



Overview

Amu 950 is a distributed network audio system processing service center, serving as a core AI audio algorithm processing platform for enterprises. It integrates powerful hardware processing capabilities with a diverse array of advanced audio algorithms. The system employs the first domestically developed audiovisual transmission protocol, TENDZONE VaiLink, specifically for the professional video field, providing ultra-low latency, uncompressed network audio input, output, and processing capabilities with a capacity of 128x128 channels. It supports 16x16 RTSP network audio streams, with audio matrix routing reaching 144x144, and is expandable and compatible with protocols like DANTE and AES67, supporting a maximum of 128x128 channels.

The Amu 950 features a configurable slot design with a total of 9 slots, allowing flexible configuration of up to 8 analog input/output cards, with optional DANTE expansion cards. The system's functions are software-defined, allowing for custom audio processing capabilities and system architecture, and it supports centralized management of network microphones, network speakers, audio nodes, and other devices.

It also incorporates a 1+1 hot backup design, supporting dual AC power supplies, ensuring the system's reliability in applications of any scale. With its exceptional performance and reliability, it presents an outstanding solution.

The Amu 950's extensive network audio channel access capacity and powerful processing capabilities make it suitable for a variety of large applications, including major command centers, conference centers, administrative centers, corporate headquarters, theme parks, and transportation hubs.

Feature

Network Audio Capabilities:

128x128 VaiLink Channels: Supports uncompressed, ultra-low-latency audio over the VaiLink protocol.

144x144 Audio Matrix Routing: Full-scale audio routing with precision and flexibility.

16x16 RTSP Stream Support: Integration of real-time audio streaming over IP.

DANTE and AES67 Compatibility: Expandable for industry-standard protocols up to 128x128 channels.

Flexible Hardware Architecture

Modular Slot Design: 9-slot chassis design, configurable with:

Up to 8 analog input/output cards

Optional DANTE expansion cards

Open Architecture Audio Designer Platform :

Using Audio Designer integrated tools, software module drag-and-drop operation, custom audio processing topology and control logic. Build and visualize signal processing workflows,

Designed for engineers with a fully operational interface.

Customizable elements, styles, and interaction methods.

Support to define the interface start page, page jump, create multiple controls, change the control location and size. a single interface can support multiple pages

Advanced Audio Processing

16 AEC Modules: With up to 512ms tail length echo cancellation.

40 AFC Modules: Flexible allocation for adaptive feedback control.

Hundreds of Audio Algorithms:

5A Algorithms

AI Noise Reduction

AI Auto Gain Control

Automatic Mixing (Threshold & Gain-Sharing)

Dynamic Compression and more.

Intelligent Microphone Management

MAOM System (Microphone Array Optimization Management):

Manage up to 128 microphones

Automatic interference analysis to prevent comb filtering

Intelligent gain allocation to prevent feedback

All microphones can remain open without screeching

Preset and Workflow Customization

Modular Preset Management: Add/exclude processing modules as needed.

Refined Presets: Apply settings at individual matrix crosspoints.

Offline Design Tools: Simulate workflows and system behavior prior to deployment.

User Interface & Control

Visual Management Interface:

Real-time dynamic audio level feedback
 Crosshair cursor for precise signal routing
 Multi-Platform UI: Compatible with Windows, Kylin, Android, iOS
 Custom Interface Design: Personalized UI for different user roles
 Multi-User Support: Hierarchical control authority, simultaneous operation

System Integration and Reliability

Open API: For seamless third-party integration
 Industrial-Grade Architecture:
 Linux OS-based system for security and stability
 Immune to common viruses
 1+1 Hot Backup Design:
 Dual AC power supply for uninterrupted operation

Specification

	Specification
Audio processor	Automatic gain AGC, threshold automatic mixer, gain sharing automatic mixer, ducker, noise gate, expander, limiter, parametric equalizer, graphic equalizer, compressor, high-pass filter, low-pass filter, crossover, delay, matrix mixer, router, selector, level control, etc.
Featured audio processing	Microphone fully open intelligent management AI-AGC, AI denoising, AI dereverberation Adaptive echo cancellation AEC , adaptive noise cancellation ANC Notch type feedback elimination AFC-N, frequency shift feedback elimination
Processing performance	Routing matrix capacity: 144*144 Can run 16 AEC processing modules (tail length up to 512ms) Can run 40 AFC processing modules Can run 2 AI algorithm modules
Preset Management	200 preset management
System Design	Customize the audio signal processing flow of the audio processing service platform Customize the signal routing relationship between network audio devices Freely configure presets, which can include all processing modules or some modules
User interface operation	Support Feiteng Kylin , Windows, Android , iOS, Hongmeng, Tongxin uos and other platform controls Support users to design personalized interface; multiple interface modes

User interface design	Users can customize the operating interface elements, styles, interaction methods, etc. Supports defining the interface startup page, page jump, creating multiple controls, and changing the control position and size A single interface can support multiple pages Support dynamic modification of control properties during operation Support interface preview and interface project upload Supports multiple data controls, date and time controls, custom control colors, fonts and icons
User interface interaction	Supports visual audio routing scheduling , audio plan saving and calling Support audio channel gain, mute control and level display Support system mute control Support matrix mixing control
Device Management	Support Vai Link audio interface machine access Support Vai Link amplifier access Support Vai Link audio microphone access Support Vai Link network audio processor access Support Vai Link audio and video integrated interface machine access
Communication Mode	Support unicast and multicast configuration
Permission Management	Supports hierarchical management of user permissions, supports multi-user online management operations at the same time, and supports multi-terminal management of the same permission
Scalability	Open API interface, support customized docking
System Tools	Supports auxiliary tools for system operation and debugging such as level meter, signal generator, spectrum analyzer, and monitoring
Maintenance function	Support log management, can view/export logs, automatically count server resource usage, and report to the management platform Important event alarms such as server failure events, network transmission redundancy events, and network transmission abnormal events facilitate operation and maintenance management Support web page upgrade, configuration backup/restore, import/export, network configuration, system restart, system reset
Network Interface	RJ45 Gigabit Ethernet port x 2 , optical port x 2
Serial control port	One RS-232 interface, one RS-485 interface, Phoenix terminal, support bidirectional control;
GPIO interface	1 GPIO interface that can define input and output;

power supply	100-240VAC, dual power supply
Device power consumption	80 W
Ambient temperature	Operating time: 0°C~45°C Non-operating : -40 °C ~ 55 °C
Relative humidity	95 % when not in operation , non-condensing at 25°C~30 °C
Installation	Support 2U independent cabinet installation
Device size	482mm * 300mm * 88mm (width*depth * height)

Modular I/O Cards

128-channel Dante expansion card Amu 128D



characteristic

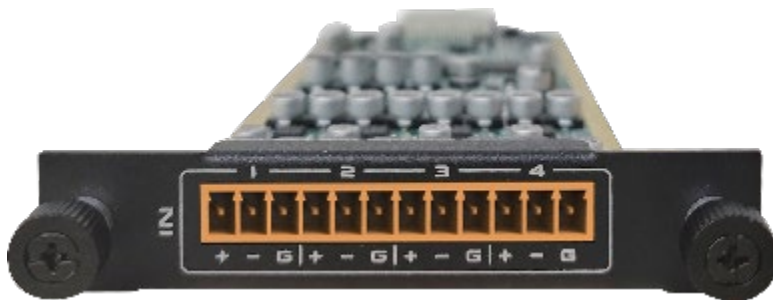
64 input and 64 output Dante channels

Support AES67 protocol

Dual RJ-45 Gigabit ports support redundant backup transmission, used to connect the main link and backup link respectively

Used with Amu 950 host, supports up to two cards

4-way microphone/line input card Amu 4I



characteristic

Supports 4-channel microphone/line level analog audio signal input and provides 48V phantom power;

5-level input gain adjustable: 0, -10dB, -20dB, -30dB, -40dB;

Dynamic range: > 108dB (A-weighted);

Frequency response (± 0.2 dB): 20Hz~20kHz;

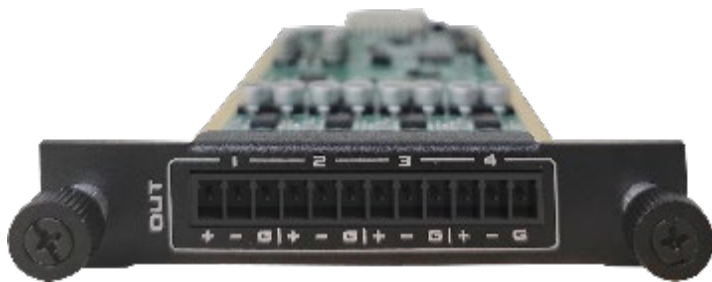
Input impedance: 2k Ω

Total harmonic distortion (THD+N): <0.005% @1kHz, +4dBu

Maximum input level: +18dBu

Interface type: 12pin3.81 Phoenix terminal

4-line output card Amu 4 O



characteristic

Support 4 balanced line level outputs

Dynamic range: > 108dB (A-weighted);

Frequency response (± 0.2 dB): 20Hz~20kHz;

Output impedance: 200 ohms

Total harmonic distortion (THD+N): <0.005% @1kHz, +4dBu

Maximum output level: +18dBu

Interface type: 12pin3.81 Phoenix terminal

Rear Panel

