70mm=0.002m ³ / 9.3 x 4.9 x 2.8in
20-piece outer box packaging dimensions (L x W x H)	490 x 270 x 385mm=0.051m ³ / 19.3 x 10.6 x 15.2in

AES67 Digital Audio Processor Node

NA02P



Introduction

Supports full-featured DSP, with each channel allowing for free definition.

EQ: Optional 1-16 segments, delay, compression, feedback suppression, high and low pass, HR, limiter, expander, noise gate, signal generator, automatic mixer (threshold type, gain sharing type), matrix mixer.

Features

1. Supports the input of two AES67 network audio signals and the output of two analog audio signals;
2. Two XLR male connectors for outputting analog audio signals;
3. One RJ45 network port is connected to a switch for receiving AES67 network audio signals;
4. Convert AES67 network audio signals into analog audio signals for output.

Specifications

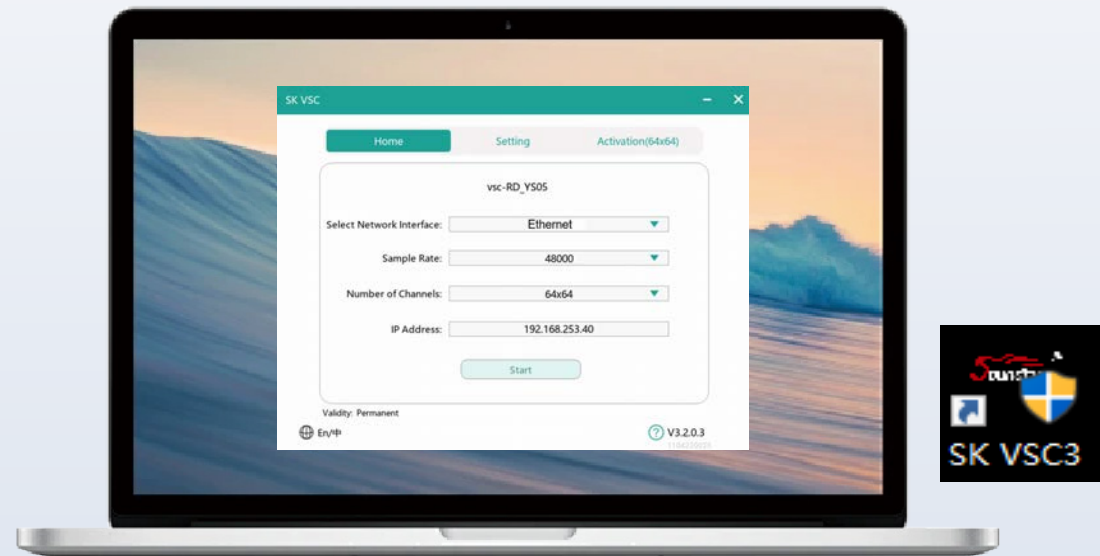
Maximum signal level (balanced)	18dBu
Frequency response	20Hz-20kHz, ±0.5dB
Impedance	110Ω (balanced type)
Dynamic range	≥110dB @+18dBu, A-weighted
Signal-to-noise ratio	≥99dB@+4dBu
THD+N	<0.005%@1kHz, A-weighted
Connector	RJ45 and 2 XLR-M
Power supply	Class 802.3 af PoE
Sampling rate	48kHz
Bit depth	24bit
AES67 device latency	1, 2, 5, or 6ms (configurable using skwkaesm win)
Network transmission	AES67 RTP audio
Product dimensions (L x W x H)	176 x 42.9 x 57.4mm / 6.9 x 1.7 x 2.3in
Product weight	0.3kg
1 Inner box packaging dimensions (L x W x H)	235 x 125 x 70mm=0.002m ³ / 9.3 x 4.9 x 2.8in
20-piece outer box packaging dimensions (L x W x H)	490 x 270 x 385mm=0.051m ³ / 19.3 x 10.6 x 15.2in

System Topology Diagram



SK VSC

64 x 64 channels AES67 virtual sound card



- **64x64** channels.
- **96kHz** is the highest sampling rate.
- **WDM Driver** is compatible with a wide range of devices.
- **Windows 10/11** is compatible with Windows systems (22H2 and above) and seamlessly integrates into your workflow using WDM audio format.
- **The X86 platform** breaks down the boundaries between computers, large screens, and AES67 audio systems.

SK AES67 Controller



Network routing



DSP config



Intelligent monitoring



Open DSP programming



IoT control



Offline design



Electronic Catalog APP of Soundking



IOS



ANDROID

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