# DIGISYNTHETIC®

## 2025 Product Brochure

RAVENNA AES67 & ST 2110 built-in

SND Network

AES67

NETGEAR® AV

# 2025

# DIGISYNTHETIC®

Wechat

WhatsApp

Tiktok

Website

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China

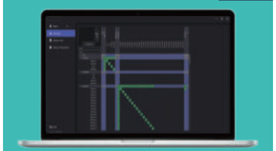
www.digisynthetic.com

V 1.1 2025

# PRODUCT LIST

**Software** | **AES67 Modules** | **Finished Product** | **Audio Station**

## Software

03

AES67 GUI

04

AES67 VSC

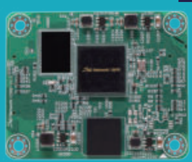
## AES67 Modules

NEW 05

DL-02S/04S

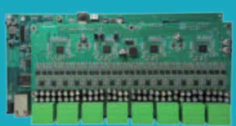
06

DL02/04

07

DL08/16/32 BTB

07

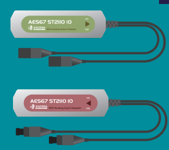
DL08/16/32 Mini-PCI

17

DL08/16/32 Mini-PCI

## Finished Product

08

DMX208A Matrix

NEW 11

DMX02 Adapters

09

DMX232 Mixer

13

DLS Speaker series

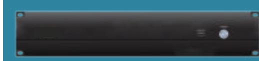
NEW 10

DMX0808/1616
Matrix

14

DSM Micphone series

## Audio Station

ODM 15

DESCORE 2U

ODM 16

DESCORE 4U

**PROUDLY SUPPORTING:**

RAVENNA AES67 & ST 2110 built-in | ND Network | AES67 | NETGEAR AV

# Our team

Since our founding in 1996, we've adhered to the philosophy of "Democratizing Technology", beginning with Karaoke effects and DSP processors and consistently pushing boundaries in fundamental audio innovations.

Now the heart of DIGISYNTHETIC lies a team of young graduated from Sun Yat-sen University. We approach every product with engineering precision, transforming cutting-edge technologies into high-value solutions. Our relentless curiosity about acoustic essence and persistent pursuit of networked audio technologies drive us to break conventional limits, delivering smarter and more connected audio experiences globally.

# Leadership

**Gary Sheng**
**CEO**

中山大學
SUN YAT-SEN UNIVERSITY

**ADUCATION**
Dual Bachelor's Degrees: Software Engineering & Business, Sun Yat-sen University
**ENTREPRENEURIAL CAREER**
2014: Pioneered ventures in real-time communication tech, garnering 10M global users
2017: Honored in Forbes China 30 Under 30 Regional List (Technology Category)
2018: Listed in Forbes Asia 30 Under 30 Regional List (Industry & Manufacturing)
**TECH LEADERSHIP AT DIGISYNTHETIC**
AES67 Ultra-Low Latency Networking: Sub-10ms edge computing frameworks
AI-driven noise suppression (60dB attenuation) and spatial audio encoding
CI/CD pipelines achieving 5x faster time-to-market vs. industry benchmarks

# WHAT IS AES67 ?

## What is AES67?

**The Ultimate Cross-Platform AOIP**

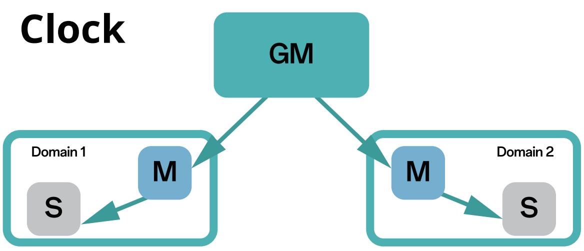
AES67 is a standard developed by the Audio Engineering Society (AES) to ensure interoperability between different Audio over IP (AoIP) systems. The standard provides a framework for devices from different manufacturers to communicate seamlessly over IP networks.

## Device Control

Device discovery is another important aspect of device control in an AES67 system. Discovery protocols such as Bonjour and SAP (Session Announcement Protocol) enable devices to automatically find each other without manual intervention. These protocols ensure that new devices connected to the network can be integrated seamlessly, allowing for rapid deployment and reconfiguration of audio streams.

## Clock Synchronization

Precision Time Protocol (PTP): Implements IEEE 1588-2008 PTP to ensure all devices share a unified time reference (<1μs network-wide accuracy).
Media Clock Synchronization: Guarantees 48kHz/96kHz audio sample alignment via RTP timestamps, eliminating drift for frame-accurate multi-device workflows (e.g., broadcast mixing consoles).




## Connection Management

AES67 Connection Management utilizes SDP to synchronize audio parameters (codec/sample rate/IP ports) and employs SIP/SAP for dynamic stream setup/adjustments, ensuring efficient device interconnection (<10ms latency) with hot-swappable channels and broadcast-grade stability. Key steps: 1) SDP metadata negotiation; 2) PTP clock sync; 3) DiffServ QoS prioritization; 4) IGMP multicast optimization. Typical use: Multi-vendor mixer-DSP plug-and-play.

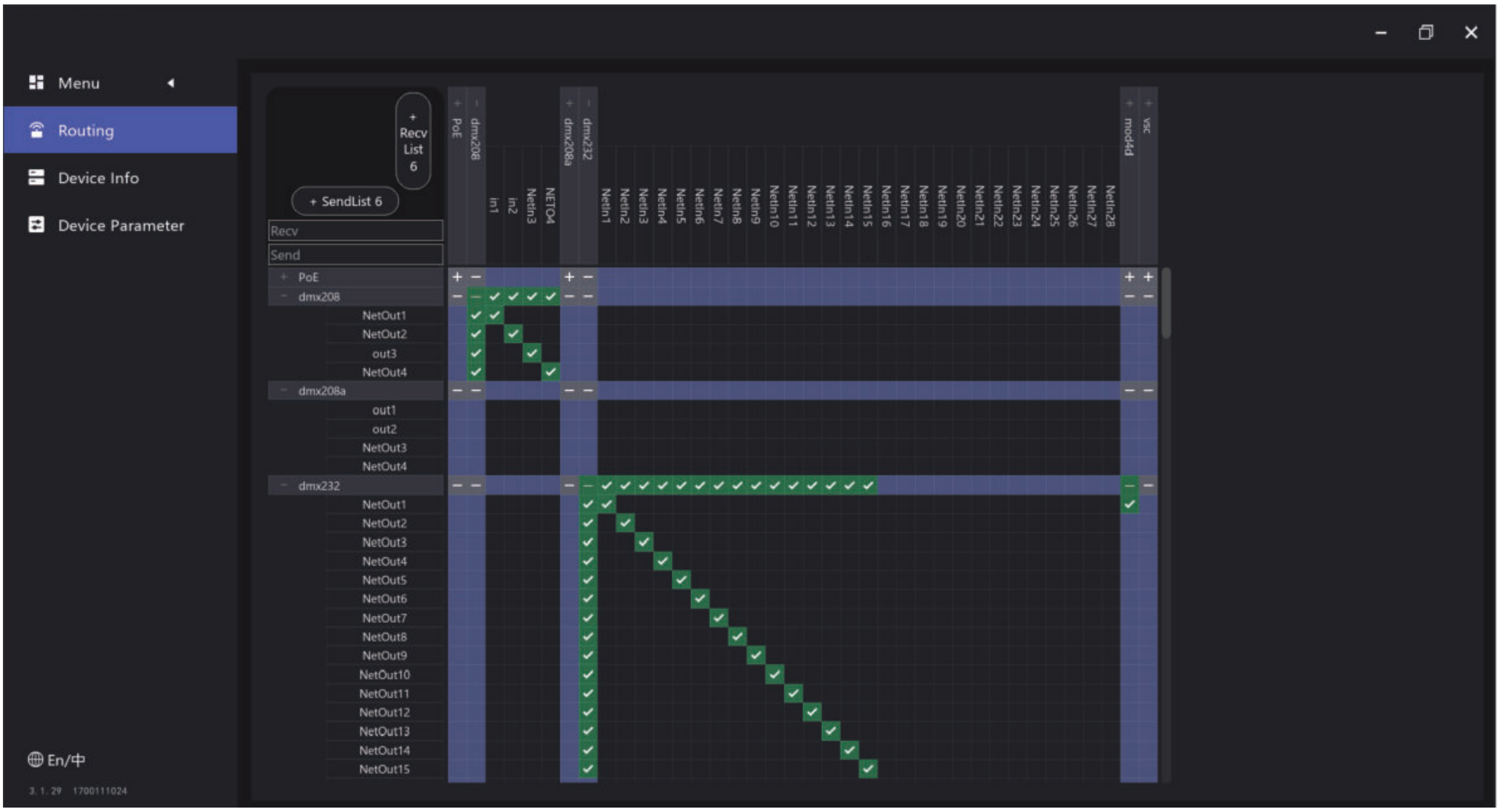
```
v=0
o=- 759457124 759457124 IN IP4 192.168.10.137
s=44D-01
c=IN IP4 239.69.248.177/32
t=0 0
m=audio 5004 RTP/AVP 96
a=rtpmap:96 L24/48000/4
a=recvonly
a=ptime:0.25
a=mediaclk:direct=0
```

## Transport

AES67 uses RTP for low-latency, synchronized audio streaming with packet sequencing. QoS (DiffServ) prioritizes audio traffic to prevent loss/delay. Multicast (IGMP) enables efficient one-to-many distribution for large setups. Key features:

- <10ms latency (PTPv2 sync)
- 48/96kHz/24-bit uncompressed audio
- Zero packet loss via QoS tagging

# AES67 LINK Feartures




- ## Device Routing

  You can use AES67 LINK to browse all AES67 LINK-supported devices in the network and create or delete audio and video connections between these devices.

- ## Powerful DSP Blocks

  Offers a comprehensive DSP blocks including:
  EQ, delay, feedback suppression, FIR filters, compressor, limiter, expander, ducking, noise gate, signal generator, AEC, AFC, ANS, AGC, ANC and matrix mixer for automatic mixing.

- ## Monitoring Bandwidth and Latency

  AES67 LINK enables you to track the real-time status of devices and overall network health, including:
  Bandwidth usage – Monitor data flow to ensure optimal performance.
  Latency metrics – Detect and address delays for synchronized audio/video transmission.

- ## Sampling Rate and Clock Settings

  It identifies any issues related to sample rates or network connections to ensure the quality and stability of your network, maintaining optimal performance.

- ## Offline desgin mode

  AES67 LINK supports importing and exporting archived configurations for all online devices within the local area network (LAN), including network routing and DSP settings. In offline scenarios, the system allows users to design complete system solutions using devices from the offline product library. This enables comprehensive pre-configuration of entire systems even without network connectivity.


- ## Custom Device Names

  You can assign custom names to devices for easier navigation, monitoring, and management within the AES67 LINK network. Additionally, you can filter devices based on parameters such as:Device name, Sample rate, Latency.

AES67 Standard | Windows WDM | RAVENNA Partner


## Applications

腾讯会议 Tencent Meeting | CUBASE

## Features

- **Outstanding Quality:** Supports up to 96kHz Hi-Fi and 64 x 64 channels, delivering an ultimate immersive panoramic sound experience.
- **Windows Compatibility:** Seamlessly integrates into your workflow using the WDM audio format.
- **User-Friendly Operation:** Features an intuitive and easy-to-use interface, allowing effortless setup and control—even beginners can get started quickly.
- **Security Protection:** Equipped with a unique device-locking function to safeguard your data and effectively prevent unauthorized access.
- Full Compatibility with Leading Audio Network Devices:

RAVENNA AES67 & ST 2110 built-in

QSC Dante Dolby

# DIGISYN VSC

64 x 64 Virtual Soundcard

## From 64×64 Audio to AI Assistants: How DIGISYN VSC Simplifies Workflows

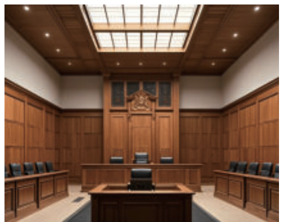
With just a Windows PC, users can effortlessly achieve 64 × 64 channel audio recording and playback. The DIGISYN VSC Virtual Sound Card supports high-quality multi-track recording and is compatible with Tencent Meeting, Microsoft Teams, or Zoom for video conferencing—eliminating the need for additional video hosts or USB sound cards.

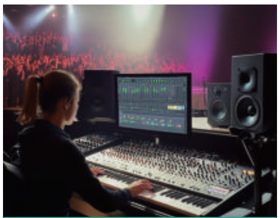
Moreover, this sound card enables high-fidelity recording for concerts and public address systems while seamlessly integrating with AI voice assistants via VSC. No extra hardware is required to access leading AI voice models such as Deepseek and ChatGPT.

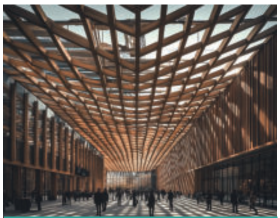
## Use cases


Boardcasting


Conference


Court recording


Live music


Public Address

Immersive Cinema

## System requirements for VSC

VSC

- CPU
  - [Intel] 8th Gen i3 or higher [AMD] Ryzen 5 1400 or higher
- RAM
  - 4GB or more
- Network
  - Standard wired Ethernet interface (100Mbps or 1Gbps)
  - Gigabit (1Gbps) required for: 32×32 channels @48kHz or 16×16 channels @96kHz
  - Wireless (Wi-Fi) not supported
- Storage
  - High-speed drives recommended for multi-track audio recording/playback:
  - HDD: 7200 RPM or faster (required for 16+ channels)
  - SSD: M.2 PCIe 3.0 (recommended for large-scale projects)
- Licensing Options
  - Single-Machine License: 8×8 channels / 64×64 channels /60-Day Trial License
  - Details: https://www.digisynthetic.com/aes67-virtual-soundcard

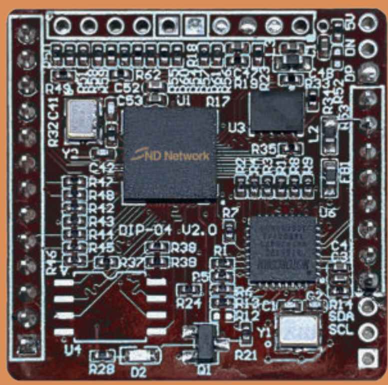
# DL-02S/04S

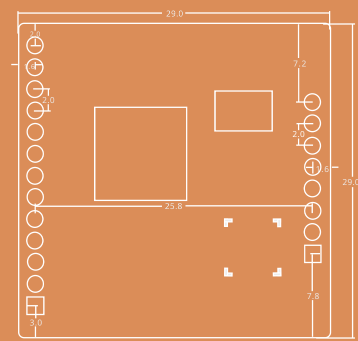
2 x 2 / 4 x 4 AES67 Network Module

AES67 Module
Cost-Performance
RAVENNA Partner

## Ideal for AES67 Network Audio

- Cost-effective small-channel AES67 Network module.
- High-performance, low-latency uncompressed audio over IP
- Compatible and interoperable with thousands of AES67 devices.
- Real-time control of routing, latency, and monitoring each devide via DIGISYN LINK3 or Allonis Http MyServer6.
- ltra High Cost-Performance Ratio to Meet the Most Demanding Budget Requirements of Any Audio Manufacturer

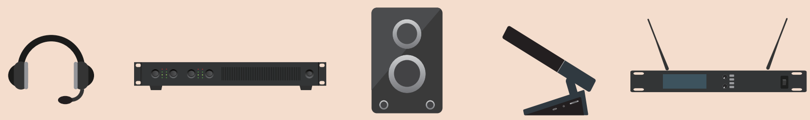




## Fully Open API for Easy Integration

Integrate DIGISYN's flexibility, and simplicity into your products to inspire creativity. The DL-02S/04S series supports AES67 features like auto-discovery, label-based routing, and plug-and-play operation. Compatible with standard Ethernet, it offers exceptional DSP audio performance. The free DIGISYN LINK3 software provides easy setup for DL modules with AES67/RAVENNA integration.

## Application Cases



Headphone receiver · Amplifier · Active speaker · Microphone · Microphone receiver

- DL-02S offers AES67/ RAVENNA interfaces for network audio modules supporting 2 x 2 channels at 48kHz, 96kHz, and 192kHz sample rates.

- DL-04S provides AES67 RAVENNA interfaces for network audio modules supporting 4 x 4 channels at 48kHz, 96kHz, and 192kHz sample rates.

## Technical Specifications

| | |
|---|---|
| **Audio Features** | |
| Sampling Rate | 48 / 96 / 192 kHz |
| Audio I/O Channels (48kHz) | Up to 4 x 4 channels |
| Audio I/O Channels (96kHz) | Up to 4 x 4 channels |
| Audio I/O Channels (192kHz) | Up to 4 x 4 channels |
| Audio Stream Input/Output | Up to 4 x 4 audio streams |
| Digital Audio Format | I²S |
| Audio Transmission Format | AES67, RAVENNA network audio |
| Bit Depth | 16, 24, or 32 bits per sample |
| Clock | Onboard word clock |
| **Hardware** | |
| Dimensions | 2.9cm x 2.9cm |
| Clock | High-quality, low-jitter onboard clock |
| Physical Connectors | 2.0mm pin header |
| Power Supply | 3.3V DC |
| **Interfaces** | |
| Control Interfaces | UART |
| Network | Ethernet PHY (MDI) |

## Features

**Compact Design:** 29mm x 29mm ideal for space-constrained devices like active speakers and wall controllers.

**Low Channel Count:** Supports 2x2 or 4x4 channels up to 192kHz perfect for endpoint audio products.

**Low Latency – Achieves:** ≤1ms receive latency, ensuring real-time audio synchronization.

**I2S Audio Interface:** Directly connects to ADCs, DACs, DSPs, and amplifiers for streamlined integration.

**RMII Ethernet:** Compatible with cost-effective PHY solutions and Power over Ethernet (PoE) designs.

**Plug-and-Play:** Auto-discovery and DIGISYN LINK3 GUI Controller compatibility for effortless setup.

**Cost-Effective:** Reduces BOM costs with high integration for low-channel applications.

**DIGISYN-AES67 API:** All API documentation and development resources are freely accessible to audio enthusiasts and manufacturers.

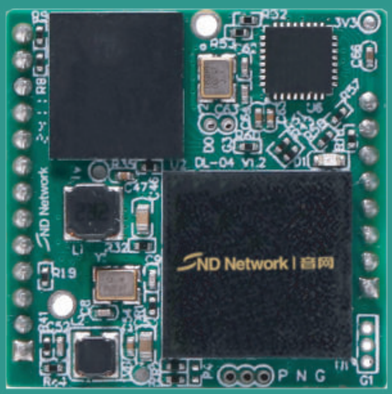
AES67 Protocol

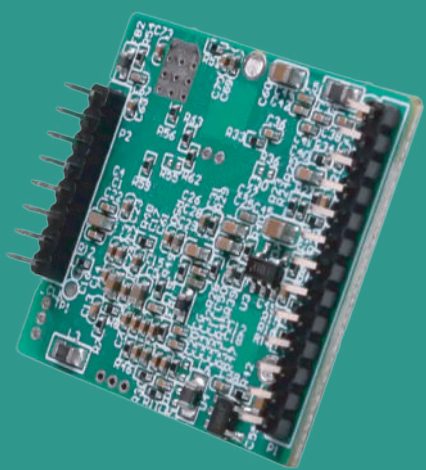
Extensive DSP

RAVENNA Partner

# DL-02D/04D

2 x 2 / 4 x 4 AES67 DSP Network Module





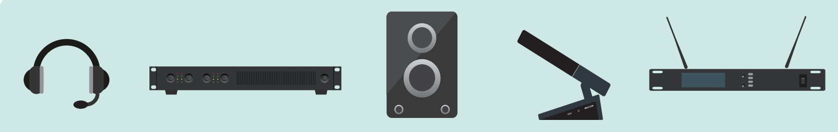
## Ideal for AES67 Network Audio

- DSP embedded small-channel AES67 Network module.
- High-performance, low-latency uncompressed audio over IP
- Compatible and interoperable with thousands of AES67 devices.
- Real-time control of routing, latency, DSP, and monitoring each devide via DIGISYN LINK3 or Allonis Http MyServer6.
- Powerful and Customizable DSP for every input and output.

## Fully Open API for Easy Integration

Integrate DIGISYN LINK's flexibility, powerful DSP, and simplicity into your products to inspire creativity. The DL-02/04 series supports AES67 features like auto-discovery, label-based routing, extensive DSP, and plug-and-play operation. Compatible with standard Ethernet, it offers exceptional DSP audio performance. The free DIGISYN LINK3 software provides easy setup for DL modules with AES67/RAVENNA integration.

## Application Cases




- DL-02 offers AES67/ RAVENNA interfaces for network audio modules supporting 2 x 2 channels at 48kHz, 96kHz, and 192kHz sample rates.

- DL-04 provides AES67 RAVENNA interfaces for network audio modules supporting 4 x 4 channels at 48kHz, 96kHz, and 192kHz sample rates.

## Features

**Network Performance**

• Ultra-Low Latency: ≤1ms network delay

• Sampling rates up to 192kHz 24-bit depth

**AES67 Virtual Sound Card (VSC) Functionality**

• Multi-Channel Distribution: Route 64 x64 channels from host PC to all DL embedded audio devices across local area network

**Standards-Based Interoperability**

•Full compliance with AES67 protocol for seamless integration with existing AoIP infrastructure

**• DSP Expansion:**

EQ: 16-Band | Delay: 2 Seconds | Compression

High-pass/Low-pass | FIR (2048 Taps)

Limiter | Noise Gate | Expander | Ducker

Auto-Mixing (Threshold/Gain Sharing)

## Technical Specifications

| | |
|---|---|
| **Audio Features** | |
| Sampling Rate | 48 / 96 / 192 kHz |
| Audio I/O Channels (48kHz) | Up to 4 x 4 channels |
| Audio I/O Channels (96kHz) | Up to 4 x 4 channels |
| Audio I/O Channels (192kHz) | Up to 4 x 4 channels |
| Audio Stream Input/Output | Up to 4 x 4 audio streams |
| Digital Audio Format | $I^2S$ |
| Audio Transmission Format | AES67, RAVENNA network audio |
| Bit Depth | 16, 24, or 32 bits per sample |
| Clock | Onboard word clock |
| **Hardware** | |
| Dimensions | 2.9cm x 2.9cm |
| Clock | High-quality, low-jitter onboard clock |
| Physical Connectors | 2.0mm pin header |
| Power Supply | 3.3V DC |
| **Interfaces** | |
| Control Interfaces | UART |
| Network | Ethernet PHY (MDI) |

# DL-08/16/32

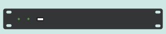
8x8/16x16/32x32 AES67 Network Module

## Ideal for AES67 Network Audio

- AES67 / DIGISYN LINK / RAVENNA delivers top-tier lossless, ultra-low-latency audio over standard IP networks, with flexible, reliable switches and devices.
- The DL-08/16/32 series supports OTA upgrades.
- DIGISYN LINK opens DSP/control protocols with open APIs for easy software development in audio routing, DSP, custom control, and monitoring.
- Free DIGISYN LINK3 Windows control software simplifies management.
- Powerful DSP with 4096-tap FIR filter, AEC, AGC and ANS.

## Application Cases


**Audio matrix**


**DSP processor**


**Amplifier**


**Line Array**


**Digital mixer**

- DL-08 offers AES67 / RAVENNA interfaces for network audio modules supporting up to 8x8 channels at 192kHz sample rate.

- DL-16 provides AES67/ RAVENNA interfaces for network audio modules supporting up to 16x16 channels at a 48kHz sample rate.

- DL-32 provides AES67 / RAVENNA interfaces for network audio modules supporting up to 32x32 channels at a 48kHz sample rate.

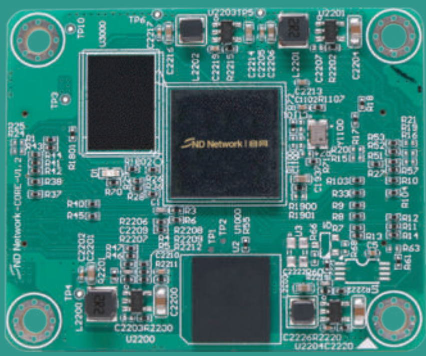
## Technical Specifications

| Audio Features | |
|---|---|
| Sampling Rate | 48 / 96 / 192 kHz |
| Audio I/O Channels (48kHz) | Up to 32 x 32 channels |
| Audio I/O Channels (96kHz) | Up to 8 x 8 channels |
| Audio I/O Channels (192kHz) | Up to 8 x 8 channels |
| Audio Stream Input/Output | Up to 4 x 4 audio streams |
| Digital Audio Format | TDM, $I^2S$ |
| Audio Transmission Format | AES67, RAVENNA network audio |
| Bit Depth | 16, 24, or 32 bits per sample |
| Clock | Onboard word clock |
| **Hardware** | |
| Dimensions | 4.5cm x 5.5cm |
| Clock | High-quality, low-jitter onboard clock |
| Physical Connectors | Mini-PCI, BTB connector |
| Power Supply | 5V DC |
| **Interfaces** | |
| Control Interfaces | GPIO, UART |
| Network | Standard RMII interface for switch chips |

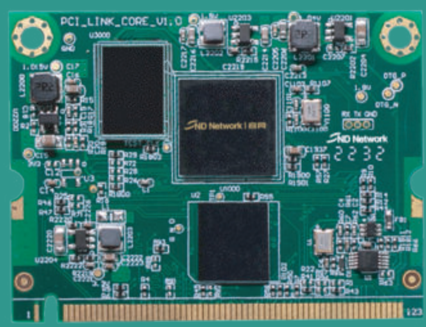
**AES67** AES67 Protocol

**DSP** Extensive DSP

**RAVENNA** Partner



## DL-08(BTB)



## DL-08(MINI-PCI)

## Features

- Ultra-Low Latency: ≤1ms network delay
- Sampling rates up to 192kHz (24-bit depth)
- Full compliance with AES67 protocol for seamless integration with existing AoIP infrastructure
- Optional DSP Expansion:

EQ: 16-Band | Delay: 2 Seconds | Compression

High-pass/Low-pass | FIR (4096 Taps)

Limiter | Noise Gate | Expander | Ducker

Auto-Mixing (Threshold/Gain Sharing)

Acoustic Echo Cancellation | Automatic Noise Suppression

Adaptive Feedback Control | Automatic Gain Control

AES67 Module

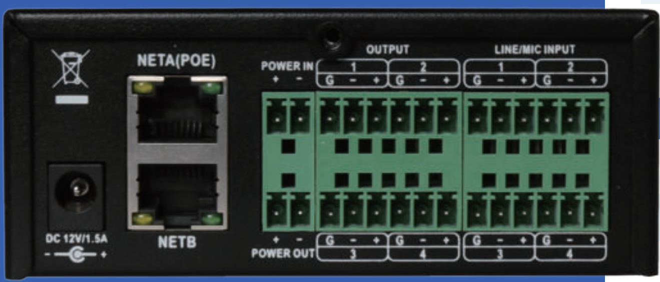
Built-in DSP

RAVENNA

Partner

# DMX208A

## 4 x 4 AES67 DSP Matrix Processor


## Technical Specifications

| | |
|---|---|
| **Audio Features** | |
| Sampling Rate | 48 / 96 kHz |
| Network Latency | 1 ~ 6 ms |
| Network Bit Depth | 16 / 24 / 32 bit |
| Channel Isolation | 84 dB A-weighted re+4 dBu |
| Frequency Response | ±0.3 dB 20 Hz - 20 kHz |
| Max Input/Output Level | 14 dBu @ 1 kHz, THD+N ≤ 1% |
| Dynamic Range | 110 dB, A-weighted |
| Mic Preamp Gain | 0 / 15 / 20 / 24 dB |
| Common-Mode Rejection Ratio (CMRR) | ≥50 dB @ 1 kHz, re+4 dBu |
| Total Harmonic Distortion + Noise (THD+N) | 0.005% @ re+4 dBu, A-weighted, 1 kHz |
| **Hardware** | |
| Analog Channels | 4 inputs 4 outputs |
| Network Channels | 4 inputs 4 outputs |
| Power Supply | 12V DC 1.5A / PoE power |
| Phantom Power | 48V 10mA |
| Analog Interface | Phoenix Terminal |
| Dimensions | 108 x 149 x 45 mm |
| Net Weight | 0.7 kg |

## Features

- Proprietary DSP 5A Algorithm: FIR@4096 Taps

  ANC AGC AFC AM AEC
- 100% Self-Developed OS
- Input Support: 48V Phantom Power
- 8 × 8 Auto Mixing Matrix (Auto Gain + Threshold-based)
- Integrates with DMX232 Mixing Matrix or DSCORE Audio Workstation to form large-scale AES67 & ST2110 Audio Networks
- Fully Compatible with Leading Audio Network Devices, Including:

RAVENNA AES67 & ST 2110 built-in

QSC Dante Dolby

## Comprehensive DSP Functions

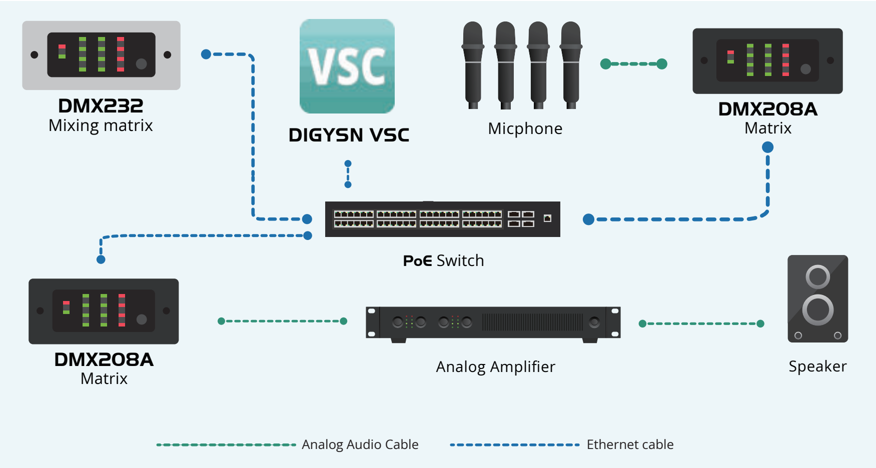
EQ: 16-Band | Delay | Compression | High-pass/Low-pass

FIR (4096 Taps) | Limiter | Noise Gate | Signal Generator | Ducker

Auto-Mixing (Threshold/Gain Sharing) | Adaptive Feedback Control

Acoustic Echo Cancellation | Automatic Noise Suppression | Automatic Gain Control

## Use cases



# DMX232

32 x 32 AES67 Mixing Matrix

AES67 Module

Auto Mixer

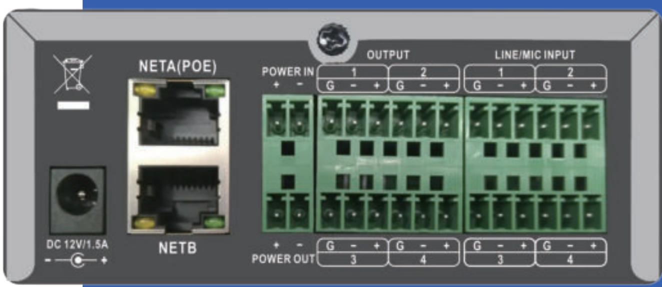
RAVENNA

Partner

## DMX232: Distributed network mixing hub

The DMX232 is a unique networked audio matrix mixer built upon the core technology of the DSCORE audio workstation. It extends the DIGISYN-AES67 ecosystem to smaller-scale audio applications such as background music (BGM), education, and compact meeting rooms. Operating at the AES67 protocol level, it communicates with other networked devices and integrates with the DMX208A matrix processor in AES67 or ST2110 audio networks to deliver mixing capabilities for up to 4 analog + 28 network channels.

The DMX208A handles per-channel DSP processing (e.g., AEC, ANS, AGC), while the DMX232 specializes in pure network mixing—processing network channels from the DMX208A and returning mixed network channels to the DMX208A nodes, achieving a synergistic 1 + 1 >2 audio enhancement effect.


## Network I/O audio stream channels

The DMX232 provides 28 × 28 network + 4 × 4 analog audio I/O streams, designed for centralized processing in multi-room interconnected and IP-based audio networks.

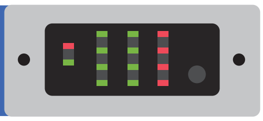
It supports: DIGISYN  AES67 networked devices (e.g., DLS series speakers, DS series processors, DSM series mics, DMX series matrices, DMA series adapters). Space-saving deployment in tight spaces (e.g., under desks) with a high-quality silent turbo fan for near-noise-free operation.

## Platform compatibility

| Dante | RAVENNA AES67 & ST 2110 built-in | QSC |
|---|---|---|
| Dante Network | RAVENNA NETWORK | Q-LAN NETWORK |

Compatible with AES67 Devices from Various Manufacturers

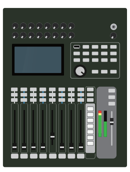
Supports Mixing of Up to 28 Network Channels



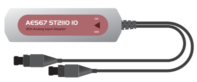
**DMX232**
Mixing Matrix

**DMX208A**
Matrix


**BI36**
Array Micphone



**Stuiomaster**
Mixing Console



**DMA04**
Adapter

## Features

- Threshold-based Auto Mixing
- Gain-sharing Auto Mixing
- 100% Self-developed OS
- 4 × 4 Analog Mixing Channels
- 28 × 28 AES67 Network Mixing Channels
- Max 32 × 32 Mixing Matrix
- Full Compatibility with Leading Audio Network Devices, Including:

RAVENNA
AES67 & ST 2110 built-in

QSC Dante Dolby

AES67 Module

Built-in DSP

RAVENNA

Partner

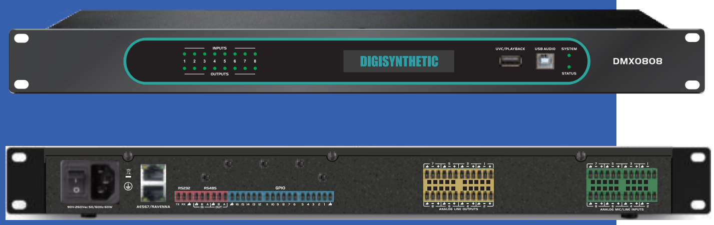
# DMX1616/0808

## 16 x 16 / 8 x 8 AES67 DSP Matrix

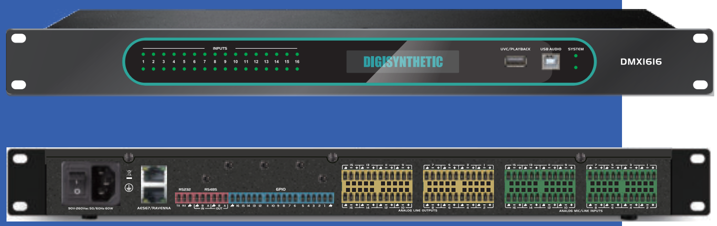
## NEW-GEN Network Audio Matrix

The 8x8 AES67 Matrix integrates 8×8 Phoenix terminal blocks with balanced analog I/O (including 8 channels of 48V phantom power) and 8×8 AES67 IP audio channels (48/96kHz/24-bit), forming a hybrid analog-network audio hub. Equipped with a user-programmable DSP library supporting 5A algorithms for acoustic echo cancellation (AEC), adaptive noise suppression (ANS), and feedback control (AFC), it enables real-time processing across all channels. Dual USB ports provide UAC and USB Audio, while 8 GPI and 8GPO ports allow seamless integration with third-party systems via contact closures or RS-485 protocols.

For larger deployments, the 16x16 AES67 Matrix scales up with 16×16 analog/AES67 channels (16 phantom-powered inputs) and an expanded DSP engine capable of managing 16 independent zones. Both models feature , sub-millisecond latency, and PoE-ready GPIO for triggers or sensor inputs, making them ideal for mission-critical environments like broadcast facilities, performing arts venues, and enterprise conference rooms.




DMX1616




DMX0808

## Features

- Up to 48 kHz Sampling Rate 32-bit Bit Depth
- Proprietary DSP 5A Algorithm: FIR@4096 Taps
  ANC AGC AFC AM AEC
- Fully Customizable DSP Algorithm Library
- Input Support: 48V Phantom Power
- 16 × 16 8 × 8 Auto Mixing Matrix (Auto Gain + Threshold-based)
- Dual USB Audio Interface with UAC Compliance
- 8-Channel GPIO Interface
- Fully Compatible with Leading Audio Network Devices, Including:

RAVENNA
AES67 & ST 2110 built-in

QSC Dante Dolby

## Technical Specifications

| Audio Features | DMX0808 | DMX1616 |
|---|---|---|
| Sampling Rate | 48 kHz | |
| Network Latency | 1 ~ 6 ms | |
| Bit Depth | 16 / 24 / 32 bit | |
| Channel Isolation | 84 dB，A-weighted，re+4 dBu | |
| Frequency Response | ±0.3 dB 20 Hz - 20 kHz | |
| Max Input/Output Level | 14 dBu @ 1 kHz, THD+N ≤ 1% | |
| Dynamic Range | 110 dB, A-weighted | |
| Mic Preamp Gain | 0 / 15 / 20 / 24 dB | |
| CMRR | ≥50 dB @ 1 kHz, re+4 dBu | |
| THD+N | 0.005% @ re+4 dBu, A-weighted, 1 kHz | |
| Hardware | | |
| Analog Channels | 8 x 8 | 16 x 16 |
| Network Channels | 8 x 8 | 16 x 16 |
| Power Supply | 100–240 V AC 50/60 Hz | |
| Phantom Power | 48V 10mA Ripple ≤10mV | |
| Analog Interface | Phoenix Terminal | |
| Network | 1 Gbps | |
| Dimensions | 108 x 149 x 45 mm | |
| Net Weight | 2.1 kg | 2.4 kg |

# DMA IO series

## AES67 to Analog DSP I/O Adapters

DL-04 **AES67 Module**

DSP **Built-in DSP**

RAVENNA **Partner**

### Smart & Flexible AES67 Device

The DMA DSP Processor supports analog audio signal input/output and can seamlessly integrate into AES67 network systems, transforming mixers, microphones, and other equipment into networked audio devices. Featuring customizable DSP functionality, it enables lossless, low-latency transmission with support for up to 48kHz sampling rate, built-in Hi-End ADC/DAC Additionally, the entire DMA series supports OTA (Over-the-Air) updates for continuous feature enhancements. Fully interoperable with all existing AES67 devices.

### Built-in Advanced DSP Functions

EQ: 16-Band | Delay: 2 Seconds | Compression | High-pass/Low-pass

FIR (2048 Taps) | Limiter | Noise Gate | Expander

Auto-Mixing (Threshold/Gain Sharing) | Ducker | Signal Generator

### Bluetooth Panel Supports PoE Power Supply, Ultra-Low Latency <2ms

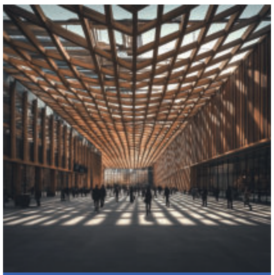
Supports 2×2 wireless Bluetooth transmission and AES67 protocol (2 channels) for high-quality networked audio. Features stereo RCA ports, 48kHz/24bit lossless audio, PoE (802.3af Class1) power, and <2ms ultra-low latency. Ideal for professional applications like conference rooms and live performances.

### Specifications

| | DMA02-OUT | DMA02-IN |
|---|---|---|
| Sampling Rate | 48kHz | |
| Bit Depth | 24 bit | |
| Maximum Signal Level | 18 dBu | |
| Frequency Response | 20Hz-20kHz,±0.5dB (Balanced) | |
| Impedance | 20kΩ（Balanced） | 110Ω（Balanced） |
| Connector | RJ45 / XLR-F | RJ45 / XLR-M |
| Dynamic Range | > 100dB | |
| Signal-to-Noise Ratio | > 100dB | |
| Number of Channels | 2 x 2 | 2 x 2 |
| PoE Power Supply | 802.3af PoE 15W Power supply | |
| Noise Floor | -90dBu | |
| Net Weight | 0.3 kg | |

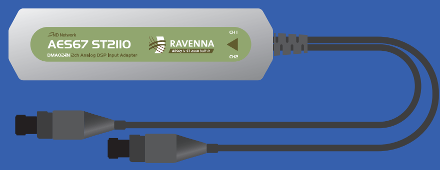
### Application Scenarios

Live Performance



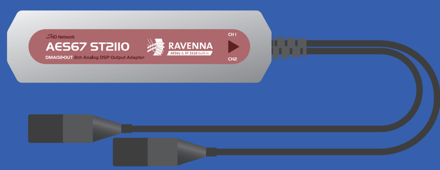
Public Address

Smart Education




## DMA02-IN

2x2 Analog to AES67 XLR Network Processor




## DMA02-OUT

2x2 AES67 to Analog XLR Network Processor


## DMA-BT22

2×2 Bluetooth to AES67 Wall Panel

### DMA-BT22 Product Features

- Supports 2×2 channel wireless Bluetooth transmission
- AES67 protocol compatible with ≥2 channels
- Stereo RCA interface supported
- ≥48kHz/24bit sampling rate & bit depth
- PoE (802.3af) power supply supported
- ≤2ms device latency

### DMA-BT22 Product Parameters

Bluetooth channels: 2；AES67 channels: 2；Stereo channels: 2；Sampling rate: 48kHz；Bit depth: 24bit；Power supply: Class 1 802.3af PoE；Device latency: <2ms

AES67 Module

Built-in DSP

RAVENNA

Partner

# DMA IO series

## AES67 to Analog DSP I/O Adapters


## Smart & Flexible AES67 Device

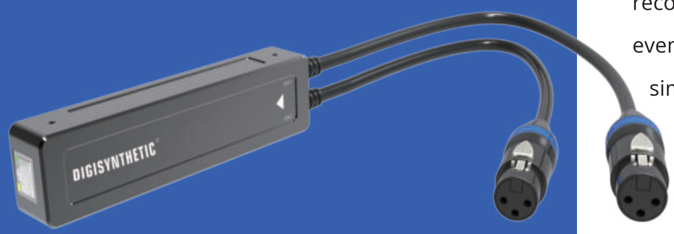
The DMA DSP Processor supports analog audio signal input/output and can seamlessly integrate into AES67 network systems, transforming mixers, microphones, and other equipment into networked audio devices. Featuring customizable DSP functionality, it enables lossless, low-latency transmission with support for up to 48kHz sampling rate, built-in Hi-End ADC/DAC Additionally, the entire DMA series supports OTA (Over-the-Air) updates for continuous feature enhancements. Fully interoperable with all existing AES67 devices.

**DMA04-P**

4x2 AES67 to Analog XLR Network Processor

## USB Plug-and-Play (No Driver Required)

The DMA24-USB Mini DSP Processor converts any computer into a networked audio device, connecting to AES67 audio networks for seamless playback and recording with any audio application. Ideal for conference rooms and lecture events, it allows a laptop to serve as a simple interface node, enabling simultaneous 24 x 24-channel recording and playback from PCs, Macs, or Linux-based computers.




**DMA04-D**

2x4 Analog to AES67 XLR Network Processor

## Built-in Advanced DSP Functions

EQ: 16-Band | Delay: 2 Seconds | Compression | High-pass/Low-pass

FIR (2048 Taps) | Limiter | Noise Gate | Expander

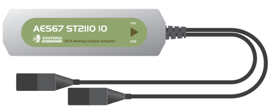
Auto-Mixing (Threshold/Gain Sharing) | Ducker | Signal Generator


**DMA24-USB**

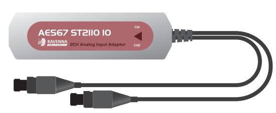
USB to AES67 Soundcard

## NEW-GEN DSP Adapters

DMXO2-IN




DMXO2-OUT




## Specifications

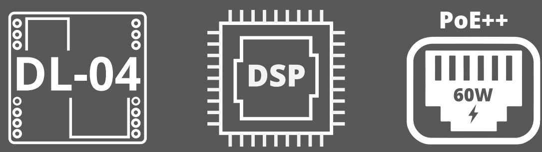
| | DMA04-P | DMA04-D | DMA24-USB |
|---|---|---|---|
| Sampling Rate | 48kHz | | |
| Bit Depth | 24 bit | | |
| Maximum Level | 18 dBu | | |
| Frequency Response | 20Hz-20kHz,±0.5dB (Balanced) | | |
| Impedance | 20kΩ(balanced) | 110Ω(balanced) | 10kΩ (Unbalanced) |
| Connector Type | RJ45 / XLR-F | RJ45 / XLR-M | RJ45 / USB |
| SNR | > 100dB | | |
| Dynamic Range | > 110dB | | |
| Channel Count | 2 x 4 | 4 x 2 | 24 x 24 / 2 x 2 (DSP mode) |
| PoE Support | PoE, PoE+, PoE++ | | USB Power |
| Net Weight | 0.3 kg | | |

**DMA-W22**

4x4 Analog to AES67 Wall Panel

# DLS PoE Speaker Series

AES67 DSP Network PoE Speakers




AES67 Module | Built-in DSP | PoE Power

## Application & Design Philosophy

The DLS-001 Ceiling Speaker, DLS-002 Wall-Mount Speaker and DLS-403Z address the aesthetic and spatial challenges of traditional audio systems in modern architectural spaces. Designed for language-focused environments (courtrooms, lecture halls, conference rooms) and background music applications (museums, transit hubs, public plazas), both models eliminate visual clutter through minimalist, low-profile designs. The DLS-001's flush ceiling integration and DLS-002's slim wall-mount form (≤60mm depth) ensure seamless blending with interiors while maintaining acoustic performance.

## Acoustic Engineering & Deployment Flexibility

Utilizes precision-tuned neodymium drivers for natural vocal reproduction and extended frequency response (80Hz–20kHz). Engineered rear-chamber acoustics enhance bass clarity without bulky enclosures.UL94V-0 fire-rated ABS housing with UV-resistant finish (IP54 moisture protection optional). Quick-install torsion clamps and tilt-adjustable brackets simplify deployment.

## Technical Specifications

| Audio Fearures | DLS-001 | DLS-002 | DLS-403Z |
|---|---|---|---|
| Rated Power | 30W @4Ω | 30W @8Ω | 30W @8Ω |
| Peak Power | 60W @4Ω | 60W @8Ω | 60W @8Ω |
| Effective Frequency Range | 20Hz-20kHz | | 80Hz~18kHz |
| THD-N | < 0.3% | | |
| Signal-to-Noise Ratio (S/N) | >100dB | | |
| Noise Floor | <200μV | | |
| Audio Transmission Protocol | AES67, ST2110, SoundNet | | |
| Sensitivity | 90dB | | 87dB |
| Output Interface | 3PIN 4.0mm Terminal (Positive/Negative/Ground) | | |
| Protection Functions | Overcurrent/Overheat/Overload/Short Circuit Protection | | |
| Power Input | PoE 802.3af / 802.3at / 802.3bt | | |
| Color | White/Black | White/Black | Black |
| Size | Φ264 x 242mm | 480 x 68 x 200mm | 530 x 98 x 153mm |
| Weight | 4.45KG | 3.5KG | 4KG |

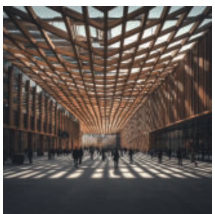
## Comprehensive DSP Functions

EQ: 16-Band | Delay: 2 Seconds | Compression | High-pass/Low-pass

FIR (2048 Taps) | Limiter | Noise Gate | Expander

Auto-Mixing (Threshold/Gain Sharing) | Ducker | Signal Generator

## Application Scenarios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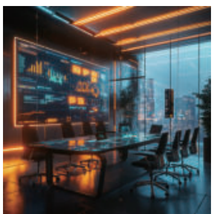
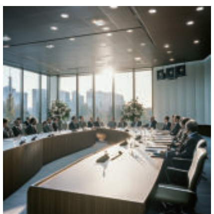

Smart Education


Conference


Command Center

**DLS-403Z**
PoE Column
DSP Speaker

**DLS-002**
PoE Network
Wall-Mount Speaker

**DLS-001**
PoE Network
Ceiling Speaker

## Features

- Integrated 4 × 4 Mixing Matrix
- 48kHz Sampling Rate
- Supports PoE-BT protocol 60W peak
- High sensitivity constant impedance design
- Extensive DSP Presets Selectable

DL-04

AES67 Module

DSP

Built-in DSP

RAVENNA

Partner

# DLM-003/004

AOIP Networked Microphone

## DLM-003

Directional Adjustable AOIP Networked Microphone

## DLM-004

AOIP Networked Microphone

## Features

- Exquisite and minimalist structural design highlights a modern vibe and conforms to ergonomics.
- Special electret condenser dual-microphone core design, with clear-bright sound quality and high sensitivity.
- Supports mainstream network audio protocols. Connect to the switch via a network cable, and transmit audio over the network to achieve signal interconnection and intercommunication.
- Powered by PoE (Power over Ethernet).
- Customizable analog backup for a more stable system.
- Resistant to radio-frequency and mobile phone interference.
- The base is equipped with damping rubber to effectively reduce the noise generated by external vibrations on the microphone.

EQ: 16-Band | Delay: 2 Seconds | Compression

High-pass/Low-pass | FIR (2048 Taps)

Limiter | Noise Gate | Expander | Ducker

Auto-Mixing (Threshold/Gain Sharing)

AEC | AGC

ANS | FBE

## Professional Audio Performance

Dual-capacitor cardioid design reduces noise. 90Hz-18kHz frequency response, 108dB dynamic range & 130dB max SPL for clear audio. Resists RF/mobile interference.

## Highly Integrated Network & Power Design

AOIP Support:
Connect via Ethernet (RJ45) for digital audio transmission, compatible with Dante/AES67 for low-latency networking.
PoE Powered (802.3af/at):
Single cable delivers data + power, no external supply needed.
Analog Backup Option:
Optional analog output ensures reliability during network issues.

## Intelligent Operation & Anti-Interference

Voice-activated operation (auto on/off). Remote control via software (customizable). Blocks mobile interference. Vibration-proof base. Combines smart sensing + rugged design for reliable performance in high-end venues.

## Technical Specifications

| Product Model | DLM-003 | DLM-004 |
|---|---|---|
| Frequency Response | 90Hz-18kHz | |
| Local Noise | 16dB (A-weighted) | |
| Dynamic Range | 108dB @1kHz (Max SPL) | |
| Maximum Sound Pressure Level | 130dB (1% THD @1kHz, 0dB SPL=$2\times10^{-5}$Pa) | |
| Microphone Sensitivity | -32dBV (25mA) | |
| Connection Method | RJ45/1m cable | |
| Power Supply Method | POE Switch (IEEE 802.3af/at compliant) | |
| Mode Selection | 1、Standard toggle; 2、Auto power-on ; 3、Press & hold to turn on ; 4、Press & hold to turn off (release to confirm); 5、Voice-activated mode; | |
| Microphone Type | Dual-capacitor | |
| Sampling Rate | 48K | |
| Maximum Power Consumption | 2W | |
| Directivity Characteristics | Omnidirectional ,Cardioid ,Supercardioid ,Hypercardioid ,Figure-8 | Cardioid |
| Size | Straw: 200*34.5*23mm Base: 138*98*54mm | Straw: 380mm Base: 138*98*54mm |
| Net Weight | 930G | 900G |

# AES-CORE 2U

CORE 2U 128 x 128 Audio Station

## SecureCore Architecture

Utilizes domestically developed chipset technology. Supports 128 channels @48kHz full parallel processing per unit, achieving 1ms end-to-end ultra-low latency through mixed-precision floating-point operations (FP16/FP32). Compatible with domestic operating systems like Kylin OS and UOS. Integrates AES67 protocols and distributed computing architecture to deliver highly reliable, fully networked audio routing and control solutions for large conferences, airports/railway stations, command centers, stadiums, and government/enterprise campuses.

## Networked I/O Distributed Deployment

Endpoint Flexibility: Analog/XLR/GPIO interface units (e.g., DL-series modules) can be deployed via IP networks at optimal physical locations (e.g., airport boarding gates, courtroom witness stands) for minimal signal path latency.Simplified Topology: Replaces traditional audio cabling with single-mode fiber, enabling ultra-low-latency (<1ms) transmission across buildings/campuses (up to 40km).

## Configurable Multi-Tier Processing Engine

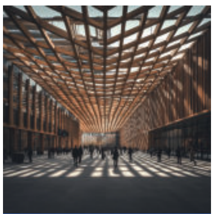
Enterprise-Grade DSP Pool: 128-channel parallel processing with dynamic allocation for:Acoustic Optimization: Echo cancellation (teleconferencing), adaptive feedback suppression (large-venue mic howling prevention). Priority Routing: Per-channel ducking (e.g., security alerts automatically overriding PA systems in stadiums).

## Premium ODM Development

Custom hardware solutions including: Analog interface cards (16x16 balanced I/O) GPIO control cards (32 channels) Serial communication cards (RS485/232 passthrough) Bespoke development for specialized require-ments.

| Audio Features | |
|---|---|
| Power Supply | 90 ~ 250 V AC 50/60 Hz |
| Audio Processing Latency | ≤ 5ms AD➞DSP➞DA (EQ) |
| Network Audio I/O (AES67) | 128 × 128 @ 48kHz |
| AEC Channels | 40 channels |
| USB Audio Interface Channels | 8 x 8 |
| Dimensions (L x W x H) | 482 x 285 x 99 mm |
| Supported Protocols | RAVENNA 、AES67、SMPTE ST2110 |

## Application Scenarios



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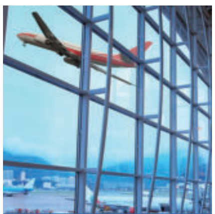
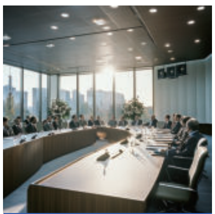
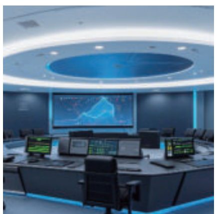


Airport



Conference

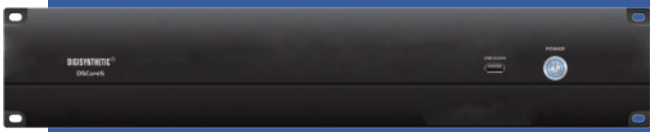


Command Center

AES67 OS

Premium Development

Partner



## AES-CORE 2U

### Features

- Arm Chipset: 128-channel full parallel processing
- Ultra-low Latency: Mixed-precision computing, end-to-end delay ≤1ms
- Fully Networked: Deep integration of RAVENNA/AES67 protocols
- Wide-area Deployment: IP-based distributed I/O, cross-region transmission <1ms
- Redundant Architecture: Dual power supply + dual NIC, failover <1ms
- Acoustic Optimization: Echo cancellation, feedback suppression, auto gain control
- Multi-scenario Support: Covers conferencing, transportation, security, and venues
- Deep Customization: ODM development for analog/GPIO/serial port cards
- Scalability: Supports specialized hardware customization

AES67 CORE

ODM

RAVENNA

AES67 OS

Premium Development

Partner


# AES-CORE 4U

## Features

- X86 Chipset: 1024-channel parallel processing
- Ultra-low Latency: Mixed-precision computing, end-to-end delay ≤1ms
- Fully Networked: Deep integration of RAVENNA/AES67 protocols
- Wide-area Deployment: IP-based distributed I/O, cross-region transmission <1ms
- Redundant Architecture: Dual power supply + dual NIC, failover <1ms
- Acoustic Optimization: Echo cancellation, feedback suppression, auto gain control
- Multi-scenario Support: Covers conferencing, transportation, security, and venues
- Deep Customization: ODM development for analog/GPIO/serial port cards
- Scalability: Supports specialized hardware customization

# AES-CORE 4U

CORE 4U 1024 x 1024 Audio Station

## SecureCore Architecture

Utilizes domestically developed chipset technology. Supports 1024 channels @48kHz full parallel processing per unit. Compatible with domestic operating systems like Kylin OS and UOS. Integrates AES67 protocols and distributed computing architecture to deliver highly reliable, fully networked audio routing and control solutions for large conferences, airports/railway stations, command centers, stadiums, and government/enterprise campuses.

## Networked I/O Distributed Deployment

Endpoint Flexibility: Analog/XLR/GPIO interface units (e.g., DMX208A matrix) can be deployed via IP networks at optimal physical locations (e.g., airport boarding gates, courtroom witness stands) for minimal signal path latency.Simplified Topology: Replaces traditional audio cabling with single-mode fiber, enabling ultra-low-latency (<1ms) transmission across buildings/campuses (up to 40km).

## Configurable Multi-Tier Processing Engine

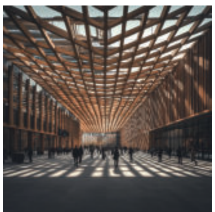
Enterprise-Grade DSP Pool: 1024-channel parallel processing with dynamic allocation forAcoustic Optimization: Echo cancellation (teleconferencing), adaptive feedback suppression (large-venue mic howling prevention). Priority Routing: Per-channel ducking (e.g., security alerts automatically overriding PA systems in stadiums).

## Premium ODM Development

Custom hardware solutions including: Analog interface cards (16x16 balanced I/O) GPIO control cards (32 channels) Serial communication cards (RS485/232 passthrough) Bespoke development for specialized require-ments.

| Audio Features | |
|---|---|
| Power Supply | 90 ~ 250 V AC 50/60 Hz |
| Audio Processing Latency | ≤ 5ms AD➞DSP➞DA (EQ) |
| Network Audio I/O (AES67) | 1024 x 1024 @ 48kHz / 512 x 512 @ 96kHz |
| AEC Channels | 160 channels |
| USB Audio Interface Channels | 8 x 8 |
| Dimensions (L x W x H) | 482 x 190x 408 mm (with handles) |
| Supported Protocols | RAVENNA 、AES67、SMPTE ST2110 |

## Application Scenarios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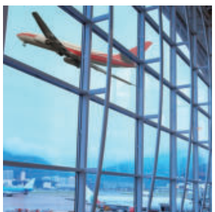
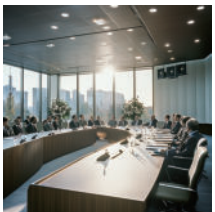
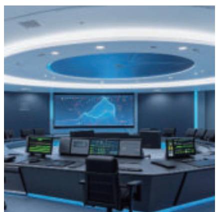

Airport


Conference


Command Center

# DB-32

32×32 DL Module Dedicated Development Board

## Included schematic diagrams accelerate product development.

This development board is designed for testing and evaluating the DIGISYN LINK AES67 network audio (DSP) module, enabling rapid functional validation to simplify customer development processes and improve efficiency.The board features a two-layer structure secured with brass standoffs and pin headers.Upper layer: Analog signal input board for receiving analog signals

## DL-04 Adapter Board Instructions

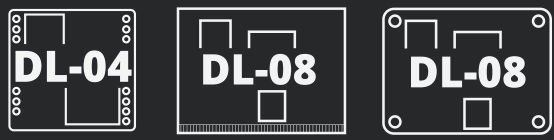
The DL-04 module has two asymmetrical pin rows – align with corresponding slots on the adapter.Critical note: Use ONLY the adapter's Ethernet port (labeled "DL-04") for networking.
The development board's Ethernet port will not function with this configuration.

## DL-08 (Mini-PCI) Adapter Board Instructions

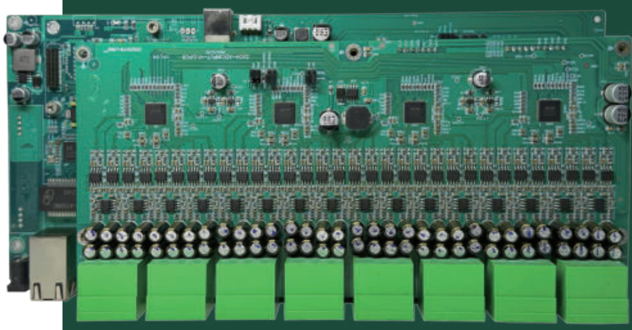
Insert the DL-08 module into the adapter's Mini-PCI slot.After connection:Use the development board's main Ethernet port for networking.

## Technical Specifications

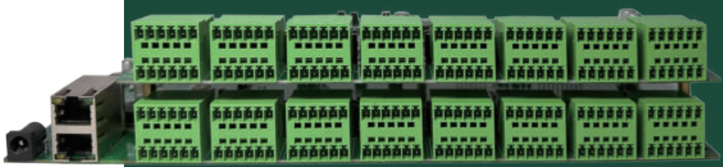
| Audio Equipment Specifications | Audio Channels |
| --- | --- |
| Analog Audio Channels | Max 32-in/32-out (Supports 16-in/16-out, 8-in/8-out, 4-in/4-out) |
| Network Audio Channels | Max 32-in/32-out (Supports 16-in/16-out, 8-in/8-out, 4-in/4-out) |
| Digital Audio Support | I2S: Max 8×8 @48kHz/TDM: Max 32×32 @48kHz |
| Sampling Rate | 48 kHz |
| SNR | 96dB@re+4dB (A-weighted) |
| THD | 0.005%@re+4dB (A-weighted) |
| Crosstalk | 90dB @ +4dBu (A-weighted) |
| Frequency Response | 20 - 20kHz ±0.2dB |
| Max Input Level | 14dBu 1 kHz THD+N<1% |
| Max Output Level | 14dBu 1 kHz THD+N<1% |
| Dynamic Range | 104dB (A-weighted, ref. max input level) |
| Power Supply | 12V DC/2A adapter |
| Supported Modules | BTB Module / Mini-PCI Module (with adapter) / DL-04 (with adapter) |




PIN Mini-PCI BTB



DB-32 Development Board



DB-32 Development Board

## Features

- Provides a standardized verification platform for DL module embedded device circuits.
- Can serve as the PCB mainboard for 32×32 interface boxes.

## Semi-finished solutions